NEC

NEC Infrontia

Xn 120

Consolidated Manual

Easy Edit Help

Character Set
The following character set is supported by the system.

Most characters are available via the keys of the keyboard:

! " # $ £ % & ( ) * + , - . / : ; < > = ? @
[ ] \ ^ _ ` { } ¦
0 1 2 3 4 5 6 7 8 9
A B C D E F G H I J K L M N O P Q R S T U V W X Y Z
a b c d e f g h i j k l m n o p q r s t u v w x y z

Others must be entered by the following method:
1. Enable Num Lock on the keyboard to allow the use of the number pad.
2. Hold down the ALT key and enter the four digit code with the number pad shown below and then release the ALT key.

Character	ALT +		Character	ALT +
À	**0192**		û	**0251**
Á	**0193**		ú	**0250**
Â	**0194**		ÿ	**0255**
Ã	**0195**		Ö	**0214**
Ê	**0202**		Ü	**0220**
õ	**0245**		¢	**0162**
á	**0225**		¥	**0165**
¡	**0161**		£	**0163**
Ó	**0211**		¸	**0184**
ú	**0250**		´	**0180**
ñ	**0241**		•	**0149**
Ñ	**0209**		ä	**0228**
ô	**0244**		Ö	**0214**
ö	**0246**		ü	**0252**
Ò	**0210**		¿	**0191**
Ç	**0199**		«	**0171**
ç	**0231**		»	**0187**
Î	**0206**		æ	**0230**
Ä	**0196**		Æ	**0198**
Å	**0197**		É	**0201**

Easy Edit Help

DDI Time Mode Table (3232)
The DDI time mode table is used to intercept specified incoming DDI calls and route them to an entry in the DDI Routing Table before they are routed o the DDI Table Area as defined in DDI Table Area Target.

There are 100 tables available, for each table you can specify up to eight time periods during which the received DDI number will be routed directly to an entry in the DDI Routing Table.
The time periods are independent of any of the Night Modes that may be setup for the system.
For any times outside of the specified time periods the system will route the DDI calls via the DDI Table Area Target.

Any incoming DDI numbers that are not entered in the DDI Time Mode Table will be routed via the DDI Table Area Target.

Table No - The DDI Time Mode table number 1 to 100.
Received DDI - Enter the DDI number received by the system.
Start Time - Enter the start time for any period that you want to intercept the received DDI number.
End Time - Enter the end time for any period that you want to intercept the received DDI number.
DDI Table Number (1-2000) - Enter the 'Table No' within the DDI Routing Table that will be used to specify the routing of the received DDI number.

Easy Edit Help

Reject Call by Caller ID (3230)
The system can reject incoming ISDN calls if the callers number matches an entry within a defined Abbreviated Dial Group.

The 'VRS Message for Caller ID Refuse' option in VRS Options will assign the DSPDB message number that will be played to the incoming caller when the call is rejected by the 'Flexible Ringing by Caller ID' feature. The message will be repeated twice and then the call will be disconnected automatically.
If this option is not set the incoming caller may hear a network tone or announcement indicating that the number is not obtainable.

Caller ID Refuse Target Area - The Abbreviated Dial Group number that will contain the Caller ID numbers of calls that the system will reject.
The trunk port must also be enabled in the 'Caller ID Refuse Setup' option in Trunk Basic Data Setup.
The feature must also be switched on by a user with either the 'Set Caller ID Refuse' service code or function key, refer to the feature 'Flexible Ringing by Caller ID'.

Easy Edit Help

Remote Conference Setup (3236)
A user can join in a conference by dialling the pilot number of a conference circuit. The system will prompt the user for a password before allowing then to join the conference.
The DSPDB card must be installed for this feature to be available. Ensure you select 'Standard DSPDB' as the VM Type for the DSPDB card in IntraMail Basic Setup, if this option is set to IntraMail the Remote Conference feature is not available.
Each conference must have a pilot code assigned.
Users must have the Remote Conference option enabled in within Remote Conference COS.

Remote Conference Group Number - There are four conference groups available.

Pilot Number - Enter the pilot code for each remote conference.
Conference Name - Enter the name to identify the conference group. The name will be shown at the keyphone display.
Password - Each conference must have a four digit password.
Maximum Participants - Enter the maximum quantity of callers that are allowed to join the conference. Enter 0 to 32.
Maximum Conference Duration - Enter the maximum duration (seconds) allowed for a conference before it will be disconnected. Enter 0 to 64800. The system can give a warning tone before the conference is disconnected.
End Tone Alert Time - Enter the duration (seconds) before the 'Maximum Conference Duration' that the system will play the warning tone to indicate that the conference will be disconnected. Enter 0 to 64800.

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Remote Conference (3237)
A user can join in a conference by dialling the pilot number of a conference circuit. The system will prompt the user for a password before allowing then to join the conference.
The DSPDB card must be installed for this feature to be available. Ensure you select 'Standard DSPDB' as the VM Type for the DSPDB card in IntraMail Basic Setup, if this option is set to IntraMail the Remote Conference feature is not available.
Each conference must have a pilot code assigned in Remote Conference.

For each Class of Service (COS) enable/disable the Remote Conference feature.

Easy Edit Help

ISDN Call Forward Method (3238)
Select the method to be used for off premise call forward for ISDN trunks, refer to the Call Deflection feature for further information.

Trunk Port - Ensure you set all trunks that the incoming calls may be received on.
ISDN Call Forward Method - Select the method from the list.
Normal Operation = The system will use a separate outgoing trunk port to perform the off premise call forward.
Call Rerouting = The system will use the Network service 'Call Rerouting' as per ETSI EN 300 207 Clause 10.5.
Call Deflection = The system will use the Network service 'Call Deflection' as per ETSI EN 300 207 Clause 9.2.4.5.

Easy Edit Help

Timer Class for Extensions (3241)
Certain timers can be selected based on a timer class that is assigned to each night mode.
If the timer is extension based the class will be defined by this program (refer to Timers for Class of Service for details of which timers are extension based).

Class of Service - Enter the extension based timer class (0-15) for each night mode. If 0 is entered the timer will be defined for the system in Timers.

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Timer Class for Trunks (3240)
Certain timers can be selected based on a timer class that is assigned to each night mode.
If the timer is trunk based the class will be defined by this program (refer to Timers for Class of Service for details of which timers are trunk based).

Class of Service - Enter the trunk based timer class (0-15) for each night mode. If 0 is entered the timer will be defined for the system in Timers.

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Timers for Class of Service (3242)
Certain timers can be selected based on a timer class that is assigned to each night mode and is based on the trunk or extension port, if the timer is not shown in this list then it is a system based timer which can be found in Timers
There are 15 classes available, if the class is set to 0 in Timer Class for Extensions or Timer Class for Trunks the Timer for Class of Service is not used and the system based timer will be used.
Each timer is based on the trunk or extension port related to the call, the list below indicates whether the Trunk or Extension Timer Class will define the class number.
Enter timer values are in Seconds from 0 to 64800.

Description	Trunk or Station Based
Camp-On trunk call back time - When a trunk Camp-On rings back it will ring for this duration.	Station
Camp-On cancel time - Any Camp-On that is not completed will be canceled after this time.	Station
Call Coverage Delay Interval - The delay before the Virtual extension will ring when set to Delayed Ringing.	Station
Internal Call Inter-digit time - The Interdigit time when placing an internal call.	Station
External Call Inter-digit time - The Interdigit time when placing a trunk call. When this timer expires the trunk call will be classed in the answered state and the call timer will start.	Trunk
Hotline call start time - Set the delay after the user goes off hook before the hotline call is placed, during the delay the user hears dial tone and can dial to override the hotline call. If the time is set to 0 the hotline call is placed immediately.	Station
Incoming ring no answer alarm start - Set the duration (Seconds) that a trunk call must ring before the alarm tone starts.	Trunk
Normal DIL Incoming no answer time - Set the duration (Seconds) that a trunk call must ring before the call steps to the second target. This timer operates for trunks set as either Normal or DIL in Incoming Service Type Setup. The first and second targets for Normal type trunks are assigned in IRG Assignment Normal. The first target for DIL type trunks is assigned in DIL Target Assignment, the second target is assigned in DIL Step on Target Assignment.	Trunk

DID no answer time - Set the duration (Seconds) a DDI call must ring before the call steps from Target 1 to Targets 2 or 3. The DDI must also have a No Answer Transfer Option selected. The targets and Transfer options are assigned in the DDI Routing Table . Trunks are set as DDI type in Incoming Service Type Setup.	Trunk
Normal Hold recall time - The duration a call must be on hold before it will ring back to the extension that placed the call on hold.	Station
Normal Hold call back time - The duration the ring back tone will ring at the extension.	Station
Exclusive Hold recall time - The duration a call must be on hold before it will ring back to the extension that placed the call on hold.	Station
Exclusive Hold call back time - The duration the ring back tone will ring at the extension.	Trunk
Park Hold time - The duration a call must be on Park hold before it will ring back to the extension that placed the call on hold if the extension's Class of Service option for Normal/Extended Park Hold is set to off.	Station
No answer time for call forward - Set the duration the call will ring before being forwarded when the user has set Call Forward No Answer.	Station
Ring Inward Recall time - Set the duration that a call will ring after being transferred by a user. After this time the call will ring back to the extension that performed the transfer.	Station
DUD/DISA No Answer time - The duration the system will ring at the target extension. When this timer expires the caller will be disconnected or step on to the Auto Attendant Fall Over.	Trunk
Disconnect after DUD/DISA re-transfer to IRG - The duration the system will ring at the Auto Attendant Fall Over. When this timer expires the call will be disconnected.	Trunk
DISA Conversation Warning tone time - The duration the system will wait before sending a warning tone to a trunk to trunk call via DISA. After this time expires the user can enter the continue code (if enabled) or the trunk to trunk call will be disconnected.	Trunk
DISA Conversation Disconnect time - The duration the system will wait before disconnecting a trunk to trunk call via DISA, this timer starts when the trunk to trunk call is established. After this time expires the trunk to trunk call will be disconnected unless the caller has entered to continue code. The duration of this timer must be greater than the combined times of the DISA Conversation Warning tone time plus the Continue Duration for DISA trunk to trunk time.	Trunk
DISA Internal Paging duration - The duration the system will allow the DISA caller to be connected to the internal paging zone. When this time expires the call will be disconnected.	Trunk
DISA External Paging duration - The duration the system will allow the DISA caller to be connected to the external paging zone. When this time expires the call will be disconnected.	Trunk
External Paging Timer - Enter the maximum duration the system will allow an external page call when the timer expires the page call will be disconnected.	Station

Easy Edit Help

Extension Call - Extension Message (3243)
Extension Call is available to Auto Attendant (DUD/DISA) callers only. The system must have the DSPDB card installed.
The system can play a DSPDB message to the incoming caller if the target extension is busy or ring no-answer. The caller may then have further single digit options available if assigned in Auto Attendant Single Digit Options.

The setting in Extension Call - System Message will take priority over any settings for the extension made here.

Extension Call Busy Message - Enter the DSPDB message number (0-48) to be played to the incoming auto attendant caller if the extension is busy.
Extension Call No Answer Message - Enter the DSPDB message number (0-48) to be played to the incoming auto attendant caller if

the extension does not answer. The no answer time is set by the 'No Answer Time for Call Forward' in Timers.

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Extension Call - System Message (3244)
Extension Call is available to Auto Attendant (DUD/DISA) callers only. The system must have the DSPDB card installed.
The system can play a DSPDB message to the incoming caller if the target extension is busy or ring no-answer. The caller may then have further single digit options available if assigned in Auto Attendant Single Digit Options.

The setting made here will take priority over any made in Extension Call - Extension Message.

Extension Call Busy Message - Enter the DSPDB message number (0-48) to be played to the incoming auto attendant caller if the extension is busy. Enter 0 if you want to use the settings for the extension made in Extension Call - Extension Message.
Extension Call No Answer Message - Enter the DSPDB message number (0-48) to be played to the incoming auto attendant caller if the extension does not answer. The no answer time is set by the 'No Answer Time for Call Forward' in Timers. Enter 0 if you want to use the settings for the extension made in Extension Call - Extension Message.

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Virtual Extension Enhanced Setup (3247)
Refer to 'Multiple Directory Numbers' in the Features section for further information.

Answering a Call to a Virtual or Call Coverage key
The operation of each virtual or call coverage key can be set to either release or remain busy when a call is answered.
When set to remain busy further calls can not be made to the virtual / call coverage extension number.

Placing Outgoing Calls from the Virtual or Call Coverage Key
Calls made by pressing the virtual / call coverage key can use the extension number of the key or the extension name of the phone that is placing the call.

Extn / Virtual Port - The extension port number of the virtual or call coverage key. Virtual extension numbers are indicated by a V before the port number.
Virtual Extension Key Operation Mode - Select the operation of the key when calls to the virtual / call coverage extension are answered.
Release = The key will go out. Further calls can be placed to the virtual / call coverage extension.
Remain Busy = The key will not go out. Further calls can not be placed to the virtual / call coverage extension.
The setting of 'Option when answered' in Virtual Extension Options must be set to 'Release VE after answer' for the settings made in this program to operate.
Display Mode when Placing a Call on Virtual Extension Key - Outgoing calls placed by pressing the virtual / call coverage key can

use the extension number of the key or the extension name of the phone.

Virtual Extension Number = Uses the number of the virtual / call coverage key.

Actual Station Name = Uses the name of the phone from which the call is made.

The 'Virtual Mode' in Keyphone Options must be set to 'Outgoing' for this option to be available.

Easy Edit Help

1 Digit Access Setup (3239)

The Service Codes and Option Codes used while accessing the VRS voice mail can be either fixed or flexible.
Select the method to be used in Voice Mail Basic Setup. The DSPDB card must be installed to provide the VRS Voice Mail.

The flexible codes are shown in the following table.
The single digit # can not be used as a 1 Digit Code.
You do not need to enter the # after the DSPDB Service Code or Option Code.

1 Digit Code	DSPDB Voice Mail Service Code	DSPDB Voice Mail Option Code
1	1	1
2	32	9
3	65	7
4	62	2
5	7	4
6	61	3
7	31	6
8		28
9		
0	50	
*	0	*

- The 1 Digit Code is dialled by the user when accessing the VRS voice mail.
- The Option Codes are available while listening to your messages i.e. the Play Message option codes.
- The VRS Voice Mail Service Codes and Option Codes are shown in the table below.
  Note - Service Code 9# to End Recording of a Message can not be assigned to a 1 Digit Code, you must always dial 9# to end recording.
- The 1 digit codes can have both a Service Code and Option Code if required.
- You will not be able to assign a 1 digit code to all Service Codes as there are more service/option codes than 1 digit codes.

VRS Voice Mail Fixed Service Codes and Option Codes Table.

Function	Service Code	Optional Function	Option Code
Help	0#		
Play Message	1#	Replay Message	1#
		Pause/Restart the playback	4#
		Erase message and play next message (during playback)	7#
		Save message and play next message	9#
		Copy the message	2#
		Skip playback forward 8 seconds	3#
		Skip playback backward 8 seconds	6#
		Broadcast the message	28#
		Exit	*#
Erase ALL messages	7#	Confirm Erase	0#

End recording of message (This Service code can not be assigned in the 1 Digit Code translation table)	9#		
Broadcast message to a multiple address group	2#		
Greeting Message 1	Play	31#	
	Record	32#	
	Erase	37#	
Greeting Message 2	Play	35#	
	Record	33#	
	Erase	38#	
Greeting Message 3	Play	36#	
	Record	34#	
	Erase	39#	
Message Notification	61#	Notify to an extension	1#
		Notify to an external number	2#
		Cancel notification	0#
		Exit	*#
Set Automated Attendant	62#		
Play messages 'First in First out'	63#		
Play messages 'Last in First out'	64#		
Password setting	65#		
Turn On/Off message recording (including Conversation Recording)	66#	Voice Mail key flashes when recording is off.	
Exit	*#		
Escape from mail box (for incoming trunk callers)	50#		

<u>VRS Voice Mail Flexible Service Codes and Option Code Table.</u>
The table below can be used to printout and record any changes you have made to the VRS 1 Digit Codes.

Function	**Service Code**	**Optional Function**	**Option Code**
Help	*		
Play Message	1	Replay Message	1
		Pause/Restart the playback	5
		Erase message and play next message (during playback)	3
		Save message and play next message	2
		Copy the message	4
		Skip playback forward 8 seconds	6
		Skip playback backward 8 seconds	7
		Broadcast the message	8
		Exit	*
Erase ALL messages	5	Confirm Erase Exit	0# *#
End recording of message	9#		
Broadcast message to a multiple address group			
Greeting Message 1	Play	7	
	Record	2	
	Erase		

Greeting Message 2	Play		
	Record		
	Erase		
Greeting Message 3	Play		
	Record		
	Erase		
Message Notification	6	Notify to an extension	1#
		Notify to an external number	2#
		Cancel notification	0#
		Exit	*#
Set Automated Attendant	4	confirm exit	0# *#
Play messages 'First in First out'			
Play messages 'Last in First out'			
Password setting	3	confirm exit	0# *#
Turn On/Off message recording (including Conversation Recording)		Voice Mail key flashes when recording is off.	
Exit			
Escape from mail box (for incoming trunk callers)	0		

Easy Edit Help

Automatic Switching Patterns (1001) The system will follow the pattern number assigned in the Weekly/Holiday Service Switching if enabled in Night Service Options.

Pattern 1-10 - The Night Service Pattern number assigned in Weekly/Holiday Service Switching.
Start Time - The start time in 24 hour clock format for the chosen Mode.
End Time - The end time in 24 hour clock format for the chosen Mode.
Mode - The night mode number 1-8 that the system will use between the start and end times.

The system will use Night Mode 1 for any time that is un-defined.
Enter the same time for the end of one mode and the start of the next otherwise the system will use Mode 1 as the gap is an un-defined time.
Do not overlap the end and start times otherwise the system will ignore the start time and continue in the current night mode until the next start time.

If the night mode is changed manually the system will remain in this mode until the next automatic time setting.

Easy Edit Help

Night Service Options (1002)

Day/Night Mode No - The night mode number 1-8 used throughout the configuration of the system.
Display Text - The system will display this text at any display keyphone that is able to manually change the night modes.
Manual Night Switching - Is manual switching of the night modes available on the system. For an extension to manually change the night mode it must be enabled in Class of Service.
Auto Night Switch - Is automatic switching of the night modes available on the system. The system will follow the Weekly and Holiday Switching.

Easy Edit Help

Weekly Night Service Switching (1003)
The system will use the Night Service Pattern defined for the day of week unless there is a Holiday pattern set in Holiday Night Service Switching.
The start and end times for each pattern are defined in Automatic Switching Patterns.

Day of Week - The day of the week of the system clock. You can set the system time and date while connected to the system with PCPro, select the clock-face icon from the toolbar.
Night Service Pattern - Enter the night service pattern number (0-10) that the system will use for each day of the week. If you enter 0 the system will not use any automatic night service switching pattern during that day, it will remain in the last mode set for the previous day.

Easy Edit Help

Holiday Night Service Switching (1004)
The system will use the Night Service Pattern defined for the day of week unless there is a Holiday pattern set.
The start and end times for each pattern are defined in Automatic Switching Patterns.

Day - The date of the month. You can set the system time and date while connected to the system with PCPro, select the clock-face icon from the toolbar.
Month - Enter the night service pattern number (0-10) that the system will use for the date within any month. If you enter 0 the system will use the pattern set in Weekly Night Service Switching.

Easy Edit Help

Keyphone Options (1006)
Options that can be set for each keyphone.

Display Language - Select the display language text. The user can also select the language with Service Code 778.
DC Key Mode - Select the operation of the DC key (Common or Group Abbreviated dialling).
Transfer Key - Select the operation of the Transfer key (TRFR).
Transfer = Hold and Transfer a call.
Series Call = Hold and Transfer a call and wait for the caller to return after the call is completed.
Flash = Send a Hook Flash during an analogue trunk call.
Hold Key Mode - Select the operation of the Hold key.
Normal Hold = The held call can be retrieved by other users.
Exclusive Hold = The held call can only be retrieved by the phone that placed the call on hold.
Park Hold = The call is held under a free Park Hold Function Key (The keyphone must have a Park Hold key available when the HOLD key is pressed).
Trunk Key Mode - Select the operation when the user presses a CO key (CO key is a programmable function key set to CO Key mode) while they already have a call in progress.
Hold = The current call is held and the new trunk is seized.

Disconnect = The current call is disconnected and the new trunk is seized.
Pre-Select/One Touch - Select the operation of the keyphone for DSS/One Touch keys (Programmable function key set as DSS/One Touch) while the keyphone is on hook (idle).
Pre-Select = The key press is not activated until the user goes off hook. The pre-selection is remembered for the period set by Pre-Selection time set in System Options for Keyphones
One Touch = The key press is activated immediately and the call is placed.
Auto Answer for Trunk Call - Turn on/off automatic answer of a trunk call ringing at the extension when the keyphone goes off hook.
Call Back - Turn on/off automatic answer of a Call Back (Camp On).
Off Hook Signal - Select the alerting indication for off hook ringing indication.
Call List - Should the Last Number Dial (LND) store either internal and external calls or just external calls.
Store Caller ID for Answered Calls - Should the Incoming Calls list show both answered and un-answered calls. If disabled then only un-answered calls are listed.
Virtual Mode - Select the operation of Virtual Extension/Call Coverage keys (programmable function key set as Virtual Extension Key).
DSS = Placing and answering calls by pressing the key
Outgoing = Placing calls by pressing the key (answering calls is not possible)
Ignore = Answering calls by pressing the key (Placing calls is not possible)
ABB Preview - Select the operation of the DC key for display keyphones. (Keyphones without a display must go off hook before pressing the DC key).
Preview = The Abbreviated dial name and number is displayed and the user must go off hook to place the call.
Outgoing Immediately = The outgoing call is placed automatically after the user enters the Abbreviated dial number.
Multi-language calendar display on LCD - For keyphones that are not set to English display text you must turn this item on for the day of week to be displayed in the chosen language.
VE Toll - Defines the toll restriction class used when an outgoing call is made on a Virtual Extension key.
Use Virtual Toll = uses the toll restriction class assigned to the virtual extension.
Use Extension = uses toll restriction class assigned to the keyphone.
Call Register Mode - Defines the Caller ID scroll store stores trunk calls only or extension and trunk calls.
MIC Lamp Status Change - Defines whether the MIC lamp should be on or off to indicate activation.
VRS Message Number - Defines the VRS message to be used instead of ringback tone.
Topaz Screen Saver Start Time - Defines the start time for the screen saver.

Easy Edit Help

F-Route Time Schedule On/Off (1008)
The Flexible Routing (F-Route) tables can be linked to the day and time the call is placed.
If the time scedule is set to 'not used' then the following F-Route options are not used.
F-Route Selection for Time Schedule
F-Route Time Schedule Setup
F-Route Weekly Time Schedule
F-Route Holiday Time Schedule

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F-Route Dial Analyse Table (1009)
This table is used to compare the digits. This is the first table used by all numbers set as F-Route in System Numbering. This table will process the first 8 digits of the number, if you need to process more than the first 8 digits then you must pass the number to the F-Route Additional Dial Analyse Table which can process up to 24 digits.

Dial Data - The digits being processed by the system, enter up to 8 digits.
Service Type - For each Dial Data entry select the operation within F-Route.
Extension Call = The number will be routed back to System Numbering. You must translate this number otherwise it will never exit the F-Route process. The 'Additional Data' for this type will define the number of digits to be deleted.
F Route Table = The number will be routed to an F-Route Table for translation.
If Time Schedule is used you will use F-Route Selection for Time Schedule to select the F-Route table to be used for the current time.
If F-Route Time Schedule On/Off is set to 'not used' you will use the F-Route table specified in the 'Additional Data'.
Additional Dial Analyse Table = This is used if you need to process more than 8 digits. The 'Additional Data' for this type will define the Table Area number 1-4 in F-Route Additional Dial Analyse Table.
Additional Data - The function of the additional data is dependant on the setting for the 'Service type':
Extension Call = Specify the quantity of digits to be deleted 0-255, 255 will delete all digits.
F Route Table = Specify the 'F Route Selection' number (1-500) to be used in F-Route Selection for Time Schedule if Time Schedule

is used. Specify the 'F Route Table' number (1-500) in F Route Table if Time Schedules are not used.
Dial Extension Analyse Table = Specify the Table Area number 1-4 in F-Route Additional Dial Analyse Table.
Dial Tone Simulation - The system can generate dial tone if required, to be used in place of the dial tone received from the outgoing trunk. The system will use 2-Internal Dial Tone in Service Tones.

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F-Route Additional Dial Analyse Table (1010)
The Dial Extension Analyse Table is used when the number is set to 'Additional Dial Analyse Table' in F-Route Dial Analyse Table.
Each table area has 250 entries plus there are two further options available in F-Route Additional Entry 251 and F-Route Additional Entry 252.

Extension Table Area 1-4 - The table area number specifed by the 'Additional Data' in F-Route Dial Analyse Table.
Dial Analysis Table - Each Table Area has 250 entries.
Dial Data - The digits being processed by the system, enter up to 24 digits.
If a corresponding entry can not be found then the system will check F-Route Additional Dial: Fall-Over for the fall-over F-Route table number or F-Route Additional Dial: Next Table for the next Table Area to be searched.
F Route Table - Specify the 'F Route Selection' number to be used in F-Route Selection for Time Schedule if Time Schedule is used.
Specify the 'F Route Table' number in F-Route Table if Time Schedule is not used.

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F-Route Additional Dial: Fall-Over (1011)
The F-Route Additional Dial: Fall-Over is used if a corresponding entry is not found when searching the .

Extension Table Area - The table Extension Table Area that was searched in .
F-Route Table - The table number (1-500) that will be used for the fall-over calls.
Specify the 'F Route Selection' number to be used in F-Route Selection for Time Schedule if Time Schedule is used.
Specify the 'F Route Table' number in F Route Table if Time Schedules are not used.

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F-Route Additional Dial: Next Table (1012)
The F-Route Additional Dial: Next Table is used if a corresponding entry is not found when searching the F-Route Additional Dial Analyse Table.
This enables you to link the table areas together to build a larger table. There are four table areas each with 250 entries.
If you have also specifed a Fall-Over for the table area then the next table area specified here will not be searched.

Extension Table Area - The Extension Table Area that was searched in F-Route Additional Dial Analyse Table.
Next Table Area - The table number (1-4) that will be searched next.

Easy Edit Help

F-Route Selection for Time Schedule (1013)
This table is only used if F-Route time schedules are used.
Each day of the week can be split into F-Route mode numbers, there are 8 mode numbers available.

The mode number is used here to specify the F-Route table that should be used.

F-Route Selection - This is the number that is specified in any of the following tables:
F-Route Dial Analyse Table in the Additional Data for Service Type = F-Route Table.
F-Route Additional Dial Analyse Table in the F-Route Table number.
F-Route Additional Dial: Fall-Over in the F-Route Table number.
Mode 1-8 - The F-Route mode number specified for the time of day that the call is placed. Refer to F-Route Time Schedule Setup for the mode number.
F-Route Table - The F-Route table number (1-500) that will be used to translate the dialled digits.

Easy Edit Help

F-Route Table (1014)
The F-Route Table contains the translations and destination trunk group for all numbers routed via F-Route.
Each F-Route table has four translations (Detour 1 to 4) which are used in order 1 to 4, if all trunks are busy in the first Detour then the next Detour is used.

F-Route Table - The F-Route table number to be used depends on the number being processed and is defined by entries of the following tables:
F-Route Dial Analyse Table if time schedules are not used and less than 8 digits are being processed.
F-Route Additional Dial Analyse Table if time schedules are not used and more than 8 digits are being
F-Route Selection for for Time Schedule if time schedules are in use.
Detour 1-4 - The system will use the translations in Detour order 1 to 4. The next Detour is used when all trunks are busy in the trunk group defined by the current Detour.
Trunk Grp - The trunk group number used to route the call, trunk groups are defined in Trunk Groups.
Del Digits - The quantity of leading digits to be deleted.
Add Digits - The digits to be inserted before the leading digits (any digits to be deleted will be removed before the additional digits are inserted). Enter the Additional Digit table number, the digits to be inserted are defined in F-Route Add Digits.
Beep Tone - The system can give a beep tone to the caller, this would typically be used to indicate a specific Detour has been selected for the call, for example a high cost route.
Gain - ICM - Enter the gain table that will be used when an extension places a call that is routed via F-Route.
Gain - Tandem - Enter the gain table that will be used when an incoming trunk call is routed via F-Route (Trunk to Trunk connection).

Easy Edit Help

F-Route Add Digits (1015)
The Add Digits table defines the additional digits specified by the F-Route Table.

Additional Dial Table - The Additional dial table number specifed by 'Add Digits' in the F-Route Table.
Additional Data - The digits that will be inserted before the leading digits of the number being processed (the system will remove any digits specifed by the 'Del Digit' option in the F-Route Table before additional digits are inserted). Enter up to 24 digits.

Easy Edit Help

F-Route Time Schedule Setup (1016)
This table is only used if F-Route time schedules are used.
Each day of the week has an F-Route Schedule number assigned in F-Route Weekly Time Schedule or F-Route Holiday Time Schedule, there are 10 schedule numbers available.
Each schedule number is split into F-Route mode numbers, there are 8 mode numbers available.
The mode number is used in F-Route Selection for Time Schedule to specify the F-Route table that should be used.

Schedule 1-10 - The F-Route schedule number assigned in F-Route Weekly Time Schedule or F-Route Holiday Time Schedule.

Table 1-20 - There are 20 entries available for you to split up the schedule into F-Route mode numbers.
Start Time - The start time in 24 hour clock format for the chosen Mode.
End Time - The end time in 24 hour clock format for the chosen Mode.
Mode - The F-Route mode number 1-8 that the system will use between the start and end times.

Easy Edit Help

F-Route Weekly Time Schedule (1017)
This table is only used if F-Route time schedules are used.
Each day of the week is assigned an F-Route schedule number (1-10). The schedule number is used in F-Route Time Schedule Setup to determine the F-Route mode number to be used for the time that the call is placed.

Day of Week - The day of the week of the system clock. You can set the system time and date while connected to the system with PCPro, select the clock-face icon from the toolbar.
Schedule No - The F-Route schedule number that will be used.

Easy Edit Help

F-Route Holiday Time Schedule (1018)
This table is only used if F-Route time schedules are used.
The system will use the F-Route Schedule number defined for the day of week unless there is a Holiday schedule set.
Each day of the week is assigned an F-Route schedule number (1-10). The schedule number is used in F-Route Time Schedule Setup to determine the F-Route mode number to be used for the time that the call is placed.

Day - The date of the month. You can set the system time and date while connected to the system with PCPro, select the clock-face icon from the toolbar.
Month - Enter the schedule number (0-10) that the system will use for the date within any month. If you enter 0 the system will use the schedule set in F-Route Weekly Time Schedule.

Easy Edit Help

DISA Class of Service Options (1019)
The Class of Service number is assigned to each DISA user in DISA Toll & COS.

Class 1-15 - The DISA Class of Service number (not the DISA user number).
Erase 1 digit of input dial - Should the system ignore the first digit dialled by the incoming caller.
Trunk Route access - Access an outgoing trunk by dialling the trunk access code.
Trunk Group access - Access an outgoing trunk by dialling the trunk group service code (default=804).
Outgoing Common speed dial - Access an outgoing trunk by dialling the Common Abbreviated Dial service code (default=813).
Operator call - Access the operator extension by dialling the operator service code.
Internal Paging - Access the internal page groups by dialling the paging service code (default=801).
External Paging - Access the external page groups by dialling the paging service code (default=803).
Specific trunk access - Access an outgoing trunk by dialling the specific trunk access code (defauilt=805)
Forced Trunk disconnect - Release a trunk port by dialling the forced trunk disconnection service code (default=724).
Call Forward edit via DISA - Set/cancel an extension's call forward by dialling the call forward service code (defaults = 848, 843, 845, 846) followed by the extension number, 0 to cancel or 1 to set and then the target for the call forward.
Barge In - Barge in on a busy extension by dialling the barge-in service code (default=810) followed by the extension number. This feature will require the DUD/DISA Error message set.

Easy Edit Help

DUD/DISA Talkie (1020)
DUD is used as Automated Attendant and allows the caller to directly dial an extension or Pilot number.
DISA is also used as Automated Attendant but can be password protected and gives the caller access to certain system features and outgoing trunk ports.
The DUD/DISA Talkie is the greeting played to the incoming trunk caller.
A trunk can be set as DUD or DISA in Incoming Service Type, a DDI is routed to DUD or DISA in DDI Routing Table.
Note that for DDI calls you must assign the Talkie to all trunks within the trunk group routed to the DDI Table Area Target, this is because the Talkie is assigned to the trunk and not to the DDI.

Trunk No - The trunk port that the DUD/DISA call will arrive on.
Mode 1-8 - The night mode number.
Talkie Type - The device used to play the greeting to the incoming caller.
No Talkie = The incoming caller will hear a system dial tone, no greeting will be played. The dial tone is specifed by Service Tone 44-External Dial Tone (DUD/DISA Dial Tone) in Service Tones.
DSPDB = The caller will hear the DSPDB message number (1-48) specified in the Message No entry. The message is recorded with the Operation for VRS Message service code (default=716) assigned in 3 Digit Codes.
ACI = The caller will hear the audio input of the ACI port. Specify the ACI Group number in the Message No entry. The ACI ports are assigned to the ACI group in ACI Port Setup.
The ACI device will play the greeting message to the incoming caller and then the system will send DUD/DISA dial tone for the caller to dial their DTMF digits. The time is set by Guidance message by ACI Talkie Duration in System Timer for DUD/DISA Service.
SLT = The caller will be answered by an automatic answer device connected to an SLT port. Specify the Department Group number in the Message No entry. The SLT ports are assigned to the Department Group in Department Group Assignment.
The SLT device will play the greeting message to the incoming caller and then the system will send DUD/DISA dial tone for the caller to dial their DTMF digits. The time is set by Guidance message by Automatic Answering Telephone Duration in System Timer for DUD/DISA Service.
Message No - Specify the DSPDB message number, ACI Group number or Department Group number of the Talkie Type.

Easy Edit Help

Auto Attendant Fall Over (1021)
The system can route DUD/DISA calls to a fall over target for the following reasons:
-Incoming Caller dials a wrong number.
-Incoming Caller does not dial any number. The DUD/DISA Dial tone timer is set in System Timer for DUD/DISA.
-The target is busy.
-The target does not answer. The DUD/DISA No answer timer is set in System Timer for DUD/DISA.
Note that for DDI calls you must assign the fall over to all trunks within the trunk group routed to the DDI Table Area Target, this is because the fall over is assigned to the trunk and not to the DDI.

Mode 1-8 - The night mode number.
IRG Wrong or No Dial - The fall over target number if the caller dials the wrong number or does not dial any number.
IRG Busy or No Answer - The fall over target number if the target extension is busy or does not answer. Note - if the DUD/DISA Talkie is set to DSPDB in DUD/DISA Talkie and the VAU Fixed Messages are used in VRS Options the incoming caller will be prompted to dial a new number when the target extension is busy, the call will not fall over.

The IRG number can be any of the following:
0 = There is no fall over operation, the call will be disconnected.
1-25 = The IRG number. The IRG members are assigned in Incoming Ring Group Setup.
101 = DSPDB Voice Mail box to leave a message. The message will be left in the extension's mail box if it has a mail box assigned in , if not the mail box number is assigned in Voice Mail Automated Attendant Data Setup.
102 = External Voice Mail. Setup in Voice Mail External.

Due to the integration between various system components and applications possible, please remember that although the DSPDB (or "VAU") Auto Attendant is capable of using the Dial - in Conversion Table to transfer to Extension numbers, ACD Group numbers, F-Route numbers, and Station Group numbers (including NEC Connect / external Voicemail) - the destination available for scenarios which might occur such as wrong dial / no answer / busy is limited to a Ring Group.

This has always been the case and has worked well because the Ring Group typically contains the Operator extension(s) as a fallback, however please take this into account at the planning stage when integrating the DSPDB with various other components / applications provided by NEC Infrontia or third parties.

Easy Edit Help

DUD/DISA Error Message Setup (1022)
The error message will be played to the incoming caller if they dial a wrong number.
The DUD/DISA call must have been answered by the DSPDB message in DUD/DISA Talkie.
Note that for DDI calls you must assign the error message to all trunks within the trunk group routed to the DDI Table Area Target, this is because the error message is assigned to the trunk and not to the DDI.

Mode 1-8 - The night mode number.
DSPDB Message Number - Enter the DSPDB message number (1-48) to be played when the caller dials a wrong number.
You can replay the same message assigned as the DUD/DISA talkie if required. Also, the caller can redial a new target extension number during the error message.

Easy Edit Help

VRS Options (1023)
VRS (Voice response Service) features are provided by the DSPDB card.

VAU Fixed Message - Should the system use the fixed messages. For example call a busy extension and hear "Station 200 is busy, for call back dial 850". The language used for the fixed messages is selected in Default Menu Language in the Voice Mail section.
General Message Setup - Specify the DSPDB message number (1-48) that will be used to save the VRS General Message. The General message is recorded with the Record and Erase General Message service code (default=712) and listened to with the General Message Playback service code (default=711).
Transfer to IRG when VAU doesn't answer - Enter the IRG number that incoming trunk calls will ring if there are no DSPDB resources available when a DIL or DUD/DISA call routes to an extension with VRS Personal Greeting set. The system will wait until the VAU No answer time expires for a resource to become available. VRS Personal Greeting is set with the VAU/Off Premise Call Forwarding service code (default=713).
VAU No Answer - The duration the system will wait for a DSPDB resource to become available before the call steps on to the 'Transfer to IRG when VAU doesn't answer target'.
Message Re-send duration for Park and Page - The system will repeat the page announcement after this time. Park and Page is set with the VAU/Off Premise Call Forwarding service code (default=713).
VAU Waiting Message Operation - The IRG queue announcements can be started automatically at the First Waiting Message Start time or manually by a user. Manual operation is started by pressing the programmable function key set to 'Auto Ans with delay mesg' Function Key Programming.
VAU Waiting Message Repeat time - The interval between the queue announcements.
VRS Message for Caller ID Refuse - Enter the DSPDB message number (1-48) that will be played to the incoming caller when rejected by the Flexible Ringing by Caller ID feature.

Easy Edit Help

Pre-Amble Message Assignment (1024)
The Pre-Amble message is played to the incoming trunk caller when the call is answered by the user. When the Pre-Amble message is complete the callers hear two beeps and the call is connected. The Pre-Amble will be played for trunks set as Normal, DIL, DDI or DUD/DISA in Incoming Service Type.

Trunk No - The trunk port that the incoming call is presented on.

VAU Message Number Mode 1-8 - The DSPDB message number (1-48) played to the incoming caller for each night mode. Enter 0 to disable Pre-Amble.

Easy Edit Help

SMDR Service Options (1025)
There are two SMDR ports available on the system, you assign the extensions and trunks to one of the ports in the SMDR Group section.

Output port Type - Select the physical output for each SMDR port. You can use the same output for both ports. Both outputs are located on the EXIFU card.
Setup the ports in SMDR Port Settings.
Header Language - Select the language used for each page header of the SMDR output.
Omit Digits - The least significant (last digits) of the dialled number can be omitted from the output, an x will be inserted for each digit omitted.
Minimum Digits - Enter the minimum number of digits that must be dialled to line for the outgoing call to be output to the SMDR.
Minimum Duration - Enter the minimum duration (Seconds) for the call to be output. Enter 0 to output all calls, if you enter any other duration the system will not output outgoing calls via ISDN lines that are not answered (as the call duration will be 0 seconds).
Minumum Ring Duration - Enter the minimum duration (Seconds) an un-answered incoming call must ring for the call to be output. Enter 0 to output all calls.

Easy Edit Help

SMDR Port Settings (1026)
The Station Message Detail recorder (SMDR) is used to output call records from the system.
The records can be output by the LAN interface for Serial interface of the EXIFU card. (An EXIFU-A1 is required for LAN output, the EXIFU-B1 only has the serial interface).

TCP Port - Enter the TCP port number the system will use to allow the external device to connect via the LAN interface to receive SMDR records.
The IP Address to communicate with the system is setup in IP/EXIFU Network Setup.
COM port BAUD rate - Select the BAUD rate of the serial interface of the EXIFU card.
Other settings of the serial interface are fixed:
Data bits = 8
Parity = None
Stop bits = 1
Flow Control = None

Easy Edit Help

SMDR Output Options (1027)

Toll Restriction Call - Should outgoing calls that have been restricted be output.
PBX Call - For trunks connected behind another PBX the system can output all calls or just those that have the PBX trunk access digit.
Trunk Name/Number - The system can output the trunk name from Trunk Basic Setup or the 3 digit trunk port number.
Daily Summary - The system can output a summary at Mid-night each day.
Weekly Summary - The system can output a summary at Mid-night every Saturday.
Monthly Summary - The system can output a summary at Mid-night on the last day of each month.
Cost Charging - The system can inlude the call charge information for outgoing calls (if the service is available from the network).
Incoming Calls - Should incoming calls be output.
Extension Number/Name - The system can output the extension name or number from Extension Basic Setup.
All Line Busy - The system can output the ALB event when all lines are in use for any trunk group.

Walking Toll Restriction Table - The system can output the Walking Toll Entry Number from Walking Toll Passwords for a call made using this feature.
DID Table Name - The system can output the DDI name from in place of the line name/number for incoming DDI calls.
CLI Output when DDI to Trunk - The system can output the received Caller ID in the STATION field of the outgoing call when a DDI is routed back out on another trunk.
Date Output - The system can output the date (month and date dependent on SMDR Date Format) and trunk port number in the LINE field for each record. If this option is set to output then the 'Trunk Name/Number' will be ignored.
CLI/DID Number Switching - The system can output either the CLI number, DDI Number or CLI name in the DIALLED NO./CLI field.
Trunk Name/Received Number - The system can output either the DDI number, line name/number or both in the LINE field. When set to both the system will output TTT DDDDDD where T is the trunk port number and D is the DDI number.
Display First or Last Digit - The system can display either the first 15 or last 15 dialled digits for outgoing calls.
External CFW Information - The system can output either the extension identification or incoming trunk information for calls that are forwarded off premise by an extension. This setting will effect the outgoing call record.

Easy Edit Help

SMDR Date Format (1028)
Select the date format for all SMDR records.

Easy Edit Help

SMDR Group Assignment for Trunks (1029)
Each trunk group is assigned to an SMDR port number. This SMDR port will be used for incoming calls to any trunk within the group. Outgoing trunk calls will use the setting for the extension in SMDR Group Assignment for Department Group.

SMDR Port - Enter the SMDR port number.

Easy Edit Help

SMDR Group Assignment for Department Groups (1030)
Each Department group is assigned to an SMDR port number. This SMDR port will be used for outgoing calls to any trunk.
Incoming trunk calls will use the setting for the trunk in SMDR Group Assignment for Trunks.

SMDR Port - Enter the SMDR port number.

Easy Edit Help

Account Code Setup (1032)
Account codes are used to identify trunks calls by the user entering the code before or during the call. Account codes are output to the SMDR. The account code is dialled by the user with the digit * as start and end identifiers.
Account codes can be enabled/disabled for each trunk in Trunk Basic Data Setup.

Class of Service - The Class of Service assigned to the extension in Class of Service per Night Mode.
Account Code Mode - Select the operation of the user.
Disabled = Account Codes are not allowed.
Optional = The user can enter an account code of up to 16 digits long, the account code is not verified.
Required but not verified = An account code must be entered, it is not verifed so the user may enter any account code.
Required and verified = An account code must be entered, it is verifed so the user must enter an account code specified in the

Verified Account Code Table. The verified account code can be from 3 to 16 digits long.
Account Code Entry for Incoming Calls - An account code can be entered after answering an incoming call. The account code is not verified for incoming calls.
Display/Hide Account Codes - The account code can be hidden on the keyphone display, a * character will be displayed. The account code will be output to the SMDR when hidden on the keyphone display.

Easy Edit Help

Verified Account Code Table (1033)
The system can verify the account code with an entry in this table, if the account code does not correspond the call will be disconnected.

Verified Account Code - Enter the account codes between 3 and 16 digits long. Do not use the digit * as this is used as the start and end identifiers when the user enters the account code.
You can also enter a 'wildcard' entry with the @ character (for example @234 will allow a user to enter 0234 to #234).

Easy Edit Help

Paging Options (1034)
Extensions are placed into internal paging groups in Internal Paging Group.

Internal All Call Paging Name - Enter the name that will be shown on the keyphone display when placing a call to the Internal All Call zone (default service code = 801+0)
Privacy Release Time - The duration the system will allow users to join the Meet Me Conference after the paging call is placed.

Easy Edit Help

Internal Paging Options (1035)
Extensions are placed into internal paging groups in Internal Paging Group.

Paging Group Name - Enter the name that will be shown on the keyphone display when placing a call to the Internal paging zone (default service code = 801+zone).
Paging Splash Tone Type - Select the splash tone that will be used at the keyphones being paged when a page call is placed.

Easy Edit Help

ACI Port Setup (1036)
Audio Communication Interfaces (ACI ports) are provided by the PGDU card. Refer to PGDU Setup for the PGDU port type (ports 3 and 4 of each PGDU card can be assigned as ACI ports) and logical port number.

ACI Extension Number - An ACI port must have an extension number assigned. Note - This can be used to place a call to the ACI port for testing.
ACI Port Type - Select the type of operation for the ACI port.
Not Set = The ACI port is not in use.
Input = The ACI port will be used as an input to the system, the system will not allow audio to the external device. Select this type if the ACI is used for queue announcements.

Input & Output = The ACI port will be used an both an input and output to the system, the system will allow two-way audio to the external device. Select this type if the ACI is used for conversation recording.
BGM(Input) = The ACI port will be used as the Background music input for the system. Only one BGM input is supported on the system.
EXMOH(Input) = The ACI port will be used as an external music on hold input. The ACI port can be assigned to trunks in Outgoing MOH Source or to incoming DDI calls in DDI Routing Table.
Group No - The ACI ports are placed into ACI Groups, these are used when there are multiple connections to the same external device for example for conversation recording or DUD/DISA greetings. The ACI Group is assigned a pilot number in ACI Group Pilot Numbering.
Priority - The ports will be accessed in priority order when the ACI group is used.

Easy Edit Help

ACI Group Pilot Numbering (1037)

ACI Group Pilot No - Enter the pilot number used to identify the ACI Group. ACI ports are assigned to the group in ACI Port Setup.

Easy Edit Help

Daylight Saving Setup (1038)
The daylight saving setup will adjust the system time by one hour to coincide with the daylight saving times of the local time zone.

Daylight Saving - Turn the feature on/off.
Switching Time - The time of day the system will adjust the clock.
Start Month - The month that the system will advance the clock by one hour.
Start Week - The week within the month the system will advance the clock. Select a fixed week by entering 1 to 5 for the week within the month or enter 0 to have the system automatically detect the last week of the chosen month.
Start Day - The day within the week the system will advance the clock.
End Month - The month that the system will revert to the standard time.
End Week - The week within the month that the system will revert to the standard time. Select a fixed week by entering 1 to 5 for the week within the month or enter 0 to have the system automatically detect the last week of the chosen month.
End Day - The day within the week that the system will revert to the standard time.

Easy Edit Help

DSP Resource Selection (1039)
Select the type of call the resources of the VOIPU card can be used for. You can select the resource for each slot within the system, check the slots used by VOIPU cards in Card Configuration.
It is recommended that you set these to 'Both IP Extensions/Trunks' as this mode will make the resource available to all call types including IP System Feature Networking.
Both IP Extensions/Trunks = Available to all call types including IP System Feature Networking.
IP Extensions = Available to calls between IP extensions.
IP Trunks = Available to calls via an IP trunk.
Network = Available to IP System Feature Networking only.

Easy Edit Help

System Information Setup (1040)
Setup the system for connection to an external SIP server.

Domain Name - The domain that the system belongs to. If there is no connection to an external SIP provider this item can be set to an valid value.
Host Name - The Host name that will be combined with the Domain name. If there is no connection to an external SIP provider this item can be set to an valid value.
Transport Protocol - This must be set to UDP.
User ID - The SIP UserID. If there is no connection to an external SIP provider this item can be set to an valid value.
Domain Assignment - The SIP messages can use the system IP address or the Domain name for addressing.

Easy Edit Help

SIP Server Setup (1041)
These settings are required when the system will register to an external SIP server.

Default Proxy (transmit) - Default = Off. Set to On for connection to an external SIP Proxy server.

Default Proxy (receive) - Default = Off. Set to On for connection to an external SIP Proxy server.
Default Proxy IP Address - Default = 0.0.0.0 enter the IP address of the external SIP Proxy server.
Default Proxy Port No - Default = 5060, the UDP port used by the SIP Proxy server.
Registrar Mode - Default = None. If a SIP Proxy server is used this should be set to None. If a SIP Carrier is in use this should be set to Manual.
Registrar IP Address - Default = 0.0.0.0 enter the IP address of the SIP server. Alternatively use the Domain Name Server (DNS) settings below.
Registrar Port Number - Default = 5060.
DNS Server Mode - Default = Off. Set to On if you want to use DNS to register to the SIP server.
DNS Server IP Address - Default = 0.0.0.0 enter the IP adress of the DNS server.
DNS Server Port Number - Default = 53.
Registrar Domain Name - Enter the fully qualified domain name (FQDN) for the SIP registration server (example sipserver.company.com)
Domain name - Enter the domain name of the SIP registration server (example company.com)
Host Name - Enter the Host name of the SIP registration server (example sipserver)
SIP Carrier Choice - Default = 0. This item alters the format of outbound CLIP.

For Networking Mode, this item should be set to 2

For Carrier Mode, please refer to the SIP Certificate of Compatibility

Registration Expiry Time - Default = 3600 Seconds. Enter the expiry time before the system will re-register to the SIP server.

Easy Edit Help

Authentication Information (1042)
This information is provided by the SIP Server maintainer.
This allows the entry of one registration, if more than one is required additional entries are made in Trunk Registration Information.

Easy Edit Help

Registrar/Proxy Setup (1043)
Setup the registration for SIP extensions.

Registration Expiry Time - Default = 120
Authentication Mode - Default = Disabled. If enabled the SIP extension must register with the 'Authentication Password' entered in IP Extension Setup.
Registrar/Proxy Domain Name - Default = none specified. The SIP extensions can register with the Domain name or IP address.

Registrar/Proxy Host Name - Default = none specified. The SIP extensions can register with the Domain name+Host Name or IP address.

Easy Edit Help

Remote Destinations (1045)
Specify the IP address of the remote systems and the number dialled to reach the system.
This table is used for SIP and H.323 Networks.

System Interconnection - Set to Yes to enable the entry of the remote IP address and dial number.
IP Address - Enter the IP address of the remote system.
Dial Number - Enter the leading digit(s) of the number dialled to reach the remote system. For example if extensions on the remote system are numbered 400 to 450 then enter 4 as the Dial Number. Use F-Route to route numbers 400 through to 450 to the trunk group number of the IP trunks.

Easy Edit Help

Calling Party number for Trunk (1046)
The Trunk Caller ID is used for outgoing SIP and H.323 calls. If the extension also has caller ID set in Calling Party Number for Extension it will be used in preference to the trunk caller ID (Extension caller ID has seperate entries for SIP or H.323 calls).

Calling Party No - Enter the calling party number for calls made via the trunk.

Easy Edit Help

Calling Party number for Extension (1047)
The Extension Caller ID is used for outgoing SIP calls. If the extension has caller ID set it will be used in preference to the trunk caller ID.

Calling Party No - Enter the calling party number for calls made via a SIP trunk.

Easy Edit Help

IP Extension Setup (1048)

Terminal Type - Shows the type of extension assigned.
IP Address - Shows the IP Address of the IP phone.
Authentication Password - Enter the password that must be used when the IP phone is installed. The password will not be displayed. The password is only required if enabled in 'Authentication Mode' in Registrar/Proxy Setup
Calling Party Display Info - Select the type of caller ID display.
IP Duplication Allowed Group - The system can accept more than one registration for the same IP address. This is required when the SIP device has multiple extension ports.

Easy Edit Help

Basic Information Setup Setup (1049)
Setup the codec used for H.323 networking.

Audio Capability Priority - Select the codec type the system will use for all H.323 calls. Setup the relevent codec type (G.711, G.729 or G.723).
G.711 Audio Frame Number - Default = 3
G.711 Voice Activity Detection Time - Default = Disable
G.711 Type - Default = A-Law
G.729 Audio Frame Number - Default = 3
G.729 Voice Activity Detection Time - Default = Disable
G.729 Jitter Bufer (min) - Default = 20 (mSeconds)
G.729 Jitter Buffer (Type) - Default = 40 (mSeconds)
G.729 Jitter Buffer (max) - Default = 60 (mSeconds)
G.723 Audio Frame Number - Default = 1
G.723 Voice Activity Detection Time - Default = Disable
Audio Delayed Jitter - Default = 60 (mSecond)
Jitter Buffer Mode - Default = Adaptive Immediately
G.711 Jitter Bufer (min) - Default = 30 (mSeconds)
G.711 Jitter Buffer (Type) - Default = 60 (mSeconds)
G.711 Jitter Buffer (max) - Default = 120 (mSeconds)
G.723 Jitter Bufer (min) - Default = 30 (mSeconds)
G.723 Jitter Buffer (Type) - Default = 60 (mSeconds)
G.723 Jitter Buffer (max) - Default = 120 (mSeconds)
VAD Threshold - Default = 0dB (-30dBm)
Idle Noise Level - Default = 7000
TX Gain - Default = 10 (-4dBm). Enter 0 to 28 (-14dBm to +14dBm)
RX Gain - Default = 10 (-4dBm). Enter 0 to 28 (-14dBm to +14dBm)
Band Limitation Mode - Default = Disabled
Band Max - Default = 0
FAX Relay Function - Default = Disabled
Echo Canceller Config Type - Default = Auto
DTMF Relay Mode - Default = VOIPU. Select from the list:
VOIPU = Specified by the VOIPU card.
RFC2833 = DTMF tones are sent via RTP protocol.
H.245 = DTMF tones are sent via H.245 specification.
Disable = DTMF tones are sent within the speech path.
Fast Start Mode - Default = Enabled

Easy Edit Help

VOIPU Card Setup (1052)
Each VOIPU card must have an appropriate fixed IP Address assigned. The VOIPU card provides the speech resources for VOIP calls to/from a non-IP extension.

Slot No - The slot number within the system.
IP Address - Enter the IP Address of the VOIPU card.
LAN Setting - Select to type of connection for the LAN interface of the VOIPU card

Easy Edit Help

VOIPU Setup (1053)

RTP Port - Default = 10020.
RTCP Port - Default = 10021.
H.245 Port - Default = 10100.
DTMF Behaviour - Select the method used to send DTMF digits for H.323 trunks (Default = Out of Band). This setting is used when 'VOIPU' is selected as the DTMF Relay Mode in H.323 Networking Basic Information Setup.
Relay Disabled = Send DTMF tones within the speech packets.
In Band = In band DTMF Relay between VOIPU cards (This is the recommended setting).
Out of Band = Does not pass DTMF tones as speech packets.
Ready Ready Answer Port - Default = 4000.
ICMP redirect - Determines whether the VOIPU cards should accept (and respond to) ICMP Redirect messages received from other network equipment. May cause problems for VOIP and System Feature Networking if this option is enabled.

Easy Edit Help

Layer 2 QoS and VLAN (1054)

VLAN Mode - Enable/Disable VLAN tagging. This will require a 'Managed Switch' capable of VLAN operation.
VLAN ID - Enter the VLAN ID, this is not always required as it is usually set on the port of the HUB/Switch.
Priority - Enter the priority number that the system will mark each frame with. Enter 0 to 7, priority 0 is lowest, 7 is highest.

Easy Edit Help

Layer 3 QoS (1055)
There are Layer 3 QoS settings for the following types of frame: DRS, Protims, H.323, RTP/RTCP and SIP.
The system must be powered off and then on for changes to take effect.

ToS Mode - Select the Quality of Service (QoS) type.
Priority (Diffserve) - Enter the priority if DiffServe is selected in 'ToS Mode'. Enter 0 to 63.
IP Precedence Priority - Enter the priority for IP Precedence (Class Selector value). Enter 0 to 7, voice is commonly set to 5 or 6.
IP Precedence Delay - Select the delay for IP Precedence, this will require the Network to act upon this value for it take effect.
IP Precedence Throughput - Select the Throughput for IP Precedence, this will require the Network to act upon this value for it take effect.
IP Precedence Reliability - Select the Reliability for IP Precedence, this will require the Network to act upon this value for it take effect.
IP Precedence Cost - Select the Cost for IP Precedence, this will require the Network to act upon this value for it take effect.

Easy Edit Help

H.323 Codec Setup (1056)
G.711 Audio Frame Number - Default = 3
G.711 Voice Activity Detection Time - Default = Disable
G.711 Type - Default = A-Law
G.711 Jitter Bufer (min) - Default = 30 (mSeconds)
G.711 Jitter Buffer (Type) - Default = 60 (mSeconds)
G.711 Jitter Buffer (max) - Default = 120 (mSeconds)
G.729 Audio Frame Number - Default = 3
G.729 Voice Activity Detection Time - Default = Disable
G.729 Jitter Bufer (min) - Default = 30 (mSeconds)
G.729 Jitter Buffer (Type) - Default = 60 (mSeconds)
G.729 Jitter Buffer (max) - Default = 120 (mSeconds)
G.723 Audio Frame Number - Default = 1
G.723 Voice Activity Detection Time - Default = Disable

G.723 Jitter Bufer (min) - Default = 30 (mSeconds)
G.723 Jitter Buffer (Type) - Default = 60 (mSeconds)
G.723 Jitter Buffer (max) - Default = 120 (mSeconds)
Jitter Buffer Mode - Default = Adaptive Immediately
VAD Threshold - Default = 20 (0dB). Enter 0 to 30 (-19dB to +10dB)
Idle Noise Level - Default = 7000
TX Gain - Default = 10 (-4dBm). Enter 0 to 28 (-14dBm to +14dBm)
RX Gain - Default = 10 (-4dBm). Enter 0 to 28 (-14dBm to +14dBm)
DTMF Relay Mode - Default = VOIPU. Select from the list:
Disable = Send DTMF tones within the speech path.
RFC2833 = Transmits DTMF via RTP protocol.
VOIPU = DTMF relay is determined by the setting of 'DTMF Behaviour' for the VOIPU card in VOIPU Setup.

Easy Edit Help

SIP Networking Codec Setup (1057)
Setup the codec used for SIP networking.

Audio Capability Priority - Select the codec type the system will use for all SIP calls. Setup the relevent codec type (G.711, G.729 or G.723).
G.711 Audio Frame Number - Default = 3
G.711 Voice Activity Detection Time - Default = Disable
G.711 Type - Default = A-Law
G.711 Jitter Bufer (min) - Default = 20 (mSeconds)
G.711 Jitter Buffer (Type) - Default = 40 (mSeconds)
G.711 Jitter Buffer (max) - Default = 60 (mSeconds)
G.729 Audio Frame Number - Default = 3
G.729 Voice Activity Detection Time - Default = Disable
G.729 Jitter Bufer (min) - Default = 20 (mSeconds)
G.729 Jitter Buffer (Type) - Default = 40 (mSeconds)
G.729 Jitter Buffer (max) - Default = 60 (mSeconds)
G.723 Audio Frame Number - Default = 1
G.723 Voice Activity Detection Time - Default = Disable
G.723 Jitter Bufer (min) - Default = 30 (mSeconds)
G.723 Jitter Buffer (Type) - Default = 60 (mSeconds)
G.723 Jitter Buffer (max) - Default = 120 (mSeconds)
Jitter Buffer Mode - Default = Adaptive Immediately
VAD Threshold - Default = 20 (0dB). Enter 0 to 30 (-19dB to +10dB)
Idle Noise Level - Default = 7000
TX Gain - Default = 10 (-4dBm). Enter 0 to 28 (-14dBm to +14dBm)
RX Gain - Default = 10 (-4dBm). Enter 0 to 28 (-14dBm to +14dBm)
DTMF Payload Number - Default = 110 (enter 96 to 127)
DTMF Relay Mode - Default = Disable. Select from the list:
Disable = DTMF tones are sent within the speech path.
RFC2833 = DTMF tones are sent via RTP protocol.

Easy Edit Help

Codec Setup (1063)
Setup the codec used for SIP extensions.

Peer To Peer Mode - Switches SIP Extension Peer to Peer mode on or off. This allows RTP (speech) to be sent directly between SIP extensions, which reduces the number of DSP resources required. Please refer to the SIP Extension Features section of the online Features Manual for further information.
Audio Capability Priority - Select the codec type the system will use for all SIP extension calls. Setup the relevent codec type (G.711, G.729 or G.723).
G.711 Audio Frame Number - Default = 3
G.711 Voice Activity Detection Time - Default = Disable

G.711 Type - Default = A-Law
G.711 Jitter Bufer (min) - Default = 30 (mSeconds)
G.711 Jitter Buffer (Type) - Default = 60 (mSeconds)
G.711 Jitter Buffer (max) - Default = 120 (mSeconds)
G.729 Audio Frame Number - Default = 3
G.729 Voice Activity Detection Time - Default = Disable
G.729 Jitter Bufer (min) - Default = 30 (mSeconds)
G.729 Jitter Buffer (Type) - Default = 60 (mSeconds)
G.729 Jitter Buffer (max) - Default = 120 (mSeconds)
G.723 Audio Frame Number - Default = 1
G.723 Voice Activity Detection Time - Default = Disable
G.723 Jitter Bufer (min) - Default = 30 (mSeconds)
G.723 Jitter Buffer (Type) - Default = 60 (mSeconds)
G.723 Jitter Buffer (max) - Default = 120 (mSeconds)
Jitter Buffer Mode - Default = Adaptive Immediately
VAD Threshold - Default = 20 (0dB). Enter 0 to 30 (-19dB to +10dB)
Idle Noise Level - Default = 7000
TX Gain - Default = 10 (-4dBm). Enter 0 to 28 (-14dBm to +14dBm)
RX Gain - Default = 10 (-4dBm). Enter 0 to 28 (-14dBm to +14dBm)
DTMF Payload Number - Default = 96 (enter 96 to 127)

Easy Edit Help

Basic Information Setup (1064)
Setup the SIP extension information.

Registrar/Proxy Port - Default = 5070
Session Timer Value - Default = 180
Minimum Session Timer Value - Default =180
Called Party Info - Default = Request URI
Expire Value of Invite - Default = 60

Easy Edit Help

VoIPU Setup (1065)
The VoIPU card provides the speech conversion for Voice over IP calls.

Trunk Type - Select either H.323 or SIP.
Ensure you assign all VoIPU trunk ports into a single trunk group in the Trunk Group screen. Do not mix other trunk types in the group.
Do not use trunk group 1 for the VoIPU group as this is the default group for any new trunks added to the system.
Outgoing calls are routed to the VoIPU trunk group with F-Route. Incoming calls are usually routed as DDI calls so set all VoIPU trunks as DDI in the Incoming Service Type screen.
The remaining VoIPU setup can be found in the VoIP section of Easy Edit.

Easy Edit Help

Trunk Registration Information (1070)
Specify additional SIP Server registrations.

Registration - Enable/disable the registration ID entry.
User ID - The user name or number supplied by the SIP server administrator.
User Authentication ID - The user authentication name or number supplied by the SIP server administrator. This is usually the same

as the User ID.
Authentication Password - The password supplied by the SIP server administrator.

Easy Edit Help

Gatekeeper Setup (1072)

Gatekeeper Mode - Select the mode from the list:
No Gatekeeper = Gatekeeper is not used.
Search Automatically = The system will use the setting og 'Preferred Gatekeeper' to search for the gatekeeper.
Manual defined = The system will use the setting of 'Gatekeepr IP address' for the gatekeeper.
Gatekeeper IP Address - Specify the IP address of the Gatekeeper when 'Manual Defined' is selected. To use the system's own internal gatekeeper enter the IP Address of the NTCPU set in NTCPU IP Network Setup.
Preferred Gatekeeper - Specify the Gatekeeper Request (GRQ) when 'Manual Defined' is selected.
Alias Address - Enter the Alias Address fro the system, this will be passed to the Gatekeeper to identify the numbers that can be received by the system.

Easy Edit Help

Calling Party number for Extension (1073)
The Extension Caller ID is used for outgoing H.323 calls. If the extension has caller ID set it will be used in preference to the trunk caller ID.

Calling Party No - Enter the calling party number for calls made via an H.323 trunk.

Easy Edit Help

Department Group Options (1206)
Pilot Number - Assign a number that can be used to call the Department Group. You can also route DDI calls and Auto Attendant calls to the pilot number.
Group Name - When an internal extension is queued at a busy department group the name will be displayed at the system phone's display. The user will see: WAITING (group name).
Calling Cycle
With Priority Routing, an incoming call routes to the highest priority extensions first. Lower priority extensions ring only if all higher priority extensions are busy.
With Circular Routing, each call rings a new extension.
Routing when Busy
If a user directly dials a busy extension within a Department Group, the system can optionally route the call to the first available group member.
Hunting Mode
Ringing calls can step around members of the group once only and stop at the last member or repeatedly search for a free member.
Extension Group All Ring mode operation
All idle members of the department group can ring simultaneously for internal and outside calls to the pilot number. Calls do not cycle between group members. Simultaneous ringing can be automatic (all members ring when the call is placed to the pilot number) or manual (the call to the pilot number will step around each member of the group until the caller selects simultaneous ring mode).
Note, when automatic is selected the operation of Enhanced Hunting Type is effected. Calls to the pilot number will not receive ringback tone if all members are busy, they will receive busy tone and will not wait for a member to become free.
Extension Group Withdraw mode - This option has no operation.
Call Recall Restriction for STG
Calls transferred to the pilot number that do not get answered will recall at the extension that transferred the call. When the transfer recall is restricted the transferred call will ring the group until it is answered or the caller clears down.

Maximum Queuing Number of Extension Group
The quantity of ISDN DDI trunk calls queuing at a busy Department Group can be limited when all members of the group are busy. When the queue limit is reached further calls will receive busy indication. The number of calls in the queue can be set by Department Options. If the queue limit is set to 0 then no ISDN DDI calls will be queued. Note that the queue limit will be ignored if enhanced hunting is set or you have enabled Busy step on for the DDI call in Dial In Conversion Table Data Setup.
Extension Group Call No Answer time
An un-answered call ringing at a member of a department group will step on to the next available member after a preset time. If the timer is set to 0 the step on will be disabled.
Enhanced Hunting
Hunting sets the conditions under which calls to a Department Group pilot number will cycle through the members of the group.
No Hunting
A call to the pilot number will hunt past a busy group member to the first available extension.
The call will continue to ring the extension until it is answered or the calling party hangs up, it will also step on to the next available member after the No Answer Step On Time.
Calls to the group when all members are busy will receive busy tone.
Busy
A call to the pilot number will ring the first idle member of a Department group, following the priority or circular routing. The call will continue to ring the extension until it is answered or the calling party hangs up, it will not step on to the next available member. If the Department Group has Priority Routing enabled, and the highest priority member is busy, the call will step on to the next available member.
Calls to the group when all members are busy will receive ring back tone and wait for a member to become free.
No Answer A call to the pilot number will ring the first idle member of a Department group, following the priority or circular routing. The call will continue to ring the extension until it is answered or the calling party hangs up, it will also step on to the next available member after the No Answer Step On Time. If the Department Group has Priority Routing enabled, and the highest priority member is busy, the call will wait for the extension to become free and will not step on to the next available member.
Calls to the group when all members are busy will receive ring back tone and wait for a member to become free.
Busy & No Answer
A call to the pilot number will ring the first idle member of a Department group, following the priority or circular routing. The call will continue to ring the extension until it is answered or the calling party hangs up, it will also step on to the next available member after the No Answer Step On Time.
If the Department Group has Priority Routing enabled, and the highest priority member is busy, the call will step on to the next available member and continue to step on after the
No Answer Step On Time.
Calls to the group when all members are busy will receive ring back tone and wait for a member to become free.
Note that enhanced hunting will effect the operation of simultaneous ringing and maximum queue limit.

Easy Edit Help

Secondary Department Groups (1207)
Allows an extension to be a member of more than one Department Group. Each group can have up to 16 secondary members.
Note. When a Department Group has a secondary member then All Ring mode is not available for the group.

Member number - These are the secondary members for each Department Group. In the Member number column enter the extension number (not the port number) of the user that you want to add as a secondary member to the Department Group.
Group number - These are the Department Groups.
Order - The ring order number of the secondary member.

Easy Edit Help

Department Group - Departmental Call Restriction (1208)
Displays the settings for Departmental Call Resriction.

Department Group - Displays the department group nember.
Pilot Number - displays the department group pilot number for the selected department group.
Group Name - displays the name department group.
Restrict group - Entry 1-8 - Enter the department group number to which calls from the selected department group are

restricted.
This will prevent members of the selected department group from making calls to the members of the restricted group.

Easy Edit Help

Department Group Timers (1209)
Times are in Seconds.

Ring Inward to busy extension group recall time - Set the duration that a call will queue after being transferred to a busy Department Group. After this time the call will ring back to the extension that performed the transfer.
Extension Group delayed transfer time - Set the duration that a call will ring at the Department group before it will be forwarded.

Easy Edit Help

Auto Attendant Single Digit Operation (13031)
Single digit options are available to the incoming DUD/DISA caller if the DSPDB message is assigned as the DUD/DISA Talkie.
There are seperate single digit options for each of the 48 DSPDB messages.
There are two possible destinations for the single digit:
Another DSPDB Message = The caller will hear the next DSPDB message, this allows the system to give multi-level greetings.
A target extension number = The caller will be routed to the target.

Message No - The DSPDB message number that the incoming caller is listening to when the single digit is dialled.
Received MF Digit - The incoming caller can press any DTMF digit (1-0, * and #). Note - If you use a single digit that is also the same as an extension number the incoming caller will not be able to dial the extension numbers (for example - If there is a single digit translation for digit 2 the incoming caller will not be able to dial any extension numbers beginning with digit 2).
Att Msg - This option specifies the next DSPDB message number (1-48) that will be played to the incoming caller when the single digit is pressed. Set this entry to 0 if you want the call to route to a destination extension number.
Dest No - This option specifies the destination extension number. You must set the Att Msg to 0 for the destination extension number to operate. You can also enter a Virtual extension number or Department Group pilot number.

Due to the integration between various system components and applications possible, please remember that although the DSPDB (or "VAU") Auto Attendant is capable of using the Dial - in Conversion Table to transfer to Extension numbers, ACD Group numbers, F-Route numbers, and Station Group numbers (including NEC Connect / external Voicemail) - the destination available for scenarios which might occur such as wrong dial / no answer / busy is limited to a Ring Group.
This has always been the case and has worked well because the Ring Group typically contains the Operator extension(s) as a fallback, however please take this into account at the planning stage when integrating the DSPDB with various other components / applications provided by NEC Infrontia or third parties.

Easy Edit Help

Extension Basic Setup (1326)
The extension properties basic setup screen has items that are common to all station types.

Extension Number - Enter the extension number of the telephone. You can not duplicate any numbers on the system.
Extension Name - Enter the name of the telephone user. The character set available is shown in Character Set.**Auto Trunk Seize** - Each time the user goes off hook an idle trunk will be seized. The trunk is defined as if the user had dialled the trunk access code. See the Trunk group routing for extensions screen for the outgoing line access.
Pick Up Grp - Place the station into one of the pick up groups. These are used to allow other users to pick up calls ringing at the phone.
Page Grp - Place the station into one of the internal paging groups (zones). You can place any station into a page group but only keyphones will be paged.

All Page group - Should the station be included in the All Call Page group. The station must also be a member of any Page Group for it to be included in the All Call Page group.
Park Hold Grp - Park hold groups allow stations to retrieve calls parked by other stations in the same group.
Department Grp - Department groups have a pilot number to enable users to call any available phone in the department. The pilot number is assigned in the Department Options screen.
Dept ring order - Chooses the order that you want the stations to ring when calls are made to the Department Group pilot number. You can enter any order number in the range 1-99. if you duplicate any order number then the station's port number will be used as the ring order (lowest port first).
Outgoing Disable on Incoming Line - Enable/disable the detection of DTMF digits dialled on 'incoming' analogue trunk calls. To enable the feature set this option to Supervise dial detection. Further settings are made in the Outgoing Call options.

Easy Edit Help

Internal Paging Group
An internal paging call is broadcast out of the loudspeakers of all available keyphones in the group. There is also an option to place the phones into the All Call Paging group.

Page Grp - The paging group number (zone number) the extension is a member of.
All Page Group - Should the extension also be included in the All Call paging group. The extension must be a member of any paging group to be included in the All Call Paging group.

Easy Edit Help

Park Hold Groups (13262)
Park hold groups allow stations to retrieve calls parked by other stations in the same group.
Park Hold Grp - The Park Hold group that the phone is a member of.

Easy Edit Help

Department Group Assignment (13263)
Department Grp - Assign the extension to one of the Department Groups. The extension can be a member of more than one group if you use Secondary Department Groups.
Dept ring order - Chooses the order that you want the stations to ring when calls are made to the Department Group pilot number. You can enter any order number in the range 1-99. if you duplicate any order number then the station's port number will be used as the ring order (ring oder = low to high port numbers).

Easy Edit Help

Class of Service Per Night Mode (1327)
There are 8 night modes available on the system. A station can be assigned a Class of Service (COS) for each mode.
The COS will define which features the station has access to; Paging, Call Forward etc.
You can define each COS in the Class of Service Options screen.

COS for Mode 1-8 - For each of the Night Modes assign one of the 15 Classes of Service.

Easy Edit Help

Toll Restriction Per Night Mode (1328)
There are 8 night modes available on the system. A station can be assigned a Toll restriction class for each mode.
The class will define which numbers the station can dial for outgoing trunk calls.
You can define each Toll class in the Toll Restriction screen.
You can also enable/disable toll restriction for each trunk in the Trunk Basic Data Setup screen.

Class for Mode 1-8 - For each of the Night Modes assign one of the 15 classes.

Easy Edit Help

Function Key Programming (1342)
Allows you to set the function of the programmable Function Keys of each keyphone. If you want all keyphones to have the same Function Key setup then use the Function Key Template.
You can not duplicate the same feature or additional data on the Function Keys so if you want to 'move' a feature to another key you must first set the key to Not Used and then add the feature to the new key.
Some features will require Additional Data, for example:
CO Key - Additional data defines the trunk port number
Trunk Group Access Key - Additional data defines the trunk group
Virtual Extension Key - Additional data defines the Virtual extension number (or any other station number)
Park Key - Additional data defines the Park orbit
Loop Key - Additional data defines the type 1-3
DSS/One Touch Key - Additional data defines the station number
Day/Night Mode Key - Additional data defines the night mode 0-8 (0 will scroll through all available modes each time the key is pressed)
Call Redirect Key - Additional data defines the destination number; station number or voice mail service code
Mail Box (DSPDB) Key - Additional data defines the mail box number
Voice mail service (DSPDB) Key - Additional data defines the operation during message playback (0=skip forward , 1=skip backwards)
Recording Service (DSPDB) Key - Additional data defines the mail box number to record into
Auto Attendant (DSPDB) Key - Additional data defines the station number for which the key will set call forward to voice mail
Voice Mail (In-skin VM) Key - Additional data defines the mail box number (also used for external VM systems)
Conv Record (In-skin VM) Key - Additional data defines the mail box number to record into(also used for external VM systems)
Auto Attendant (In-skin VM) Key - Additional data defines the station number for which the key will set call forward to voice mail (also used for external VM systems)

You can assign Function Keys to any station regardless of the number of Function Keys are station type.

Key - The Function Key on the keyphone
Feature - Select the feature from the list. Some features can not be duplicated.
Data - The Additional Data for certain features. You can not duplicate some features+additional data (for example, you can't have two CO keys for line 01)

Easy Edit Help

Doorphone Timer Setup (1351)
Doorphone units are connected to the PGDU card and provide a call button that will ring a group of extensions when pressed. When the doorphone call is answered the user can talk with the visitor at the door. Each doorphone can also have an associated door lock connection, the system provides a relay contact that will close when the user activates the doorlock release.

Doorphone Ring Time - The duration (Seconds) the extensions will ring when the doorphone call button is pressed.
Door Release Time - The duration (Seconds) that the associated door lock will be released.

External CFW by Doorphone disconnect timer - The system can forward the doorphone call to an external number via an ISDN trunk. The user sets the call forward with the 'External Call Forward by Doorphone' service code (default=822), the call is forwarded to an Abbreviated Dial location. The outgoing trunk is specified for the Abbreviated Dial location in Trunk Group for Abbreviated Dialling.

Easy Edit Help

Doorphone Ringing Assignment (1352)
Assign the extensions that will ring when the call button is pressed on the doorphone.
The doorphone numbers are assigned automatically by the system, refer to PGDU Setup for the doorphone port numbers.

Mode 1-8 - The night mode number. Enter the extension numbers that will ring when the doorphone call button is pressed.

Easy Edit Help

Doorphone Basic Data Setup (1353)

Chime Pattern - Select the doorphone ringing tone for the keyphones (SLT's will use the 'Doorphone Ringing for SLT' ring pattern defined in Ring Pattern if it is not set to fixed ring cadence in SLT Basic Setup).
The chime patterns are setup in Service Tones.
Ext to Doorphone transmit gain - Select the gain from the extension to the doorphone loud speaker.
Ext to Doorphone receive gain - Select the gain from the doorphone microphone to the extension.

Easy Edit Help

External Paging Options (1354)

External Speaker No - The external speaker numbers 1-6 are assigned to the audio ports of the PGDU cards in PGDU Setup.
External speaker number 7 is the external page output of MOH/PAGE socket on the Main Unit.
External Page Group - Assign each audio output to one of the six paging groups, the group number is used to identify the group when placing the paging call.
You can assign more than one External Speaker to the same External Paging Group to combined the speakers.
Paging Start Tone - Select the tone played over the external speaker when page call is placed.
Paging End Tone - Select the tone played over the external speaker when page call is completed.
Speech Path - Select the audio speech direction between the PGDU and external paging equipment.
Transmit Gain - Select the audio speech gain from the PGDU card to the external equipment. This will also effect the level of the paging start/end tones.
Receive Gain - Select the audio speech gain from the external equipment to the PGDU card.
Back Ground Music - Background music can be played over the external page speakers when there is no paging announcements in progress.
Background music is assigned to an ACI port in ACI Port Setup.

Easy Edit Help

Combined Internal/External Paging (1356)
The system can combine the internal and external paging groups.
The user dials the 'Combine Paging' service code (default=751) followed by the external group number and the system will include

the internal group specified.

External Paging group - The external group number selected by the user. The 'All External groups' is group number 0.
Internal Paging Group - Enter the internal paging group number that will be combined with the external group number.

Easy Edit Help

BRIU Setup (1361)
Configure each BRI circuit of the BRIU cards.

ISDN Card location - The cabinet and slot the card is installed in.
Cab 1 is the main unit, cab 2 is expansion unit 1, cab 3 is expansion unit 2.
The BRIU card can be installed into slot 5 or 6 of the 2OPBox.
With the cabinet wall mounted; slot 5 is the one nearest the wall.
Other BRIU settings can be made on the BRIU card, refer to the BRIU getting Started Guide for the selection of Polarity, Power Feed and Termination on/off.

ISDN Type - The type of BRI circuit (T-point or S-point) is selected in the Card Configuration screen.
Logical Trunk Port - The ports (trunk or station) assigned to the circuit. The port assigned to the first B-channel is shown.
ISDN Protocol - P-MP (Point to Multi-Point) or P-P (Point to Multi-Point).
Layer 3 Timer Type - There are five sets of Layer 3 timers. The timers are not available with Easy edit. Do not change this unless you are familiar with the Layer 3 timers.
T-point timers are set by program number 81-06, S-point timers are set by program 82-06.
CLIP Information Announcement - For T-point circuits Caller ID can be allowed/restricted.
Caller ID can be set for each trunk or each extension in the ISDN Trunks calling party number or DDI Calling party number screens.
Connection Bus Mode - The estimated length of the P-MP S0-bus cable. Short = less than 150 metres, Extended = up to 300 metres. (This setting has not effect P-P S0-bus cable length which can be up to 500 metres.)
S-Point DDI Digits - You can add additional digits onto the MSN number sent by the system on S-point circuits. You define the quantity of digits 0-4.
For example, The MSN number of the S-point circuit is set by its extension number, if you add 2 additional DDI digits the system will allow the extension number plus 00-99, giving a total of 5 digits for the MSN.
Dial Sending Mode - For T-point circuits select en-block (all dialled digits sent in a block, may cause a delay after dialling) or overlap (each digit is sent when the user dials).
Dial Information Element - Select the mode of the dialled digits for T-point circuits set for overlap dialling. Each dialled digit can be sent as either a Called Party Number or Keypad Facility.
Service Protocol for S-point - For S-point circuits you can select a specific protocol to provide additional services.
Leave this set to Keypad Facility unless you are connecting the circuit to another system.
Call Busy Mode for S-point - Calls from the S0 device that are routed to a busy destination can return either Alerting with call progress indication or Disconnect with User Busy indication.
ISDN Line Ring Back - The system can send ring back tone to incoming callers if it is not provided by the network. This feature must also be available on the ISDN network.
Type of Number - Outgoing ISDN calls contain a Type of Number setting within the Setup message. Leave this set to 'Unknown' unless the network provider supports the Type of Number being set by the system.
Numbering Plan - Outgoing ISDN calls contain a Numbering Plan setting within the Setup message. Leave this set to 'Unknown' unless the network provider supports the Numbering Plan being set by the system

Easy Edit Help

Trunk Outgoing Caller ID (1362)

ISDN Outgoing CLI - Enter the full number the system will send for all outgoing calls.
You may need to request this service from your Network provider.
The DDI Calling party number assigned to an extension will override the trunk caller ID.

Easy Edit Help

ISDN Options (1363)
This screen allows you to change the ISDN options used by all BRIU cards.

Send Release message after subscriber hangs up - Outgoing calls made on a T-point circuit to a busy destination can either release the B-channel automatically or wait for the system user to hang up before sending the Release.
Progress Indication Information Element detection - Outgoing calls made on a T-piont circuit can either detect (Service ON) or ignore (Service OFF) any Call Progress Indication. If set to Service OFF users may not hear any in-band tones or announcements after the dialled digits.
Bearer capability selection for SLT outgoing - Outgoing calls made on a T-point circuit from an SLT port can be sent as either 3.1KHz or Speech.
Send DT until user dials the first digit (overlap sending mode) - Users can hear system dial tone if set to Service ON or network dial tone if set to Service OFF.
Some Networks do not return dial tone when the system sends any Caller ID, this option allows you to use system dial tone to overcome this.
Start T305 timer after sending Disconnect Message - When set to Service ON timer T305 is controlled by the system and will force the B-channel to release after the system sends any Disconnect message.
When set to Service OFF T305 is controlled by the BRIU card and the system will wait for the Network to return a Release message.
Call Proceeding send mode - Incoming calls on a T-point circuit to a busy extension can either return Call Proceeding then Disconnect User Busy (Service ON) or Release Complete (Service OFF).
Send local busy tone when receiving Disconnect Message - Outgoing calls made on a T-point circuit to a busy destination may receive Disconnect User Busy but no in-band busy tone. The system can insert busy tone with this option set to Service ON.
Low Layer Compatibility - Should the system use (Service ON) or ignore (Service OFF) the LLC information eg Speech, 3.1KHz, Unrestricted Digital etc.
High Layer Compatibility - Should the system use (Service ON) or ignore (Service OFF) the HLC information eg Telephony, G2/3 FAX, G4 FAX etc.
S-Point Terminal seizes analogue trunk - Should an S-point device seize an analogue trunk for outgoing calls.
Clock Adjustment Mode - Should the system adjust the system time each day. The adjustment is made once per day according to the first connected call received after 01:00
TRK-TRK Transfer Info - Incoming calls forwarded back out automatically return Connect when the outgoing call receives Alerting. Allows the incoming caller to hear any in-band tones or announcements.
Busy tone mode - Outgoing calls made on a T-point circuit to a busy destination may receive Release but no in-band busy tone. The system can insert busy tone with this option set to Service ON.

Easy Edit Help

Port Setup (1365)
This screen will show the ports assigned to the main/expansion units and the 008/308 cards.
This information is of use when you download an existing configuration from the system.
The terminal type is assigned automatically, you do not need to pre-set the type.

Terminal Type - Shows the type of phone assigned to the port. The keyphone (Multi-Line Telephone) must be connected to show the terminal type.
If the keyphone is un-plugged the terminal type will default to SLT automatically.
64 Button consoles are shown as DSS. To remove a DSS console and reuse the port for another terminal type you must set the type to 'not used' and then plug in the new terminal.
Logical Port - The port assigned by the system.
Transmit Gain - Transmit gain is from the system to the terminal (System --> Terminal).
Receive Gain - Receive gain is from the terminal to the system (System <-- Terminal).

Easy Edit Help

Outgoing MOH Source (1366)
You can specify the music on hold source for incoming and outgoing trunk calls (There is a different option for incoming DDI calls in DDI Routing Table).

MOH Resource Type - Select the music on hold source for each trunk port.
Internal/External MOH = The system will use the music on hold source set in Music on Hold Setup.
BGM Source = The system will use the source connected to the Background music input. The BGM input is defined in ACI Port Setup.
ACI port = The system will use the source connected to the ACI port specified by the 'ACI Port Number'.
ACI Port Number - The ACI port number used when the MOH Resource Type is set to 'ACI port'. The ACI ports are set as external MOH in ACI Port Setup.

Easy Edit Help

Incoming Caller ID Setup (1367)
Times are in Seconds.

Caller ID displaying format - If the system receives more than 12 digits for the caller ID should the keyphone display the first 10 digits or the last 10 digits.
Caller ID Information waiting time - The system will accept caller ID information for this duration after the call arrives on an analogue trunk.
Caller ID edit mode - Should the system edit the received caller ID with the Network Number Plan settings in Location Setup within the general trunk setup options. Used for ISDN trunks. Set this option to Yes if you want the caller ID received on an ISDN trunk to use the Abbreviated Dial store to display the caller name.
Wait facility IE time - Leave this time set to 10 Seconds
Caller ID Sender active time - Set the duration the system will wait for a free resource when sending caller ID to a single line telephone SLT, the ringing will be delayed until the resource is available. If set to 0 the system will ring immediately, if no resource is available then no caller ID will be sent.
Private Call - Assign the display information when the received caller ID has an absence message of type P. Used for analogue trunks.
Call from out of service area - Assign the display information when the received caller ID has an absence message of type O. Used for analogue trunks.
Call information with error - Assign the display information when the received caller ID is not sent and there is no absence code. Used for analogue trunks.

Easy Edit Help

Incoming Calls (1368)
Setup the options common to all telephones that are related to incoming calls.

Priority for Incoming call - Choose which type of call should take priority when ringing at the extension. The priority call will be answered when the user goes off hook (Auto answer is rerquired for keyphones in Keyphone Options).
Incoming Call ring no answer alarm - Enable/disable the audible alarm when a ringing call is not answered. The alarm is a distinctive ring tone.
Incoming ring no answer alarm start - Set the duration (Seconds) that a trunk call must ring before the alarm tone starts.
Normal DIL Incoming no answer time - Set the duration (Seconds) that a trunk call must ring before the call steps to the second target. This timer operates for trunks set as either Normal or DIL in Incoming Service Type Setup. The first and second targets for Normal type trunks are assigned in IRG Assignment Normal. The first target for DIL type trunks is assigned in DIL Target Assignment, the second target is assigned in DIL Step on Target Assignment.
DID no answer time - Set the duration (Seconds) a DDI call must ring before the call steps from Target 1 to Targets 2 or 3. The DDI must also have a No Answer Transfer Option selected. The targets and Transfer options are assigned in the DDI Routing Table. Trunks are set as DDI type in Incoming Service Type Setup.
DID Incoming Ring Group no answer time - Set the duration (Seconds) a DDI call must ring before the call steps from an IRG in Target 2 or 3. The DDI must also have a No Answer Transfer Option selected.
DID Incoming Pilot Call no answer time - Set the duration (Seconds) a DDI call must ring before the call steps from a Department Group in Targets 1, 2 or 3. The DDI must also have a No Answer Transfer Option selected.
DID to Trunk to Trunk no answer time - Set the duration (Seconds) a DDI call must ring before the call steps from a Trunk to Trunk call (incoming DDI call roued to an outside number) in Targets 1, 2 or 3. The DDI must also have a No Answer Transfer Option

selected.
VRS Waiting Message operation - The VRS messages for Queue Announcements can be played automatically (set by the system timer) or manually (with programmable Function Key 'Auto Answer with delay message'). It is recommended that this option is set to Automatic.
VRS Waiting Message interval time - Set the interval (Seconds) between each VRS announcement. Used for VRS Queue Announcements.

Easy Edit Help

Hold and Transfer (1369)
Setup the options common to all telephones that are related to holding and transferring calls.

Transfer to busy extension - Enable/disable the ability for an extension to transfer the held call to a busy extension. The held call will queue at the busy extension.
Ring-back tone to transferred calls - Select the tone the held caller will hear when the call is transferred.
No answer time for call forward - Set the duration (seconds) the call will ring before being forwarded when the user has set Call Forward No Answer.
Ring Inward Recall time - Set the duration (Seconds) that a call will ring after being transferred by a user. After this time the call will ring back to the extension that performed the transfer.
Ring Inward to busy Extension Group - Set the duration (Seconds) that a call will queue after being transferred to a busy Department Group. After this time the call will ring back to the extension that performed the transfer.
Trunk to Trunk Transfer Warning Tone time - Set the duration (Seconds) before the trunk to trunk warning tone will be played. After this time the caller can dial the continue code set in Trunk to Trunk Routing.
Extension Group delayed transfer time - Set the duration (Seconds) that a call will ring at the Department group before it will be forwarded.
Trunk to Trunk Transfer Disconnect Time - Set the duration (Seconds) before the call will be disconnected if the caller does not dial the continue code. This timer starts after the 'Trunk to Trunk Warning Tone Time' has expired.
No Answer Time for Step Transfer - The duration the system will wait for the outgoing call to be answered before stepping to the next Abbreviated Dial set in Automatic Trunk to Trunk Transfer Target if 'Step Transfer' is selected for the trunk port (of the associated incoming call) in the 'Automatic Trunk to Trunk Transfer Mode' in Trunk Basic Data Setup.
No Answer Time for Automatic Trunk to Trunk Transfer - The duratin the system will wait for the incoming call to be answered before routing the call to the Abbreviated Dial entry for the Trunk to Trunk Forwarding Feature.

Easy Edit Help

LCR Dial Data (1370)
Least Cost Routing (LCR) is used to access an alternate network carrier. The system will compare the digits dialled on the trunk to the Dial Table, if a corresponding entry is found the call will follow the LCR dial translations. If a corresponding entry is not found the digits can be sent to line without any translation.

Trunk No - LCR is enabled for each trunk port.
There is also the option to send only the cost center code before the dialled digits, this option will not compare the dialled digits to the Dial table.

Dial table. There are 200 entries within the Dial Table.
Dial Data - The dialled digits that will used to select the Carrier Table for LCR dial translation.
Enter the leading digits of the numbers that will be routed via the alternate network carrier and enter the LCR Carrier Table that you will use to perform the LCR dial translation.
Carrier Table - Enter the Carrier Table number (1-25) that will be used to perform the digit translation to access the alternate carrier.

Carrier Table - There are 25 Carrier Tables available, each can be used to access a different carrier.
Delete Digits - Enter the quantity of leading digits to be deleted.
LCR Code - Enter the Access Code that must be dialled to access the indirect carrier. Leave blank if you want the call to be routed by the direct carrier (the Network Supplier).
Enter P to add a pause of 3 seconds.
Enter @ to change from dial pulse to DTMF dialling on an analogue trunk or 'Wait for Connect' on an ISDN trunk, further digits will then be sent as DTMF in the speech path.
Auth Table No - Certain indirect carriers require an authorisation code, enter the Authorisation table number (1 to 10) specified in

LCR Options. The Authorisation code is sent after the LCR Access Code.
Cost Center Code - The system can send the Cost Center Code after the LCR Access code and optional Authorisation code.

Easy Edit Help

LCR Options (1371)

Authorisation Code - Enter the Authorisation Codes if specifed in the 'Carrier Table'. Enter up to 10 digits.

Manual Exemption Table - Enter any numbers (for example Emergency codes) that you do not want to be routed via an indirect carrier. This table will filter out any numbers except those that begin with digit 0. Any numbers dialled by a user that correspond to an entry in this table will be dialled to line with no LCR translation.

Easy Edit Help

Cost Center Codes (1372)
The system can send a Cost Centre Code for trunk calls routed via either the direct or indirect carrier.
To send to the direct carrier select 'LCR On (Cost Center Code only)' for the trunk in LCR Dial Data.
To send to the indirect carrier enable the 'Cost Center Code' option for the Carrier Table in LCR Dial Data.

Cost Center Code - Enter the cost center code for each extension. Ensure you enter the quantity of digits expected by the network carrier or the calls will mis-dial.

Easy Edit Help

PGDU Setup (1375)
Setup the ports of any PGDU cards you have installed in the system.
The PGDU card provides two doorphone ports with associated relays and two audio ports that can be used for external paging or external music on hold sources.
Each main and expansion unit can have one PGDU card installed giving a maximum of three cards.

Type - Port 1 and 2 must be doorphones, do not change the port type. The doorphone number is shown by the logical port number. Ports 3 and 4 can be set to either external paging, tone ringer or ACI. Port 3 is identified as Audio 1 on each PGDU card and Port 4 is Audio 2.
External Paging - Connect the audio port to an external paging amplifer. The paging zone number is shown by the logical port number.
Tone Ringer - Connect the audio port to an external paging amplifier to have ring tone for incoming trunks calls played over the paging system. Easy Edit does not support the ringing assignment, this is done with program 31-05.
ACI - Connect the audio port to an external music source. The ACI number is shown by the logical port number.
An ACI port used for external music on hold can be assigned to trunks Outgoing MOH Source and incoming DDI calls DDI Routing Table.
The ACI ports can also be used to record conversations, refer to Conversation Recording for Trunks or Conversation Recording for Extensions.
ACI port mode is setup in ACI Port Setup
Logical Port Number - Automatically assigned by the system. Defines the port number of the associated type. Note - if you change the Type then you must upload the configuration to the system and then download again to see the correct port number.
Relay - There are two relay contacts on each PGDU card. You can assign the relays to any of the doorphone or audio ports.

Easy Edit Help

Location Setup (1376)
Used by the system to edit the Caller ID received on ISDN trunks. You must also set the Caller ID Edit Mode to On in Incoming Caller ID Setup in the Extensions section for these settings to be implemented.

Country Code - Enter the International Prefix for your country (for example UK=44, France=33, Germany=49). The system will remove this code from the calling party number with the International number type set. This setting is optional.
International Access Code - Enter the International access code, the default is 00. The system will insert this code when the International number type is set in the received calling party number.
Other Area Code - Enter the National access code (to call outside of your local area), the default is 0. The system will insert this code when the National number type is set in the received calling party number.
Local Area Code - Enter the National area code for your own area (omit the National access code from the area code). The system will remove this code from the calling party number with the National number type set. This setting is optional.
Trunk Access Code - Enter the trunk access code for your system. This will be inserted into the calling party number if the call is routed to an S0 device or DECT handset. This setting is optional.

Easy Edit Help

Answer Trunk Ringing on External Speakers (1380)
The system can use the external speakers to indicate that a trunk call is ringing. This feature will only operate for trunks set to Normal incoming type in Incoming Service Type Setup.
The ringing trunk can be answered at any extension with the 'Answer for non-ringing lines' service code (default=872).
The extension can also automatically answer the ringing trunk by going off hook if the 'Auto Off Hook Answer' option is enabled in Class of Service.
The trunks ringing on the external speakers that the extension can answer are assigned in Trunk Ringing on External Speaker.

Mode 1-8 - The night mode number. For each night mode enter the trunk route number that contains the trunk group(s) of the trunks that can be answered.
Trunk groups are assigned to trunk routes in Trunk Group Routing.

Easy Edit Help

OPAC Key Operation (1381)
This feature defines the features available via the OPAC on system telephones.
The features available will only be available dependant on the status of the terminal i.e. some functions are only available when the station is idle or when the station is on a call.

OPAC + Digit - Defines the digit to be dialled in addition to the OPAC key to select the feature
Feature - Defines the feature accessed by the OPAC +Digit selection

*** Short List -** OPAC * is fixed for the configuration of function keys. The single digit can be assigned to commonly configured keys to speed-up programming, the shortcuts are accessed via the volume keys. This list is also used if the function key is assigned via service code 851 or 852
Function - Defines the keys to be included in the shortcut.

To configure a key via OPAC key operation
*1. With handset on-hook, press OPAC + **
2. Press function key to be programmed
3. Using vol. keys scroll to required selection
4. Press hold
5. Enter additional information if required
6. Press hold.

 # Short List - Defines the expanded option list short code
Function - Defines the function to be used when OPAC + # + code is entered.

Easy Edit Help

OPAC Key Operation (1382)
This feature defines the features available via the OPAC on system telephones.
The features available will only be available dependant on the status of the terminal i.e. some functions are only available when the station is idle or when the station is on a call.

OPAC + Digit - Defines the digit to be dialled in addition to the OPAC key to select the feature
Feature - Defines the feature accessed by the OPAC +Digit selection

*** Short List -** OPAC * is fixed for the configuration of function keys. The single digit can be assigned to commonly configured keys to speed-up programming, the shortcuts are accessed via the volume keys. This list is also used if the function key is assigned via service code 851 or 852
Function - Defines the keys to be included in the shortcut.

To configure a key via OPAC key operation
*1. With handset on-hook, press OPAC + **
2. Press function key to be programmed
3. Using vol. keys scroll to required selection
4. Press hold
5. Enter additional information if required
6. Press hold.

 # Short List - Defines the expanded option list short code
Function - Defines the function to be used when OPAC + # + code is entered.

Easy Edit Help

3 Digit Service Codes (1410)
Most three digit service codes are available while you are listening to dial tone, some can be used while listening to busy tone or ring back tone (eg 850 for Camp-on).
Although all service codes are three digits long at default you can change them if you edit the System Numbering plan.
Three digit service codes are independent of the One Digit service codes.
Note. If you want to find a service code number then click on the Service Code column header and the codes will be sorted low-high, click again for high-low.

Left Column - The program reference number. Not used within Easy Edit.
Service Description - The feature that will be set by the service code.
Service Code - Enter the service code to access the feature.
Do not duplicate any service codes, click the column header to list the service codes.
The service code is not validated against the System Numbering Plan by Easy Edit so be sure to check the amount of digits you should enter if you have made changes in the System Numbering screen.

Easy Edit Help

1 Digit Service Codes (1411)
One digit service codes are available while you are listening to busy tone or ring back tone.
For example you dial a busy extension and then press a single digit to set Camp-On.
One digit service codes do not need defining within the System Numbering Plan.

Left Column - The program reference number. Not used within Easy Edit.

Service Description - The feature that will be set by the single digit.
Service Code - Enter the single digit to access the feature.

Easy Edit Help

Toll Restriction Common Tables (1420)

Common Restriction - The numbers for the Common Restriction table.
Common Permit - The numbers for the Common Permit table. These numbers are exemptions to the Common Bar table.

Easy Edit Help

Toll Restriction International Tables (1421)

International Restriction - The numbers for the International Restriction table.
International Permit - The numbers for the International Permit table. These numbers are exemptions to the International Bar table.

Easy Edit Help

Toll Restriction Exemption Tables (1422)

Permit Table 1-4 - The numbers for each Exemption table. The exemptions are to the restricted numbers in the corresponding restriction table 1-4.

Easy Edit Help

Toll Restriction Tables (1423)

Restriction Table 1-4 - The numbers for each Restriction table.

Easy Edit Help

Toll Table Assignment (1424)
Each extension is assigned a Toll Class number in the Toll Restriction per Night mode screen within the Extension Properties menu. The Toll Restriction classes share a set of restriction and permit tables, these tables are assigned to each class in this screen.

Toll 1-15 - There are 15 Toll Restriction classes available.
International Call Restriction Table - There is one restriction table available in the International screen.
International Call Permit Table - There is one permit table available in the International screen. This table contains exemptions to any numbers in the International Restriction Table.
Maximum Digit Table Assignment - To limit the quantity of digits that are dialled to line. There are 4 tables available, set to 0 if no

limit is required.
Common Permit Code Table - There is one table available that can be assigned to each class. The Common Permit table contains exemptions to any numbers in the Common Restriction table.
Common Restriction Code Table - There is one table available that can be assigned to each class.
Permit Code Table - There are four tables available, one can can be assigned to each class. The Permit tables contain exemptions to any numbers in the same numbered Restriction table. Enter 0 if you do not want any table assigning.
Restriction Table - There are four tables available, one can can be assigned to each class. Enter 0 if you do not want any table assigning.
Common Abbreviated Dial restriction - Should numbers in the Common Abbreviated dial area have toll restriction applied.
Group Abbreviated Dial restriction - Should numbers in the Group Abbreviated dial area have toll restriction applied.
Internal Call Restriction - Extensions in a class with this option enabled can not receive internal calls.
PBX Call Restriction - With this option set to disable calls via a PBX line will be restricted only if the PBX trunk access code is dialled, calls to PBX extensions are not restricted.
If you enable this option calls via a PBX line are restricted if the PBX trunk access code is dialled or not, calls to PBX extensions are also restricted.
Transfer of Incomplete Dial - Should outgoing calls that have not had enough digits dialled to complete the toll restriction process be transferred to other extensions.
Hold of Incomplete Dial - Should outgoing calls that have not had enough digits dialled to complete the toll restriction process be placed on hold.

Easy Edit Help

Toll Restriction Maximum Digits Dialled (1425)

Max digits - The maximum quantity of digits that can be dialled to line if the table is assigned to the Toll Restriction class.

Easy Edit Help

Dial Block (1426)

Dial Block Toll Table (1-15) - The Toll Restriction class that the extension will be set to when Dial Block is set (if the Common Dial Block Toll Table is set to 0). This assigned class is also used in case of "Toll restriction by Credit".
Supervisor Dial Block Password - The password that will allow a user to set Dial Block for any other extension.
Common Dial Block Toll Table - The Toll Restriction class that any extension will be set to when Dial Block is set (set to 0 if you want to use the Dial Block Toll Table for each extension).

Easy Edit Help

Toll Restriction Override Passwords (1427)

Override Password - The password for each extension to allow the user to override the Toll Restriction.
Toll Restriction Override Release time - The duration the user has until the toll restriction override is cancelled.
Interdigit time for Toll Restriction path control - How long the system will wait before connecting the transmit speech path to the trunk. Set this timer greater than the duration (Seconds) that the Network will supply dial tone.

Easy Edit Help

Walking Toll Passwords (1428)

Password - The password the user must enter to override the toll restriction at any extension.
Toll Class - The Toll restriction class that will be used for each entry.

Easy Edit Help

Extension Ring On/Off (1429)
Calls ringing via an IRG can be set to ring audibly or silently for each extension.

Mode 1-8 - For each night mode select Ring or No ring.

Easy Edit Help

Incoming Ring Group Setup (1430)
Incoming Ring Groups (IRG) are used to route incoming trunks to one or more telephones. All available telephones in the group will ring simultaneously. If more than one trunk call is ringing at the same IRG they will be queued with the longest ringing first.

Ring Group - The ring group number is used to route the trunk call to the IRG. The IRG number can be entered for Normal type trunks in IRG Assignment (Normal), DIL Step on Target Assignment or DDI Routing Table.
Extn No - The extension numbers of each member of the IRG. You can also enter Virtual extension numbers. An extension can be a member of more than one group.

Easy Edit Help

IRG Assignment Normal (1431)
Each trunk set to Normal type in Incoming Service Type Setup is routed to an IRG for each night mode.
There are two targets available, the system will step unanswered calls from the first to second target after the Normal DIL Incoming Call No Answer Time in Incoming Call.

IRG First Target - Enter the IRG number for the first target.
IRG Second Target - Enter the IRG number for the second target. Enter 0 if you do not want any IRG step on.

Easy Edit Help

DISA External Access (1437)
If the DISA User ID is set to ON in DISA Basic Options the passwords must be assigned here. If DISA User ID is turned OFF the system user the settings for DISA User 1 for the incoming DISA call.

DISA User 1-15 - The DISA User number used by the system when the incoming caller enters the corresponding password. The

DISA User number is used to assign the Toll Restriction class and Class of Service in DISA Toll & COS.
Password - The password that must be entered by the incoming caller (minimum of 6 digits).
Trk route Mode 1-8 - The trunk route used to seize the outgoing trunk when the incoming caller dials the trunk access code set in 1 Digit Codes.
Individual Trk route Mode 1-8 - The trunk route used to seize the outgoing trunk when the incoming caller dials the Alternate trunk access code set in 1 Digit Codes.

Easy Edit Help

DISA Toll & COS (1438)
Each DISA user number is assigned a toll restriction class and class of service.

DISA User 1-15 - The DISA user number assigned by the DISA password the incoming caller has entered.
Password - The DISA password is shown here. The passwords are setup in DISA External Access.
Toll Mode 1-8 - The toll restriction class assigned to the DISA user when the call is routed to an outgoing trunk. The toll restriction tables are setup in Toll Restriction.
COS Mode 1-8 - The class of service number assigned to the DISA user. The DISA COS options are setup in DISA Class of Service Options.

Easy Edit Help

Music on Hold Setup (1441)
Music on hold is heard by the caller when they are placed on hold. The system has a built in melody or can have an external music device connected to the MOH socket of the Main Unit or an audio input of a PGDU card.
Refer to ACI Port Setup to setup the audio ports of the PGDU card for external music on hold connection. Select 'Input' for the ACI type.
Incoming DDI calls have a separate option to select the music on hold tone in DDI Routing Table.
Other trunk calls have an option to select the music on hold tone in Outgoing MOH Source.

Select Type - Select the music on hold source, this will be used for internal calls, trunks set to 'Internal/External MOH' in Outgoing MOH Setup and incoming DDI calls set to 'MOH Tone' in DDI Routing Table.
Internal Source = Will use the system's built in melody.
External Source = Will use the external source connected to the MOH socket of the Main Unit.
Service Tone = Will use '64-MOH Tone' in Service Codes.
DSPDBU = Will use the DSPDB message number defined in the 'Select Tone' option.
Select Tone - Turn music on hold on/off or define the DSPDB message number.
If 'Select Type' is set to Internal:
    0 will play silence.
    1-3 will play one of the built in melodies
If 'Select Type' is set to External or Service Tone:
    The value entered in this option is not used.
If 'Select Type is set to DSPDBU:
    0 will play silence.
    1-48 will play the DSPDB message number (requires the DSPDB card to be installed).
Volume Level - Select the gain of the music on hold tone (internal melody and MOH socket of the Main Unit).

Easy Edit Help

Extension Outgoing Caller ID (1445)

Extn CLIP - Enable or disable the sending of each extension's caller ID.
ISDN Calling Party Number - Enter the full number the system will send for all outgoing calls.

You may need to request this service from your Network provider.
The DDI Calling party number assigned to an extension will override the trunk caller ID.

Easy Edit Help

IRG Queue Announcement Setup (1446)
The system can play the DSPDB messages to incoming callers whilst the call is ringing at an IRG (Incoming Ring Group). The call will continue to ring at the IRG while the queue announcement is being played.
The system can repeat up to two seperate announcements (first and second). The interval between announcements is set by VAU Waiting Message Repeat Time in VRS Options.
IRG queue announcements can be started automatically after the First Waiting Message Start time or manually by pressing the programmable function key set to 'Auto Ans with delay mesg' in Function Key Programming. Automatic/manual operation is selected by 'VAU Waiting Message Operation' in VRS Options.

IRG - The Incoming Ring Group number the trunk call is ringing at.
First Waiting Message Start time - The duration the system will wait before answering the trunk to play the first announcement.
First Waiting Message No - The DSPDB message number 1-48 for the first announcements. The DSPDB messages are recorded with the Operation for VRS Message service code (default=716).
First Message Sending Count - How many times will the first announcement be repeated. When this quantity of First Message announcements are complete the system will begin sending the Second Waiting Message announcements.
Second Waiting Message No - The DSPDB message number 1-48 for the second announcements.
Second Message Sending Count - How many times will the second announcement be repeated.
Message Interval Tone Kind - During the interval the system can send one of the following tones:
Ring-back tone = The system will send 14-Ring Back tone in Service Tones.
MOH Type = The system will send the music on hold tone set in Music on Hold Setup.
BGM Source = The system will send the music on hold tone connected to the ACI port set as BGM (Background Music) input. The BGM input is specified in ACI Port Setup.
After end of VRS Waiting Message Disconnect time - The system can disconnect calls if they are not answered after this time. This timer starts when the system has completed the sending of all second messages.

Easy Edit Help

Department Group Queue Announcement Setup (1447)
The system can play the DSPDB messages to incoming trunk callers whilst the call is queued at a busy Department Group. The call will continue to queue at the group while the queue announcement is being played.
The system can repeat up to two seperate announcements (first and second). The interval between announcements is set by VAU Waiting Message Repeat Time in VRS Options.
Analogue trunks must be set as DIL type in Incoming Service Type to be routed to a Department Group.

Department Group - The Department Group number the trunk call is queued at.
First Waiting Message Start time - The duration the system will wait before answering the trunk to play the first announcement.
First Waiting Message No - The DSPDB message number 1-48 for the first announcements. Message number 49 is a pre-recorded announcement. The DSPDB messages are recorded with the Operation for VRS Message service code (default=716).
First Message Sending Count - How many times will the first announcement be repeated. When this quantity of First Message announcements are complete the system will begin sending the Second Waiting Message announcements.
Second Waiting Message No - The DSPDB message number 1-48 for the second announcements.
Second Message Sending Count - How many times will the second announcement be repeated.
Message Interval Tone Kind - During the interval the system can send one of the following tones:
Ring-back tone = The system will send 14-Ring Back tone in Service Tones.
MOH Type = The system will send the music on hold tone set in Music on Hold Setup.
BGM Source = The system will send the music on hold tone connected to the ACI port set as BGM (Background Music) input. The BGM input is specified in ACI Port Setup.
After end of VRS Waiting Message Disconnect time - The system can disconnect calls if they are not answered after this time. This timer starts when the system has completed the sending of all second messages.
Enhanced Hunting Type - This is the Enhanced Hunting option for the Department Group in Department Group Options.
For an analogue trunk to hear the queue announcements this option must be set to 'No Hunting'.

For a DDI call to hear the queue announcements this option must not be set to 'No Hunting'. If you require the No Hunting set for the Department Group and also want queue announcements then you must set a No Answer Transfer option and Max Number of Calls not set to 0 for the DDI in the DDI Routing Table.

Easy Edit Help

Virtual Extension CLI and Fixed Forward
Setup the Virtual Extension numbers.

Virtual Port - The Virtual extension ports available on the system.
Virtual No - The extension number for the virtual extension, this number is used to route calls to the virtual extension. You can not duplicate the extension number with any other extension number or pilot number on the system. The virtual extension numbers can be any number set as Extension type in System Numbering.
ISDN Calling Party Number - Enter the full number the system will send for all outgoing calls.
You may need to request this service from your Network provider.
The DDI Calling party number assigned to an extension will override the trunk caller ID.

Fixed call forwards can be defined for each virtual extension number.
External fixed call forward will take priority over the internal setting if you set both External and Internal fixed call forwards.
Each virtual extension can have either a fixed call forward to an internal destination (another station, Department Group pilot number or virtual extension) or to an external destination.
INTERNAL - Fixed call forward - Select the type of internal fixed call forward.
INTERNAL - Fixed call forward destination - Enter the destination number (up to 4 digits).
EXTERNAL - Fixed call forward destination - Enter the outside number, remember to enter the trunk access code (up to 24 digits).

Easy Edit Help

Fixed Call Forward (1449)
Fixed call forwards can be defined for each station. When defined the user can not turn off the fixed call forward but they can choose to temporarily overide it with any other call forward using any of the call forward service codes. When the user cancels the call forward the fixed call forward will be activated automatically. External fixed call forward will take priority over the internal setting if you set both External and Internal fixed call forwards.
Each user can have either a fixed call forward to an internal destination (another station, Department Group pilot number or virtual extension) or to an external destination.
INTERNAL - Fixed call forward - Select the type of internal fixed call forward.
INTERNAL - Fixed call forward destination - Enter the destination number (up to 4 digits).
EXTERNAL - Fixed call forward destination - Enter the outside number, remember to enter the trunk access code (up to 24 digits).

Easy Edit Help

DISA Basic Setup (1450)

DUD/DISA Dial in mode - The system can either use the digits directly as they are received or search the DDI Routing Table.
Use Extn/Service code = The system will use the digits received to route directly to the destination extension number or Auto Attendant Single Digit Operation table.
Use Dial Conversion Table = The system will use the received digits to search the DDI Routing Table.
The trunk will be routed to the DDI Tables as if the trunk was set as DDI Type in Incoming Service Type Setup.
Note - If you select the Dial Conversion Table you will not be able to use the Auto Attendant Single Digit Operation.
DISA User ID - The DISA password can be turned on or off.
On = The caller must enter a valid password assigned in DISA External Access to gain access to the system.
Off - The caller does not need to enter any password to gain access to the system. The caller will use the DISA Class 1 settings in DISA Class of Service Options.
DUD/DISA Transfer Alarm - When the DUD or DISA call steps on to the fall over IRG the system can use a different ring pattern.

Normal = The call will use the DUD/DISA ring pattern set in Ring Pattern.
Alarm = The call will use the Call-Back ring pattern set in Ring pattern.

Easy Edit Help

DISA Service Options (1451)

DSPDB Message Access Password - The incoming DISA caller can access the DSPDB messages (1-48).
The incoming caller dials the Operation for VRS Message service code (default=716) followed by the 6 digit password. They then have access to the messages (listen 5, record 7 or erase 3).
Continue code for DISA trunk to trunk - Specify the DTMF digit the caller must dial to continue the trunk to trunk call.
Disconnect code for DISA trunk to trunk - Specify the DTMF digit the caller can dial to disconnect the trunk to trunk call.

DISA trunk to trunk continue/disconnect operation will also require both trunk ports having the Continue/Disconnect trunk to trunk conversation option enabled in Trunk Basic Data Setup.
The timers are specified in System Timer for DUD/DISA Service.

Easy Edit Help

System Timer for DUD/DISA Service (1452)

DUD/DISA Dial tone - The duration the system will wait for the incoming caller to complete dialling. When this timer expires the caller will be disconnected or step on to the Auto Attendant Fall Over.
DUD/DISA No Answer time - The duration the system will ring at the target extension. When this timer expires the caller will be disconnected or step on to the Auto Attendant Fall Over.
Disconnect after DUD/DISA re-transfer to IRG - The duration the system will ring at the Auto Attendant Fall Over. When this timer expires the call will be disconnected.
Calling time to Automatic Answering telephone - The duration the system will wait for the SLT port to answer the DUD/DISA call. When this timer expires the caller will be answered by the DUD/DISA dial tone.
Guidance message by Automatic Answering telephone - The duration the system will connect the SLT device. When this timer expires the caller will be answered by the DUD/DISA dial tone.
Guidance message by ACI Talkie duration - The duration the system will connect the ACI device. When this timer expires the caller will be answered by the DUD/DISA dial tone.
DISA Conversation Warning tone time - The duration the system will wait before sending a warning tone to a trunk to trunk call via DISA. After this time expires the user can enter the continue code (if enabled) or the trunk to trunk call will be disconnected.
DISA Conversation Disconnect time - The duration the system will wait before disconnecting a trunk to trunk call via DISA, this timer starts when the trunk to trunk call is established. After this time expires the trunk to trunk call will be disconnected unless the caller has entered to continue code.
The duration of this timer must be greater than the combined times of the DISA Conversation Warning tone time plus the Continue Duration for DISA trunk to trunk time.
Continue Duration for DISA trunk to trunk - The duration the system will wait before re-sending the warning tone, this timer starts after the DISA Conversation Warning tone expires.
The trunks must have the Continue/Disconnect trunk to trunk conversation option enabled in Trunk Basic data Setup.
DISA Internal Paging duration - The duration the system will allow the DISA caller to be connected to the internal paging zone. When this time expires the call will be disconnected.
DISA External Paging duration - The duration the system will allow the DISA caller to be connected to the external paging zone. When this time expires the call will be disconnected.
DUD/DISA Answer delay - The duration the system will wait before answering the incoming DUD/DISA call.
DUD/DISA Busy tone - The duration the system will send busy tone to the incoming caller when the destination extension thay have dialled is busy.
The DUD/DISA Talkie must be set to No Talkie in DUD/DISA Talkie and the Busy fall over must be set to 0 in Auto Attendant Fall Over.
Delayed DUD Answer time - The duration the system will wait before answering the incoming Delayed DUD call. The trunk is set to Delayed DUD type in Incoming Service Type Setup. Before this timer expires the system will route the call as a Normal type (for example to an IRG).

Easy Edit Help

CTI LAN Port Setup (1453)
Any CTI Applications will connect to the system via the LAN.

3rd Party CTI
CTI - TCP Port - Enter the TCP port used by the CTI Server to connect to the system (8181 recommended).
Keep Alive Timer - It is recommended that this is set to 30 seconds.

1st Party CTI
CTI - TCP Port - Enter the TCP port used by the 1st party CTI clients to connect to the system (8282 recommended).
Keep Alive Timer - It is recommended that this is set to 30 seconds.

Disconnect Supervision - This should be enabled for 1st Party CTI.

XN120 MYCTI License procedure

The XN120 CTI Application is licensed based on a unique PC Hardware ID. This Hardware ID is generated on installation of XN120 CTI.

A 45-day evaluation is automatically generated on installation. During this period it is possible to use all features, it is recommended to license the software before the evaluation period expires.

In order for the License code(s) to be generated you are required to provide the following:

Purchase Order
Company Name
Company Address
Hardware Code (Generated on installation of NEC MYCTI)

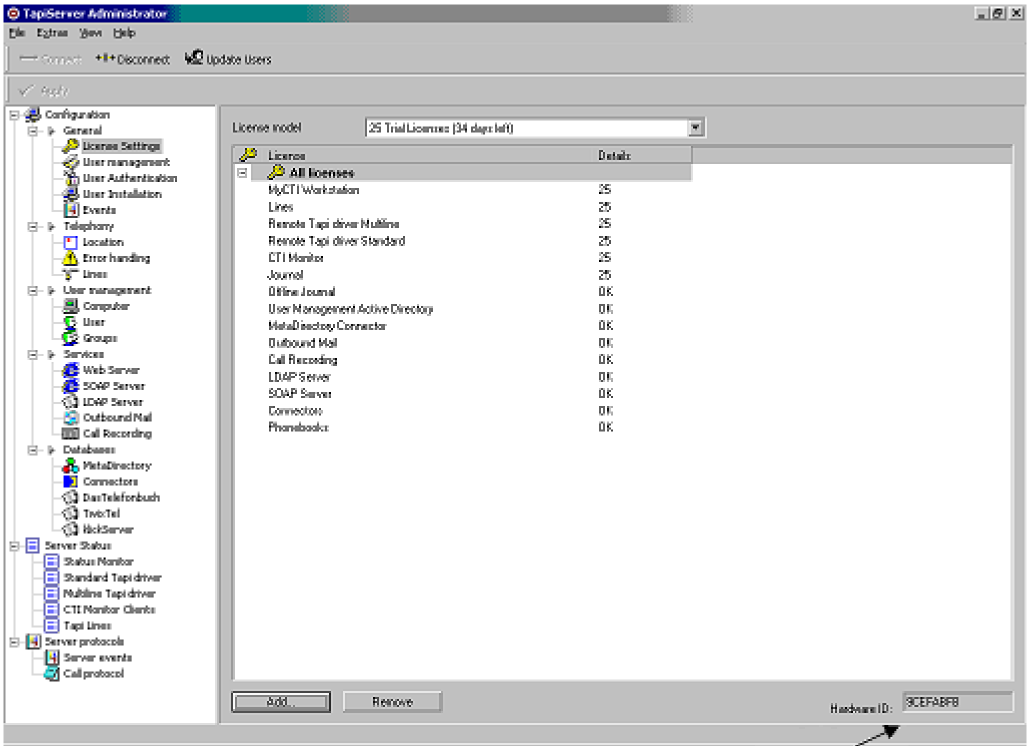
Methods used to send Purchase Order and Hardware ID:

eMail	licensingrequests@necinfrontia.co.uk
Fax	+44 (0) 1509 610206
Telephone	+44 (0) 1509 643100

Acquiring Hardware ID

In order to display the Hardware-ID bound to your installation of TapiServer, perform the following at the NEC TAPIServer.

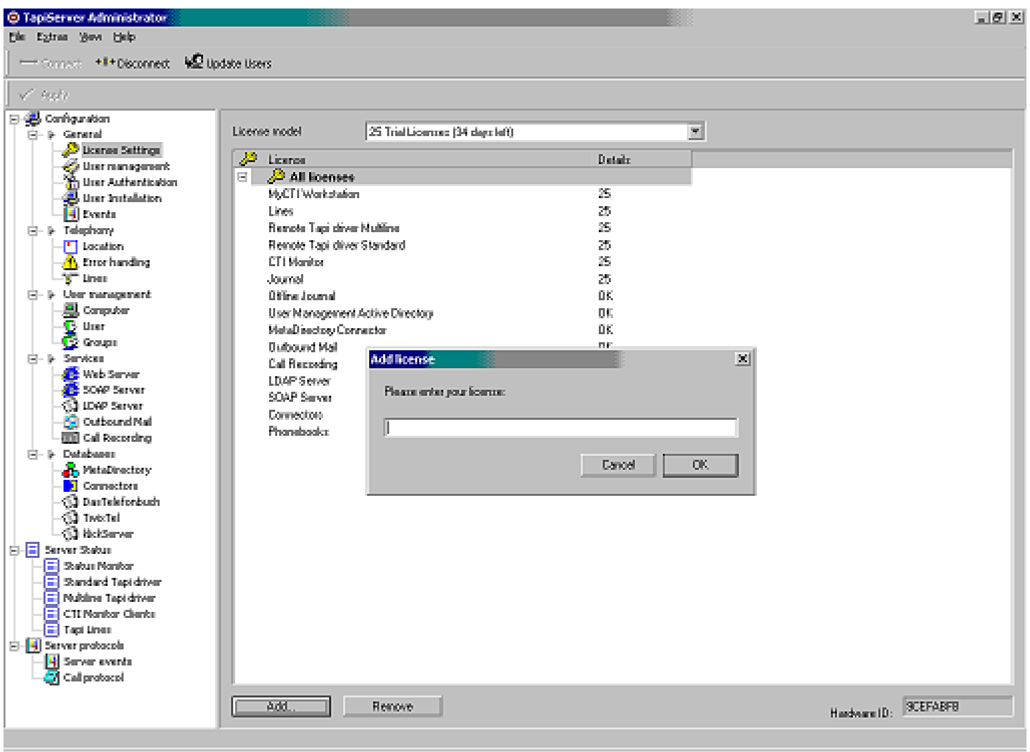
Click Start/Run/NEC/TapiServer/TapiServer Admin
Click on to Configuration/General/License Settings.



The Hardware ID is located in the lower right of the screen

Entering License

Click the Add button under Configuration/General/License settings.



Enter the license exactly as provided by your authorized reseller, click ok to apply.

Example Code

ABCD1234-ABCD1234-ABCD1234-ABCD1234

Easy Edit Help

Service Tones (1455)
Service tones are built up from a range of Basic Tones that are added to the Service Tone as Units for a specified duration. Each Service Tone has eight Units available. The eight units can then be repeated either a set number of times or continuously.
Examples.
To build a continuous tone: Add one unit that contains the required Basic Tone with any duration and repeat this continuously.
To build a continuous 500mSon/500mSoff tone: Add one unit that contains the required Basic Tone with a duration of 500mS and another unit of silence for 500mS then repeat continuously.
To build a tone that contains two beeps of 100mS tone seperated by 500mS that is not repeated: Add one unit for the 100mS tone, a unit of 500mS silence and another unit of 100mS tone, repeat count of one.

The system must be powered off and on for any changes to Service Tones to take effect.

Service Tone - The Service Tones are numbered 1-64, this is the number used by the system when assigning the tone to a feature.
Repeat Count - How many times the Service Tone will be repeated. Enter 0 to repeat continuously.
Unit 1-8 - The units that build up the Service Tone. The units are played once in order 1 to 8. Each unit specifies the Basic Tone, its duration and level.
Any unit with a Basic Tone and Duration of 0 will not be included in the Service Tone.
Basic Tone - Select the Basic Tone (1-32) from the Basic Tone table below.
Duration - Enter the duration of the unit in 100 milli-Seconds blocks (for example enter 5 for a 500mS duration).
Gain Level - Enter the level (1-63) from the Gain table below.

Basic Tone Table

Basic Tone	Frequency (Hz)	Level (dB)		Basic Tone	Frequency (Hz)	Level (dB)
1	420	-13		17	520/650	-13/-19
2	520	-13		18	650/780	-13/-19
3	580	-13		19	780/1040	-13/-19
4	660	-13		20	1040	-13
5	700	-13		21	Not used	-
6	800	-13		22	Not used	-
7	880	-13		23	Not used	-
8	1050	-13		24	Not used	-
9	430	-13		25	Not used	-
10	440/480	-13/-13		26	Not used	-
11	480/620	-13/-13		27	Not used	-
12	440	-16		28	Not used	-
13	Not used	-		29	Not used	-
14	520/650	-19/-13		30	Not used	-
15	650/780	-19/-13		31	Not used	-
16	780/104	-19/-13		32	Not	-

	0				used	

Gain Table

Enter	Gain (dB)
1	-15.5
2	-15.0
3	-14.5
~	~
31	-0.5
32	0
33	+0.5
~	~
62	+15.0
63	+15.5

Easy Edit Help

Trunk Call Charging (1456)
The system can receive Advice of Charge (AOC-D or AOC-A) from the Network supplier for outgoing calls.

COS Table - Class of Service assigned to the extensions in Class of Service per Night Mode.
COS AOC Option - Should extensions in each Class of Service display the Advice of Charge information (only available for display keyphones).
Multiplier for AOC - Should the AOC Charge cost be multiplied. The value is entered as a percentage (100 = x1, 150 = x1.5). Default = 100, enter a value from 100 to 500.
The AOC Charge cost will also multiply the Charge Cost per Unit.
Setting of charge cost per unit - Set the cost for each AOC Unit received from the Network. The value is entered as a percentage (100 = 1.00, 150 = 1.50) to convert the charge unit into Pounds and Pence or Euros and Cents.
For unit count set this to 0.
To convert the unit count to a charge cost set this to a value between 1 and 65535 (1 = 0.01, 100 = 1.00).
Advice of Charge for Telephone Display - Select the decimal point used at the keyphone display.
Advice of Charge for SMDR - Select the decimal point used for the SMDR (call logging output).

Easy Edit Help

Hotline Setup (1470)
Hotlines are numbers that are automatically called when the user goes off hook.

Hotline destination - Enter the number the system will dial when the user goes off hook. Enter up to 24 digits, for external calls enter the trunk access code and for Abbreviated dial calls enter the Abbreviated dial service code plus the location. The extension must also have hotline enabled in Class of Service.
Hotline start timer - Set the delay after the user goes off hook before the hotline call is placed, during the delay the user hears dial tone and can dial to override the hotline call. If the time is set to 0 the hotline call is placed immediately.

Easy Edit Help

Service Tone Timer (1471)
Setup the durations for tones the users will hear when using system features. All times are in Seconds.

Extension dial tone - The duration the system will send dial tone when a user goes off hook, the user will then hear Busy tone.
Busy Tone - The duration the system will send busy tone when a user calls a busy extension, the user will then hear lock out tone tone.
Congestion Tone - The duration the system will send congestion tone whe a user can not place a call due to no system resource available.
Warning Tone - The duration the system will send Warning tone when a user dials an invalid number, the user will then hear lock out tone.
Confirmation Tone - The duration the system will send Confirmation tone for example when a user cancels a feature.
Interval of call waiting tone - The interval between Call Waiting tones.
Intrusion tone - The interval between Intrusion tones.
Conference tone interval - The interval between Conference tones.
Warning beep tone signalling interval - The interval between Warning tones for trunks if enabled in Trunk Basic Data Setup.

Easy Edit Help

SLT Basic Setup (1472)
Setup the options for each Single Line Telephone.

DP/DTMF - Select the type of dialling that the SLT port will accept. The DTMF setting will accept both DTMF and Dial Pulse digits. The Dial Pulse setting will only accept Dial Pulse digits.
Terminal Type - Set to special if you want to send DTMF tones from a keyphone to the SLT port during an internal call. Also set to special if you are connecting an external Voice Mail system.
Flashing - Enable/disable the detection of hook flash or Timed Break Recall (TBR).
Caller ID function - Enable/disable the sending of Caller ID to the telephone.
Caller Name - If Caller ID is enabled the system can also send the name.
Caller ID type - If Caller ID is enabled you can select either DTMF or FSK as the sending type.
Fixed Cadence - If the cadence (ringing pattern) is fixed the system will only send single ring to the telephone (1 second on 2 seconds off). If set to Normal the ring pattern will be defined for each type of call in Ring Pattern.

Caller ID on Analogue VM Extension Ports
Any analogue voicemail connected to a telephone system via extension ports uses DTMF Tones (in-band signalling) to communicate information between them.
It is important therefore that Caller ID is disabled for these voicemail extension ports.
This is because the FSK (Frequency Shift Keying) audio signaling method used on the Aspire, XN120 and other systems to send the Caller ID to an analogue handset can be detected by the analogue voicemail.
This can lead to erratic operation of the voicemail.
This applies equally whether a third party telephone system or voicemail is connected to NEC Infrontia equipment, or whether only NEC Infrontia equipment is concerned.
The relevant command both on the Aspire and XN120 is 15-03 Item 9, "Caller ID Function" and this should be set as Disabled for all voicemail extension ports.
It is important to note that the XN120 Quick Install sets Caller ID to Enabled on all Single Line Telephone ports.
This will not be changed as the integral voicemail is more commonly used on the XN120
In summary any extension ports used for connection of an analogue voicemail must be checked to ensure Caller ID is not in use.

Easy Edit Help

SLT Options (1473)
Setup the options that are common to all Single Line Telephones.

Call Waiting answer method (SLT only) - Select the operation that an SLT user must use to answer a Call Waiting.
Hooking = The SLT presses Recall (or Hook Flash) to place the current call on hold and answer the waiting call.
Hooking = Service Code = The SLT presses Recall (or Hook Flash) and dials the Call Waiting Answer service code (894) to place the current call on hold and answer the waiting call.
Ignore DP at DTMF port - You can choose to ignore dial pulse digits at any DTMF port. The DTMF ports are setup in SLT Basic Setup.
SLT DTMF Dial to trunk lines - Select the method that digits are dialled to trunk for calls made from an SLT port.
Store + Forward = The digits dialled by the SLT are stored by the system and dialled to the trunk after a pause time. The pause time is measured for the digits dialled by the SLT.
Direct = The digits dialled by the SLT are not stored by the system, they are dialled to line immediately.
Trunk call dial sending start time by SLT - If the SLT DTMF dial to trunk lines option is set to Store+Forward then the pause time is set by this option. When the SLT leaves a pause of this duration in their dialing then the system will process all digits received.
SLT Operation Mode - Select the operation for SLT ports when the Recall key (Hook Flash) is pressed while they are placing an outgoing call and also have a call on hold. It is recommended that you set this option to Extended Mode 2.
Normal Mode = The outgoing call is placed on hold and the SLT is reconnected to the held call.
Extended Mode 1 = Do not select this option
Extended Mode 2 = The outgoing call is cleared and the SLT is reconnected to the held call. This gives the SLT user the option to clear the outgoing call if busy tone is received, alternatively the SLT user can go on hook to transfer the held call.
Headset ringing start timer (for SLT) - Set the duration that the SLT will receive dial tone before headset mode begins. The SLT must also have the Restriction of Headset Earpice Ringing option in Class of Service set to OFF (Headset earpiece ringing enabled).

Easy Edit Help

Outgoing Call (1474)
Setup the options common to all telephones that are related to outgoing calls.

Seizure Trunk line mode - Select the type of trunk seizure within each trunk group for outgoing trunk calls.
Priority Route = The trunks are seized in the priority order defined for each trunk within the trunk group. Each trunk is seized in order, the next trunk seized may be the same trunk if it is idle.
Circular Route = The trunks are seized in priority order. Each trunk is seized in circular order, the next trunk seized is the next priority order.
Internal Call Inter-digit time - The Interdigit time (Seconds) when placing an internal call.
External Call Inter-digit time - The Interdigit time (Seconds) when placing a trunk call. When this timer expires the trunk call will be classed in the answered state and the call timer will start.
Dial Tone Detection - The duration the system uses to measure dial tone. This timer can be left at default (5 Seconds).
Disconnect time when dial tone not detected - If dial tone detection is enabled for the analogue trunk port (in Analogue Trunk Data Setup) this time is used to set the duration (Seconds) the system will wait for dial tone. If dial tone is not detected during this time the line will be disconnected. This timer will be cancelled if digits are dialled out so you must set this timer to a value equal to or less than the Dial Pause at 1st digit timer. Set this timer to 0 if you do not want to disconnect the line. It is recommended that this timer is set to 3 or higher as the system needs at least 1.5 seconds to detect dial tone.
Dial Pause at 1st digit - Set the pause duration (Seconds) before the first digit is dialled out on an analogue trunk. If dial tone is detected on the trunk this timer is not used, the digits will be dialled to line when dial tone is detected. If dial tone is not detected on the trunk (or is disabled) the digits will be dialled to line when this timer expires.
Toll Restriction Override release - This time sets the duration (Seconds) the user has to to place the outgoing trunk call if they are using the Toll Restriction Override service.
Preset dial display hold time - Sets the duration (Seconds) the display of the keyphone will wait when the user preview dials a number.
Hotline call start time - How long the system waits (Seconds) after the user goes off hook and the Hotline call is placed.
Dial digits for Toll Restriction path control - How many digits must be dialled from the keyphone's own keypad before the system will connect the transmit speech path to the trunk. Used to prevent the user placing a DTMF tone dialler at the microphone to bypass the system Toll Restrictions.
Interdigit time for Toll Restriction path control - How long the system will wait before connecting the transmit speech path to the trunk. Set this timer greater than the duration (Seconds) that the Network will supply dial tone.
Forced Account Code interdigit time - The system waits this time for the user to start to enter and between each digit when Forced Account Codes are enabled.
Outgoing Disable on incoming line - Turn on/off the Outgoing disable on incoming line feature. This feature can also be enabled/ disabled for each extension in Extension Basic Setup.
Outgoing Disable on incoming line - Timer - Set how long (Seconds) the system will detect DTMF digits for Outgoing Disable on incoming line feature.
Outgoing Disable on incoming line - Digits - How many DTMF digits the system will allow to be dialled to line before the trunk is disconnected. This is usually set to 5 digits or less digits to allow users to access dial up services but not able to dial a complete Network number.

Easy Edit Help

SLT Data Setup (1475)
Setup the system data for SLT ports.
Minimum Break Time - The minimum duration of a dial pulse break (default=10mS)
Maximum Break Time - The maximum duration of a dial pulse break (default=100mS). For Time Break Recall detection set this timer to 13 (65mS). Do not set this item to less than 13 (65mS).
Minimum Make Time - The minimum duration of a dial pulse make (default=10mS)
Maximum Make Time - The maximum duration of a dial pulse make (default=100mS)
Minimum Hook Flash Time - The minimum duration of a Hook Flash/Time Break Recall TBR (default=105mS). For Time Break Recall detection set this timer to 14 (70mS).
Maximum Hook Flash Time - The maximum duration of a Hook Flash/Time Break Recall TBR (default=1000mS). For Time Break Recall detection set this timer to 25 (125mS)

Easy Edit Help

DSS Console Assignment (1476)
If you have more than one console you must locate it with the correct keyphone in the system configuration and on the users desk!. When you connect the DSS console to the extension port the system will automatically give it a console number, refer to Port Setup for the DSS console number.

DSS Console - The console number.
Associated Extn - The extension number of the keyphone that the console will be used with. You can assign more than one DSS console to the same extension number.
DSS Operation Mode - Select the mode of the DSS console. This will select the type of extensions that will show lamp flash patterns for the keys (for example, a Hotel DSS console will show Hotel room extension status).

Easy Edit Help

DSS Lamp Table (1477)
The busy lamp indication for DSS keys can be changed for each state of the telephone. This setting also applies to programmable function keys set as DSS type.

Lamp Pattern Data - Enter the lamp flash pattern number from the list below.

0= Off
1= On:250mS, Off:250mS (fast flash)
2= On:500mS, Off:500mS (slow flash)
3= On:875mS, Off:125mS (long flash)
4= Same as pattern 1
5= On:125mS, Off:125mS, On:125mS, Off:625mS (double flash)
6= Same as pattern 1
7= On

Easy Edit Help

Installation date (1480)
This is for information only to indicate the date of the installation.

Easy Edit Help

Trunk Busy Control (1482)
Trunk Busy control can block the trunk port for outgoing calls, the trunk will still respond to incoming calls. If the trunk is in use when the block is attempted the system will wait for the call to end before blocking the trunk.

Set Condition - Select the status of the trunk port.
Busy Out When Idle - The trunk will block outgoing calls.
Normal (reset busy condition) - The trunk will allow outgoing calls.

Easy Edit Help

System Alarm Setup (1484)
The system can report alarms to either PCPro, the serial port of the EXIFU card or an e-mail address.
To view alarms with PCPro select Advanced/Maintenance/Alarm while connected to the system.
To setup the Serial port use System report COM Setup.
To setup e-mail notification use System Alarm E-Mail Setup.

Alarm Description - The cause of the system alarm.
Alarm Status - Select the status for the alarm (Major or Minor), there is no priority assigned to the status it is for information only when viewing the report. Set to 'No Alarm Set' to stop the alarm being reported.
Report Type - The alarm can be e-mailed by the system (if suitable equipment is connected, available and setup in System Alarm E-Mail Setup).
No Report = The alarm will not be e-mailed. It will be reported by PCPro or serial port.
Output Report = The alarm will be e-mailed. It will also be reported by PCPro or serial port.

Easy Edit Help

System Alarm E-mail Setup (1485)
The system can report an alarm by sending an e-mail. This will require a connection to an SMTP server for delivery of the e-mail to its destination address.
Note - If e-mail reporting is enabled the system will not report the alarms to the serial port.
Alarm reports will be sent automatically when they occur. The system can also send a report that contains all alarms, use System Alarm Report Notification Time Setup to set the notification dates.

System Alarm Display Telephone - Enter the extension number of the display keyphone that will show the following two alarms only.
Low Battery! = The systems's memory backup battery is discharged or not installed.
SMDR n Full! = The buffer is full for SMDR port n, the external device is either disconnected or faulty.
Report Method - Set to E-Mail to turn on the e-mail reporting.
SMTP Host Name - If there is no DNS server enter the IP address of the SMTP host, otherwise enter the machine name.
SMTP Host Port - Enter the SMTP port number (default=25).
To Address - Enter the destination e-mail address.
CC Address 1-5 - Enter optional e-mail CC address if required.
Reply Address - Enter the optional reply e-mail address.
From Address - Enter the 'From' address that will be inserted into the e-mail header.
DNS Primary Address - Enter the optional DNS primary IP address.
DNS Secondary Address - Enter the optional DNS secondary IP address.
Customer Name - Enter the name of the customer, this will be inserted into the 'From' address in the e-mail header.

Easy Edit Help

System Report COM Setup (1486)

Alarms Output Port Type - The system alarms can be output to the serial interface of the EXIFU card. Alarms are output as they occur if set to automatic in 'Output Mode'.
Traffic Report Output - The system can generate a summary report of the traffic for extensions and trunks on the system. The traffic report is setup in Traffic Report Data Setup. The report can be viewed with PCPro while connected to the system or output to the serial port of the EXIFU every hour.
Information Output Port Type - The system can generate a report of the cards installed in the system. The report can only be generated by entering PRG90-13-03 at a display keyphone. When using PCPro the same information can be found in the Cards section.
Output Mode - The alarm reports can be output automatically as they occur or manually. When set to manual the alarm report is output using PRG90-12-03 or 04 at a display keyphone. When using PCPro the same information can be found in the Alarm Report (select Advanced/Maintenance/Alarm from the toolbar).

Easy Edit Help

Main Software and Firmware (1487)
Shows the version of firmware on the various parts of the system.

Software Information - The version of software loaded onto the Main Unit CPU.
Firmware Version - The version of firmware loaded onto each of the cards.

Easy Edit Help

Traffic Report data Setup (1488)
The system can generate a summary report of the traffic for extensions and trunks on the system. The report can be viewed with PCPro while connected to the system or output to the serial port of the EXIFU card every hour if set in System report COM Setup.

Call Traffic Output - Turn the Traffic Reporting on/off. The Traffic Report can be viewed with PCPro by selecting Advanced/ Maintenance/Traffic from the toolbar.
All Line Busy Output - Will show when all lines in any trunk group are busy. Enter the quantity of trunks that must be busy in any trunk group to trigger the output, enter 0 to turn off the output. The All Line Busy report can be viewed with PCPro by selecting Advanced/ Maintenance/Resource from the toolbar.

Easy Edit Help

System Alarm Report Notification Time Setup (1490)
The system can generate a summary e-mail report that will be sent to the address in System Alarm E-Mail Setup.
For each month enter the date and time the system will send the summary report.

Easy Edit Help

Loop Key (1491)
Loop keys are used for general purpose keys for for trunk calls if you do not have a CO key for the trunk being used. The loop key is also used when the trunk call is on hold.
The loop key can be used for either incoming only, outgoing only or both incoming and outgoing calls. The trunk group accessed is specified for each key.
For both incoming and outgoing operation specify the trunk group in both options.

Group Out option - The trunk group that a trunk will be picked from for outgoing calls.
Group In option - The trunk group that will be shown for ringing calls.

Easy Edit Help

Virtual Extension Options (1492)

Option when Answered - The virtual extension can be released or made busy while a call is in progress. When released the virtual extension is available to further calls. The operation of the virtual extension function keys can be further selected by the Virtual Mode option in Keyphone Options.
This option must be set to 'Release VE after Answer' if you want to use the related option in Virtual Extension Enhanced Setup.
Call Coverage Delay Interval - The delay before the Virtual extension will ring when set to Delayed Ringing.

Easy Edit Help

DLS Key Programming (1498) The DLS console provides an additional 24 Programmable Function Keys for the keyphone.

Extn No - The extension number of the keyphone that the DLS console is connected to.
CODE - The Function Key Code, select from the list of available functions.
Add Data - Certain function key types require additional data, for example to define the extension number or voice mailbox number.

Easy Edit Help

Class of Service Options (1502)
Each extension is assigned a Class of Service (COS) number in the Class of Service per Night mode screen within the Extension Properties menu.

Class 1-15 - There are 15 classes available.
Option - The feature that is enable/disabled for each class of service.
To assist in finding an option you can click a column header to list the entries in alphbetical order.

Easy Edit Help

System Numbering (1503)
All numbers processed by the system are defined in this screen.
The system will define the type of number by either the first digit or first two digits dialled, this is shown in the Dial column.
A number defined by the first digit will ignore any entries you make for the first and second digits
For each entry you must define how many digits the system will need and what type the number will be used for.

For example, if you want to have all numbers beginning with 1 to be two digit extension numbers then set as shown below.

Dial	How many digits	Type
1xx	2	Extension number

If you want to have all numbers beginning with 31 to be two digit extension numbers but all numbers beginning with 32 to be four digits then set as shown below.

Dial	How many digits	Type
3xx	0	Not used
31xx	2	Extension number
32xx	4	Extension number

The default system numbering plan is for the UK:
(x is any digit)

Dial	How many digits	Type
1xx	3	Extension number
2xx	3	Extension number
3xx	3	Extension number
4xx	3	Extension number
5xx	3	Extension number
6xx	3	Extension number
7xx	3	Service code
8xx	3	Service code
9	1	Trunk route access
0	1	Operator access
*xx	4	Service code
#xx	4	Service code

The default system numbering plan is for the other areas of Europe:
(x is any digit)

Dial	How many digits	Type
1xx	3	Extension number
2xx	3	Extension number
3xx	3	Extension number
4xx	3	Extension number
5xx	3	Extension number
6xx	3	Extension number
7xx	3	Service code
8xx	3	Service code
9	1	Operator access
0	1	Trunk route access
*xx	4	Service code
#xx	4	Service code

Dial - The first digit only entries are shown by 1xx, 2xx, 3xx etc.
The first and second digit entries are shown by 11x, 12x, 13x etc.
If you want to use any first and second digit entries you MUST set the corresponding single digit entry to Number of digits=0 and Type=Not used.
See the examples above.
Number of digits - This is how many digits the system will wait for before starting to route the call.
Type - The destination for the numbers dialled eg Extension number, trunk access etc.

Easy Edit Help

Trunk Access Code (1504)
There are two trunk access codes available on the system, each code has seperate trunk routes for seizing an outgoing trunk. Refer to Trunk Group Routing for each trunk access code.

Trunk Access Code - The standard trunk access code for the system.
The same digit entered here must also be set as Trunk type in System Numbering.
The trunk route is defined in Trunk Group Routing for Extensions.
Individual Trunk Access Code - The second trunk access code for the system.
The same digit entered here must also be set as Individual Trunk type in System Numbering.
The trunk route is defined in Alternate Trunk Group Routing for Extensions.

Easy Edit Help

System Options for Keyphones (1505)
Select the options that affect all keyphones on the system.

Trunk Group Key Operation Mode -
Keep Lamp = The lamp will remain on while a call is made using the Trunk Group key.
Extinction = The lamp will go off while a call is made or placed on hold when using the Trunk Group key, it will then show further calls ringing while you are on a call.
Trunk Group Access Key Operation Mode - Select the mode of the trunk group keys when the user presses the key: Outgoing/ Incoming will answer a ringing call or seize a free trunk to place an outgoing call, Outgoing will only seize a free trunk to place an outgoing call and Incoming will only answer ringing calls within the trunk group.
Retrieve Line After Transfer - Allows a user to retrieve a call after being transferred, but before it's answered, by pressing the line key for the held trunk.
Pre-Selection time - If the keyphone is set to Pre-Selection mode in Keyphone Options this timer sets how long the system will remember the pre-selection for Function keys set as DSS/One touch keys.
Date-Time Display Mode - Select the clock mode displayed at the keyphone.
1 = 12Hour clock. Default setting.
5 = 24Hour clock.
Disconnect Supervision - Forces the keyphone to idle when the called party clears down, single line telephones will receive dial tone. With this option disabled the keyphone and single line telephone will receive busy tone when the called party clears down.
Incoming Call from Extension Mode - Select the default operation of keyphones when they receive an internal call. Voice calling allows the called extension to talk-back to the caller. Signal is normal ringing tone, the caller must go off hook to answer the call. The user can change this setting by service code 821 to select Voice calling mode or 823 to select Signal mode.
You can also prevent the caller changing from Signal to Voice calling with Class of Service option Signal/Voice Call Switching.

Easy Edit Help

Ring Pattern (1506)
The ring patterns define the on/off pattern for the various type of call. These are used for both keyphones and single line telephones (SLT's must have the Fixed Cadence option set to Normal in SLT Basic Setup).
If the SLT has Caller ID enabled in SLT Basic Setup then ringing will be fixed to pattern 3 for all calls.

There are 13 patterns available: All times are in Seconds.
1 = On (continuous)
2 = On:2.0 / Off:4.0
3 = On:1.0 / Off:2.0
4 = On:0.5 / Off:0.5
5 = On:0.25 / Off:0.25
6 = On:0.5 / Off:0.5 / On:0.5 / Off:1.5
7 = On:0.25 / Off:0.25 / On:0.25 / Off:5.25
8 = On:0.375 / Off:0.25 / On:0.375 / Off:2.0
9 = On:0.25 / Off:0.125 / On:0.25 / Off:0.125 / On:0.25 / Off:2.0
10 = On:1.0 / Off:4.0
11 = On:0.25 / Off:0.25 / On:0.25 / Off:4.25
12 = On:1.0 / Off:3.0
13 = On:0.25 / Off:0.25 / On:0.25 / Off:2.25

Single ring incoming call on trunk - Used for ISDN non-DDI trunks and analogue trunks when a single ring pattern is detected.
Double ring incoming call on trunk - Used for analogue trunks when a double ring pattern is detected.
Internal incoming call - Used for internal calls.
DUD/DISA incoming call - Used for calls via DUD/DISA.
DDI/DID - Used for ISDN DDI calls
E&M Tie line - Used for Tie line calls
Doorphone ringing for SLT - Used for doorphone calls to an SLT.
Virtual extension ring - Used when a virtual extension is ring at the keyphone.
Call-Back - Used when a Call-Back (Camp-On) is returned.
Alarm Clock - Used at the keyphone when the alarm clock is activated.

Easy Edit Help

Virtual Extension Ringing Option (1507)
You can select the ring tone for calls to the virtual extension. There are 5 tone patterns available.

Ring PTTN - Select the ring pattern.

Easy Edit Help

Operator Setup (1510) Operators are any extension that you want extension users to ring when they dial the Operator extension number set in System Numbering.

Operator - Operator number used to route calls to operators if Step Call Access Mode is selected.
Operators Ext - The extension number of the telephone that will receive Operator calls.
Operators Access Mode - If there are two or more operator extensions you can select the method that calls are distributed:
Step Call = Each Operator call will ring the operator extensions in the Operator number order.
Circular = Each Operator call will ring at a new operator extension.

Easy Edit Help

Text Messages (1511) Text messages are available for display keyphones to inform other keyphone users that they are not available. The user can leave the time or contact number if they wish.
There are 10 pre-set messages, you can edit these.
To allow the user to enter a digit when they are setting the Text Message you must enter the # character for each digit in this option.
Calls to the extension with a Text Message set can either ring or return busy indication.

Message Number - The number used to select the Text Message when the user sets the feature.
Message Data - The text displayed to the caller. Enter # where you want the user to enter their own numbers.
Text Message Mode - Should calls to the extension ring or return busy indication.

Easy Edit Help

System Timers (1514)
General system timers. All settings are in Seconds. To disable a timer set it to 0.

DTMF receive active time - The duration that the system will detect DTMF digits for Analogue extensions and trunks that require DTMF detection, for example DISA trunks.
Alarm clock duration - The duration that the extension alarm will ring for.
Camp-On extension call back time - When an extension Camp-On rings back it will ring for this duration.
Camp-On trunk call back time - When a trunk Camp-On rings back it will ring for this duration.
Camp-On cancel time - Any Camp-On that is not completed will be cancelled after this time.
Trunk Guard time - The duration the system will prevent users seizing an analogue trunk after it has been cleared, during the guard time the CO key will show red (busy). This gives the Network time to clear the circuit.
Long Conversation alarm 1 (until sending 1st alarm) - The duration a trunk call must be in progress for before the warning tone is played if enabled in Class of Service.
Long Conversation alarm 2 (until sending next alarm) - The interval between each warning tone.
Long conversation cut off for incoming - The duration an incoming trunk call must be in progress for before the call is disconnected

if enabled in Class of Service and Trunk Basic Data Setup.
Long conversation cut off for outgoing - The duration an outgoing trunk call must be in progress for before the call is disconnected if enabled in Class of Service and Trunk Basic Data Setup.
Internal Call Inter-digit time - The Interdigit time when placing an internal call.
External Call Inter-digit time - The Interdigit time when placing a trunk call. When this timer expires the trunk call will be classed in the answered state and the call timer will start.
Dial Tone Detection - The duration the system uses to measure dial tone. This timer can be left at default (5 Seconds).
Disconnect time when dial tone not detected - If dial tone detection is enabled for the analogue trunk port (in Analogue Trunk Data Setup) this time is used to set the duration the system will wait for dial tone. If dial tone is not detected during this time the line will be disconnected. This timer will be cancelled if digits are dialled out so you must set this timer to a value equal to or less than the Dial Pause at 1st digit timer. Set this timer to 0 if you do not want to disconnect the line.
Dial Pause at 1st digit - Set the pause duration before the first digit is dialled out on an analogue trunk. If dial tone is detected on the trunk this timer is not used, the digits will be dialled to line when dial tone is detected. If dial tone is not detected on the trunk (or is disabled) the digits will be dialled to line when this timer expires.
Toll Restriction Override release - This time sets the duration the user has to to place the outgoing trunk call if they are using the Toll Restriction Override service.
Preset dial display hold time - Sets the duration the display of the keyphone will wait when the user preview dials a number.
Hotline call start time - How long the system waits after the user goes off hook and the Hotline call is placed.
Dial digits for Toll Restriction path control - How many digits must be dialled from the keyphone's own keypad before the system will connect the transmit speech path to the trunk. Used to prevent the user placing a DTMF tone dialler at the microphone to bypass the system Toll Restrictions.
Interdigit time for Toll Restriction path control - How long the system will wait before connecting the transmit speech path to the trunk. Set this timer greater than the duration that the Network will supply dial tone.
Forced Account Code interdigit time - The system waits this time for the user to start to enter and between each digit when Forced Account Codes are enabled.
Outgoing Disable on incoming line - Turn on/off the Outgoing disable on incoming line feature. This feature can also be enabled/ disabled for each extension in Extension Basic Setup.
Outgoing Disable on incoming line - Timer - Set how long the system will detect DTMF digits for Outgoing Disable on incoming line feature.
Outgoing Disable on incoming line - Digits - How many DTMF digits the system will allow to be dialled to line before the trunk is disconnected. This is usually set to 5 digits or less digits to allow users to access dial up services but not able to dial a complete Network number.
Time of Redial - Set how many times (not in Seconds) the system will attempt the Repeat Redial feature.
Interval of Redial - The interval between each Repeat Redial Attempt.
Redial Calling Time - The duration the system will call the outgoing number for the Repeat Redial feature.
ISDN Calling Party Busy Tone - The duration the system will send busy tone to the extension if the Repeat redial is placed via an ISDN line and the called party is busy.
Normal Hold recall time - The duration a call must be on hold before it will ring back to the extension that placed the call on hold.
Normal Hold call back time - The duration the ring back tone will ring at the extension.
Exclusive Hold recall time - The duration a call must be on hold before it will ring back to the extension that placed the call on hold.
Exclusive Hold call back time - The duration the ring back tone will ring at the extension.
Long hold condition forced release time - The duration a call must be on hold before it will be disconnected if enabled in Trunk Basic Data Setup.
Park Hold time - The duration a call must be on Park hold before it will ring back to the extension that placed the call on hold if the extension's Class of Service option for Normal/Extended Park Hold is set to off.
Park Hold time extension - The duration a call must be on Park hold before it will ring back to the extension that placed the call on hold if the extension's Class of Service option for Normal/Extended Park Hold is set to on.
No answer time for call forward - Set the duration the call will ring before being forwarded when the user has set Call Forward No Answer.
Ring Inward Recall time - Set the duration that a call will ring after being transferred by a user. After this time the call will ring back to the extension that performed the transfer.
Ring Inward to busy Extension Group - Set the duration that a call will queue after being transferred to a busy Department Group. After this time the call will ring back to the extension that performed the transfer.
Trunk to Trunk Transfer Warning Tone time - Set the duration before the trunk to trunk warning tone will be played. After this time the caller can dial the continue code set in Trunk to Trunk Routing.
Extension Group delayed transfer time - Set the duration that a call will ring at the Department group before it will be forwarded.
Trunk to Trunk Transfer Disconnect Time - Set the duration the system will wait after sending the Trunk to Trunk Transfer Warning Tone before the system will disconnect the call.
No Answer Timer for Step Transfer -
No Answer time for automatic trunk to trunk transfer -

Easy Edit Help

Trunk Access Map for Extension (1515)
Trunk access maps are used to allow/prevent access to trunks for each extension.
The extensions are assigned an access map number for each night mode. Each access map number then has access properties in Trunk Access Map for Extensions.

Mode 1-8 - The access map number for the extension in each night mode.

Easy Edit Help

Trunk Group Routing (1516)
Trunk routes are used when the user dials the trunk access code (trunk access code is set in System Numbering).

Route Table - The route number assigned to the extension. Extensions are assigned a route in each night mode in Trunk Group Routing for Extensions.
Priority 1-4 - Enter the trunk group number that contains the trunks to be seized. Trunk groups are searched in priority order 1 to 4, if all trunks in the trunk group are busy then the trunk group in the next priority will be searched.

Easy Edit Help

Trunk Access Map Setup (1517)
Trunk access maps are used to allow/prevent access to trunks for each extension.
The extensions are assigned an access map number for each night mode in Trunk Access Map for Extensions. Each access map number then has access properties in this screen.

Trunk - The trunk ports are shown down the left column.
Access Map - Each access map number is assigned a property, the property can be different for each trunk port.
No Access = This trunk is not available to any extension with this access map number.
Outgoing Access only = Only outgoing calls can be made.
Incoming Access only = Only incoming calls can be received.
Access only when trunk on hold = An incoming/outgoing trunk that has been placed on hold can be accessed.
Outgoing access when trunk on hold = An outgoing trunk that has been placed on hold can be accessed.
Incoming access when trunk on hold = An incoming trunk that has been placed on hold can be accessed.
Incoming/Outgoing access = An incoming/outgoing trunk can be accessed. A trunk that has been placed on hold can not be accessed.
Incoming/Outgoing access when trunk on hold = Full Access, an incoming/outgoing trunk can be accessed. A trunk that has been placed on hold can also be accessed.

Easy Edit Help

Trunk Group Routing for extensions (1518)
Trunk routes are used when the user dials the trunk access code (trunk access code is set in System Numbering).
The route will then specify the trunk groups(s) to be searched in Trunk Group Routing.
The trunk group will finally specify the trunk(s) to be seized in Trunk Group

Mode 1-8 - The route number assigned to the extension for each night mode.

Easy Edit Help

Alternate Trunk Group Routing for extensions (1519)
Trunk routes are used when the user dials the Alternate trunk access code (Alternate trunk access code is set in System Numbering).
The route will then specify the trunk groups(s) to be searched in Trunk Group Routing.
The trunk group will finally specify the trunk(s) to be seized in Trunk Group

Mode 1-8 - The route number assigned to the extension for each night mode.

Easy Edit Help

Trunk Group (1520)
Trunk groups are used for outgoing trunk access (Trunk Group Routing) and incoming for DDI routing.

Trunk Group - Enter the trunk group number.
Priority - Used for outgoing access the trunk are seized in order from lowest to highest priority. If the priority is duplicated within the same group then the lowest trunk port number will be used.

Easy Edit Help

Outgoing Route Setup (1521)
Trunk routes are used when the system is routing an incoming trunk call directly back out of the system, for example a DDI call routed to an outside number.
The route will then specify the trunk groups(s) to be searched in Trunk Group Routing.
The trunk group will finally specify the trunk(s) to be seized in Trunk Group

Mode 1-8 - The route number assigned to the incoming trunk for each night mode.

Easy Edit Help

Automatic Trunk to Trunk Transfer target (1522)
Specify the Abbreviated dial location that will be used to store the destination number for Trunk to Trunk Forwarding.

Mode 1-8 - The Abbreviated dial location for each incoming trunk.

The function of modes 1 to 8 depend on the setting of 'Automatic Trunk to Trunk Transfer Mode' in Trunk Basic Data Setup.
The mode is either the Night Mode number for 'Normal Transfer' or the sequence number for 'Step Transfer'.
The 'No answer timer for Step Transfer' is set in Trunk to Trunk Options.

Easy Edit Help

Analogue trunk initial data setup (1523)

Setup the system data for analogue trunks.
Clear Signal (Open Loop) Detection - (timer = value x 8mS) The minimum time the system will detect a break in loop current to indicate the trunk has been cleared (Disconnect Clear signal). The trunk must be set to detect the clear signal in Analogue Trunk data Setup. Default = 37 (296mS).

Ringing signal detection time minimum - (timer = value x 8mS) The minimum ringing pulse duration the system will detect to start the incoming call. Default = 13 (104mS).
Single ringing detection minimum - (timer = value x 8mS) The minimum duration of a ring pulse for the system to detect as a single ring pattern. Default = 82 (656mS).
Double ringing detection minimum off time 1 - (timer = value x 8mS) The minimum duration between each ring pulse of a double ring pulse for the system to detect as a double ring pattern. Default = 13 (104mS).
Double ringing detection maximum off time 2 - (timer = value x 8mS) The maximum duration between each ring pulse of a double ring pulse for the system to detect as a double ring pattern. Default = 50 (400mS).
Ringing signal no detection minimum time - (timer = value x 8mS) The minimum duration between the two ring pulses of a double ring pulse for the system to detect as a double ring pattern. Default = 88 (704mS).
Time ringing signal stop detection time - (timer = value x 8mS) The maximum duration between any single or double ring pattern for the system to begin a new incoming call (Ring signal abandon time). Default = 47 (3008mS).
Hook flash time 1 - (timer = value x 16mS) The Hooking duration for trunks set to use Hooking type = Flashing in Analogue Trunk Data Setup. Default = 50 (800mS).
Hook flash time 2 - (timer = value x 16mS) The Hooking duration for trunks set to use Hooking type = Disconnect in Analogue Trunk Data Setup. Default = 156 (2496mS).
Pulse break time (10pps) - (timer = value x 5mS) The break time for 10 pulse per second dial pulse digits. Default = 13 (65mS).
Pulse make time (10pps) - (timer = value x 5mS) The make time for 10 pulse per second dial pulse digits. Default = 7 (35mS).
Inter-digit time (10pps) - (timer = value x 32mS) The inter-digit time for 10 pulse per second dial pulse digits. Default = 19 (608mS).
Pulse break time (20pps) - (timer = value x 5mS) The break time for 20 pulse per second dial pulse digits. Default = 6 (30mS).
Pulse make time (20pps) - (timer = value x 5mS) The make time for 20 pulse per second dial pulse digits. Default = 4 (20mS).
Inter-digit time (20pps) - (timer = value x 32mS) The inter-digit time for 20 pulse per second dial pulse digits. Default = 16 (512mS).

Easy Edit Help

Analogue Trunk data Setup (1524)
Options for analogue trunk ports.

DP/DTMF - Set the dial type for outgoing calls.
Incoming call detection type - When ringing is detected on the analogue trunk:
Normal = The system will wait until it has detected either a single or double ring before presenting the call to the extension.
Ring Immediately = The system will present the call to the extension immediately, the ring pattern may change when the single or double ring pattern is detected.
Hooking type - The duration of the hook flash sent to line can be set to either Flashing (short flash) or Disconnect (long flash).
Flashing = Short flash, the duration is set by Hook flash time 1 in Analogue Trunk Initial Data Setup.
Disconnect = Long flash, the duration is set by Hook flash time 2 in Analogue Trunk Initial Data Setup.
DTD after line seized in manual dial mode - Will the system detect dial tone before dialling out digits for outgoing calls. The timers for dial tone detection are in System Timers. If dial tone detection is not used the system will wait for the Pause at 1st digit time before dialling.
The system will detect any continuous tone as dial tone.
Pause at 1st digit after line seized in manual dial mode - If dial tone detection is not used the system will wait for this duration before dialling the digits to line.
Change DP to DTMF - For lines set to Dial Pulse type you can select when the system will allow the user to send additional digits as DTMF.
Auto = The system will send DTMF digits after the External Call Inter-digit time set in Outgoing Call.
Both = The system will send DTMF digits after the External Call Inter-digit time or when the user dials #.
Manual = The system will send DTMF digits when the user dials #.
Anwering Condition - The system can use line reversal to determine that an outgoing call has been answered or wait for the External Call Inter-digit time set in Outgoing Call.
Caller ID - Will the trunk detect Caller ID.
Outgoing trunk skip on no dial tone - If the trunk is set to detect dial tone the system can automatically try the next trunk when dial tone is not detected.
Will not operate when the user seizes the line directly, the user must dial the trunk access code, Abbreviated dial or last number dial.
The Disconnect time when dial tone not detected in Outgoing Call must be set (3 seconds or higher) to enable the system to disconnect the trunk.
Detect Network disconnect signal - Will the trunk detect the disconnect clear signal to determine the end of the incoming call.
Disconnect clear signal must be supplied by the Network. This option must be enabled for trunk to trunk transfer (also set the trunk to trunk transfer option in Trunk Basic Data Setup).

The minimum Clear Signal (open loop) detection time is set in Analogue Trunk Initial Data Setup.
Caller ID Type - Select the type of caller ID the system will detect on the trunk.

Easy Edit Help

Analogue Trunk Options (1525)

DTMF receive active time - The duration that the system will detect DTMF digits for Analogue extensions and trunks that require DTMF detection, for example DISA trunks.
Camp-On trunk call back time - When a trunk Camp-On rings back it will ring for this duration.
Camp-On cancel time - Any Camp-On that is not completed will be cancelled after this time.
Trunk Guard time - The duration the system will prevent users seizing an analogue trunk after it has been cleared, during the guard time the CO key will show red (busy). This gives the Network time to clear the circuit.

Easy Edit Help

DIL Target (1526)
Direct Inward Lines (DIL) are assigned as the incoming trunk type in Incoming Service Type.
A DIL is routed directly to the extension number and can therefore follow any off-premise call forwards. A DIL can also be routed to a virtual extension numbers or Department Group pilot number. A different target can be assigned for each night mode.
If a DIL is not answered it can step on to a second target after the DIL No Answer Time, the second target is assigned in DIL Step On Target. The second target is not used if a call arrives when the DIL target is busy, the call will route to the IRG set in IRG Assignment (Normal).

Target Mode 1-8 - Enter the extension number, virtual extension number or Department group pilot number.

Easy Edit Help

DIL Step On Target(1527)
Direct Inward Lines (DIL) are assigned as the incoming trunk type in Incoming Service Type.
A DIL is routed directly to the extension number for each night mode in DIL Target.
If a DIL is not answered it can step on to a second target after the DIL No Answer Time. The second target is not used if a call arrives when the DIL target is busy, the call will route to the IRG set in IRG Assignment (Normal).

Target IRG Mode 1-8 - Enter the IRG number for the second target. The members of the IRG are assigned in Incoming Ring Group Setup. Enter 0 if you do not want the call to step on from the DIL Target.

Easy Edit Help

DIL Step On Timer (1528)
If a DIL is not answered it can step on to a second target after this time.

DIL No answer time - Enter the time (Seconds) that the incoming trunk call will ring at the DIL Target, if not answered the call will step on to the DIL Step On Target

Easy Edit Help

Trunk Basic Data Setup (1529)
These settings will effect analogue and ISDN trunks.

Trunk Name - Enter the name that will be shown on the keyphone display when the trunk is seized. Can also be shown on the SMDR printout.
Transmit Gain - The gain from the system to the Network. Used when the trunk is connected to an extension.
Receive Gain - The gain from the Network to the system. Used when the trunk is connected to an extension.
Transmit Gain in Conference and Transfer mode - The gain from the system to the Network. Used when the trunk is part of a multi trunk conference or Trunk to Trunk transfer.
Receive Gain in Conference and Transfer mode - The gain from the Network to the system. Used when the trunk is part of a multi trunk conference or Trunk to Trunk transfer.

(Gain settings: Enter a value from 1 to 63, the gain levels are from -15.5dB to +15.5dB. Each increment in the value is equivalent to 0.5dB. For example, 26=-3dB, 32=0dB and 38=+3dB.)

NOTE;
Should external calls appear faint or echo at any XN120 site with analogue trunks then it is possible to compensate for any issues with the network.

The default settings on the XN120 are:
Program 81-07-01 is set to 0 for each analogue trunk port
Program 82-07-01 is set to 0 for each extension port
These set the filters to “not used” so that they will not effect the gain.

Altering the gain of the trunks can then make further improvement to the external volume for the XN120 users.
Program 14-01-02 (TX gain XN120 -> trunk) is set at default to -3dB (26).
Program 14-01-03 (RX gain trunk -> XN120) is set at default to +3dB (38).

Please note that although the above are the factory settings Commands 14-01-02 and 14-01-03 can of course be varied according to the site conditions.

Note however that the TX and RX gain must be balanced (a gain in one direction should have an equal loss in the opposite direction otherwise the side tone will be louder) however please be careful not to set the RX gain too high (and therefore the TX has more loss to balance the side tone) as the trunk party may then report the calls are too quiet

Due to all Analogue trunks being different, padding of the Analogue Trunks in PRG 81-07 and PRG 14-01 may be necessary. Even after the pad changes are made, an echo may still be present the first few seconds of the call while echo cancellers are learning the characteristics of the circuit on this call.
- For best performance, it is recommended to use digital trunks when using IP Terminals.

SMDR Printout - Should calls on the trunk be output to the call logger (SMDR).
Outgoing - Can outgoing calls be made on the trunk.
Toll Restriction - Will outgoing calls be subject to Toll Restriction settings.
DTMF tone for outgoing calls - Give confirmation tone to the user when DTMF digits are dialled.
Account Code - Will account codes be used on the trunk. Account codes are setup in the Account Codes section with SMDR.
Trunk to Trunk Transfer - Can the trunk be transferred to another trunk. Analogue trunks must also have Detect Network Disconnect Signal enabled in Analogue Trunk Data Setup.
Long Conversation cut-off - Will long conversations be disconnected. The timer is set in System Timers, the extension must also have the Outgoing or Incoming Long Conversation cut-off enabled in Class of Service.
Long Conversation Alarm before cut-off - Play a warning tone to the caller before the line is disconnected. The timers are set in System Timers, the extension must also have the Long Conversation alarm enabled in Class of Service.
Long time holding forced disconnect - Will calls placed on hold be disconnected if they are not retrieved. The timer is set in System Timers.
Trunk to Trunk Long Conversation Alarm - Play a warning tone to the callers during a trunk to trunk connection, the tone is not repeated. The timer is set in Hold and Transfer.
Warning Beep tone - Play a tone to the callers for incoming calls, the tone is repeated. The timer is set in Service Tone Timers.
Privacy mode toggle option - Privacy Release allows others extensions to conference into the trunk call by pressing the flashing line key.
Block Outgoing Caller ID - The system can dial the Caller ID Block Code before the dialled digits. The Block Code must be recognised by the Network for the code to disable the sending of Caller ID to the called party otherwise the calls may mis-dial. The

extension must also be enabled in Class of Service.
Caller ID Block Code - Enter up to 8 digits that will be dialled by the system.
Caller ID to Voice Mail - Send the Caller ID received on incoming calls to the voice mail, when the trunk is answered by the external voice mail system (Caller ID is not sent to the DSPDB voice mail).
Least Cost Routing - Should outgoing calls be routed to an indirect carrier by using the least cost routing tables within the system. The least cost routing tables are setup in the LCR section. This option also allows the selection of just the Cost Centre Code being sent to line before the dialled digits, this setting will not route via an indirect carrier.
Trunk to trunk outgoing caller ID through mode - When an incoming call on an ISDN trunk is routed out over another ISDN trunk the system can pass through the received caller ID to the outgoing call.
Continue/Disconnect trunk to trunk conversation - If Long Conversation Cut-off is enabled the system can allow the caller to dial a DTMF digit to delay the cut-off and continue the call. The caller will hear a warning tone before the call is disconnected prompting them to dial the continue code. The continue code and delay time are setup in Trunk to Trunk Data Setup. This option is also required for DISA trunk to trunk conversation continue/disconnect operation.

Easy Edit Help

Incoming Service Type (1530)
Select the type of the trunk port for incoming calls. The type is set for each night mode.

Type - Select from the list.
Normal = Can be set for analogue and ISDN non-DDI trunks. The incoming call will be routed to the Incoming Ring Group (IRG) set in IRG Assignment in the Ring Groups section.
DUD = Direct Universal Dialing. Can be set for analogue and ISDN non-DDI trunks. The incoming call will be routed to the Voice Response System for Automated Attendant type operation (DSPDB card required to play the greeting message otherwise DUD/DISA dial tone will be played). The DSPDB greeting message is assigned in DUD/DISA Talkie. Other options are setup in the Auto Attendant section.
DISA = Dial In System Access. Can be set for analogue and ISDN non-DDI trunks, DISA has options to give the caller access to system features and outgoing trunks. The incoming call will be routed to the Voice Response System for Automated Attendant type operation (DSPDB card required to play the greeting message otherwise DUD/DISA dial tone will be played). The DSPDB greeting message is assigned in DUD/DISA Talkie. Other options are setup in the Auto Attendant section.
DID = Direct Inward Dial. Can be set for ISDN trunks only with Direct Dial In (DDI) service. The incoming call will routed via the DDI table in the DDI section.
DIL = Direct Inward Line. Can be set for analogue and ISDN non-DDI trunks. Direct Inward Lines (DIL) can be routed directly to an extension, virtual or Department group pilot number via DIL Target Assignment.
Leased Line = Tie Line setting - used for SIP trunks. Follows settings of system numbering and f-route.
Delayed DUD = Can be set for analogue and ISDN non-DDI trunks. The incoming call will be routed to the Incoming Ring Group (IRG) set in IRG Assignment, if the call is not answered it will step on to DUD type and be routed to the Voice Response System for Automated Attendant type operation (DSPDB card required to play the greeting message otherwise DUD/DISA dial tone will be played). The VRS greeting message is assigned in DUD/DISA Talkie. Other options are setup in the Auto Attendant section.
The Delayed DUD Answer timer is set in System Timer for DUD/DISA Service.
DDI Time Mode = DDI operation for time of day. Can be set for ISDN lines with Direct Dial In (DDI) service. The incoming call will be checked against the entries in the DDI Time Mode Table.

Easy Edit Help

Trunk Ring Tone setup (1531)
Select the ring tone for each trunk when ringing at a keyphone. The ring pattern (on/off time) are set set in Ring Pattern in the Extensions section.

Ring Tone - Select from the list.
Tone Pattern 1 = 600/450Hz with 16Hz modulation.
Tone Pattern 2 = 450Hz with 16Hz modulation.
Tone Pattern 3 = 600Hz.

Easy Edit Help

DDI Routing Table (1532)
The trunks are set as DDI type in Incoming Service Type Setup.
DDI trunks are placed into a trunk group in Trunk Group, the trunk group number is used to route to the DDI table area for each night mode.
The DDI Table Area is defined in DDI Table Area Setup. This defines the DDI Table entries that will be searched.
Each DDI table entry has up to three targets and options defined in this section.

Table No - The DDI Table entry number.
Received DDI - The DDI number received from the Network, enter up to 8 digits.
DDI Name - The name of the DDI, enter up to 12 characters.
Target 1 - The first target for the DDI call, enter an extension number, virtual extension number or Department Group pilot number.
You can also route to an external number by entering the trunk access code followed by the external number. An outgoing trunk will be seized in the route defined by Outgoing Route Setup.
Transfer Option - You can specify the operation of the DDI call if the target is busy or does not answer. The no-answer timers for each type of call are specified in Incoming Call.
No Transfer = The call will return busy indication or continue to ring as appropriate.
Busy = If the target is busy the call will step on to the next target. If the target does not answer the call will continue to ring, it will not step on.
No Answer = If the target does not answer the call will step on to the next target after the DID No Answer Time in Incoming Call. If the target is busy the call will return busy indication, it will not step on.
Busy/No Answer = If the target does not answer the call will step on to the next target after the DID No Answer Time in Incoming Call. If the target is busy the call will step on to the next target.
Target 2 - The second target for the DDI Call. The following can be entered:
None (0). The target is not used, the call will automatically try the next target.
Incoming Ring Group number (1-25). The IRG members are shown in Incoming Ring Group Setup.
DSPDB Voice Mail (101). The call will be answered by the mail box defined in the Mail Box Entry for Message in Voice Mail Automated Attendant Data Setup for the caller to leave a message. Note this mail box is defined for the trunk port so must be defined for all trunk ports within the trunk group for the DDI.
External Voice Mail (102). The call will be answered by the external voice mail system setup in Voice Mail External.
Department Group Number (201-232) 201 is group 01, 202 is group 02. The Department Group members are shown in Department Group Assignment.
DUD (400). The call will be routed to DUD Auto Attendant. The greeting message is defined in DUD/DISA Talkie. Note this greeting message is defined for the trunk port so must be defined for all trunk ports within the trunk group for the DDI.
DISA (401). The call will be routed to DISA Auto Attendant. The greeting message is defined in DUD/DISA Talkie. Note this greeting message is defined for the trunk port so must be defined for all trunk ports within the trunk group for the DDI.

New for v6.00 - VRS messages (501-548), directly targets VRS messages for DISA or Auto Attendant. To target message 2 enter 502. To enable this feature for Auto Attendant, disable DISA User ID for ISDN trunks in DISA Basic Options.

New for v6.00 - DSPDB mailboxes (601-699), directly targets first 100 DSPDB mailboxes as assigned in Mail Box Setup. To target mailbox 2 enter 602.

Abbreviated Dial (1000-1999) 1000 is Abbreviated Dial 000, 1001 is Abbreviated Dial 001. The call will be routed via another trunk port to the Abbreviated Dial number. The trunk route is specified in Outgoing Route Setup.
Target 3 - The third target for the DDI Call. The same entries shown for the second target are available.

Call Waiting - The DDI call can wait (DDI Camp-On) at the busy extension set in Target 1. If the No Answer Transfer option is set the call will step on if the extension does not become available.
Max number of calls - Specify the maximum quantity of un-answered calls the system will queue at any of the targets. Set to 0 if you do not want to limit the maximum quantity.
MOH Type - The system can play music on hold to incoming callers. Outgoing calls will use the MOH type defined for the trunk port in Outgoing MOH Source in the MOH/BGM section.
MOH Tone = The system will use the general MOH setting in Music on Hold Setup
BGM Source = The system will use the Background music source defined in ACI port setup.
ACI Port = The system will use the ACI port number specified in the Sound Source below.
Sound Source - Specifies the ACI port number (1-6) to be used for the MOH Type. The ACI port number is shown in Port Setup for each 2PGDAD Unit installed.
Fall Over IRG - There is a fall over IRG available for ALL DDI's within each DDI Table Area. This fall over IRG is available if you enable the transfer option for the DDI and enable this option.

Easy Edit Help

DDI Fall Over IRG (1533)
The trunks are set as DDI type in Incoming Service Type Setup.
DDI trunks are placed into a trunk group in Trunk Group, the trunk group number is used to route to the DDI table area for each night mode.
The DDI Table Area is defined in DDI Table Area Setup. This defines the DDI Table entries that will be searched.
Each DDI Table Area has a common fall over option that will step un-answered calls to the IRG specified in this section. Refer to the DDI Routing Table for the step on options and related timers.

Table Area Number - The DDI Table Area number.
IRG Mode 1-8 - The Incoming Ring Group number for the fall over.

Easy Edit Help

DDI Basic Setup (1534)
The trunks are set as DDI type in Incoming Service Type Setup.
DDI trunks are placed into a trunk group in Trunk Group, the trunk group number is used to route to the DDI tables, you make seperate entries for each trunk group.

Dial In receive digits - Enter how many DDI digits will be received from the Network supplier.
Received vacant number operation - When the system receives a DDI number that has no corresponding entry in the DDI Routing Table it can either receive busy indication or transfer to the DDI Fall Over IRG.
Sub-Addressing mode - Select the operation of Sub-Address digits received by the system.
Extension Number Specify = The call routes directly to the extension number that corresponds to the Sub-Address.
DDI Routing Table = The DDI Routing Table is searched to find a corresponding entry.
DDI Receiving Mode - Select the operation of the system for the DDI digits.
Enblock Receiving = All DDI digits are sent within the SETUP message.
Overlap Receiving = Not all DDI digits are sent within the SETUP message, further digits may be sent in seperate information messages.
Overlap receiving mode local code digits - The local code digits for Overlap Receive.
Overlap receiving mode local code - The local code for Overlap Receive.
Overlap receiving mode pilot code - The pilot code for Overlap Receive.
Overlap receiving mode T302 time-out operation - Select the operation if the Network does not send all digits before timer T302 expires. Options are:-
Disconnect - the call can receives busy indication
Transfer - transfers to the DDI Fall Over IRG or
Search - search the table with an incompletely received DDI. This happens when either Timer T302 expires or the PBX receives a SendingCompleteIE and has not yet collected the configured amount of DDI digits, as assigned in 'Dial In Receive Digits'. This

setting allows the ability to have DDI's that are shorter than configured, i.e. DDI's with different lengths.

Easy Edit Help

DDI Table Area Setup (1535)
The trunks are set as DDI type in Incoming Service Type Setup.
DDI trunks are placed into a trunk group in Trunk Group, the trunk group number is used to route to the DDI table area in DDI Table Area Target.
The area of the DDI Routing table is defined in this section, the area defines the DDI Routing table entries to be searched in the DDI Routing Table.
There are two areas available, the second is optional. The DDI will search the first area and if a corresponding entry is not found the second area will be searched.
You can duplicate or overlap DDI Routing Table entries for any of the table areas, this enables you to make more efficient use of the DDI Routing Table entries.
Enter 0 for any unused Tables Areas. There are 2000 DDI Routing Tables entries available.

Table Area Number - The target DDI Table area number that the trunk group is routed to in DDI Table Target Area.
1st Area setup (Start Address) - The DDI Routing table entry number for the start of the Table Area.
1st Area setup (End Address) - The DDI Routing table entry number for the end of the Table Area.
2nd Area setup (Start Address) - The DDI Routing table entry number for the start of the Table Area.
2nd Area setup (End Address) - The DDI Routing table entry number for the end of the Table Area.

Easy Edit Help

DDI Table Area Target (1536)
The trunks are set as DDI type in Incoming Service Type Setup.
DDI trunks are placed into a trunk group in Trunk Group, the trunk group number is used to route to the DDI table area for each night mode.
The DDI Table Area is defined in DDI Table Area Setup.

Trunk Group - The trunk group number that the DDI trunks are assigned to.
Table Area Mode 1-8 - The DDI Table area that will be searched.

Easy Edit Help

Mail Box Setup
Assign the mail box ID numbers and passwords for the DSPDB voice mail.

Entry Number - The DSPDB mail box number (1-300).
VM Box ID Number - The mail box ID number is used to gain access to the mail box. You can enter up to 8 digits. This is the number entered as the additional data for the programmable function keys.
Password - Enter the optional 4 digit password that the user must enter to gain access to the mail box. Users can set/delete/change their passwords with Voice Mail service code 65#.
Language - Select the language for the voice mail box prompts. The flexible language is determined by the flexible language loaded onto the compact flash card in the DSPDB card.

Easy Edit Help

Voice Mail Basic Setup (1538)
DSPDB Voice mail settings.

Exclusive channel for voice mail - The system can reserve this number of channels for voice mail access. This will limit the number of simultaneous calls to DUD/DISA and queue announcements.
Time Stamp - Each voice mail message will have the time that the message was recorded.
Conversation recording mode for after transfer - Should conversation recording continue to record after the call is held and transferred.
Automated attendant voice mail for non-existing extension - The system can allow calls to a vacant (not installed) extension number to route to a voice mail box. The vacant extension number must have a mail box assigned in Mail Box Setup.
Maintenance Time - Specify the time that the system will erase out of date messages and 'clean-up' the compact flash memory on the DSPDB card. Set to 00:00 to disable. The DSPDB will be taken out of service during the maintenance time.
Automatically Erase Message - Specify the number of days a message will be kept if the Maintenance Time is enabled. If the Maintenance time is disabled the messages will not be erased.
Escape from DSPDB-VM while Recording - Incoming callers that are forwarded to the DSPDB voice mail as a result of a user setting DSPDB Voice Mail Automated Attendant can have the ability to escape from the user's mail box and be able to dial another extension number.
1 Digit Access - Enable this option if you want to use the DSPDB Voice Mail 1 digit access codes assigned in VRS 1 Digit Translation . If this option is disabled the DSPDB voice mail service codes and option codes are fixed.

Easy Edit Help

Conversation Recording Destination for Trunk (1539)
Setup the conversation recording operation for each trunk port. The system has an option to continue to record calls after they are held and transferred to another extension in Voice Mail Basic Setup.
If the extension also has conversation recording destination setup then the settings for the trunk in this section will be overidden.

Recording Destination Extension - To record to the DSPDB voice mail enter the 'Voice Mail Centre Access' service code (default=884).
You can also enter an ACI Extension or ACI pilot number as the recording destination. The conversation will be output via the associated audio port of the PGDU.
Automatic Recording - Automatically record all incoming calls to the trunk.
Record Contents Storing Method - The destination DSPDB mail box for the recorded conversation.
Save to Own Mailbox = The recorded conversation will be saved to the mail box of the extension that answers the call. If the extension does not have a mailbox then it will be saved to the mail box set in Live Recording Setup.
Save to Dialed Mailbox = The recorded conversation will be saved to the mail box set in Live Recording Setup.
Automatic Recording for Outgoing Call - Automatically record all outgoing calls on the trunk.

Easy Edit Help

Conversation Recording Destination for Extension (1540)
Setup the conversation recording operation for each extension. The system will record all trunk calls to the extension, internal calls are not recorded. The system has an option to continue to record calls after they are held and transferred to another extension in Voice Mail Basic Setup.
If the trunk also has conversation recording destination setup then the settings for the extension in this section will be used.
These settings are used for automatic conversation recording and manual conversation recording via the programmable function key 69+0.

Recording Destination Extension - To record to the DSPDB voice mail enter the 'Voice Mail Centre Access' service code (default=884).
You can also enter an ACI Extension or ACI pilot number as the recording destination. The conversation will be output via the

associated audio port of the PGDU.
Automatic Recording - Automatically record all incoming trunk calls.
Record Contents Storing Method - The destination DSPDB mail box for the recorded conversation.
Save to Own Mailbox = The recorded conversation will be saved to the mail box of the extension that answers the call. If the extension does not have a mailbox then it will be saved to the mail box set in Live Recording Setup.
Save to Dialed Mailbox = The recorded conversation will be saved to the mail box set in Live Recording Setup.
Automatic Recording for Outgoing Call - Automatically record all outgoing trunk calls.

Easy Edit Help

Message Recording Setup (1541)
Message recording options for calls forwarded to a mail box.

Recording Time - The maximum duration the system will allow for each message. The system does not limit the maximum quantity of messages only the duration of each message.
Denied Recording Guidance Message - If the mail box has set Message Restriction with Voice Mail service code 66# then calls forwarded to the voice mail box can not leave a message. The caller will hear either a fixed greeting "We have no attendant available now, please call back again" or the mail box greeting message recorded by the user.

Easy Edit Help

Live Recording Setup (1542)
Message recording options for calls recorded to a mail box. The system has an option to continue to record calls after they are held and transferred to another extension in Voice Mail Basic Setup.

Undefined Destination Operation Mode - Conversation recording for trunks and extensions has an option to save the recording to a dialed mailbox. The dialed mail box is defined in this option.
Temporary Mailbox = The message will be saved to the mailbox defined in the 'Temporary Mail Box entry number'.
Call-Back Mailbox = The system will call the extension when the recording has finished and prompt the user for the mailbox number to save the message.
Temporary Mail Box entry number - The mail box entry number (1-300) that the conversation will be saved into when 'Temporary Mailbox' is set. If 0 is entered the recorded conversations will not be saved.
Live Recording Display - The display of the keyphone can show 'CONV.REC' when the conversation is being recorded.

Easy Edit Help

Call Information Setup (1543)
The DSPDB Voice mail can notify the user of a new message in their mail box. The is additional to the Voice Mailbox programmable function key 67+mailbox and interupted dial tone.
The mailbox user must enable Message Notification with voice mail service code 61#, they can select either internal or external notification.
A mailbox password must be set as this will be required to listen to the message when the user is notified.

Maximum Simultaneous Outgoing Calls - How many simultaneous calls can be made for message notification.
Trunk Route - The trunk route number for external notification. Trunk routes are setup in Trunk Group Routing.
ISDN Calling Party - For external notification the system will send this Calling Party number for the outgoing ISDN trunk call.
Call Interval for Intercom Call - If the notification is not answered within 60 seconds the system will try again after this time (minutes).
Call Interval for External Call - If the notification is not answered within 60 seconds the system will try again after this time (minutes).
Maximum Intercom Call - If the notification is not answered or the message is not listened to the system will make this many notification attempts.
Maximum External Call - If the notification is not answered or the message is not listened to the system will make this many

notification attempts.

Easy Edit Help

Voice Mail Automated Attendant Data Setup (1544)
Voice Mail Automated Attendant is used to route trunk calls to a voice mail box to leave a message.
The trunk call is routed to the DSPDB voice mail with IRG number 101.

Mode 1-8 - The night mode 1-8.
Trunk No - The trunk port that the call is presented on.
Option Mode - The incoming caller can be routed directly to a mail box to leave a message or be prompted to enter the mail box number to access the mail box. Allows the incoming caller to listen to their messages.
Automated Attendant = The incoming caller will be routed to the mail box specified in the 'Mail Box entry for message' option. The caller can leave a message in the mail box. There is the option for the incoming caller to dial 50# to exit the mail box and dial an extension number.
Mail Box Access = The incoming caller will be prompted to enter a mail box number to gain access to the mail box.
Guidance Message - The incoming caller can hear one of the DSPDB messages when the call is answered in Automated Attendant mode. If this is set to 0 the caller will hear a fixed greeting "We have no attendant available now, please leave your message".
This option is not used in 'Mail Box Access mode'.
Mail Box entry for Message - The mail box entry number (1-300) that the incoming caller will be routed to in 'Automated Attendant' mode.
The incoming caller can exit from the mail box by dialling a service code which is fixed to 50# when VRS 1 Digit Access Setup is disabled in Voice Mail Basic Setup but can be flexible if VRS 1 Digit Access Setup is enabled. The VRS 1 Digit Codes are setup in 1 Digit Access Setup.
When the incoming caller dials this service code they will hear a fixed message "Please dial extension number". They will then hear the DUD/DISA Talkie set for the trunk port in DUD/DISA Talkie.

Easy Edit Help

Default Menu Language (1545)
Select the language for the system prompts for the DSPDB voice mail. This option also selects the language for the system's VRS Fixed Messages.
The language of the individual mail box prompts are selected in Mail Box Setup.

Voice Prompt Language - Select the language.
The 'Flexible' language is determined by the flexible language loaded onto the compact flash card installed in the DSPDB card.

Easy Edit Help

Voice Mail Broadcast Lists
A broadcast list allows a user to record a message into multiple mailboxes with voice mail service code 2#. The mailboxes are created in Mail Box Setup.

Entry - The entry number 1-100 within the broadcast list.
Group 1-10 - The Broadcast list number (Group 1 is Broadcast list 0, Group 2 is Broadcast list 2). Enter the mail box ID numbers that will be members of the Broadcast List.

Easy Edit Help

Voice Mail External (1548)
An external voice mail can be connected to SLT ports of the system. The external voice mail system is independent of the DSPDB voice mail system.
The SLT ports connected to the external voice mail must be placed into their own Department Group in Department Group Assignment, the Department Ring Order is not valid for voice mail SLT ports.
The Department Group pilot number is used as a method of placing calls to the external voice mail system. Assign the pilot number for the voice mail department group in Department Group Options.
The SLT ports must be set to Special in the Terminal Type option of SLT Basic Setup.
Caller ID can also be sent to the voice mail system if the SLT ports are enabled in SLT Basic Setup. You must also ensure the voice mail system is configured to receive the caller ID information.
Ensure that the system is setup to accept the correct Timed Break Recall signal duration in SLT Data Setup, the voice mail system will use recall to transfer calls.

Voice Mail Extension Group - Enter the Department Group number that will have the SLT ports assigned in Department Group Assignment.
Voice Mail Master Name - Enter the name that will be displayed at the keyphone when calling the voice mail system.
Voice Mail Call Screening - The system can accept call screening commands from the voice mail system (Recall + 1 + extension number). You should normally set this option to ON unless you have extension numbers on the system that begin with the digit 1.
Park and Page - The system can accept Park and Page commands from the voice mail system (Recall + * + extension number + Paging string). You should normally set this option to ON.
Message Wait - The system can accept Message Waiting commands from the voice mail system (# + extension number + Message count). You should normally set this option to ON.
Record Alert Tone Interval Time - The interval between voice mail conversation record tones.

Caller ID on Analogue VM Extension Ports
Any analogue voicemail connected to a telephone system via extension ports uses DTMF Tones (in-band signalling) to communicate information between them.
It is important therefore that Caller ID is disabled for these voicemail extension ports.
This is because the FSK (Frequency Shift Keying) audio signaling method used on the Aspire, XN120 and other systems to send the Caller ID to an analogue handset can be detected by the analogue voicemail.
This can lead to erratic operation of the voicemail.
This applies equally whether a third party telephone system or voicemail is connected to NEC Infrontia equipment, or whether only NEC Infrontia equipment is concerned.
The relevant command both on the Aspire and XN120 is 15-03 Item 9, "Caller ID Function" and this should be set as Disabled for all voicemail extension ports.
It is important to note that the XN120 Quick Install sets Caller ID to Enabled on all Single Line Telephone ports.
This will not be changed as the integral voicemail is more commonly used on the XN120
In summary any extension ports used for connection of an analogue voicemail must be checked to ensure Caller ID is not in use.

Easy Edit Help

Abbreviated Dial Function Setup (1549)

Speed Dial Auto Outgoing Mode - You can choose the mode of the dialled digits within all Abbreviated dial bins.
Select Trunk Access mode and the Abbreviated dial will seize a free trunk (defined in Trunk Group for Abbreviated Dial).
Select Intercom Access Mode and the Abbreviated dial will make an internal call, so you will need to preceed each Abbreviated dial with the trunk access digit if you want the call to seize a trunk.
Individual Speed Dial - Enable or disable the users access to Station Abbreviated speed dials (Accessed by Abbreviated dials 900 to 919). Use the Station Abbreviated Dial number and name screen to enter each extension's speed dials.
Common Abbreviated Table Size - Set the quantity of common speed dials, There are a total of 2000 speed dials stores available. If you choose more than 900 then the users will not be able to use their Station Speed Dials 900-919. The remaining speed dials stores can be used for Group Abbreviated Dialling.

Easy Edit Help

Group Abbreviated Dial Area Setup (1550)

Speed Dial Group - Shows the groups available. Stations are assigned to one of the groups in the Abbreviated Dial Group Assignment for Extensions screen.
Start Address of SPD Bin - This sets the first Abbreviated dial location (0-1990). The value must be a multiple of 10 and must not be within the Common Abbreviated Table Size.
End Address of SPD Bin - This is the last Abbreviated dial location (9-1999). The value must be in multiples of 10.

Easy Edit Help

Group Abbreviated Dial Assignment (1551)

Extension Number - The extension numbers of all stations on the system.
Group Number - The Abbreviated Dial group number the station is assigned to.

Easy Edit Help

Abbreviated Dial Name and Number (1552)

Location - The Abbreviated Dial location number that the user will dial to access each bin.
Abb Dial Number - The number dialled by the system. Up to 24 digits long.
Abb Dial Name - The name associated with the Abb dial. Used for Keyphone display, Directory Dialling and CLIP alpha tagging. Up to 12 characters long.
Transfer Mode - This option is used to enable routing of the call based on the Caller ID, refer to 'Flexible Ringing by Caller ID'. The caller ID is matched to the 'Abb Dial Number'.
Not Defined = Routing based on caller ID is disabled, calls will be routed as defined in Incoming Service Type Setup.
Internal Dial = The call will be routed to the 'Destination Number'.
Inc Ring Grp = The call will be routed to the 'Destination Number' which will define the IRG number.
Destination Number - Enter the destination number (extension, Department Group pilot number, DSPDB Access number) or the IRG number.
Incoming Ring Pattern - You can select a different ring pattern for the calls routed by Caller ID, this can help to identify the calls when ring at the keyphones.
Enter a value from 0 to 9:
0 = Use the ring pattern defined for the trunk in Trunk Ring Tone Setup.
1 to 9 = Use ring tone 1 to 9 as defined in Ring Tone Setup.
ISDN Call Forward Method - Enter the method to be used for Call Forward Off Premise.
0 = Use a separate outgoing trunk port to perform the off premise call forward.
1 = Use the 'Call Deflection' feature if the ISDN trunk port is also enabled in ISDN Call Forward Method. This option has no effect on analogue trunks that are forwarded off premise. The system will use the 'Abb Dial Number' as the redirected number.
VRS Message Number - Defines the VRS message number to be played instead of ringing for incoming calls matching the CLI. See Flexible Ringing by Caller ID for more details.

Easy Edit Help

Trunk Group for Abbreviated Dial (1553)

Choose which trunk group the Abbreviated Dial call should use.
Location - The Abbreviated Dial location number that the user will dial to access each entry.
Abb Dial Number - The number dialled by the system. Up to 24 digits long.
Abb Dial Name - The name associated with the Abbreviated dial. Used for Keyphone display, Directory Dialling and CLIP alpha tagging. Up to 12 characters long.
Trunk Group - The Trunk group number. Enter trunk group 0 to use the same trunk group used if the user was to dial the trunk access code.

Easy Edit Help

Station Abbreviated Dial Name and Number (1554)

There are 20 personal speed dials available at each extension. They are accessed by Abbreviated dial locations 900-919 by the user.
Extension No - The extension numbers of all station on the system.
Abb Dial - The number dialled by the system. Up to 24 digits long.
Name - The name associated with the Abbreviated dial. Used for Keyphone display, Directory Dialling and CLIP alpha tagging. Up to 12 characters long.

Easy Edit Help

IP/EXIFU Network Setup (1555)
The EXIFU-A1 card has the LAN interface for the system.
The system must be powered off and on for any changes to take effect.

IP Address - Set the IP Address for the system. This will be used for PCPro, Voice Over IP, SMDR and CTI connections to the system.
Subnet Mask - Select the appropriate mask.
Default Gateway - Enter the IP address of the Default Gateway (router).
Time Zone - Set the time zone for the system. Enter 0 to 24 (-12 hours to +12 hours), 12 = GMT.
NIC Setting - Select the setting for the NIC (Network Interface Card) built into the EXIFU card.
NAT Router - Is a NAPT router used to for VOIP calls (SIP only).
Default Gateway (WAN) - If a NAT router is used enter the Public IP Address of the NAPT router.
ICMP redirect - Will the system use ICMP Redirect information. May cause problems for VOIP if this option is enabled.

XN120 MYCTI License procedure

The XN120 CTI Application is licensed based on a unique PC Hardware ID. This Hardware ID is generated on installation of XN120 CTI.

A 45-day evaluation is automatically generated on installation. During this period it is possible to use all features, it is recommended to license the software before the evaluation period expires.

In order for the License code(s) to be generated you are required to provide the following:

Purchase Order
Company Name
Company Address
Hardware Code (Generated on installation of NEC MYCTI)

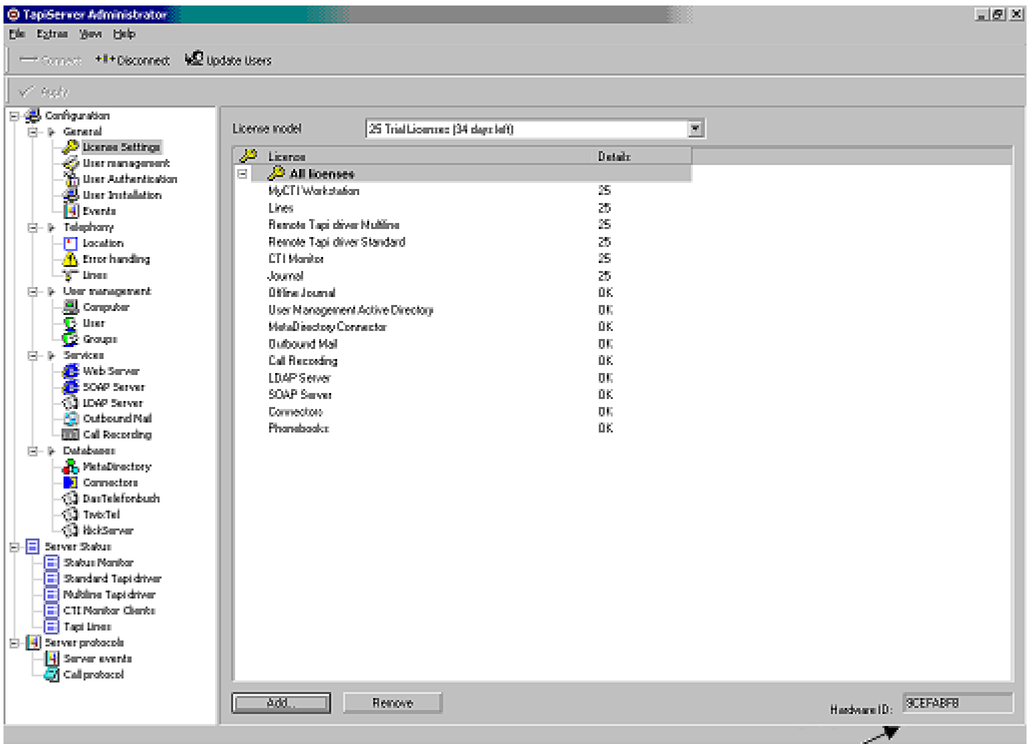
Methods used to send Purchase Order and Hardware ID:

eMail	licensingrequests@necinfrontia.co.uk
Fax	+44 (0) 1509 610206
Telephone	+44 (0) 1509 643100

Acquiring Hardware ID

In order to display the Hardware-ID bound to your installation of TapiServer, perform the following at the NEC TAPIServer.

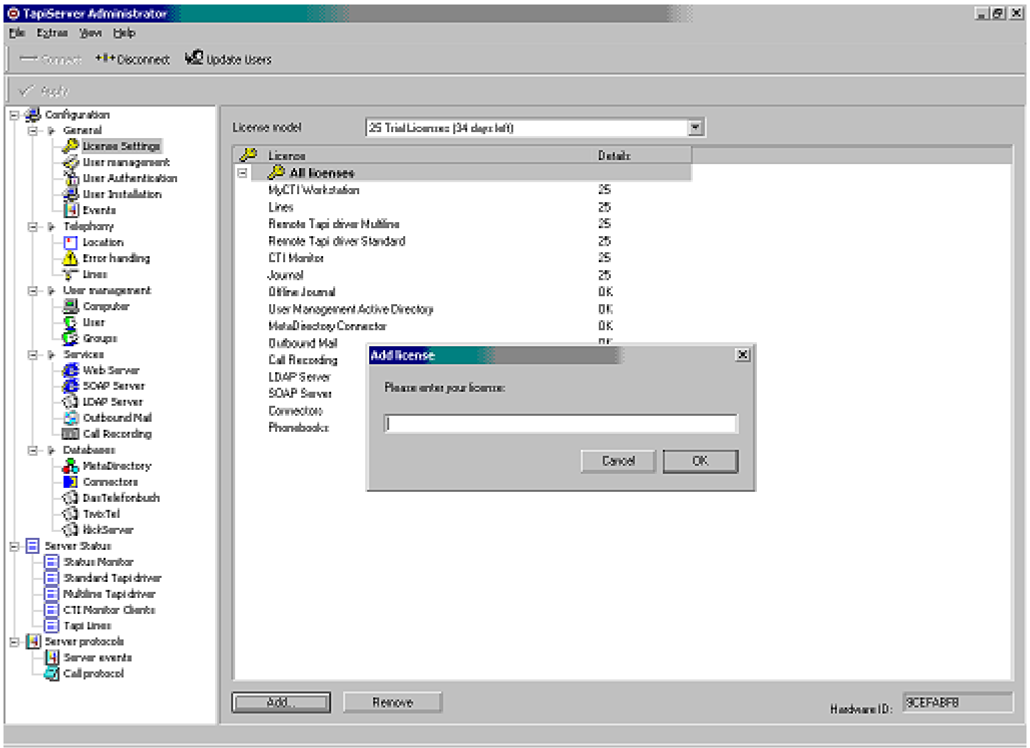
Click Start/Run/NEC/TapiServer/TapiServer Admin
Click on to Configuration/General/License Settings.



The Hardware ID is located in the lower right of the screen

Entering License

Click the Add button under Configuration/General/License settings.



Enter the license exactly as provided by your authorized reseller, click ok to apply.

Example Code

ABCD1234-ABCD1234-ABCD1234-ABCD1234

Easy Edit Help

Function Key Template (1600)
The template is used if you want to set the programmable Function Keys of all keyphones to the same features.
You can not duplicate the same feature or additional data on the Function Keys so if you want to 'move' a feature to another key you must first set the key to Not Used and then add the feature to the new key.
Some features will require Additional Data, for example:
CO Key - Additional data defines the trunk port number
Trunk Group Access Key - Additional data defines the trunk group
Virtual Extension Key - Additional data defines the Virtual extension number (or any other station number)
Park Key - Additional data defines the Park orbit
Loop Key - Additional data defines the type 1-3
DSS/One Touch Key - Additional data defines the station number
Day/Night Mode Key - Additional data defines the night mode 0-8 (0 will scroll through all available modes each time the key is pressed
Call Redirect Key - Additional data defines the destination number; station number or voice mail service code
Mail Box (DSPDB) Key - Additional data defines the mail box number
Voice mail service (DSPDB) Key - Additional data defines the operation during message playback (0=skip forward , 1=skip backwards)
Recording Service (DSPDB) Key - Additional data defines the mail box number to record into
Auto Attendant (DSPDB) Key - Additional data defines the station number for which the key will set call forward to voice mail
Voice Mail (In-skin VM) Key - Additional data defines the mail box number (also used for external VM systems)
Conv Record (In-skin VM) Key - Additional data defines the mail box number to record into(also used for external VM systems)
Auto Attendant (In-skin VM) Key - Additional data defines the station number for which the key will set call forward to voice mail (also used for external VM systems)

You can assign Function Keys to any station regardless of the number of Function Keys are station type.

Key - The Function Key on the keyphone
Feature - Select the feature from the list. Some features can not be duplicated.
Data - The Additional Data for certain features. You can not duplicate some features+additional data (for example, you can't have two CO keys for line 01)

Easy Edit Help

DSS Key Assignment (1601)
The keys of the DSS console can be assigned with the same features as the programmable Function Keys on the keyphone.

Key number - The key on the DSS console
Feature - The Feature assigned to the key. For Busy Lamp Indication for othet extensions set to DSS type.
Data - Certain Feature types require additional information, for example a DSS key will require the extension number that it will show busy lamp indication for.

Easy Edit Help

Trunk to Trunk Options (2013)
Setup the options for trunk to trunk transferred calls.

Trunk to Trunk Transfer Warning Tone time - Set the duration (Seconds) before the trunk to trunk warning tone will be played. After this time the caller can dial the continue code if they want to prevent the call being disconnected.
Trunk to Trunk Transfer Disconnect Time - Set the duration (Seconds) before the call will be disconnected if the caller does not dial the continue code. This timer starts after the 'Trunk to Trunk Warning Tone Time' has expired.
No Answer Time for Step Transfer - The duration the system will wait for the outgoing call to be answered before stepping to the next Abbreviated Dial set in Automatic Trunk to Trunk Transfer Target if 'Step Transfer' is selected for the trunk port (of the associated incoming call) in the 'Automatic Trunk to Trunk Transfer Mode' in Trunk Basic Data Setup.
No Answer Time for Automatic Trunk to Trunk Transfer - The duratin the system will wait for the incoming call to be answered before routing the call to the Abbreviated Dial entry for the Trunk to Trunk Forwarding Feature.
Conversation Continue Code - Enter the DTMF digit the caller can dial during a trunk to trunk call to prevent the system disconnecting the call. The trunks must also be enabled in 'Continue/Disconnect Trunk to Trunk Conversation' option of Trunk Basic Data Setup.
Conversation Disconnect Code - Enter the DTMF digit the caller can dial during a trunk to trunk call to disconnecting the call. The trunks must also be enabled in 'Continue/Disconnect Trunk to Trunk Conversation' option of Trunk Basic Data Setup.
Conversation Extend Time - Enter the duration the trunk to trunk call will be extended for after the user dials the 'Conversation Continue Code'.

Easy Edit Help

SMDR Output for Extensions (2014)
Turn on/off SMDR output for each extension's calls. You can also turn the SMDR output on/off for each trunk in Trunk Basic Data Setup
.

Easy Edit Help

Trunk Ringing on External Speakers (2021)
The system can use the external speakers to indicate that a trunk call is ringing. This feature will only operate for trunks set to Normal incoming type in Incoming Service Type Setup.
The system will use '63-External Speaker Ring Back Tone' in Service Tones as the ringing tone for the external speakers.
For each external speaker you can assign ringing for each trunk port in each night mode.
The ringing trunk can be answered at any extension with the 'Answer for non-ringing lines' service code (default=872).
The extension can also automatically answer the ringing trunk by going off hook if the 'Auto Off Hook Answer' option is enabled in Class of Service.
The trunks ringing on the external speaker that the extension can answer are assigned in Answer Trunk Ringing on External Speaker.

Mode 1-8 - The night mode number. For each night mode select 'Ring' to have the trunk port give ringing indication over the external speaker. Set to a 'blank' entry if you do not want the trunk to ring over the external speaker.

Easy Edit Help

Hotel Wake-Up Call Setup (2092)

Answering Message Mode for Wake-Up Call - Select the operation when the hotel room extension answers the wake-up call (for single line telephones only):
MOH = The system will play the Music On Hold assigned in Music on Hold Setup.
VRS Message = The system will play the VRS message (DSPDB) assigned in 'Wake-Up Call Message Assignment'.
VRS Message+Time = The system will play the VRS message (DSPDB) assigned in 'Wake-Up Call Message Assignment' and announce the time.
The system will ring the room extension for the duration set by 'Alarm Clock Duration' in System Timers and will use the ring pattern set by 'Alarm Clock' in Ring Pattern.
Wake-Up Call Transfer - The system can call the operator extension if the room extension does not answer their wake-up call. The operator extension is assigned in Operator Extension.
Setup Message Mode for Wake-Up Call - Select the operation when the hotel room extension sets the wake-up call (for single line telephones and keytelephone):
Check Tone = The system will give a confirmation tone.
VRS Message = The system will play the VRS message (DSPDB) assigned in 'Wake-Up Call Message Assignment'.
VRS Message+Time = The system will play the VRS message (DSPDB) assigned in 'Wake-Up Call Message Assignment' and announce the time.

Easy Edit Help

Hotel Room Extension Setup (2093)

Hotel Mode - Each hotel room extension must be assigned as 'Hotel'. 'Normal' is a non-hotel room extension.
Toll Restriction Class when Check-In - Assign the Toll Restriction class number (1 to 15) that will be used when the extension is checked-in. When checked out the toll restriction class is assigned in Toll Restriction Class per Night Mode.

Easy Edit Help

Hotel Class of Service Setup (2094)
Select the Hotel features for the class of service number assigned to the extensions in Class of Service per Night Mode.

Easy Edit Help

Hotel Service Code Setup (2095)
Shows the Hotel service codes.

Easy Edit Help

Hotel Single Digit Routing (2096)
Hotel room extensions can go off hook and dial a single digit that is translated in this table into an extension number or service code. The single digit does not use the System Numbering Plan and will require the user to dial the single digit code and wait for the 'Internal Call Inter-digit Time' set in System Timers to expire before the system will translate the digit and place the call.

Department Group - The Department Group that the hotel room extension is assigned to in Department Group Assignment.
Press 1 to # - Enter the translation for each single digit that the user will press. Enter up to 4 digits.

Easy Edit Help

Hotel Report Printing (2097)
Hotel reports are printed to the serial port of the EXIFU-A1 card installed in the Main Unit. They are output by the user with the 'Room Status Printout' service code, the extension must also have 'Room Status Output' enabled in Hotel Class of Service Setup.

Output port type - Select COM to enable output via the EXIFU serial port.
Wake-up call no answer data - Enable the output of un-answered wake-up calls.
Check out sheet - Enable the output of the check-out sheet.

Easy Edit Help

Timers (2098)

System Timers
General system timers. All settings are in Seconds. To disable a timer set it to 0.
DTMF receive active time - The duration that the system will detect DTMF digits for Analogue extensions and trunks that require DTMF detection, for example DISA trunks.
Alarm clock duration - The duration that the extension alarm will ring for.
Camp-On extension call back time - When an extension Camp-On rings back it will ring for this duration.
Camp-On trunk call back time - When a trunk Camp-On rings back it will ring for this duration.
Camp-On cancel time - Any Camp-On that is not completed will be cancelled after this time.
Trunk Guard time - The duration the system will prevent users seizing an analogue trunk after it has been cleared, during the guard time the CO key will show red (busy). This gives the Network time to clear the circuit.
Long Conversation alarm 1 (until sending 1st alarm) - The duration a trunk call must be in progress for before the warning tone is played if enabled in Class of Service.
Long Conversation alarm 2 (until sending next alarm) - The interval between each warning tone.
Long conversation cut off for incoming - The duration an incoming trunk call must be in progress for before the call is disconnected if enabled in Class of Service and Trunk Basic Data Setup.
Long conversation cut off for outgoing - The duration an outgoing trunk call must be in progress for before the call is disconnected if enabled in Class of Service and Trunk Basic Data Setup.
Internal Call Inter-digit time - The Interdigit time when placing an internal call.
External Call Inter-digit time - The Interdigit time when placing a trunk call. When this timer expires the trunk call will be classed in the answered state and the call timer will start.
Dial Tone Detection - The duration the system uses to measure dial tone. This timer can be left at default (5 Seconds).
Disconnect time when dial tone not detected - If dial tone detection is enabled for the analogue trunk port (in Analogue Trunk Data Setup) this time is used to set the duration the system will wait for dial tone. If dial tone is not detected during this time the line will be disconnected. This timer will be cancelled if digits are dialled out so you must set this timer to a value equal to or less than the Dial Pause at 1st digit timer. Set this timer to 0 if you do not want to disconnect the line.
Dial Pause at 1st digit - Set the pause duration before the first digit is dialled out on an analogue trunk. If dial tone is detected on the trunk this timer is not used, the digits will be dialled to line when dial tone is detected. If dial tone is not detected on the trunk (or is disabled) the digits will be dialled to line when this timer expires.
Toll Restriction Override release - This time sets the duration the user has to to place the outgoing trunk call if they are using the Toll

Restriction Override service.
Preset dial display hold time - Sets the duration the display of the keyphone will wait when the user preview dials a number.
Hotline call start time - Set the delay after the user goes off hook before the hotline call is placed, during the delay the user hears dial tone and can dial to override the hotline call. If the time is set to 0 the hotline call is placed immediately.
Dial digits for Toll Restriction path control - How many digits must be dialled from the keyphone's own keypad before the system will connect the transmit speech path to the trunk. Used to prevent the user placing a DTMF tone dialler at the microphone to bypass the system Toll Restrictions.
Interdigit time for Toll Restriction path control - How long the system will wait before connecting the transmit speech path to the trunk. Set this timer greater than the duration that the Network will supply dial tone.
Forced Account Code interdigit time - The system waits this time for the user to start to enter and between each digit when Forced Account Codes are enabled.
Outgoing Disable on incoming line - Turn on/off the Outgoing disable on incoming line feature. This feature can also be enabled/ disabled for each extension in Extension Basic Setup.
Outgoing Disable on incoming line - Timer - Set how long the system will detect DTMF digits for Outgoing Disable on incoming line feature.
Outgoing Disable on incoming line - Digits - How many DTMF digits the system will allow to be dialled to line before the trunk is disconnected. This is usually set to 5 digits or less digits to allow users to access dial up services but not able to dial a complete Network number.
Time of Redial - Set how many times (not in Seconds) the system will attempt the Repeat Redial feature.
Interval of Redial - The interval between each Repeat Redial Attempt.
Redial Calling Time - The duration the system will call the outgoing number for the Repeat Redial feature.
ISDN Calling Party Busy Tone - The duration the system will send busy tone to the extension if the Repeat redial is placed via an ISDN line and the called party is busy.
Normal Hold recall time - The duration a call must be on hold before it will ring back to the extension that placed the call on hold.
Normal Hold call back time - The duration the ring back tone will ring at the extension.
Exclusive Hold recall time - The duration a call must be on hold before it will ring back to the extension that placed the call on hold.
Exclusive Hold call back time - The duration the ring back tone will ring at the extension.
Long hold condition forced release time - The duration a call must be on hold before it will be disconnected if enabled in Trunk Basic Data Setup.
Park Hold time - The duration a call must be on Park hold before it will ring back to the extension that placed the call on hold if the extension's Class of Service option for Normal/Extended Park Hold is set to off.
Park Hold time extension - The duration a call must be on Park hold before it will ring back to the extension that placed the call on hold if the extension's Class of Service option for Normal/Extended Park Hold is set to on.
No answer time for call forward - Set the duration the call will ring before being forwarded when the user has set Call Forward No Answer.
Ring Inward Recall time - Set the duration that a call will ring after being transferred by a user. After this time the call will ring back to the extension that performed the transfer.
Ring Inward to busy Extension Group - Set the duration that a call will queue after being transferred to a busy Department Group. After this time the call will ring back to the extension that performed the transfer.
Trunk to Trunk Transfer Warning Tone time - Set the duration before the trunk to trunk warning tone will be played. After this time the caller can dial the continue code set in Trunk to Trunk Routing.
Extension Group delayed transfer time - Set the duration that a call will ring at the Department group before it will be forwarded.
Trunk to Trunk Transfer Disconnect Time - Set the duration the system will wait after sending the Trunk to Trunk Transfer Warning Tone before the system will disconnect the call.
No Answer Time for Step Transfer - The duration the system will wait for the outgoing call to be answered before stepping to the next Abbreviated Dial set in Automatic Trunk to Trunk Transfer Target if 'Step Transfer' is selected for the trunk port (of the associated incoming call) in the 'Automatic Trunk to Trunk Transfer Mode' in Trunk Basic Data Setup.
No Answer Time for Automatic Trunk to Trunk Transfer - The duratin the system will wait for the incoming call to be answered before routing the call to the Abbreviated Dial entry for the Trunk to Trunk Forwarding Feature.

Service Tone Timer
Extension dial tone - The duration the system will send dial tone when a user goes off hook, the user will then hear Busy tone.
Busy Tone - The duration the system will send busy tone when a user calls a busy extension, the user will then hear lock out tone tone.
Congestion Tone - The duration the system will send congestion tone whe a user can not place a call due to no system resource available.
Warning Tone - The duration the system will send Warning tone when a user dials an invalid number, the user will then hear lock out tone.
Confirmation Tone - The duration the system will send Confirmation tone for example when a user cancels a feature.
Interval of call waiting tone - The interval between Call Waiting tones.
Intrusion tone - The interval between Intrusion tones.
Conference tone interval - The interval between Conference tones.
Warning beep tone signalling interval - The interval between Warning tones for trunks if enabled in Trunk Basic Data Setup.

Doorphone Timer Setup
Doorphone Ring Time - The duration (Seconds) the extensions will ring when the doorphone call button is pressed.
Door Release Time - The duration (Seconds) that the associated door lock will be released.
External CFW by Doorphone disconnect timer - The system can forward the doorphone call to an external number via an ISDN trunk.

The user sets the call forward with the 'External Call Forward by Doorphone' service code (default=822), the call is forwarded to an Abbreviated Dial location. The outgoing trunk is specified for the Abbreviated Dial location in Trunk Group for Abbreviated Dialling.

Incoming Calls

Incoming ring no answer alarm start - Set the duration (Seconds) that a trunk call must ring before the alarm tone starts.
Normal DIL Incoming no answer time - Set the duration (Seconds) that a trunk call must ring before the call steps to the second target. This timer operates for trunks set as either Normal or DIL in Incoming Service Type Setup. The first and second targets for Normal type trunks are assigned in IRG Assignment Normal. The first target for DIL type trunks is assigned in DIL Target Assignment , the second target is assigned in DIL Step on Target Assignment.
DID no answer time - Set the duration (Seconds) a DDI call must ring before the call steps from Target 1 to Targets 2 or 3. The DDI must also have a No Answer Transfer Option selected. The targets and Transfer options are assigned in the DDI Routing Table. Trunks are set as DDI type in Incoming Service Type Setup.
DID Incoming Ring Group no answer time - Set the duration (Seconds) a DDI call must ring before the call steps from an IRG in Target 2 or 3. The DDI must also have a No Answer Transfer Option selected.
DID Incoming Pilot Call no answer time - Set the duration (Seconds) a DDI call must ring before the call steps from a Department Group in Targets 1, 2 or 3. The DDI must also have a No Answer Transfer Option selected.
DID to Trunk to Trunk no answer time - Set the duration (Seconds) a DDI call must ring before the call steps from a Trunk to Trunk call (incoming DDI call roued to an outside number) in Targets 1, 2 or 3. The DDI must also have a No Answer Transfer Option selected.
VRS Waiting Message interval time - Set the interval (Seconds) between each VRS announcement. Used for VRS Queue Announcements.

System Timer for DUD/DISA Service

DUD/DISA Dial tone - The duration the system will wait for the incoming caller to complete dialling. When this timer expires the caller will be disconnected or step on to the Auto Attendant Fall Over.
DUD/DISA No Answer time - The duration the system will ring at the target extension. When this timer expires the caller will be disconnected or step on to the Auto Attendant Fall Over.
Disconnect after DUD/DISA re-transfer to IRG - The duration the system will ring at the Auto Attendant Fall Over. When this timer expires the call will be disconnected.
Calling time to Automatic Answering telephone - The duration the system will wait for the SLT port to answer the DUD/DISA call. When this timer expires the caller will be answered by the DUD/DISA dial tone.
Guidance message by Automatic Answering telephone - The duration the system will connect the SLT device. When this timer expires the caller will be answered by the DUD/DISA dial tone.
Guidance message by ACI Talkie duration - The duration the system will connect the ACI device. When this timer expires the caller will be answered by the DUD/DISA dial tone.
DISA Conversation Warning tone time - The duration the system will wait before sending a warning tone to a trunk to trunk call via DISA. After this time expires the user can enter the continue code (if enabled) or the trunk to trunk call will be disconnected.
DISA Conversation Disconnect time - The duration the system will wait before disconnecting a trunk to trunk call via DISA, this timer starts when the trunk to trunk call is established. After this time expires the trunk to trunk call will be disconnected unless the caller has entered to continue code.
The duration of this timer must be greater than the combined times of the DISA Conversation Warning tone time plus the Continue Duration for DISA trunk to trunk time.
Continue Duration for DISA trunk to trunk - The duration the system will wait before re-sending the warning tone, this timer starts after the DISA Conversation Warning tone expires.
The trunks must have the Continue/Disconnect trunk to trunk conversation option enabled in Trunk Basic data Setup.
DISA Internal Paging duration - The duration the system will allow the DISA caller to be connected to the internal paging zone. When this time expires the call will be disconnected.
DISA External Paging duration - The duration the system will allow the DISA caller to be connected to the external paging zone. When this time expires the call will be disconnected.
DUD/DISA Answer delay - The duration the system will wait before answering the incoming DUD/DISA call.
DUD/DISA Busy tone - The duration the system will send busy tone to the incoming caller when the destination extension thay have dialled is busy.
The DUD/DISA Talkie must be set to No Talkie in DUD/DISA Talkie and the Busy fall over must be set to 0 in Auto Attendant Fall Over.
Delayed DUD Answer time - The duration the system will wait before answering the incoming Delayed DUD call. The trunk is set to Delayed DUD type in Incoming Service Type Setup. Before this timer expires the system will route the call as a Normal type (for example to an IRG).

CTI LAN Port Setup

Keep Alive Timer - It is recommended that this is set to 30 seconds.

System Options for Keyphones

Pre-Selection time - If the keyphone is set to Pre-Selection mode in Keyphone Options this timer sets how long the system will remember the pre-selection for Function keys set as DSS/One touch keys.

SLT Options

Trunk call dial sending start time by SLT - If the SLT DTMF dial to trunk lines option is set to Store+Forward then the pause time is set by this option. When the SLT leaves a pause of this duration in their dialing then the system will process all digits received.
Headset ringing start timer (for SLT) - Set the duration that the SLT will receive dial tone before headset mode begins. The SLT must also have the Restriction of Headset Earpice Ringing option in Class of Service set to OFF (Headset earpiece ringing enabled).

Analogue trunk timers
Setup the system data for analogue trunks.
Clear Signal (Open Loop) Detection - (timer = value x 8mS) The minimum time the system will detect a break in loop current to indicate the trunk has been cleared (Disconnect Clear signal). The trunk must be set to detect the clear signal in Analogue Trunk data Setup. Default = 37 (296mS).

Ringing signal detection time minimum - (timer = value x 8mS) The minimum ringing pulse duration the system will detect to start the incoming call. Default = 13 (104mS).
Single ringing detection minimum - (timer = value x 8mS) The minimum duration of a ring pulse for the system to detect as a single ring pattern. Default = 82 (656mS).
Double ringing detection minimum off time 1 - (timer = value x 8mS) The minimum duration between each ring pulse of a double ring pulse for the system to detect as a double ring pattern. Default = 13 (104mS).
Double ringing detection maximum off time 2 - (timer = value x 8mS) The maximum duration between each ring pulse of a double ring pulse for the system to detect as a double ring pattern. Default = 50 (400mS).
Ringing signal no detection minimum time - (timer = value x 8mS) The minimum duration between the two ring pulses of a double ring pulse for the system to detect as a double ring pattern. Default = 88 (704mS).
Time ringing signal stop detection time - (timer = value x 8mS) The maximum duration between any single or double ring pattern for the system to begin a new incoming call (Ring signal abandon time). Default = 47 (3008mS).
Hook flash time 1 - (timer = value x 16mS) The Hooking duration for trunks set to use Hooking type = Flashing in Analogue Trunk Data Setup. Default = 50 (800mS).
Hook flash time 2 - (timer = value x 16mS) The Hooking duration for trunks set to use Hooking type = Disconnect in Analogue Trunk Data Setup. Default = 156 (2496mS).
Pulse break time (10pps) - (timer = value x 8mS) The break time for 10 pulse per second dial pulse digits. Default = 8 (64mS).
Inter-digit time (10pps) - (timer = value x 32mS) The inter-digit time for 10 pulse per second dial pulse digits. Default = 19 (608mS).
Pulse break time (20pps) - (timer = value x 8mS) The break time for 20 pulse per second dial pulse digits. Default = 4 (32mS).
Pulse make time (20pps) - (timer = value x 8mS) The make time for 20 pulse per second dial pulse digits. Default = 2 (16mS).
Inter-digit time (20pps) - (timer = value x 32mS) The inter-digit time for 20 pulse per second dial pulse digits. Default = 16 (512mS).

DIL No Answer Timer
If a DIL is not answered it can step on to a second target after this time.
DIL No answer time - Enter the time (Seconds) that the incoming trunk call will ring at the DIL Target, if not answered the call will step on to the DIL Step On Target

External Paging Timer
Enter the maximum duration the system will allow an external page call when the timer expires the page call will be disconnected.

DECT Out of Area Timer
Out of Area Detection Time - Enter the duration (Seconds) the system will try to locate the DECT handset when a call is placed. If the handset is not located when this timer expires the call will use the Out of Range options set in DECT Handset Information. If no out of range option is set the caller will hear '16-Lock-out Tone' or the 'Out of Area VRS Message'.

Tie Line Timers
1st Digit Pause - Enter the duration (Seconds) the system will wait before dialling.
Tie Line Answer Detect time - Enter the duration (Seconds) the system will wait for an answer signal, from the remote system, for outgoing calls.

Easy Edit Help

SMDR Output for Trunks (2099)
Turn on/off SMDR output for each trunk. You can also turn the SMDR output on/off for each extension in SMDR Output for Extensions.

Easy Edit Help

VOIPU Firmware Information (2101)
Shows the version of firmware on the VOIPU card.
VOIPU Firmware Version - The version of firmware loaded onto each of the cards.

Easy Edit Help

Gateway Prefix (2102)
Gateways are used to connect to a non H.323 network.

Gateway Prefix Registration - Enable/disable Gateway Prefix Registration.
Gateway Prefix - Enter the Gateway Prefix.

Easy Edit Help

DHCP Server Setup (2990)
Select the DHCP settings for the system.
DHCP Server Mode - Enable/Disable the built in DHCP server within the system.
Lease Time - Enter the lease time of the IP Address for the client.
Number of Networks - The number of networks to manage. With a single network select 'Single'. When dividing and managing the same network as a multiple networks select 'Divide'. The range of the IP address to lease is set in assigned in DHCP Managed Network Setup.
Router - Enter the Router IP Address that will be given to the client.
DNS Server - Enter the DNS Server IP Address that will be given to the client.
TFTP - Enter the TFTP IP Address that will be given to the client.
DRS - Enter the DRS IP Address that will be given to the client. The default of 172.16.0.10 is the system's IP address assigned in IP Network Setup.

Easy Edit Help

DHCP Managed Setup (2991)
Select the IP Address range that the system will use to issue IP addresses to clients.
If 'Divide' is selected in DHCP Server Setup you can use Scope numbers 01 to 10.

Easy Edit Help

DHCP Address Reservations Setup (2992)
Use this option to reserve a fixed IP Address for a client. Enter the MAC Address of the client and the reserved IP Address the system will issue. The IP Address must be within the range assigned in DHCP Managed Network Setup.

Easy Edit Help

Virtual Extension Ringing Assignment (3010) Select the ringing (on or off) for each extension's programmable Function Keys. You can select the ring assignment for each night mode. There are settings for each function key for all extensions.

Extn No - The extension number of the keyphone that has the Virtual extension assigned to it's Function Keys. You assign the Function keys in Function Key Programming.
Key 1-22 - The function key that has the Virtual extension assigned. There are settings for all function keys including the optional DLS

console.

Easy Edit Help

Voice Mail Group Assignment (3019)
The ports used for either External Voice Mail or the FMSU/VMSU are placed into a Department Group and set as 'Special' mode.

Assign Department Group to Voice Mail Ports - Assign the SLIU ports connected to the External Voice Mail or the ports of the FMSU/ VMSU card to the Department Group number specified in Voice Mail External or Voice Mail FMSU / VMSU. Ensure that only Voice Mail ports are assigned to the selected group.
Department Group Ring Order - The ring order within the group is not normally changed from default but if necessary you can select the order in which the system will access the voice mail ports.
Assign Voice mail Ports to Special - Each voice mail port must be set to 'Special' in order for the system to send the DTMF digits to the voice mail.

Caller ID on Analogue VM Extension Ports
Any analogue voicemail connected to a telephone system via extension ports uses DTMF Tones (in-band signalling) to communicate information between them.
It is important therefore that Caller ID is disabled for these voicemail extension ports.
This is because the FSK (Frequency Shift Keying) audio signaling method used on the Aspire, XN120 and other systems to send the Caller ID to an analogue handset can be detected by the analogue voicemail.
This can lead to erratic operation of the voicemail.
This applies equally whether a third party telephone system or voicemail is connected to NEC Infrontia equipment, or whether only NEC Infrontia equipment is concerned.
The relevant command both on the Aspire and XN120 is 15-03 Item 9, "Caller ID Function" and this should be set as Disabled for all voicemail extension ports.
It is important to note that the XN120 Quick Install sets Caller ID to Enabled on all Single Line Telephone ports.
This will not be changed as the integral voicemail is more commonly used on the XN120
In summary any extension ports used for connection of an analogue voicemail must be checked to ensure Caller ID is not in use.

Easy Edit Help

F-Route Gain Table (3020)
The F-Route Gain Table is used to specify the gains used when calls are placed via F-Foute.

F-Route Table - The table number specified by either the 'Gain-ICM' or 'Gain-Tandem' option of the F-Route Table.
Within the F-Route Table the Gain-ICM option will use the gains specified in the 'ICM and Tandem Outgoing' entry. The Gain-Tandem option will use gains specified by 'ICM and Tandem Outgoing' for the outgoing trunk and 'Tandem Incoming Trunk' for the incoming trunk call.
ICM and Tandem Outgoing Trunk - Select the transmit and receive gains for calls from an extension on the system.
Tandem Incoming Trunk - Select the transmit and receive gains for tandem calls (Trunk to Trunk connection).

Easy Edit Help

External Page Timer (3021)
Enter the maximum duration the system will allow an external page call when the timer expires the page call will be disconnected.

Easy Edit Help

Hotel License Information (3050)
Hotel operation is limited to 30 days unless a Hotel License code is entered into the system.
To remove the 30 day limit you must enter a valid license key which can be obtained from your supplier.
Refer to the 'Hotel License Input' guide for further information.

Service Code for entering License (set to #700) - Enter the Service Code for entering the license code, #700 is as example service code.
Current License Status - Shows the period remaining of the 30 days of Hotel operation. If a valid Hotel License code has been entered into the system this option will show 'Active(ON)'.

Easy Edit Help

Global Language (3101)
Select the language for the system.
Keyphone Display - Select the display language for all keyphones on the system. Keyphones can also be set individually in Keyphone Options.
DSPDB Prompts - Select the language for the system prompts for the DSPDB Voice Mail. This is also shown in Default Menu Language within the Voice Mail DSPDB menu.
DSPDB Mail Boxes - Select the language for the voice mail box prompts for all DSPDB voice mail boxes. Mail boxes can be set individually in Mail Box Setup.

Easy Edit Help

UserPro Extension Password (3245)

Assigns the password required to log into UserPro in UB mode.

Easy Edit Help

UserPro UA Mode Programming Commands (3246)

Assigns the programming commands available for edit when logged into UserPro in UA mode (User Programming Administrator Mode).

Easy Edit Help

Card Configuration (900)
This screen allows you to add/view the cards in the system. This gives a representation of the actual cards installed in your system.
You can add/remove a card at any time while using Easy Edit.
Ensure you add the cards into the same slots when you build the PBX hardware.

To add/remove an expansion cabinet- Move the pointer over the POW (Power Supply) and click the left mouse button.
To add a new card- 'Drag and drop' a new card from the list. Easy Edit will ensure you drop the card into an appropriate slot.

When you add the card into a slot you will see the ports assigned automatically.
When you have added all of your cards click the Auto Port Assign button, this will align the port numbers.
The 8VoIPU card will install a 4VoIPU with the 4VoIPU expansion daughter card to give you an 8VoIPU.
Do not use the 2DIOPU or 2TLIU, these are not available in Europe.
To change the settings of a BRIU card- Move the pointer over the card and click the right mouse button and select Port Settings.
You can select the mode of each BRI circuit (Trunk = T-Point, S0 = S-Point).
To remove a card- Move the pointer over the card and click the right mouse button and select Remove Card.
To remove ALL cards- Click the Init (Initialise) button.

Easy Edit Help

Mobile Extension Setup (4011)
Setup the Abbreviated Dial number options relating to the Mobile Extensions.

Abb Dial Number - Assign the external telephone number of the Mobile Extension phone in the location corresponding to the assignments in Mobile Extension Setup.
Abb dial Name - Enter the name of the Abb Dial. The character set available is shown in Character Set.
ISDN Call Forward Method - Enter the method to be used for Call Forward Off Premise.
Disabled = Use a separate outgoing trunk port to perform the off premise call forward.
Enabled = Use the 'Call Deflection' feature if the ISDN trunk port is also enabled in ISDN Call Forward Method. This option has no effect on analogue trunks that are forwarded off premise. The system will use the 'Abb Dial Number' as the redirected number
Outgoing Trunk Group - Assign the trunk group used for calls to the abdial location.
Caller ID Edit Mode - As the PBX is required to match the CLI of the Mobile Extension to the number assigned in the abdial location to judge whether it is the owner of the Mobile Extension DDI it is recommended that this is enabled to prevent mismatches.

Easy Edit Help

Mobile Extension Setup (4010)
Setup the options relating to the Mobile Extensions.

Extension Number - Assign a free extesnion number against an unequipped extension port.
Extension Name - Enter the name of the telephone user. The character set available is shown in Character Set.
Abdial Location - Assign the Abdial location that coincides with the mobile extension external number (1-1999).
Connect Supervision Type - As the majoroty of Analogue telephone networks do not provide a 'hard' signal for a outgoing call being answered, 'soft' markers need to be used. On outgoing calls to Mobile Extension this is done by by requesting call acknowledge to the call. The sending of a * indicates that the call is connected. Assign when Answer Supervision code (*) is required - Always = all calls to mobile extension, Analogue lines only = analogue CO line only, None = never..
Trunk access type - Assign the method the Mobile extension uses to access trunks. Normal = use Trunk Access Code, Individual = use Individual Trunk Access Code.
Department Grp - Assign the Mobile Extension to a Department group. Department groups have a pilot number to enable users to call any available phone in the department. The pilot number is assigned in the Department Options screen.
Dept ring order - Chooses the order that you want the stations to ring when calls are made to the Department Group pilot number. You can enter any order number in the range 1-99. if you duplicate any order number then the station's port number will be used as the ring order (lowest port first).

Easy Edit Help

Virtual LoopBack DDI Table Area Target (4005)
The LoopBack trunks are set as DDI type in Virtual LoopBack Incoming Service Type Setup.
DDI trunks are placed into a trunk group in Virtual Loopback Trunk Group, the trunk group number is used to route to the DDI table area for each night mode.
The DDI Table Area is defined in Virtual Loopback DDI Table Area Setup.

Trunk Group - The trunk group number that the Virtual Loopback DDI trunks are assigned to.
Table Area Mode 1-8 - The DDI Table area that will be searched.

Easy Edit Help

Vitual LoopBack Trunk Group (4004)
Setup the options relating to the Trunk Group of the Virtual Loopback trunk ports.

Virtual Start Port - This is the port that the system has allocated as the start port of the Virtual Loopback trunk ports.
Number Of Channels - This is the amount of channels or ports configured as virtual loopback ports.
Trunk Name - Enter the name that will be shown on the keyphone display when the trunk is seized. Can also be shown on the SMDR printout. In order to easily identify the Loopback trunks names like 'Loop trunk 1' may be used.
Trunk Group - Enter the trunk group number.
Priority - Used for outgoing access the trunk are seized in order from lowest to highest priority. If the priority is duplicated within the same group then the lowest trunk port number will be used
Trunk to trunk outgoing caller ID through mode - When an incoming call on an ISDN trunk is routed out over another ISDN trunk the system can pass through the received caller ID to the outgoing call. In this case the Caller ID of the incoming call is passed through the virtual loopback.

Easy Edit Help

Vitual LoopBack Department Group (4003)
Setup the options relating to the numbering of the Virtual Loopback Department Group.

Pilot Number - Assign a number that can be used to call the Department Group. You can also route DDI calls and Auto Attendant calls to the pilot number.
Group Name - Assign a name to the department group for easy recognition of the loopback group.

Easy Edit Help

Vitual LoopBack Basic Setup (4002)
Setup the options relating to the Virtual Loopback.

Layer 3 Timer Type - There are five sets of Layer 3 timers. The timers are not available with Easy edit. Do not change this unless you are familiar with the Layer 3 timers. T-point timers are set by program number 81-06, S-point timers are set by program 82-06.
Calling Party Number - For T-point circuits Caller ID can be allowed/restricted. Caller ID can be set for each trunk or each extension in the ISDN Trunks calling party number or DDI Calling party number screens.
Flexible Digits for DDI Routing (S-point) - Additional digits can be added onto the MSN number sent by the system on S-point circuits. You define the quantity of digits 0-4. For example, The MSN number of the S-point circuit is set by its extension number, if you add 2 additional DDI digits the system will allow the extension number plus 00-99, giving a total of 5 digits for the MSN.
Call Busy Mode For S-point - Calls from the Virtual S0 device that are routed to a busy destination can return either Alerting with call progress indication or Disconnect with User Busy indication.

Easy Edit Help

Vitual LoopBack - Port Numbering (4001)
Setup the options relating to the port numbering of the Virtual Loopback.

Virtual Start Port - This is the port that the system has allocated as the start port of the Virtual Loopback extension numbers.
Number Of Channels - This is the amount of channels or ports configured as virtual loopback ports.
Extension No. - Assign a free extension number to the virtual loopback extension port.
Extension Name - Assign a name to the virtual loopback ports. For easy recognition of the ports an example could be 'Loop 1', 'Loop 2' etc.
Department Group Number (1-64) - Assign the virtual loopback extension ports to a unique, i.e. not used, department group.

Easy Edit Help

Set / Cancel Caller ID Refuse (4015)

Set / Cancel Caller ID Refuse - Enable or disable, on a per Class of Servce basis, the ability to enable/disable Caller ID Refuse.

Easy Edit Help

Set Caller ID Refuse (4014)

Set Caller ID Refuse - Define the service code to be dialled to enable Caller ID Refuse.

Easy Edit Help

Reject VRS Message (4013)

Set VRS Message for Caller ID Refuse - Define the VRS message to be played when an Caller ID has been matched for Caller ID Refuse.

Easy Edit Help

Flexible Ringing - Reject Trunk Data (4012)
Setup on a per trunk basis the abilty to use Flexible Ringing by Call ID.

Flexible Ringing By Caller ID - Enable or disable the Flexible Ringing By Caller ID feature.
Caller ID Refuse Setup - Enable or disable the ability to reject calls based on Caller ID

Easy Edit Help

ISDN Emergency Override - Forced Disconnect (4022)
Defines the number to be used for emergency override.

Emergency Number - include trunk access code - Enter the number for emergency override including the trunk access code e.g. if the emergency number is 999 and the trunk access code is 9 enter 9999 or if the emergency number is 112 and the trunk access code is 9 enter 9112.

Easy Edit Help

Trunk to Trunk Transfer (4021)
For each trunk enable the options for trunk to trunk transfer.

Trunk to Trunk Transfer - For each trunk enable the ability to perform trunk to transfer.
Analogue trunk ONLY Disconnect Clear - As analogue trunks do not provide call supervision, it is necessary for the network to provide a disconnect signal in order to prevent the trunks 'locking up'. This is provided by a timed temporary break in the line. Enabling this parameter enables the analogue line to detect this signal. The timer to set the minimum detected break is set in Timers and is called the 'Clear Signal (Open Loop) Detection Time'.

Easy Edit Help

External Call forward - COS (4020)
For each Class of Service enable the ability to set External Call Forward.

External Call Forward (Off Premise) 20-11-02 - Defines the status of the Class Of Service entry for External Call Forward (Off Premise).

Easy Edit Help

DIL Assignment (4017)
For each trunk assign the first and second Targets for each time mode (1-8).

First Target Extn/Pilot - Direct Inward Lines (DIL) are assigned as the incoming trunk type in Incoming Service Type.
A DIL is routed directly to the extension number and can therefore follow any off-premise call forwards.
A DIL can also be routed to a virtual extension numbers or Department Group pilot number.
A different target can be assigned for each night mode.
If a DIL is not answered it can step on to a second target after the DIL No Answer Time.
IRG Second Target - Enter the IRG number for the second target. The members of the IRG are assigned in Incoming Ring Group Setup.
Enter 0 if you do not want the call to step on from the DIL Target.
The second target is not used if a call arrives when the DIL target is busy, the call will route to the IRG set in IRG Assignment (Normal).

Easy Edit Help

IRG No Answer Timer (4016)

IRG No Answer Timer - Set the timer, for a DID incoming call directed to an Incoming Ring Group, before the call steps to the next DID target.

Easy Edit Help

Trunk Toll Restriction Per Night Mode (4023)
There are 9 night modes available on the system including the power fail mode. A trunk can be assigned a Toll restriction class for each mode.
The class will define which numbers a station can dial for outgoing trunk calls.
You can define each Toll class in the Toll Restriction screen.
You can also enable/disable toll restriction for each trunk in the Trunk Basic Data Setup screen.

Toll restriction class per trunk - For each of the Night Modes assign one of the 15 classes.

Easy Edit Help

Virtual Extension Toll Restriction Per Night Mode (4040)
There are 8 night modes available on the system. A station can be assigned a Toll restriction class for each mode.
The class will define which numbers the station can dial for outgoing trunk calls.
You can define each Toll class in the Toll Restriction screen.
You can also enable/disable toll restriction for each trunk in the Trunk Basic Data Setup screen.

Class for Mode 1-8 - For each of the Night Modes assign one of the 15 classes.

Easy Edit Help

Virtual Extension Type/Toll (4041)
The settings in this page define the operational settings for the virtual extensions when assigned to a keyphone.

Extn Port - The system port number of the extension port.
Extn No - The extension number of the system extension port.
Extn name - The name of the extension port
Virtual Mode - Select the operation of Virtual Extension/Call Coverage keys (programmable function key set as Virtual Extension Key).

DSS = Placing and answering calls by pressing the key
Outgoing = Placing calls by pressing the key (answering calls is not possible)

Ignore = Answering calls by pressing the key (Placing calls is not possible)
VE Toll - Select the Toll restriction used when an outgoing call is made from a virtual extension key.
Use Virtual Toll - Use toll restriction settings assigned to the virtual extension.
Use Extension Toll - Use toll restriction settings assigned to the normal extension.

Easy Edit Help

Numbering Plan (1104)
The numbers processed by the system are defined in this screen.
The system will define the type of number by the first digit dialled, this is shown in the Dial column.
For each entry you must define how many digits the system will need and what type the number will be used for.

For example, if you want to have all numbers beginning with 1 to be a digit extension number then set as shown below.
Note: any setting in Virtual Loopback Port Setup Flexible Digits for DDI Routing will be added to this setting.

Dial	How many digits	Type
1	2	Extension number

The default system numbering plan is for the UK:

Dial	How many digits	Type
1	3	Extension number
2	3	Extension number
3	3	Extension number
4	3	Extension number
5	3	Extension number
6	3	Extension number
7	3	Service code
8	3	Service code
9	1	Trunk route access
0	1	Operator access
*	4	Service code
#	4	Service code

The default system numbering plan is for the other areas of Europe:

Dial	How many digits	Type
1	3	Extension

		number
2	3	Extension number
3	3	Extension number
4	3	Extension number
5	3	Extension number
6	3	Extension number
7	3	Service code
8	3	Service code
9	1	Operator access
0	1	Trunk route access
*	4	Service code
#	4	Service code

Dial - The first digit only entries are shown by this page.
If you require setting by 2 digits e.g. 21 = extension number and 22 = Service code please refer to System Numbering.

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Abbreviated Dial

Abbreviated Dialling gives an extension user quick access to frequently called numbers. This saves time, for example, when calling a client with whom they deal often. Instead of dialling a long telephone number, the extension user just dials the Abbreviated Dialling code. There are two types of Abbreviated Dialling: Common and Group. All co-workers can share the Common Abbreviated Dialling numbers. All co-workers in the same Abbreviated Dialling Group can share their Group's Abbreviated Dialling numbers. The system has 2000 Abbreviated Dialling bins that you can allocate between Common and Group Abbreviated Dialling.
Each Abbreviated Dialling bin can store a number up to 36 digits long.
When placing an Abbreviated Dialling call, the system normally routes the call through Trunk Group Routing, or the user can pre-select a specific trunk for the call. In addition, the system can optionally force Common Abbreviated Dialling numbers to route over a specific Trunk Group, set by Outgoing Trunk Group for Abbreviated Dialing. User pre-selection always overrides the system routing.

Common Abbreviated Dialling
At default the system has 1000 Common Abbreviated Dials set by Abbreviated Dial Function Setup. The entries of the Common Abbreviated bins can be set either with Abbreviated Dial Number and Name or with Service Code 853. They can be dialled by pressing the DC key or Service Code 813 followed by the bin number.

Group Abbreviated Dialling
At default there are no groups assigned in Group Abbreviated Dial Area Setup. There are up to 64 groups available in Abbreviated Dial Group Assignment for Extensions. You can set the quantity of bins available for each group; the quantity must be in multiples of 10. The bins used for each group cannot overlap any other group or the Common Abbreviated bins. The entries of the Group Abbreviated bins can be set either with Abbreviated Dial Number and Name or with Service Code 854. They can be dialled by Service Code 814 followed by the bin number. When using Service Codes 854 and 814 the bin number starts at 000 for each group regardless of the actual start bin number defined in Group Abbreviated Dial Area Setup.

DSS Console Chaining
DSS Console chaining allows an extension user with a DSS Console to chain to an Abbreviated Dialling number stored under a DSS Console key. The stored number dials out (chains) to the initial call. This can, for example, simplify dialling when calling a company with an Automated Attendant. You can program the bin for the company number under one DSS Console key and the client's extension number under the other DSS Console key in DSS key Assignment. The DSS Console user presses the first key to call the company, waits for the Automated Attendant to answer, then presses the second key to call the client extension.
The DSS Console user can also chain to an Abbreviated Dialling number dialled manually, from a Programmable Function Key or a One-Touch Key.

Using a Programmable Function Key
To streamline frequently called numbers, an Abbreviated Dialling Programmable Function Key can also store an Abbreviated Dialling bin number. When the extension user presses the key, the phone automatically dials out the stored number. This provides one-touch calling via a phone's function keys.

Conditions
None

Default Setting
There are no Group Abbreviated Dialling bins assigned.
There are no Abbreviated dial names or numbers assigned.
There are 1000 Common Abbreviated Dial bins assigned.
All Abbreviated Dial bins will use Trunk Group Routing.

Account Codes

Description
Account Codes are user-dialed codes that help the system administrator categorize and/or restrict trunk calls. The system has three types of Account Codes:
• Optional Account Codes
Optional Account Codes allow a user to enter an Account Code while placing a trunk call or anytime while on a call. This type of Account Code is optional; the system does not require the user to enter it.
• Forced Account Codes

Forced Account Codes require an extension user to enter an Account Code every time they place a trunk call. If the user doesn't enter the code, the system prevents the call. As with Optional Account Codes, the extension user can enter an Account Code for an incoming call. However, the system does not require it. Forced Account Codes does not block emergency assistance calls.
Once set up in system programming, you can enable Forced Account Codes on a trunk-by trunk basis.

• Verified Account Codes

With Verified Account Codes, the system compares the Account Code the user dials to a list of up to 1000 pre-programmed codes. If the Account Code is in the list, the call goes through. If the code dialed is not in the list, the system prevents the call. Verified Account Codes can be from 3-16 digits long using the characters 0-9 and #. During programming, you can use "wild cards" to streamline entering codes into system memory. For example, the entry 123W lets users dial Verified Account Codes from 1230 through 1239.

Account Codes for Incoming Calls

The system can control the ability of extension users to enter Account Codes for incoming calls. When this option is enabled, a user can dial * while on an incoming call, enter an Account Code, and then dial * to return to their caller. If the option is disabled, any digits the user dials after answering an incoming call out dial on the connected trunk.

Hiding Account Codes

Account Codes can be optionally hidden from a telephone's display. This would prevent, for example, an unauthorized co-worker from obtaining a Verified Account Code by watching the display and making note of the digits that dial out. When hidden, the Account Code digits show as the character "*" on the telephone's display.

Account Code Capacity

Account Codes print along with the other call data on the SMDR record after the call completes. Account Codes can be 1-16 digits in length using 0-9 and #. Verified Account Codes can be from 3- 16 digits long.

Redialed Numbers Do Not Contain Account Codes

When using the Last Number Redial, Save or Repeat Dial features, the system will not retain Account Code information. Any number redialed with these features, the user will need to reenter an Account Code.

Note:

If a user enters *12345*203 926 5400*67890*, if the Last Number Redial feature is used, the system dials the number as 203 926 5400*67890*. The *67890* is not treated as an Account Code.

Conditions

(A.) If a user enters a code that exceeds the 16 digit limit, the system ignores the Account Code entry.
(B.) If the system has Account Codes disabled, the digits dialed (e.g., *1234*) appear on the SMDR report as part of the number dialed.
(C.) Do not use an asterisk within a PBX access code when using Account Codes. Otherwise, after the *, the trunk will stop sending digits to the central office.

Default Setting

Account Codes are disabled.

Related Features

Abbreviated Dialing

Abbreviated Dialing bins can contain stored Account Codes.

One-Touch Calling

To simplify Account Code entry, store the Account Code (e.g., *1234*) in a One-Touch Key.
Just press the key instead of dialing the codes.

PBX Compatibility

When using Account Codes, do not use an asterisk within a PBX access code. Otherwise, after the *, the trunk would stop sending digits to the central office.

Station Message Detail Recording

Account Codes appear on the SMDR report (even if they are hidden on the phone's display).

Operation

To enter an Account Code any time while on a trunk call:

The outside caller cannot hear the Account Code digits you enter.
You can use this procedure if your system has Optional Account Codes enabled. You may also be able to use this procedure for incoming calls.
This procedure is not available at SLTs.

1. Dial *.
OR
Press your Account Code key (SC 851: code 50).
2. Dial your Account Code (1-16 digits, using 0-9 and #).
If Account Codes are hidden, each digit you dial will show an "*" character on the telephone's display.
3. Dial *.
OR
Press your Account Code key (SC 851: code 50).

To enter an Account Code before dialing the outside number:

If your system has Forced Account Codes, you must use this procedure. If it has Verified Account Codes, you can use this procedure instead of letting the system prompt you for your Account Code. You may also use this procedure if your system has Optional Account Codes.
If your system has Verified Account Codes enabled, be sure to choose a code programmed into your Verified Account Code list.

1. Access trunk for outside call.
You can access a trunk by pressing a line key or dialing a code (except 9). Refer to Central Office Calls, Placing for more information.
2. Dial *
OR
Press your Account Code key (SC 851: code 50)
3. Dial your Account Code (1-16 digits, using 0-9 and #).
If Account Codes are hidden, each digit you dial will show an “*” character on the telephone’s display.
4. Dial *.

To enter an Account Code for an incoming call:
This procedure is not available at SLTs.
1. Answer incoming call.
If Account Codes for Incoming Calls is disabled, the following steps will dial digits out onto the connected trunk.
2. Dial *.
3. Enter the Account Code.
You can enter any code of the proper length. Incoming Account Codes cannot be Forced or Verified.
4. Dial *.

To enter an Account Code while placing a trunk call:
If your system has Forced Account Codes, you must follow this procedure.
1. Access trunk for outside call.
You can access a trunk by pressing a line key or dialing a trunk access code.
With Forced Account Codes, you hear, “Please enter an Account Code.” Your display shows: ENTER ACCOUNT CODE.
2. Dial *.
3. Dial your Account Code (1-16 digits, using 0-9 and #).
4. Dial *
If the system has Forced Account Codes and you don’t enter a code, your call cannot go through. You can, however, dial ** to bypass Forced Account Code entry.
5. Dial number you want to call.

To enter an Account Code at a single line set:
1. Access trunk for outside call.
You can access a trunk by dialing a trunk access code.
2. Dial *.
3. Enter Account Code (1-16 digits).
4. Dial *.
5. Dial number you want to call.

ACI (Audio Communications Interface)

Description
The Analogue Communications Interface (ACI) feature uses a PGDU card to provide two analogue audio ports (with associated relays) for Background Music, External Paging or external music input. The system allows up to 3 PGDU cards (one in each main or expansion unit), for a maximum of 6 audio ports.

• Music on Hold
You can connect up to two customer provided music sources to each PGDU module. A maximum of 6 external music sources can be installed via the PGDU inputs. This lets you add additional music sources if the external source on the Main Unit or the internal source are not adequate. By using PGDU cards, you could have a different music source for each trunk or DDI number.
When the system switches the ACI analogue port to a trunk on Hold, the PGDAD relay associated with the ACI analogue port closes. You can use this capability to switch on the music source, if desired.

• External Paging
An ACI analogue port can also be an External Page output. A maximum of 6 external paging zones can be installed via the PGDU cards. When connected to customer-provided External Paging equipment, the ACI port provides External Paging independent of the Main Unit external paging output. To use the External Paging, an extension user just dials the external paging service code and makes the announcement. The system broadcasts the announcement from the ACI analogue port and simultaneously closes the associated PGDU relay. You can use the relay closure to control the External Paging amplifier, if required.

PGDU Ports
Each PGDU card installed in the phone system has two analogue channels. During installation, the first PGDU card you install uses ports 1 & 2; the second PGDU card uses port 3 & 4, etc.

Conditions

The devices connected to the PGDU card must be compatible with the specifications in the system Hardware Manual.
Background Music: ACI software ports cannot be Background Music sources.

Default Setting
No PGDU cards installed.

Related Features
Door Box
The PGDU card is also used for connection of door box units.
Music on Hold ACI software ports can be Music on Hold music sources.
Paging, External
ACI software ports can provide External Paging with control, independent of the External Paging circuits on the Main Unit.

Alarm

Description
Alarm lets a system phone extension work like an Alarm clock. An extension user can have Alarm remind them of a meeting or an appointment. There are two types of Alarms:
• Alarm 1 (sounds only once at the preset time)
• Alarm 2 (sounds every day at the preset time)

Conditions
Single line sets will ring when the alarm sounds and Music on Hold will be heard when the handset is lifted to silence the alarm.

Default Setting
Alarm can be used.

Operation To set the alarm:
1. At system phone, press idle CALL key.
OR
At single line set, lift handset.
2. Dial 827.
3. Dial alarm type (1 or 2).
Alarm 1 sounds only once. Alarm 2 sounds each day at the preset time.
4. Dial the alarm time (24-hour clock).
For example, for 1:15 PM dial 1315.
A confirmation tone will be heard if the alarm has been set. If the alarm was not set, an error tone will be heard instead.
5. At system phone, press SPK to hang up.
OR
At single line set, hang up.

To silence an alarm:
1. At system phone, press CLEAR.
OR
At single line set, lift handset.
The single line set user will hear Music on Hold when the handset is lifted.
To check the programmed alarm time:
1. Press CHECK.
2. Dial 827.
3. Dial alarm type (1 or 2).
The programmed time displays.
4. Press CLEAR.

To cancel an alarm:
1. At system phone, press idle CALL key.
OR
At single line set, lift handset.
2. Dial 827.
3. Dial alarm type (1 or 2).
4. Dial 9999.
5. At system phone, press SPK to hang up.
OR
At single line set, hang up.

Alphanumeric Display

Description
System display telephones have a 2-line, 16 character per line alphanumeric display that provides various feature status messages. These messages help the display telephone user process calls, identify callers and customize features.

Conditions
The contrast is not adjustable when the telephone has background music enabled.

Operation
To adjust the LCD contrast press the volume keys when the phone is idle.

Attendant Call Queuing

Description
Attendant extensions can have up to 32 incoming calls queued before additional callers hear busy tone. This helps minimize call congestion in systems that use the attendant as the overflow destination for unanswered calls. For example, you can program Direct Inward Lines and Voice Mail calls to route to the attendant when their primary destination is busy. With Attendant Call Queuing, these unanswered calls would normally “stack up” for the attendant until they can be processed. The 32 call queue total includes Intercom, DISA, DID, DDI, DIL, tie line and transferred calls. If the attendant doesn’t have an appearance for the queued call, it waits in line on a CALL key. If the attendant has more than 32 calls queued, an extension can Transfer a call to the attendant only if they have Busy Transfer enabled. Attendant Call Queuing is a permanent, non-programmable system feature.

Conditions
Attendant extensions are setup in Operator Setup.
Call Forwarding / Personal Greeting: Forwarding when unanswered or busy can only occur at the attendant if there are more than 32 calls in queue.
Busy Transfer is disabled in Hold and Transfer.

Default Setting
Enabled.

Background Music

Description
Background Music (BGM) sends music from a customer-provided music source to speakers in system phones. If an extension user activates it, BGM plays whenever the user’s extension is idle.

Conditions
Background Music requires a customer-provided music source. Refer to the system Hardware Manual.
Single Line Telephones: Background Music is not available on single line telephones.

Default Setting
Not installed.
BGM is allowed in Class of Service.

Related Features
Music on Hold: The system can broadcast music to callers on Hold.

Operation
To turn Background Music on or off:
1. Press idle CALL key.
2. Dial 825.

3. Press SPK to hang up.

Barge In

Description
Barge In permits an extension user to break into another extension user’s established call. This sets up a Conference-type conversation between the intruding extension and the parties on the initial call. With Barge In, an extension user can get a message through to a busy co-worker right away. There are two Barge In modes: Monitor Mode (Silent Monitor) and Speech Mode. With Monitor Mode, the caller Barging In can listen to another user’s conversation but cannot participate. With Speech Mode, the caller Barging In can listen and join another user’s conversation.

CAUTION
The use of monitoring, recording, or listening devices to eavesdrop, monitor, retrieve, or record telephone conversation or other sound activities, may be illegal in certain circumstances. Legal advice should be sought prior to implementing any practice that monitors or records any telephone conversation. Some form of notification to all parties to a telephone conversation may be required, such as using a beep tone or other notification methods or requiring the consent of all parties to the telephone conversation, prior to monitoring or recording the telephone conversation.

Conditions
Conference: An extension user cannot Barge In on a Conference.
Intercom: An extension user cannot Barge In on an Intercom call if one of the Intercom callers is using
Handsfree Answerback: Both Intercom parties must have lifted the handset or pressed SPK. Off Hook Signaling
If the system has Automatic Off Hook Signaling, an extension user can Barge In on an Intercom call only if the second extension appearance is busy or ringing. Transfer into Conference: Barge in must be enabled for transfer into conference operation.

Default Setting
Service Code for Barge In is 810.
'Barge in Monitor Mode' is off in Class of Service.
'Break In' is on in Class of Service.
'Broken In' (the ability for another extension to Break In to your extension) is on in Class of Service.
'Intrusion Tone' is on in Class of Service.
'Deny Multi Barge In' is off on in Class of Service.
'Intrusion Tone' repeat time is set to 0 (not repeated) in Timers.

Related Features
Programmable Function Keys: Function keys simplify Barge In operation.

Operation
To Barge In after calling a busy extension:
The call must be set up for about 10 seconds before you can Barge In.
Listen for busy/ring or busy tone.
1. Call busy extension.
2. Press Barge In key (SC 851: 34).

To Barge in without first calling the busy extension:
1. Press idle CALL key.
2. Dial 810.
OR
Press Barge In key (SC 851: 34).
3. Dial busy extension.

To Barge In to a Conference call, dial the extension number of a user active on a Conference call. When a new call is added to the conference, an intrusion tone is heard by all parties, depending on system programming, and all display system phones show the joined party.

If Barge In is not possible:
- the extension user will hear a warning tone
- the DISA user will be rerouted to the defined ring group
OR
- the tie line user will hear a busy tone.

Busy Status Display

Description
Busy status display will show the details of a busy co-worker on the system phone display. When a call is made to a co-worker that is busy on the phone the display will show the details of their caller (which extension or line they are talking to). This can be useful if the caller wants to break in to the call.

Conditions
The feature is only available at a system phone with a display.
The feature is not available if Transfer without holding is enabled and you want to press a busy line key (Transfer without holding will allow you to wait off hook for the busy line).
The feature is also not available if Automatic Override is enabled (Automatic Override will allow you to wait off hook for the busy co-worker).

Default Setting
'Display detailed state of called party' (Busy Status Display) is on in Class of Service.
'KST Automatic Override/SLT Call Waiting' is off Class of Service.
'Transfer without holding' is off in Class of Service.

Related Features
Off Hook Signalling: Wait off hook for a busy co-worker.
Transfer: Options for Non-Hold Transfer.

Operation
The operation is automatic once it is enabled.

Call Deflection

Description
Call Deflection allows the system to redirect an incoming ISDN call to an off premise location without the need to use any trunk ports for the duration of the call.
Call Deflection is initiated by the system when the incoming ISDN call is presented, the system requests the Network to redirect the call to the off premise location. The Network will route the call to the redirected number.
Each trunk can be set to use either Call Rerouting (ETSI EN 300 207-1 Clause 10.5) or Call Deflection (ETSI EN 300-207-1 Clause 9.2.4.5). The operation of both types is identical for the user.

The redirected number is stored in an Abbreviated Dial location within the system and can therefore be easily controlled by the user.
The following types of call forward off premise use an Abbreviated Dial to define the redirected number and can use the Call Deflection feature.

Call Forward set by the Extension
The extension can set call forward off premise by using the 'Call Forward to Abbreviated Dial' Feature.

Department Group Forward
ISDN calls routed to the Department Group can use the Call Deflection feature.

DDI Call Forward
A DDI call can be routed to an Abbreviated Dial by either entering the Service Code for Abbreviated Dial (default = 813) and location number in 'Target 1' in the DDI Routing Table or entering 1xxx in 'Targets 2 or 3' (xxx is the Abbreviated Dial location number 000 to 999).

Call Forwarding - Fixed.
A DDI call can be routed to an Abbreviated Dial by either entering the Service Code for Abbreviated Dial (default = 813) and location number in 'External' entry of Fixed Call Forward.

Conditions
The system supports either Call Deflection or Call Rerouting, the operations are the same but you must select the corresponding service that is provided by the Network.

If Call Deflection is not supported by the Network the system will attempt to route the call via a separate outgoing trunk port.
Call Deflection can only occur when an incoming ISDN call is at the 'Call Receiving' state. The call can not be redirected if it is in any other state, for example, if the system sends a VRS queue announcement the call is in the 'Connected' state and Call Deflection can not be used.
If the Abbreviated Dials are set to 'Internal Access Mode' in Abbreviated Dial Function Setup the system will automatically remove the trunk access code when sending the the redirected number to the network.

Default Setting
All Abbreviated Dials are set to a separate outgoing trunk port in Abbreviated Dial Number and Name.
All trunks are set to use a separate outgoing trunk port in ISDN Call Forward Method.

Related Features
Call Forwarding, Fixed: Fixed Call Forwarding can automatically forward an extensions calls to an outside number.
Call Forward to Abbreviated Dial: Call forward codes can be used to forward to abbreviated dial that will route off premise.
Department Group - Call Forward: Forward a Department Groups calls off premise.
Door Box: Door Boxes must be programmed in order for the calls to be transferred off-premise.

Operation
Refer to the user features shown above for each method of setting call forward off premise.

Call Forward DND Override

Description
An extension user can override Call Forwarding or Do Not Disturb at another extension.

Conditions
None.

Default Setting
'DND/Call Forward Override (Bypass Call)' is on in Class of Service
'DND/Call Forward Override (Bypass Call)' service code is 807.

Related Features
Programmable Function Keys
Function keys simplify Call Forwarding/DND Override operation.

Operation
To override an extension’s Call Forwarding or Do Not Disturb:
1. Call the forwarded or DND extension.
2. Press Override key (SC 851: 37) or dial 807.

Call Forwarding - Fixed

Description
Fixed Call Forwarding is a type of forwarding that is permanently in force at an extension. Calls to an extension with Fixed Call Forwarding enabled automatically reroute - without any user action. Unlike normal Call Forwarding (which is turned on and off by extension users), Fixed Call Forwarding is set by the administrator in system programming. Fixed Call Forwarding complements Voice Mail, for example. The administrator can program Fixed Call Forwarding to send a user’s unanswered calls to their Voice Mail mailbox. Each individual user no longer has to manually set this operation.
In system programming, the administrator can set the Fixed Call Forwarding destination and type for each extension and virtual extension. The forwarding destination can be an on- or off-premise extension or Voice Mail. The Fixed Call Forwarding types are:
• Fixed Call Forwarding with Both Ringing
• Fixed Call Forwarding when Unanswered
• Fixed Call Forwarding Immediate
• Fixed Call Forwarding when Busy or Unanswered
• Fixed Call Forwarding Off-Premise
Fixed Call Forwarding reroutes the following types of incoming calls:
• Ringing intercom calls from co-worker’s extensions

• Calls routed from the VRS or Voice Mail
• Direct Inward Lines
• DISA and DID calls to the forwarded extension
• Transferred calls

Fixed Call Forwarding Chaining
Fixed Call Forward Chaining allows Fixed Call Forwards to loop from one extension to the next.
For example, you could have the chain 301 + 302 + 303 + 304 set up for Fixed Call Forwarding when Busy. If extension 301 is busy, calls to 301 route to 302. If 302 is also busy, the calls route to 303 and so on. Chaining allows you to set up very basic hunting between co-workers. Keep the following in mind when setting up Fixed Call Forwarding Chaining:
• If Fixed Call Forwarding Chaining forms a complete Call Forwarding loop (i.e., 301 + 302 + 303 + 301), the system rings the last extension in the chain (303). It does not complete the loop.
• If Fixed Call Forwarding Chaining finds an extension with user-implemented Call Forwarding in the middle of a chain, it rings that extension. It does not continue routing to the other extensions in the chain.
• If one of the extensions in a Fixed Call Forwarding chain has its fixed option set for Both Ringing (1), the system rings that extension. It does not continue routing to the other extensions in the chain.
• The receiving extension's display shows:
STA AAA AAA is the extension that initially placed the call.
TRANSFER STA BBB BBB is the first extension in the Fixed Call Forwarding chain.

Conditions
Call Forwarding an extension in a Department Group will prevent that extension from receiving Department Pilot Calls.
Multiple Directory and Call Coverage Key calls follow Call Forwarding.
Ring Group calls do not follow Call Forward to voice mail.

Default Setting
'No answer time for Call Forward' is set to 10 seconds in Timers.

Related Features
Alphanumeric Display: When a call is Fixed Call Forwarded, the display at the destination shows from which extension the call was routed.
Call Forwarding: User entered Call Forwarding overrides Fixed Call Forwarding.
Call Forwarding, Off-Premise: An extension user can forward their calls to an outside telephone number.
Multiple Directory Numbers: Calls to virtual extension numbers follow the Fixed Call Forwarding assignment of their virtual port.

Operation
None

Call Forwarding with Follow Me

Description
While at a co-worker's desk, a user can have Call Forwarding with Follow Me redirect their calls to the co-worker's extension. This helps an employee who gets detained at a co-worker's desk longer than expected. To prevent losing important calls, the employee can activate Call Forwarding with Follow Me from the co-worker's phone.
Call Forwarding with Follow Me reroutes calls from the destination extension. To reroute calls from the initiating (forwarding) extension, use Call Forwarding.

Conditions
Call Forwarding an extension in a Department Group will prevent that extension from receiving Department Pilot Calls.
If a Programmable Function key is not defined for Call Forwarding (10 - 17), the DND key flashes to indicate that the extension is call forwarded.

Default Setting
'Follow Me' is on in Class of Service.

Related Features
Programmable Function Keys: Function keys simplify Call Forwarding with Follow Me operation.

Operation
To activate Call Forward Follow Me:

1. At a system phone other than your own, press idle CALL key and dial 888.
OR
Press Call Forward (Station) key (SC 851: 15).
OR
At SLT other than your own, lift handset and dial 888.
2. Dial 3 + Dial your own extension number (i.e., the source).
3. Dial Call Forwarding Type:
2 = All Calls
3 = Outside calls only
4 = Intercom calls only
4. SPK (or hang up at SLT) if you dialed 888 in step 1.
Your Call Forwarding (Station) Programmable Function Key flashes when Call Forwarding is activated.

To cancel Call Forward Follow Me:
1. At system phone, press idle CALL key and dial 888.
OR
Press Call Forward (Station) key (SC 851: 15).
OR
At SLT, lift handset and dial 888.
2. Dial 0.
3. SPK (or hang up at SLT) if you dialed 888 in step 1.
Your Call Forwarding (Station) Programmable Function Key goes out.

Call Forward Off Premise

Description
Off-Premise Call Forwarding allows an extension user to forward their calls to an off-site location. By enabling Off-Premise Call Forwarding, the user can stay in touch by having the system forward their calls while they are away from the office. The forwarding destination can be any phone number the user enters.
Off-Premise Call Forwarding reroutes the following types of incoming calls:
• Ringing intercom calls from co-worker's extensions
• Calls routed from the VRS or Voice Mail
• Direct Inward Lines
• DISA, DID and DDI calls to the forwarded extension
• Transferred calls

Off-Premise Call Forwarding does not reroute Call Coverage keys, Multiple Directory Number keys, or Ring Group calls.
Off-Premise Call Forward for Door Boxes.
Off-Premise Call Forwarding allows Door Box callers to be transferred automatically to the preprogrammed external party. The destination telephone number is stored in the Common Abbreviated Dial area. This feature may be used in case a co-worker is out of the office. All incoming calls for their extension will be automatically transferred to their external number (example: cell phone).
Off-Premise Call Forward for Door Boxes can be transferred to the external party through ISDN lines only.
Off-Premise Call Forwarding can reroute an incoming trunk call only if the outgoing trunk selected has disconnect supervision enabled.

Conditions
Call Forwarding Off-Premise requires ISDN, loop start trunks with disconnect supervision or ground start trunks.
The trunk access code and the outside telephone number combined cannot exceed 24 digits.
Normally, the system does not allow the chaining of Call Forwards. For example, extension 316 forwards to 318, and 318 in turn forwards to 320. Calls to 316 route to 318. Calls to 318
route to 320. The system does allow a single chain, however, if the second extension in the chain is forwarded off-premise (713 + 6 + trunk access code + destination telephone number).
Call Forwarding an extension in a Department Group will prevent that extension from receiving Department Pilot Calls.
If a Programmable Function key is not defined for Call Forwarding (10 - 17), the DND key flashes to indicate that the extension is call forwarded.

Default Setting
'Trunk to Trunk Transfer' is disabled in Trunk Basic Data Setup, this will be required for trunk to trunk calls.
No outgoing route is set up in Outgoing Route Setup for trunk to trunk calls.
'External Call Forward (Off-Premise)' is off in Class of Service.

Related Features
Call Forwarding, Fixed: Fixed Call Forwarding can automatically forward an extensions calls to an outside number.

Call Forward to Abbreviated Dial: Call forward codes can be used to forward to abbreviated dial that will route off premise.
Department Group - Call Forward: Forward a Department Groups calls off premise.
Door Box: Door Boxes must be programmed in order for the calls to be transferred off-premise.
Toll Restriction: The outside number Off-Premise Call Forwarding dials can only be a number normally allowed by the forwarded extension's Toll Restriction.
Trunk to Trunk Forwarding: Forward all calls received on a trunk port to an off premise number.
Voice Response System (VRS): In systems with a DSPDB daughter board for VRS, callers to an extension forwarded off-premise hear, "Please hold on, your call is being rerouted."

Operation
To activate Call Forwarding Off-Premise:
1. At system phone, press idle CALL key + Dial 713.
OR
Press Call Forward (Device) key (SC 851: 17)
OR
At SLT, lift handset Dial 713.
2. Dial 6 + trunk access code.
Trunk access code or 804 + Line Group (1-9, 01-99 or 001- 100) or 805 + Line number (e.g., 05 or 005 for line 5.
3. Dial the outside number to which your calls should be forwarded.
4. (System Phone only) Press HOLD.
5. Press SPK (or hang up at SLT) to hang up if you dialed 713 in step 1.
Your DND or Call Forwarding (Device) Programmable Function Key flashes.

To cancel Call Forwarding Off-Premise:
1. At system phone, press idle CALL key + Dial 713.
OR
Press Call Forward (Device) key (SC 851: 17)
OR
At SLT, lift handset and dial 713.
2. Dial 6 + HOLD.
3. Press SPK (or hang up at SLT) to hang up if you dialed 713 in step 1.
Your DND or Call Forwarding (Device) Programmable Function Key stops flashing.
Off-Premise Call Forwarding for Door Boxes:
These operations are performed at the Door Box Ringing Extension only.

To activate Call Forwarding Off-Premise by Door Box:
1. At system phone, press idle CALL key + Dial 822.
OR
Press Call Forward (Device) key (SC 851: 54)
OR
At SLT, lift handset Dial 822.
2. Dial the Door Box number (1-8).
3. Dial the Abbreviated Dialing number to which the calls should be forwarded.
4. Press SPK (or hang up at SLT) to hang up.
Your DND or Off-Premise Call Forwarding Programmable Function Key flashes.

To cancel Call Forwarding Off-Premise:
1. At system phone, press idle CALL key + Dial 822.
OR
Press Call Forward (Device) key (SC 851: 54)
OR
At SLT, lift handset and dial 822.
2. Dial 0.
3. Press SPK (or hang up at SLT) to hang up.
Your DND or Off-Premise Call Forwarding Programmable Function Key stops flashing.

Call Forwarding to Abbreviated Dial

Description
Call Forwarding to Abbreviated Dial allows an extension user to forward their calls to an off-site location. By enabling Call Forwarding to abbreviated dial, the user can stay in touch by having the system forward their calls while they are away from the office. Off-Premise Call Forwarding will route the off-site phone number over the trunk route set for the abbreviated dial bin.
Off-Premise Call Forwarding reroutes the following types of incoming calls:

• Ringing intercom calls from co-worker's extensions
• Calls routed from the VRS or Voice Mail
• Direct Inward Lines
• DISA, DID and DDI calls to the forwarded extension
• Transferred calls

Call Forwarding does not reroute Call Coverage keys, Multiple Directory Number keys, or Ring Group calls).
All incoming calls for their extension will be automatically transferred to their external number.
Off-Premise Call Forwarding allows Door Box callers to be transferred automatically to the preprogrammed external party. The destination telephone number is stored in the Common Abbreviated Dial area.
Off-Premise Call Forward for Door Boxes can be transferred to the external party through ISDN lines only.

Off-Premise Call Forwarding can reroute an incoming trunk call only if the outgoing trunk selected has disconnect supervision enabled

Conditions
Call Forwarding Off-Premise requires ISDN, loop start trunks with disconnect supervision or ground start trunks.
The system does not allow the chaining of Call Forwards. For example, extension 316 forwards to 318, and 318 in turn forwards to 320. Calls to 316 route to 318. Calls to 318 route to 320. The system does allow a single chain, if the second extension in the chain is forwarded off-premise (848 + 1 + 813 + bin).
Call Forwarding an extension in a Department Group will prevent that extension from receiving Department Pilot Calls.
If a Programmable Function key is not defined for Call Forwarding (10 - 17), the DND key flashes to indicate that the extension is call forwarded.

Default Setting
Trunk to Trunk Transfer' is disabled in Trunk Basic Data Setup, this will be required for trunk to trunk calls.

'External Call Forward (Off-Premise)' is off in Class of Service.

Related Features
Call Forwarding, Fixed: Fixed Call Forwarding can automatically forward an extensions calls to an outside number.
Call Forward, Off Premise: Call forward codes can be used to forward to a specific outside number.
Door Box: Door Boxes must be programmed in order for the calls to be transferred off-premise.
Toll Restriction: The outside number Call Forwarding dials can only be a number normally allowed by the forwarded extension's Toll Restriction.
Voice Response System (VRS): In systems with a DSP daughter board for VRS, callers to an extension forwarded off-premise hear, "Please hold on, your call is being rerouted."

Operation
To activate Call Forwarding Off-Premise:
1. At system phone, press idle CALL key + Dial 848.
OR
Press Call Forward key (SC 851: 10)
OR
At SLT, lift handset Dial 848.
2. Dial 1 + 813 +1.
813 is the Abbreviated dial access code, can be replaced with the Group Abbreviated dial access code 814.
3. Dial the bin number to which your calls should be forwarded.
4. Press SPK (or hang up at SLT) to hang up if you dialed 848 in step 1.
Your DND or Call Forwarding (Device) Programmable Function Key flashes.

To cancel Call Forwarding Off-Premise:
1. At system phone, press idle CALL key + Dial 848.
OR
Press Call Forward (Device) key (SC 851: 10)
OR
At SLT, lift handset and dial 848.
2. Dial 0.
3. Press SPK (or hang up at SLT) to hang up if you dialed 848 in step 1.
Your DND or Call Forwarding (Device) Programmable Function Key stops flashing.

Off-Premise Call Forwarding for Door Boxes:
These operations are performed at the Door Box Ringing Extension only.
To activate Call Forwarding Off-Premise by Door Box
1. At system phone, press idle CALL key + Dial 822.
OR

Press Call Forward (Device) key (SC 851: 54)
OR
At SLT, lift handset Dial 822.
2. Dial the Door Box number (1-8).
3. Dial the Abbreviated Dialing number to which the calls should be forwarded.
4. Press SPK (or hang up at SLT) to hang up.
Your DND or Off-Premise Call Forwarding Programmable Function Key flashes.

To cancel Call Forwarding Off-Premise by Door Box:
1. At system phone, press idle CALL key + Dial 822.
OR
Press Call Forward (Device) key (SC 851: 54)
OR
At SLT, lift handset and dial 822.
2. Dial 0.
3. Press SPK (or hang up at SLT) to hang up.
Your DND or Off-Premise Call Forwarding Programmable Function Key stops flashing.

Call Forwarding

Description
Description
Call Forwarding permits an extension user to redirect their calls to another extension. Call Forwarding ensures that the user's calls are covered when they are away from their work area. The types of Call Forwarding are:
• Call Forwarding when Busy or Not Answered
Calls to the extension forward when busy or not answered.
• Call Forwarding Immediate
All calls forward immediately to the destination, and only the destination rings.
• Call Forwarding with Both Ringing
All calls forward immediately to the destination, and both the destination and the forwarded extension ring (not for Voice Mail).
• Call Forwarding when Unanswered
Calls forward only if they are unanswered (Ring No Answer).
• Personal Answering Machine Emulation
Allows the extension to emulate an answering machine. Refer to 'Voice Mail' for more.
Call Forwarding will reroute calls ringing an extension, including calls transferred from another extension. The extension user must enable Call Forwarding from their phone.
To redirect calls while a user is at another phone, use "Call Forwarding with Follow Me".
A periodic VRS announcement may remind users that their calls are forwarded.

Conditions
Normally, the system does not allow the chaining of Call Forwards. For example, extension 316 forwards to 318, and 318 in turn forwards to 320. Calls to 316 route to 318. Calls to 318 route to 320. The system does allow a single chain, however, if the second extension in the chain is forwarded off- premise (713 + 6 + trunk access code + destination telephone number).
Periodic reminder message requires a DSP daughter board for Voice Response System (VRS).
Call Forwarding an extension in a Department Group will prevent that extension from receiving Department Pilot Calls.
If a Programmable Function key is not defined for Call Forwarding (10 - 17), the DND key flashes to indicate that the extension is call forwarded.
Ring Groups do not follow Call Forward to voice mail.
Multiple Directory and Call Coverage Key calls do not follow Call Forwarding.

Default Setting
'Call Forward Immediately' is on in Class of Service.
'Call Forward Busy' is on in Class of Service.
'Call Forward No Answer' is on in Class of Service.
'Call Forward Dual Ring' is on in Class of Service.
'Follow Me' is on in Class of Service.
'No Answer Time for Call Forward' is set to 10 seconds in Timers.

Related Features
Call Forwarding, Fixed
Fixed Call Forwarding is a permanent type of forwarding that automatically reroutes calls under certain condition - without any user action. User entered Call Forwarding overrides Fixed Call Forwarding.
Call Forwarding, Off-Premise
An extension user can forward their calls to an off-premise location.

Call Forwarding with Follow Me
While away from their desk, a user can redirect their calls to a co-worker's extension.
Call Forwarding/Do Not Disturb Override
Override Call Forwarding or DND at another extension.
Central Office Calls, Answering
When a call is transferred because of Call Forwarding No Answer, Call Forwarding Busy, or DND, the Reason for Transfer option can display to the transferred extension why the call is ringing to their phone.
Department Calling
An extension user can forward their calls to a Department number.
Direct Inward System Access - DISA
Setting call forward for an extension user via an incoming DISA line.
Do Not Disturb
If an extension user activates DND option 4, the system prevents other extensions from forwarding calls to them. If an extension already receiving forwarded calls activates DND option 4, callers to the forwarded extension hear DND tone.
Programmable Function Keys
Function keys simplify Call Forwarding operation.
Voice Response System (VRS)
The periodic reminder message requires a Voice Response System (VRS).

Operation
To activate or cancel Call Forwarding:
1. Press idle CALL key (or lift handset) + Dial 888.
OR
Press Call Forwarding key (SC 851: code 16).

2. Dial Call Forwarding condition:
1 = Personal Answering Machine Emulation (then skip to step 4 - refer also to “Voice Mail”).
2 = Busy or not answered
4 = Immediate
6 = Not answered
7 = Immediate with simultaneous ringing (not for Voice Mail)
0 = Cancel

3. Dial destination extension, Voice Mail master number or press Voice Mail key.

4. Dial Call Forwarding type:
2 = All calls
3 = Outside calls only
4 = Intercom calls only

5. Press SPK to hang up (hang up at SLT).
Your DND or Call Forwarding (Station) Programmable Function Key flashes when Call Forwarding is activated.

Also available are Service Codes 848 (Call Forward Immediate), 843 (Call Forward Busy), 845 (Call Forward No Answer, 844 (Call Forward, Busy/No Answer, or 842 (Call Forward Both Ring).
OR
Programmable Function Key: code 10 for Forward All Calls Immediately
Programmable Function Key: code 11 for Forward when Busy
Programmable Function Key: code 12 for Forward when Unanswered
Programmable Function Key: code 13 for Forward Busy/No Answer
Programmable Function Key: code 14 for Forward with Both Ringing
When you enable Call Forwarding, your Call Forwarding key flashes slowly. If you don't have a Call Forwarding key, DND flashes slowly.
You will hear interrupted dial tone when call forward is set.

Group Call Pickup

Group Call Pickup allows an extension user to answer a call ringing an extension in a Pickup Group. This permits co-workers in the same work area to easily answer each other's calls. The user can intercept the ringing call by dialling a code or pressing a programmed Group Call Pickup key. If several extensions within the group are ringing at the same time, Group Call Pickup intercepts the call based on the extension's priority within the Pickup Group.
With Group Call Pickup, a user can intercept the following types of calls:
A call ringing the user's own pickup group

A call ringing another pickup group when the user knows the group number
A call ringing another pickup group when the user doesn't know the group number

Default Setting
'Call Pickup' service code is 867.
'Call Pickup for Another Group' service code is 869.
'Call Pickup for Specified Group' service code is 868.
'Direct Call Pickup for Own Group' service code is 856.
'Direct Extension Call Pickup' service code is 715.
'Call Pickup' is on in Class of Service.
'Call Pickup for Another Group' is on in Class of Service.
'Call Pickup for Specified Group' is on in Class of Service.
'Direct Call Pickup for Own Group' is on in Class of Service.
'Call Picked-Up Telephone' is on in Class of Service.

To answer a call ringing another phone in your Pickup Group:
1. Lift handset.
2. (System Phone only) Press Group Call Pickup key (SC 851: 24).
OR
Dial 856 or 867.
Service Code 867 can pick up any call in the group, plus any Ring Group calls. Service Code 856 cannot pick up Ring Group calls.

To answer a call ringing a phone in another Pickup Group when you don't know the group number:
1. Lift handset.
2. (System Phone only) Press Group Call Pickup key (SC 851: 25).
OR
Dial 869.

To answer a call ringing a phone in another Pickup Group when you know the group number:
1. Lift handset.
2. (System Phone only) Press Group Call Pickup key (SC 851: 26 + group).
OR
Dial 868 and the group number (1-9 or 01-32).

To answer a call ringing a phone:
1. Lift handset.
2. Dial 715 and the extension number of the ringing phone.

Call Redirect

Description
Call Redirect allows a system phone user to transfer a call to a pre-defined destination (such as an operator, voice mail, or another extension) without answering the call. This can be useful if you are on a call and another rings in to your extension. By pressing the Call Redirect key, the call is transferred, allowing you to continue with your current call.
This feature works with the following types of calls:
• Normal trunk call
• DID
• DISA
• DIL
• ICM
The following types of calls cannot be redirected with the feature:
• Transferred
• Department Group (all ring mode)
• Door Box
• Virtual Extension

Conditions
After pressing the Call Redirect key, the call will not recall the extension.
The pre-defined destination has to be an extension number or voice mail pilot number.

Default Setting
'Call Redirect is off in Class of Service.
Call Redirect programmable function key is 49 + destination extension number.

Related Features
None.

Operation
To redirect a ringing call:
1. With an incoming call ringing your extension, press the Call Redirect key (SC 851: 49 + Destination Extension Number) without lifting the handset or pressing the CALL keys.
A confirmation tone is heard over the telephone's speaker.

Call Timer

Description
Call Timer lets a system phone user time their trunk calls on the telephone display. This helps users that must keep track of their time on the phone. For incoming analogue trunk calls, the Call Timer begins as soon as the user answers the call. For outgoing analogue trunk calls, the Call Timer starts about 10 seconds after the user dials the last digit or when the called party answers if line reversal is available. For ISDN trunks the call timer begins when the called party answers.

Conditions
None

Default Setting
'Call Duration Display' is on in Class of Service.
'External Call Inter-digit time' is set to 10 seconds in Timers.

Related Features
Alphanumeric Display
Line Reversal - Analogue Trunks: Line reversal will detect when the called party answers and start the call timer.

Operation
To time your trunk calls:
1. Place trunk call.
The timer starts automatically.

Call Waiting

Description With Call Waiting, an extension user may call a busy extension and wait in line (Camp-On) without hanging up. When the user Camps-On, the system signals the busy extension with two beeps indicating the waiting call. The call goes through when the busy extension becomes free. Call Waiting helps busy extension users know when they have additional waiting calls. It also lets callers wait in line for a busy extension without being forgotten.

Conditions
None.

Default Setting
Service code for 'Call Waiting Answer/Split Answer for SLT' is 894.
'Interval of Call Waiting Tone' is set to 10 seconds in Service Tone Timers.

Related Features
Callback: If an extension user Camps-On and then hangs up, the system converts the Camp On to a Callback.
Dual Line Appearance/Off Hook Signaling: If an extension busy on a call has Off Hook Signaling, an incoming Intercom calls rings the idle second line appearance.
Off Hook Signaling: The Off Hook Signaling Enhancements give an extension the ability to block a caller from dialing 804 to Camp On and/or DID callers from automatically camping on.
Programmable Function Keys: Function keys simplify Call Waiting/Camp On operation.
Transfer: An extension user may be able to Transfer a call to a busy extension.
Trunk Queuing/Camp-On: Trunk Queuing lets an extension user Camp-On to a trunk.

Operation

To Camp-On to a busy extension:
1. Call busy extension.
2. Dial 850 or Press Camp-On key (SC 851: 35).
3. Do not hang up.
To Camp-On to a trunk, see Trunk Queuing

To cancel a Camp-On request:
1. Hang up.
2. At system phone, press idle CALL key and Dial 870.
OR
At system phone, press Camp-On key (SC 851: 35).
OR
At single line set, lift handset and dial 870.

To Split (answer a waiting call) at a single line telephone:
Listen for Camp On beep.
1. Single Line Telephone:
Hook flash and dial 894.

Callback

Description
When an extension user calls a co-worker that doesn't answer, they can leave a Callback request for a return call. The user does not have to repeatedly call the unanswered extension back, hoping to find it idle.
The system processes Callback requests as follows:
1. Caller at extension A leaves a Callback at extension B.
Caller can place or answer additional calls in the mean time.
2. When extension B becomes idle, the system rings extension A. This is the Callback ring.
3. Once caller A answers the Callback ring, the system rings (formerly busy) extension B.
If caller A doesn't answer the Callback ring, the system cancels the Callback.
4. As soon as caller B answers, the system sets up an Intercom call between A and B.
Callback Automatic Answer determines how an extension user answers the Callback ring. When
Callback Automatic Answer is enabled, a user answers the Callback ring when they lift the handset.
When Callback Automatic Answer is disabled, the user must press the ringing line appearance to answer the Callback ring.

Conditions
An extension can leave only one Callback request at a time.

Default Setting
'Call Back' (Callback Automatic Answer) is on in Keyphone Options.
'Camp-On Extension call back time' is set to 15 seconds in Timers.
'Camp-On cancel time' is set to 64800 seconds (18 hours) in Timers.

Related Features
Call Waiting (Camp-On): If an extension user initiates a Callback but does not hang up, their extension Camps-On to the busy extension.
Programmable Function Keys: Function Keys simplify Callback operation.

Operation
To place a Callback:
1. Call unavailable (busy or unanswered) extension.
2. Dial 850 or Press Callback key (SC 851: 35).
3. Hang up.
4. Lift handset when busy extension calls you back.
If the unavailable extension was unanswered (not busy), the Callback goes through after your co-worker uses their phone for the first time.
If you have Callback Automatic Answer, you automatically place a call to the formerly busy extension when you lift the handset. If you don't have Callback Automatic Answer, you must press the ringing line appearance to place the call.

To cancel a Callback:
1. At system phone, press idle CALL key and Dial 870.
OR
At system phone, press Camp-On key (SC 851: 35).

OR
At single line set, lift handset and dial 870.

To test Callback at your single line phone:
1. Lift the handset.
1. Dial 899.
2. Hang up.
3. When the phone rings, lift the handset.
You hear Music on Hold.
4. Hang up.

Caller ID Sending

Description
Caller ID sending allows the system to send calling party's telephone number (CGPN) for outgoing ISDN, H.323 and SIP trunk calls.
The CGPN is passed on by the Network provider to the called party.
Note - You may need to request this service with the ISDN Network provider.
The CGPN number can be set for each trunk and/or each extension, when both are set the extension will take priority over the trunk CGPN.

Conditions
This service may be restricted by the ISDN Network provider.

Default Setting
'CLIP Information Announcement' (sending Caller ID to the Network) is enabled in BRIU Setup and PRIU Setup.
'Extn CLIP' is enabled in Extension Outgoing Caller ID.
There are no entries for trunk or extension CLIP for ISDN, H.323 or SIP trunks (Trunk Outgoing Caller ID, Extension Outgoing Caller ID , SIP Networking and H.323 Networking).
'Set Calling Party Number' is on in Class of Service. If this is set to off the system will not include any calling party number information for outgoing ISDN calls.

Caller ID

Description
Caller ID allows a display system phone or Caller ID SLT to show an incoming caller's telephone number and optional name.
The system can receive caller ID on ISDN and Analogue trunks.

Second Call Display
While busy on a call, the telephone display can show the identity of an incoming trunk or Intercom call. For incoming trunk calls (not ring groups), the display will show the Caller ID or the trunk's name if Caller ID is not installed. For incoming Intercom calls, the display will show the calling extension's name. You can set up the system to present the second call data automatically or allow the user to select the operation of second calls (with service code 779).

Once installed and programmed, Caller ID is enabled for all types of trunk calls, including:
• Ring Group calls
• Calls transferred from another extension
• Calls transferred from the VRS
• Calls transferred from Voice Mail (unscreened)
• Direct Inward Lines (DILs)
• ISDN DDI Calls

Caller ID Number to Abbreviated Dial Name Tagging
If the received Caller ID number matches a number in an Abbreviated Dial bin the name assigned to the Abbreviated Dial bin will also be displayed.

Caller ID to Single Line Analogue Telephones
The system can send the caller ID to analogue single line telephones connected via an SLIU PCB.
The system will send the calling party number and name. The name will be the extension name for internal calls or the Abbreviated

Dial bin name for trunk calls.
Note. When caller ID is enabled the ringing pattern is fixed at 2 Sec ON/4 Sec OFF for internal and external calls.
The Caller ID is sent by the system after the first ring pulse (Ring Pulse Alerting Signal).
The sender type for SLIU extensions is set in SLT Basic Setup.

Caller ID Received on Analogue Trunks (COIU trunk ports)
The system will receive caller ID on analogue COIU trunk ports. The COIU will accept the caller ID after line reversal (Line Reversal Alerting Signal) or Ring Pulse (Ring Pulse Alerting Signal).
Type of Caller ID (FSK / DTMF)
The type of caller ID can be selected for each CO trunk and SLIU extension.
The detection type for CO trunks is set in Analogue Trunk Data Setup.

System Resources for Caller ID
For both FSK and DTMF detection on CO trunks a DSP resource must be available to receive the Caller ID. There are up to 64 resources available (NTCPU=32 and DSPDBU=32).
The DSP resources are not used for sending DTMF caller ID to SLIU extensions.
When sending FSK caller ID to SLIU extensions there are 4 resources available on the system.

Caller ID Sender - Queuing for Resources
When sending FSK caller ID to SLIU extensions the system can either present the call when no resource is available (No Queuing) or wait until a resource is available (Queue).
When No Queuing is selected the system will present the call immediately to the extension, if no resource is available then no caller ID will be sent to the extension.
When Queuing is selected the system will wait for a resource to become available before ringing the extension. This will reduce the chance of calls presented to SLIU extensions without caller ID.
While a call is queuing the SLIU extension will not ring, if the extensions goes off hook the user will not hear dial tone and cannot make outgoing calls. When they go back on hook the queued call will be presented without caller ID.
If no caller ID becomes available within a preset time the call will ring at the SLIU extension without caller ID. The timer is set by 'Call ID Sender Queuing' in Incoming Caller ID Setup.

Outputting Caller ID Data
The system includes the Caller ID data on the SMDR report. The report provides the incoming caller ID in the DIALED NUMBER field.

Caller ID Digits to Voice Mail
The system can send Caller ID digits to the voice mail.
When a trunk ‘001’ receives the Caller ID as ‘12345’, the protocol becomes ‘***60001*12345*’.

ISDN Calls Display Reason for No Caller ID Information
With Caller ID enabled, the system will provide information for ISDN calls that do not contain the Caller ID information. If the Caller ID information is restricted, the telephone display will show “PRIVATE”. If the system is not able to provide Caller ID information because the Caller ID information is not available, then the display will show “No Caller Info”.

Hardware Considerations
Caller ID is provided by the NTCPU. The DSP daughter board, which plugs onto the NTCPU, can provide additional resources for Caller ID if needed.

Conditions
To have pre-answer Caller ID from the voice mail, the call must be an unscreened transfer.
Caller ID must be supported by the Network provider.

Default Setting
Sending caller ID to voice mail is disabled in Trunk Basic Data Setup.
Caller ID receive is disabled for analogue trunks in Analogue Trunk Data Setup.
Caller ID receive type is set to FSK for analogue trunks in Analogue Trunk Data Setup.
Block Outgoing Caller ID is off in Class of Service.
Block Outgoing Caller ID is off in Trunk Basic Data Setup.
'Caller ID Information Display' is on in Class of Service.

Related Features
Caller ID - Sending: The system can send caller ID for outgoing calls.
Station Message Detail Recording: Caller ID information outputs on the SMDR report.
Voice Mail: The system can send Caller ID digits to the voice mail.

Central Office Calls - Answering

Description
The system provides flexible routing of incoming CO (trunks) calls to meet the exact site requirements.
This lets trunk calls ring and be answered at any combination of system extensions. For additional information on making trunk ring, refer to the Ring Group feature.

No Answer Step On
If the trunk is not answered at its original destination it can step on to an alternative target.

Universal Answer
Universal Answer allows an employee to answer a call by going to any system phone and dialing a unique Universal Answer code. The employee doesn't have to know the trunk number or dial any other codes to pick up the ringing trunk. You'll normally set up Universal Answer along with Universal Night Answer (see "Night Service"). When a Universal Night Answer call rings the External Paging, an employee can answer the call from the first available phone.

Conditions
Trunk Access Maps can restrict calls being answered.

Default Setting
All extensions can answer ringing trunk calls.
All trunks are set as 'Normal Type' in Incoming Service Type Setup.
All trunks ring at IRG 1 in IRG Assignment (Normal).
IRG 1 contains extension number 200 in Incoming Ring Group Setup.

Related Features
Directed Call Pickup/Group Call Pickup: Using these features, ringing calls can be picked up regardless of access map programming.
Direct Inward Line: Direct Inward Lines ring an extension directly, without Ring Group or Access Map programming.
Line Preference: An extension user can answer an outside call just by lifting the handset.
Long Conversation Cutoff/Warning Tone for Long Conversation: Long Conversation Cutoff can disconnect incoming and outgoing CO calls after a set time period. Using the Warning Tone for Long Conversation feature allows users on outgoing calls to hear a warning tone prior to the call disconnecting.
Night Service: Use Universal Answer to pick up Universal Night Answer calls.
Programmable Function Keys: Line keys and loop keys simplify answering outside calls.

Operation
To answer an incoming trunk call:
1. Lift handset.
2. At system phone, press flashing line key.
If you don't have a line or loop key for a trunk call ringing your phone, it rings an idle CALL key.
If you have Ringing Line Preference, lifting the handset answers the call.
OR
1. If you know the specific line number, dial 772 + Line number (001-200).

To use Universal Answer to answer a call ringing over the Paging system:
1. At system phone, press idle CALL key.
OR
At single line set, lift handset.
Depending on system programming, this may answer the call and you can skip Step 2.
2. Dial 872.
If you hear error tone, your extension's Class of Service prevents Universal Answer.

To listen to the incoming trunk ring choices (system phone only):
1. Press idle CALL key.
2. Dial 811 + 2.
3. Select the ringing (1-8) and tone range (1-4) you want to check.
4. Go back to step 3 to listen to additional choices or press SPK to hang up.

To change the pitch of your incoming trunk ring (system phone only):
1. Press idle CALL key.

2. Dial 820 + 2.
3. Select the ringing (1-8).
4. Press SPK to hang up.

Central Office Calls - Placing

Description
The system provides flexibility in the way each extension user can place outgoing trunk calls. This lets you customise the call placing options to meet site requirements and each individual's needs.
A user can place a call by:
• Pressing Line Keys or "Loop Keys"
• Pressing a Trunk Group (i.e., loop) key
• Pressing a Trunk Group Routing (dial 9) key
• Dialing a code for a specific trunk (805 + the trunk number)
• Dialing a code for a Trunk Group (804 + group number)
• Dialing a code for Trunk Group Routing or F-Route (9)
• Dialing an Alternate Trunk Route Access Code (which you must define)

Conditions
Trunk Access Maps can restrict outgoing trunk calls.

Default Setting
Outgoing trunk access is available to all extensions.

Related Features
F-Route: The system can automatically select the correct type of line to use based on the number dialed and the time.
Dial Tone Detection: Refer to this feature for the specifics on how the system handles Dial Tone Detection.
Line Reversal - COIU Trunks: Line reversal will detect when the called party answers and start the call timer
Long Conversation Cutoff/Warning Tone for Long Conversation: Long Conversation Cutoff can disconnect incoming and outgoing CO calls after a set time period. Using the Warning Tone for Long Conversation feature allows users on outgoing calls to hear a warning tone prior to the call disconnecting.
Loop Keys: Loop keys simplify placing Central Office Calls.
Programmable Function Keys: Line keys and loop keys simplify placing outside calls.
Trunk Group Routing: Trunk Group Routing sets out bound call routing options for users that dial the Trunk Group Routing code (9) for trunk calls.
Trunk Groups: Use trunk group programming to set the order in which users access trunks within a specific trunk group.

Operation
To place a call over a trunk group:
1. At system phone, press idle CALL key.
OR
At single line set, lift handset.
2. Dial 804.
3. Dial line group number (1-9 or 001-100).
4. Dial number.
OR
1. At system phone, press trunk group key (SC 851: *02 + group).
Also see the "Loop Keys" feature.
2. Dial number.

To place a call using Trunk Group Routing:
1. At system phone, press idle CALL key.
OR
At single line set, lift handset.
2. Dial 9.
If your system has an Alternate Trunk Route Access code, you may dial that instead.
3. Dial number.
OR
1. At system phone, press Trunk Group Routing key (SC 852: *05).
Also see the "Loop Keys" feature.
2. Dial number.

Conference - Voice Call Privacy Release

Description
Voice Call Conference lets extension user's in the same work area join in a trunk Conference. To initiate a Voice Call Conference, an extension user just presses the Voice Call Conference key and tells their co-workers to join the call. The system releases the privacy on the trunk, and other users can just press the trunk's line key to join the call.
Voice Call Conference does not use the telephone system features to announce the call. The person initiating the Voice Call Conference just announces it "through the air."

Privacy Mode Toggle Option
The Privacy Mode Toggle option allows an extension user to quickly change an outside call from the non-private mode to the private mode. This would help a work group supervisor, for example, that needed to quickly monitor any group member's call. If the supervisor wanted to make a "secure" call, however, they could quickly switch the line's mode and be assured that their call would not be monitored. If the outside call is on a line key, the user just presses the line key to switch modes. If the call is on a loop key, the user presses their Privacy Release function key instead.
For systems using the Privacy Mode Toggle option, trunks initially have the privacy released. If privacy is desired for a trunk, use the toggle option or press the Privacy Release function key to switch modes.

Conditions
Line keys are required for the trunk ports.
This feature is not available at single line telephones.

Default Setting
Disabled.
'Privacy Mode Toggle Option' is disabled in Trunk Basic Data Setup.
'Privacy Release' is on in Class of Service.
'Privacy Release by pressing Line Key' is off in Class of Service.
'Privacy Release Time' is set to 90 seconds in Paging Options, Note that
this interval is also used for Meet Me Conference).

Related Features
Conference: Set up a multiple-party telephone meeting without leaving the office.
Programmable Function Keys: Voice Call Conference requires a Voice Call Conference function key and line keys.

Operation
To join a Voice Call Conference (if invited):
1. After Conference request, press indicated line key.
To exit a Voice Call Conference without affecting the other parties:
1. Press SPK to hang up.

Conference

Description
Conference lets an extension user add additional inside and outside callers to their conversation.
With Conference a user may set up a multiple-party telephone meeting without leaving the office.
The Main Unit provides 32 Conference circuits, allowing any number of internal or external parties in conference up to the block's limit of 32. This means that one extension can Conference up to 31 internal and/or external parties together (the originator would be the 32nd party reaching the maximum of 32). While this Conference call is active, another user can use the second block of Conference circuits to make the same type of call.
Each block of Conference circuits can have multiple Conference calls, providing there are Conference circuits available. It is not restricted to one Conference per block.

Split (From Conference)
Split allows a user to alternate (i.e., switch) between their callers in Conference.
Split cycles through the Conference in the same order in which the Conference was initially set up.
If a user places an outside call, conferences extension 302 followed by extension 303, Split will cycle from the trunk, to 302 and finally to 303. The Split cycle then repeats.

Barge Into Conference

If a user's extension has Barge In capability enabled, they can also Barge In on an established Conference.
This permits, for example, an attendant or supervisor to join a Conference in an emergency.
It also allows a co-worker to leave a conference and then rejoin the telephone meeting when it is convenient to do so.

Transfer Call Into Conference
An extension with Barge In capability can Transfer a call into an existing Conference. This would allow, for example, an attendant to locate co-workers and then Transfer them into an existing telephone meeting. There is no need for the attendant to locate all the parties at the same time and sequentially add them into the Conference.

Conditions
Conversation recording also uses conference circuits; each conversation recording will use 2 conference circuits.

Default Setting
Service code for split is 894 ('Call Waiting Answer/Split Answer for SLT').
Service code for conference is 826 ('Conference').
'Conference' is on in Class of Service.

Related Features
Conference, Voice Call/Privacy Release: Set up a Conference with a co-worker in your immediate work area.
Direct Inward System Access (DISA): DISA users may use the Barge In feature on a Conference call if they know the service code and are permitted in their DISA/Tie Line Class of Service.
Meet Me Conference: Meet Me Conference lets an extension user set up a Conference via Paging.
Meet Me Paging: Meet Me Paging lets an extension user set up a two-party meeting via Paging.
Programmable Function Keys: In order for system phone to have Conference, it must have a Conference function key.
Tandem Trunking: A user can set up an Unsupervised Trunk-to-Trunk Conference and then drop out of the call, allowing the remaining parties to continue the conversation.
Transfer Into Conference: Transfer a call into an existing conference.
Withdraw from Conference (Trunk to Trunk Transfer): Transfer the two trunk parties together (Trunk to Trunk Transfer) by withdrawing from the conference.

Operation
To establish a Conference:

System Phone
1. Establish Intercom or trunk call.
2. Press DND/CONF or Conference key (PGM 15-07 or SC 851: 07).
3. Dial extension you want to add.
OR
Access outside call
OR
Retrieve call from Park orbit.
To get the outside call, you can either press a line key or dial a trunk/trunk group code. You can optionally go back to step 2 to add more parties to your Conference.
4. When called party answers, press DND/CONF or Conference key twice.
If you cannot add additional parties to your Conference, you have exceeded the system's Conference limit.
5. Repeat steps 2-4 to add more parties.

Single Line Set
1. Establish Intercom or trunk call.
2. *Single Line Telephone*
Recall and dial 826.
3. Dial extension you want to add.
OR
Access trunk call.
4. *Single Line Telephone*
Recall and repeat step 3 to add more parties.
OR
Recall twice to set up the Conference.

If you cannot add additional parties to your Conference, you have exceeded the system's Conference limit.

To exit a Conference without affecting the other internal parties:

System Phone
1. Hang up.
If you press Hold while on a call with two outside callers, the outside callers hear Music on Hold.

If you are the only internal extension that is part of a conference that includes two or more exchange lines then hanging up will clear the trunk calls.

Single Line Set
1. Hang up.
If you are not permitted to use Tandem Trunking, outside callers may hear Music on Hold.

To Barge In to Conference Call:
1. Pick up the handset or press SPK and dial the service code (810=default).
If the telephone doesn't have the proper COS, a warning tone is sent. After the user hangs up, the system will automatically place a Callback to the extension.
2. Dial the extension number or press a DSS key of a telephone within a Conference call.
When a new call is added to the conference, an intrusion tone is heard by all parties in the Conference, depending on system programming, and all display system phones show the joined party. If a Conference is not possible:
- the extension user will hear a warning tone
- the DISA user will be rerouted to the defined ring group
OR
Not available for DISA trunks:
1. Dial the extension number of the internal party.
2. Dial the single digit service code.
Instead of the single digit service code, the service code 810 can also be dialled at this point.

Continued Dialling

Description
Continued Dialing allows an extension user to dial a call, wait for the called party to answer and then dial additional digits.

There are two types of Continued Dialing:
• Continued Dialing for Intercom Calls
Depending on an extension's Class of Service, a system phone user may be able to dial additional digits after their Intercom call connects. In systems with Voice Mail, for example, Continued Dialing lets extension users dial the different options after the Voice Mail answers.
Without Continued Dialing, extension users cannot access these Voice Mail options.
• Continued Dialing for Trunk Calls
Continued Dialing gives a user access to outside services like automatic banking or an outside Automated Attendant. After the outside service answers, the user can dial digits for whatever options the services allow. Without Continued Dialing, the system's Toll Restriction will cut off the call after a specific number of dialed digits.

NOTICE
Continued Dialing may make the system more susceptible to toll fraud.

Conditions
None

Default Setting
'Send DTMF While Talking on Extension' (Continued Dialing) is on in Class of Service.
'Maximum Number of Digits Table Assignment' is not assigned in Toll Table Assignment.

Related Features
Pulse to Tone Conversion: Users can place calls to services over Dial Pulse trunks - and then dial DTMF digits after the service answers.
Toll Restriction: The ability to use Continued Dialing on trunk calls is set by Toll Restriction programming.

Operation
To use Continued Dialing:
1. Place Intercom or trunk call.
2. Continue dialing after call connects.
Toll Restriction and Class of Service programming may limit Continued Dialing.

Class of Service (COS)

Description
Class of Service (COS) sets various features and dialling options for extensions.
The system allows any number of extensions to share the same Class of Service. An extension can have a different Class of Service for each of the Night Service modes. This lets you program a different set of dialling options for daytime operation, night time operation and even during lunch breaks. An extension's Class of Service can be changed in system programming or via the 'Set Station Class of Service' service code (default = 777).

Conditions
None

Default Setting
All extensions have Class of Service 1 in all Night Service modes (Class of Service per Night Modes).
Changing COS by Service Code is disabled ('COS Programming' option in Class of Service Options).
The default Service Code is 777 (3 Digit Codes).

If changing Class of Service via Service Code:
An extension can use Service Code 777 to change another extension's Class of Service ('COS Programming' option in Class of Service Options).
An extension automatically blocks another extension's attempt to change their Class of Service if the 'COS Programming' option in Class of Service Options is disabled.

Operation To change an extension's Class of Service (via Service Code 777):
1. Press SPK.
2. Dial 777.
3. Dial the extension number you want to change.
You see: MODE1:nn
Press HOLD to leave the current value unchanged. The extension you dial may be set to block your attempt to change their Class of Service.
4. Enter the Day 1 Mode Class of Service for the extension you selected in step 3 and press HOLD.
You see: MODE2:nn
Press HOLD to leave the current value unchanged.
5. Enter the Night 1 Mode Class of Service for the extension you selected in step 3 and press HOLD.
You see: MODE3:nn
Press HOLD to leave the current value unchanged.
6. Enter the Midnight 1 Mode Class of Service for the extension you selected in step 3 and press HOLD.
You see: MODE4:nn
Press HOLD to leave the current value unchanged.
7. Enter the Rest 1 Mode Class of Service for the extension you selected in step 3 and press HOLD.
You see: MODE5:nn
Press HOLD to leave the current value unchanged.
8. Enter the Day 2 Mode Class of Service for the extension you selected in step 3 and press HOLD.
You see: MODE6:nn
Press HOLD to leave the current value unchanged.
9. Enter the Night 2 Mode Class of Service for the extension you selected in step 3 and press HOLD.
You see: MODE7:nn
Press HOLD to leave the current value unchanged.
10. Enter the Midnight 2 Mode Class of Service for the extension you selected in step 3 and press HOLD.
You see: MODE8:nn
Press HOLD to leave the current value unchanged.
11. Enter the Rest 2 Mode Class of Service for the extension you selected in step 3 and press HOLD.
You see: Enter Sta#-
12. Go to step 3 and enter another extension number.
OR
Press SPK to hang up.

Cost Centre Codes

Description
Certain Network providers can accept a Cost Centre Code (CCC) that will be used to provide an itemised bill. This will give the customer a record of which extension made the call.
The CCC is inserted automatically by the system before the dialled digits and is stripped off by the Network supplier.
You must confirm the number of CCC digits required by the Network otherwise calls will route incorrectly.

Default setting
Cost Centre Codes are disabled in LCR Dial Data.
The Cost Centre Codes are assigned in Cost Centre Codes.

Related Features
LCR - Least Cost Routing: Cost Centre Codes can also be sent as part of the LCR operation.

Operation
The operation is automatic and can not be bypassed.

Direct Dial In (DDI)

Description
Direct Inward Dialling (DID) or Direct Dial In (DDI) lets outside callers directly dial system extensions. For example, DID number 926-5400 can directly dial extension 400. The caller does not have to rely on attendant or secretary call screening to complete the call.
Note: Direct Inward Dialling requires a DDI service from the Network Provider.

Overview of DID Operation
Each trunk must be set to DID mode. The trunks are then placed into a Trunk Group and routed to a DID Translation Table Area.
You can specify a different area for each night mode.
The DID Translation Table Area contains a specified quantity of DID Translation Entries, the received DID number is compared to each entry to find a match.
The DID Entry will then specify the destination extension number, Incoming Ring Group, Department Group or Voice Mail.
Each DID Entry has three destinations available, the incoming call can be made to fall over to the next destination if the call is busy or not answered.

DID operation also provides:

DID Dialled Number Translation
DID allows different tables for DID number translation. This gives you more flexibility when buying DID service from the Network Provider. If you can't buy the exact block of numbers you need (e.g., 200-299), use the translation tables to convert the digits received.
For example, a translation table could convert DDI digits 500-799 to extension numbers 200-499.

Seperate DID Routing for each Night Mode The PBX has 2000 DID Translation Table entries that you can allocate among the 20 DID Translation Table areas. There is one DID translation made in each entry.
For flexibility you can route to a different area in each Night Mode, this gives the ability to have different translations for each of the Night Nodes.

DID Naming
In addition to number conversion, each DID Translation Table entry can have a name assigned to it. When the DID call rings the destination extension, the programmed name displays, helping the user to identify the called number before they answer.

Flexible DID Service Compatibility
You can program the system to be compatible with up to 8 digit DID service. Be sure to program your system for compatibility with the provided service. For example, if the Network sends four digits, make sure you set up the translation tables to accept the four digits.

DID Fall-Over
If the DID destination is busy or does not answer the call can fall over to another destination. Each DID Translation Entry has three destinations.

DID Call Waiting
DID Call Waiting sets what happens to DID calls to busy extensions when you have Busy fall over disabled (Transfer Option within DDI Routing Table). With DID Call Waiting enabled, a call to a busy extension 'Camps-on'. If the Transfer Option within DDI Routing Table is set for No Answer step on, the call will step on after the DID Ring No Answer Time interval (System Options for Incoming Call Service). It will step on to the programmed fall over extension, Ring Group, Department Group or Voice Mail. Without DID Call Waiting the caller to the busy extension just hears busy tone.

DID Queue Limit For each DID Translation Entry you can specify how many calls the system will allow to ring the destination. When

the limit is reached callers will hear busy indication.
This setting is optional and can be set to unlimited if required.

DID Music on Hold selection
You can specify the music device for each DID Translation Entry.

DID Routing Through the VRS
DID calls can optionally route through the VRS. The DID caller hears an Automated Attendant Greeting explaining their dialling options. For example the greeting can be, “Thank you for calling. Please dial the extension number you wish to reach or dial 0 for the operator.” If the caller inadvertently dials an extension that doesn’t exist, they could hear, “The extension you dialled is unavailable. Please dial 0 for assistance.”
You assign Automated Attendant greetings (i.e. VRS Messages) to the trunks so you can have a single greeting for all DID callers.

SMDR Includes Dialled Number
The SMDR report can optionally print the trunk’s name, the DID name or the number the incoming caller dialled (i.e., the dialled DID digits). This gives you the option of analysing the SMDR report based on the number your callers dial.

Conditions
DID service must be provided from your Network Provider.

Default Setting
Trunks are set to Normal incoming type, for DDI operation change in each trunk to DID type in Incoming Service Type Setup.
All trunks are in Trunk Group 01 in Trunk Group.
All Trunk Groups are routed to DID Translation Tale Area 01 in all night modes (DDI Table Area Target).
The DID Translation Table Area is spilt into 10 areas each with 200 Translation Entries (DDI Table Area Setup).
DID Translation Table Area 1 has received DID numbers 00 to 99 routed to extensions 200 to 299, DID Fall Over disabled, Call Waiting disabled, Queue Limit is unlimited and the music device is the system tune (DDI Routing Table).

DDI Time Mode Operation

Description
DDI Time Mode Operation allows selected incoming DDI calls to be routed to different destinations depending on the time the call arrives.
There are 100 entries available each with up to 8 time modes, the time modes are independent to the system's Night Modes.
The system will automatically route the DDI calls to the defined destination as specified by the time mode. The destination is one of the DDI Routing table entries 1-2000.
In addition, there is a manual overide option that allows a user to select a different time mode for each of the 100 entries.

The diagram below shows an overview of DDI Time Mode Operation.

Trunk port setting for incoming service type	DDI	Route the caller as defined in the DDI section
|		
DDI Time Mode		
|		
Is the DDI number entered in the DDI Time Mode table	No	Route the caller as defined in the DDI section
|		
Yes		
|		
Route the call to the DDI Routing Table Number specified for the time of day.		

Conditions
It is important to name the DDI's in use within the DDI Routing Table as the name is displayed when a user manually overrides the time patterns.
The installer must supply the user with the DDI Time Mode table numbers (1-100) in use and the pattern numbers configured if manual override is required.

Default Setting
No trunks are set to DDI Time Mode in Incoming Service Type Setup.

There are no entries in the DDI Time Mode Table.
The service code for 'Dial In Mode Switching' is not assigned in 3 Digit Codes.
There are no function keys assigned for 'Dial in Mode Switching' in Function Key Programming.

Operation
The operation is automatic once the time modes are assigned.

Manual Override
A user can manually override the current automatic DDI Time Mode, the system will revert to automatic switching at the start of next time mode.

Manual Override using Service Code.
A keyphone with a display is recommended.
Note - If you select a pattern number (1-8) that is not configured by the installer then the system will return busy indication to the incoming DDI caller.

1. Press SPK.
2. Dial the Dial In Mode Switching service code.
3. You will be prompted to enter the DDI Time Mode Table number (1-100). The system installer must provide a list of the table numbers in use.
4. Enter three digits for the table number.
5. You will be prompted for the Pattern number (1-8). The current pattern number in use will also be displayed, only select a pattern number that has been assigned by the installer.

The example display below shows that DDI Time Mode Table number 003 has been selected.
The current Pattern number in use is 1 and the current DDI target name (from the DDI Routing Table) for pattern 1 is 'Tech Support'.

DDI Table003:Tech Support
Pattern(1-8)? 1-

6. Enter the new Pattern number to be used (the system will revert to automatic switching at the next automatic switching time).

Manual Override using Function Key.
A keyphone with a display is recommended.
You must have a function key assigned to the specific DDI Time Mode Table number (1-100).

To setup the Function Key:
1. Press SPK.
2. Dial the service code for Function Key Programming (851).
3. Press the function key you want to assign (you may need to clear the key first, refer to Programmable Function Keys for help).
4. Enter code 88.
5. Enter the DDI Time Mode Table number (1-100). The system installer must provide a list of the table numbers in use.
6. Press SPK to end.

To use the Function Key:
1. Press the Function Key for the DDI Time Mode Table number (1-100).
2. You will be prompted for the Pattern number (1-8). The current pattern number in use will also be displayed.
3. Enter the new Pattern number to be used, only select a pattern number that has been assigned by the installer.

Department Groups

Description
With Department Calling, an extension user can call an idle extension within a pre-programmed Department Group by dialling the group’s pilot number. The call would ring the first available extension in the group. For example, this would let a caller dial the Sales department just by knowing the Sales department’s pilot number. The caller would not have to know any of the Sales department’s extension numbers. The system allows up to 32 Department Calling Groups.

Department Group Name
When an internal extension is queued at a busy department group the name will be displayed at the system phone’s display. The user will see: WAITING (group name). The group name is set by Department Group Options.

Department Group Routing
There are two types of routing available with Department Calling: Priority Routing and Circular Routing.

With Priority Routing, an incoming call routes to the highest priority extensions first. Lower priority extensions ring only if all higher priority extensions are busy. Priority routing is selected in Department Group Options.
With Circular Routing, each call rings a new extension, providing an easy type of Uniform Call Distribution (UCD).
For example, in a Department Group set for circular routing with extensions 210 (Priority 1), 211 (Priority 2) and 212 (Priority 3).
The first call rings 210.
The second call rings 211.
The third call rings 212.
The fourth call rings 210 and the cycle repeats.
Note: When programming, the high priority extensions have low priority numbers. For example, priority 1 has a higher priority than priority 10.

Routing when Busy
Department Calling also provides overflow routing for extensions within the group. If a user directly dials a busy extension within a Department Group, the system can optionally route the call to the first available group member. Overflow routing is set by Department Group Options.

Hunting Mode
Ringing calls can step around members of the group once only and stop at the last member or repeatedly search for a free member. Hunting mode is set by Department Group Options.

Simultaneous Ringing (All ring mode)
All idle members of the department group can ring simultaneously for internal and outside calls to the pilot number. Calls do not cycle between group members. Simultaneous ringing can be automatic (all members ring when the call is placed to the pilot number) or manual (the call to the pilot number will step around each member of the group until the caller selects simultaneous ring mode). The automatic/manual option is selected by Department Group Options. To select manual mode the user must either dial the service code 780 (set by 3 Digit Codes) or the single digit service code set by set by (1 Digit Codes).
Note, when automatic is selected the operation of Enhanced Hunting Type in Department Group Options is effected. Calls to the pilot number will not receive ring back tone if all members are busy, they will receive busy tone and will not wait for a member to become free.

Recall Restriction
Calls transferred to the pilot number that do not get answered will recall at the extension that transferred the call. The transfer recall can be restricted (set to Disable(Recall) by Department Group Options). When the transfer recall is restricted the transferred call will ring the group until it is answered or the caller clears down.

Maximum Queuing Limit
The quantity of ISDN DDI trunk calls queuing at a busy Department Group can be limited when all members of the group are busy. When the queue limit is reached further calls will receive busy indication. The number of calls in the queue can be set by Department Group Options. If the queue limit is set to 0 then no ISDN DDI calls will be queued. Note that the queue limit will be ignored if enhanced hunting is set or you have enabled Busy step on for the DDI call in DDI Routing Table.

No Answer Step On Time
An un-answered call ringing at a member of a department group will step on to the next available member after a preset time, set by Department Group Options. If the timer is set to 0 the step on will be disabled.

Enhanced Hunting
Department Calling is enhanced with expanded hunting capabilities. Hunting sets the conditions under which calls to a Department Group pilot number will cycle through the members of the group. The hunting choices are:
No Hunting
A call to the pilot number will hunt past a busy group member to the first available extension.
The call will continue to ring the extension until it is answered or the calling party hangs up, it will also step on to the next available member after the No Answer Step On Time.
Calls to the group when all members are busy will receive busy tone.
Busy
A call to the pilot number will ring the first idle member of a Department group, following the priority or circular routing. The call will continue to ring the extension until it is answered or the calling party hangs up, it will not step on to the next available member. If the Department Group has Priority Routing enabled, and the highest priority member is busy, the call will step on to the next available member.
Calls to the group when all members are busy will receive ring back tone and wait for a member to become free.
No Answer A call to the pilot number will ring the first idle member of a Department group, following the priority or circular routing. The call will continue to ring the extension until it is answered or the calling party hangs up, it will also step on to the next available member after the No Answer Step On Time. If the Department Group has Priority Routing enabled, and the highest priority member is busy, the call will wait for the extension to become free and will not step on to the next available member.
Calls to the group when all members are busy will receive ring back tone and wait for a member to become free.
Busy & No Answer
A call to the pilot number will ring the first idle member of a Department group, following the priority or circular routing. The call will continue to ring the extension until it is answered or the calling party hangs up, it will also step on to the next available member after the No Answer Step On Time.

If the Department Group has Priority Routing enabled, and the highest priority member is busy, the call will step on to the next available member and continue to step on after the
No Answer Step On Time.
Calls to the group when all members are busy will receive ring back tone and wait for a member to become free.
Note that enhanced hunting will effect the operation of simultaneous ringing and maximum queue limit.

Queue Announcements
If a call in queue is an outside call, and the system has DSPDB card installed for VRS, the queued caller can hear a customer recorded announcement.
Up to two different announcements can be given to the queued caller.
The queue announcements are assigned by Department Group Queue Announcement Setup for each group.
Note, DDI calls will not receive the queue announcement if the queue limit is set to 0 in Department Group Options or enhanced hunting is set to No Hunting and the DDI has no transfer set in DDI Routing Table as calls will not queue at the busy group. You must set either a queue limit greater than 0 or set enhanced hunting to allow calls to queue at a busy group or set a transfer option in DDI Routing Table.

User Log In/Log Out
An extension user can log out and log in to a Department Calling Group. By logging out, the user removes their extension from the group. Once logged out, Department Calling bypasses their extension. When they log back in, Department Calling routes to their extension normally. All users can dial a code to log in or log out of their Department Calling Group. A system phone can optionally have a function key programmed for one-button log in and log out operation.

Secondary Group Allocation
An extension can be a member of one Department Group by allocation in Department Group Options. An extension can also be a secondary member of other Department Groups by allocation in Secondary Dept Assignment. Each Department Group can have up to 16 secondary members.
When a Department Group contains a secondary member allocated then the simultaneous ring option (automatic and manual) is not available.

Call restriction Between Department Group
Calls between members of different Deaprtment groups can be restricted on a per group basis.
Each department group can restrict calls to up to 8 department groups in Department Group - Departmental Call Restriction

Conditions
When a DIL rings to a Department Group, the DIL will not follow overflow programming set in DIL Step On Target Assignment and DIL No Answer Time).
If an extension sets call forward it will be removed from any department groups. Department group calls cannot follow the call forward.
If an extension sets DND External it will not receive any outside calls to the department group.
If an extension sets DND Internal it will not receive any internal calls to the department group.
If an extension sets DND All it will not receive any outside or internal calls to the department group.

Default Setting
All extensions are members of Department Group 01 (Department Group Assignment).
There are no Department Group Pilot numbers assigned (Department Group Options).

Operation
To call a department:
1. Lift handset.
2. Dial the Department Group pilot number.
The system routes the call to the first free phone in the department.
To log out of your Department Calling Group:
While you are logged out, Department Calling cannot route calls to your extension.
1. Lift handset.
2. Dial 750 + 1.
OR
1. Press Department Calling Log In key (SC 851: 46).
The key lights while you are logged out.
To log back in to your Department Calling Group:
While you log back in, Department Calling will route calls to your extension.
1. Lift handset.
2. Dial 750 + 0.
OR
1. Press Department Calling Log In key (SC 851: 46).
The key goes out when you log back in.

Department Group Forward

Description

Department Group Call Forward allows a user to set call forward for calls routed to a department group.
The forward can be set for each department group and each department group can have its own destination number that the calls are forwarded to. The destination can be either an internal extension / pilot number or an off premise number.
The feature will operate for calls via ISDN DDI and internal calls to the pilot number.
The call forward is controlled by service codes and function keys. The destination can also be changed by the user for each night mode (1-8).
When call forward is set all incoming calls to the department group will be forwarded immediately or after a delay. Call forward after a delay can only be set via function key 59 (SC 851+59) and will only operate for incoming DDI calls to the department group. The delay time is set by Hold and Transfer.
The destination of the call forward is saved in Abbreviated Dial bin 1999. To route the call off premise you must enter a trunk access digit (e.g. 9) before the destination number. If there is no trunk access digit the call will route internally. The internal number can be an extension or another department group pilot number.
It is also possible to chain the call forwards, for example if department group 01 is forwarded to department group 02 and group 02 is also forwarded extension 250 then calls to department group 01 will ring at extension 250.
A call that is forwarded off premise will be disconnected after the 'DISA Conversation Warning Tone Time' and 'DISA Conversation Disconnect Time' in Timers.
The outgoing trunk route is defined by Outgoing Route Setup. A free trunk within this route will be used when a call is forwarded off premise.

Conditions
Analogue trunks must have disconnect clear enabled when call forward off premise is set, this is to ensure the lines are cleared when a call is disconnected.
The call forward destination cannot be set from an analogue SLT via Service Code 704.

Default Setting
The service code to enable department group forwarding is 702 ('Automatic Transfer Setup per Extension Group').
The service code to disable department group forwarding is 703 ('Automatic Transfer Cancellation per Extension Group').
The service code to set/change the destination number is 704 ('Automatic Transfer Destination per Extension Group').
Abbreviated Dial bin 1999 is used as the call forward destination for all night modes.
The disconnect timers are set to give a warning tone after 30 seconds and then disconnect after another 15 seconds.
The outgoing trunk route is not defined.
Disconnect clear is not set.
'Set Automatic Transfer at Extension Group Call' is on in Class of Service.

Operation
To set the destination number for each department group:
(Cannot be set from an analogue SLT.)
1. Press SPK to go off hook.
2. Dial 704.
3. Dial the department group number (01 to 64).
4. Dial the night mode number (1 to 8).
5. Dial the off premise destination number (include any trunk access digits for off premise).
6. Press HOLD to set the destination for other night modes.
7. Press SPK to hang up.

Note that at step 5 the destination number will be saved to the Abbreviated Dial bin number specified by Program 24-05-01 for the chosen night mode.
If the same Abbreviated Dial bin is also used for other night modes then you do not need to set them separately. For example at default bin number 1999 is used for all night modes so when a night mode destination number is set for any night mode it will be used for all night modes.

To set the call forward for incoming calls to a department group:
1. Press SPK to go off hook.
2. Dial department group call forward service code - 702.
3. Dial the department group number (01 to 64).
4. Press SPK to hang up.
5. Repeat steps 1 to 4 for further department groups.

Or

6. Press the department group call forward function key (851+58+group number). The key will flash red when the call forward is set.

To cancel the call forward for incoming calls to a department group:
1. Press SPK to go off hook.
2. Dial department group call forward cancel service code - 703.
3. Dial the department number (01 to 64).
4. Press SPK to hang up.
5. Repeat steps 1 to 4 for further department groups.
Or
6. Press the department group call forward function key (851+58+group number). The key will go out when the call forward is cancelled.

To set delayed call forward for incoming DDI Calls:
1. Press the department group delayed call forward function key (851+59+group number).
The key will flash red when the call forward is set.
2. Incoming DDI calls will ring at the department group until a timer expires (Program 24-02-08)
3. When the timer expires the call to the department group will forward to the destination number.

To cancel delayed call forward for incoming DDI Calls:
1. Press the flashing department group delayed call forward function key (851+59+group number). The key will go off when the call forward is cancelled.

Department Group Step Calling

Description

After calling a busy Department Calling Group member, an extension user can have Department Step Calling quickly call another member in the group. The caller does not have to hang up and place another Intercom call if the first extension called is unavailable. Department Step Calling also allows an extension user to cycle through the members of a Department Group.

Conditions
None

Default Setting
'Step Call' service code is 808.

Operation
To make a Step Call:
1. Place call to busy Department Group member.
OR
Place call to Department Group pilot number.
2. Press Step Call key (SC 851: 36).
3. Repeat step 2 to call other Department Group members.

Dial Number Preview

Description
Dialing Number Preview lets a display system phone user dial and review a number before the system dials it out. Dialing Number Preview helps the user avoid dialing errors.

Conditions
An extension user cannot edit the displayed number.
If the DSPDB is installed the user must dial * to begin preview dial.

Default Setting
'Preset Dial' is on in Class of Service.

Related Features
Central Office Calls, Placing: In order to place an outgoing call, an extension user must have outgoing access to a line, loop or trunk

group key.

Operation

To use Dial Number Preview to place a call (system phone only):

1. Do not lift the handset or press a CALL key.
2. To preview any number, press * (if DSPDB installed).
To preview an Abbreviated Dial number, press DIAL.
3. Dial number you want to call.
The number displays.
4. To dial out the displayed trunk number, press a line/loop/trunk group key.
If the previewed number as a trunk access code (e.g., 9), you can press CALL instead.
OR
To dial out the displayed Intercom number, press a CALL key.
OR
To cancel the number without dialing it out, press HOLD.

Dial Pad Confirmation Tone

Description

For an extension with Dial Pad Confirmation Tone enabled, the user hears a beep each time they press a key. This is helpful for Intercom calls and Dial Pulse trunk calls, since these calls provide no Call Progress tones.

Conditions

None

Default Setting

Disabled

Related Features

Single Line Telephones: Dial Pad Confirmation Tone does not apply to single line telephones.

Dial Tone Detection

Description

If a trunk has Dial Tone Detection enabled, the system monitors for dial tone from the network when a user places a call on that trunk. If the user accesses the trunk directly (by pressing a line key or dialing 805 and the trunk's number), the system will drop the trunk if dial tone does not occur. If the user access the trunk via a Trunk Group (by dialing a trunk group code or automatically through a feature like Last Number Redial), the system can drop the trunk or optionally skip to the next trunk in the group.

Dial Tone Detection is available for the following features:

• Automatic Route Selection
• Abbreviated Dialing
• Central Office Calls, Placing
• Last Number Redial
• Loop Keys (out bound)
• Save Number Dialed
• Trunk Group Routing
• Trunk Groups

Conditions

None

Default Setting

Disabled for manually dialed calls in Analogue Trunk Data Setup.
'Outgoing trunk Skip on No Dial Tone' is disabled in Analogue Trunk Data Setup.
'Dial Tone Detection Time' is set to 5 seconds in Outgoing Call.
'Disconnect Time when Dial Tone Not Detected' is set to 0 seconds in Outgoing Call.

Related Features

See Description above.

Operation
Dial Tone Detection is automatic if enabled in programming.

DIL (Direct Inward Line)

Description
A Direct Inward Line (DIL) is a trunk that rings an extension, virtual extension or Department Group directly. Since DILs only ring one extension or group (i.e., the DIL destination), employees always know which calls are for them. The DIL does not ring other extensions.

DIL Delayed Ringing
Extensions in a Ring Group can have delayed ringing for another extension’s DIL. If the DIL is not answered at its original destination, it rings the DIL No Answer Ring Group.

Conditions
If unanswered, a DIL without delayed ringing rings an extension until the outside party hangs up.
If a DIL rings a Department Group and all agents are busy, the system routes the call according to the Ring Group assignments.
The DIL follows call forwarding programming, even to voice mail.

Default Setting
No DIL trunks are assigned.

Related Features
Call Forwarding: Call Forwarding reroutes DILs.
Central Office Calls, Answering: When a call is transferred because of Call Forwarding No Answer, Call Forwarding Busy, or DND, the Reason for Transfer option can display to the transferred extension why the call is ringing to their phone.
Central Office Calls, Placing: You can place DILs in trunk groups to make outgoing DIL calls easier.
Department Calling: A DIL can have an Extension (Department) Group as its destination.
Do Not Disturb: If an DILs destination extension is in DND, an incoming call rings according to Ring Group programming.
Group Call Pickup: A user can activate Group Call Pickup to intercept a DIL ringing another extension.
Name Storing: Program a name for a DIL. This makes it easier to identify the incoming call.
Private Line: To simulate Private Line operation, create a unique Access Map for the DIL that allows full access only for the destination. Give all other extensions only Hold access.
Off Hook Signaling: If a system phone’s first channel is busy, a DIL always signals the idle second channel if available.
If the second channel already has a call waiting, DIL waits in line for a channel to
become free. The outside caller hears ring back tone while this occurs.
Programmable Function Keys: If an extension has a line key for a DIL, the call will ring the key. If not, the call rings an available line appearance. For other extensions, the DIL indicates as busy.
Ring Groups: A DIL will ring its assigned extension without Ring Group programming. A DIL only rings its assigned extension. It will not ring other extensions in a Ring Group.

Operation
To answer a call on your Direct Inward Line:
1. Lift handset.
2. At system phone, press flashing line key for the trunk.
If you don’t have a line key for the trunk, the DIL rings an idle CALL key.
If you have Ringing Line Preference, lifting the handset answers the call.
If you don’t answer the call, it may ring other extensions (i.e., the DIL No Answer Ring Group).

To place a call on your Direct Inward Line:
1. Lift handset.
2. At system phone, press line key for the trunk
OR
Dial 805 and the DIL trunk number (e.g., 005).
OR
Dial 804 and the DIL trunk group number (e.g., 05).
OR
Dial 9 for Trunk Group Access
3. Dial number.

Directed Call Pickup

Description
Directed Call Pickup permits an extension user to intercept a call ringing another extension. This allows a user to conveniently answer a co-worker's call from their own telephone. With Directed Call Pickup, an extension user can pick up:
• Trunk calls (i.e., Ring Group calls)
• Direct Inward Lines
• Transferred trunk calls
• Transferred Intercom calls
• Ringing and voice-announced Intercom calls

Conditions
Directed Call Pickup does not pick up calls recalling an extension (such as Hold and Transfer recalls) or calls on Hold.
An extension can use Directed Call Pickup to intercept calls to which it is denied access in Trunk Access Maps.

Default Setting
'Directed Call Pickup for Own Group' is enabled in Class of Service.

Related Features
For other features which let you cover a co-worker's calls, refer to:
• Department Calling
• Group Call Pickup
• Hotline
• Multiple Directory Numbers
• Secretary Call Pickup
Park: Personal Park also uses the Directed Call Pickup code.

Operation
To use Directed Call Pickup to intercept a call to a co-worker's extension:
1. At system phone, press idle CALL key.
OR
At single line set, lift handset.
2. Dial 715.
3. Dial number of extension whose call you want to intercept.
If more than one call is coming in, the system sets the priority for which call it will answer first.

Directory Dialing

Description
Directory Dialing allows a display system phone user to select a co-worker or outside
call from a list of names, rather than dialing the phone number. There are four types of Directory Dialing:
• ABBc - Company (Common) Abbreviated Dialing
• ABBg - Department (Group) Abbreviated Dialing
• EXT. - Co-worker's extensions
• OneT - Personal Abbreviated Dialing (One-Touch Keys)

Conditions
Directory Dialing sorts directory names in alphabetical order (based on the first four characters of
the name) when the system starts up or restarts. In addition, the system will re-sort extension
names when:
- You change an extension's name (Extension Numbers and Names).
- Any user dials 800 and changes their extension's name.
Directory Dialing follows all the programmed options and conditions for Abbreviated Dialing, Intercom Calling and One-Touch Calling.
If a user waits longer than the 'Internal Call Inter-digit Time' Directory Dialing steps, Directory Dialing automatically cancels.

Default Setting
Enabled.

Related Features
None.

Operation
To use Directory Dialing from a display system phone:
1. Do not lift handset or press SPK.
2. Press Directory Dialing Soft Key.
3. Press Soft Key for Directory Dialing type:
ABBc = Common Abbreviated Dialing.
ABBg = Group Abbreviated Dialing.
EXT. = Co-worker's extension numbers.
OneT = Your One-Touch Keys (1-10).
Directory Dialing follows any feature restrictions that your system may have enabled.
For example, if your extension cannot normally use Common Abbreviated Dialing, Directory Dialing can't access it either.
4. Dial letter/number range for the party you want to call (e.g., dial 2 for A, B, C or 2).
5. Press the Down Arrow Soft Key to jump to that section.
6. Press VOLUME UP or VOLUME DOWN to scroll through the list.
If you wait too long between your selections, Directory Dialing automatically cancels.
7. Lift handset or press DIAL, CALL1 or SPK to place the selected call.
If you selected an outside call, the call will route according to your system's Trunk Group Routing/ARS setup.

To cancel Directory Dialing:
1. Press CLEAR.

DISA - Remote Feature Setup

Description
This software provides an option for remotely setting various system functions for the specified extension by dialing the extension number and service code using a DISA line. This option is available for keyphones, single line telephones, and IP telephones.
When the outside caller, using an analog or ISDN trunk, places a call to a DISA line and dials the service code for this function, the system will respond with a fixed message prompting the entry of the extension number ("Please dial the extension number."). After the outside caller dials the desired extension number, the system will respond with another fixed message ("Please enter the required Service Code."), then the outside caller dials the required service code to set/cancel the function.
The following features can be set using service codes with this option:

Function Name	**Service Code**	**Description**
day / night mode switching for own night group	818	change the operation mode for each night group
setting the automatic trunk transfer for each trunk	833	set the automatic trunk transfer for each trunk
canceling the automatic trunk transfer for each trunk	834	cancel the automatic trunk transfer for each trunk
setting the destination for automatic trunk transfer	835	register the destination telephone number for automatic trunk transfer. with the remote feature setup, after dialing the destination telephone number, *# must be entered (ex: 2035551234 *#).
register the destination telephone number for automatic trunk transfer. with the remote feature setup, after dialing the destination telephone number, *# must be entered (ex: 2035551234 *#).	716	record / playback / erase VRS messages
VRS - general message playback	711	playback general message
VRS - record / erase general message	712	record / playback / erase general message

call forward - immediate	848	set/cancel call forward immediate (service code + 1 [set] / 0 [cancel] + transferred destination extension number)
call forward - busy	843	set/cancel call forward when busy (service code + 1 [set] / 0 [cancel] + transferred destination extension number)
call forward - no answer	845	set/cancel call forward when no answer during pre-assigned period (service code + 1 [set[/ 0 [cancel] + transferred destination extension number)
call forward - busy/no answer	844	set/cancel call forward when no answer during pre-assigned period (service code + 1 [set] / 0 [cancel] + transferred destination extension number)
call forward - both ring	842	set/cancel call forward both ring (service code + 1 [set] / 0 [cancel] + transferred destination extension number)
call forward - follow me	846	set/cancel call forward - follow me (service code + 1 [set] / 0 [cancel] + appropriate extension number)
DND (do not disturb)	847	set/cancel DND (service code + 0 [cancel] / 1 [external call] / 2 [internal call] / 3 [all call] / 4 [CFW transferred call])

Default Setting
Service code for 'Remote Access from DISA' is 800.

Conditions
• While the outside caller is setting the function via DISA, no one can use the extension which is being set.
• The outsider caller can not set/cancel a function via DISA when the selected extension is being used except during incoming ringing. If the extension is busy, the call will be terminated.
• The VRS is required to send the fixed messages heard during the feature setup.
• The DISA feature must be enabled for this function.

Operation
1. An outside caller dials in on a DISA trunk.
2. The system answers the call.
3. The outside caller dials the Remote Access from DISA service code (default =800).
The caller hears "Please dial the extension number".
If an incorrect extension number is dialed, the caller hears "That is an invalid entry. Please dial the extension number."
4. Dial the extension number for which a feature is to be activated/deactivated.
The caller hears "Please enter the required service code".
If an incorrect service code is dialed, the caller hears "That is an invalid entry. Please enter the required service code."
5. The outside caller can now dial the service code for the feature to be activated/ deactivated.
When the function setting via DISA has succeeded, the caller hears the fixed message "The setting has been activated".
The display indication at the extension changes in accordance with the set function.

DISA (Direct Inward System Access)

Description
DISA permits outside callers to directly dial system extensions, trunks and selected features. This could help an employee away from the office that wants to directly dial co-workers or use the company's trunks for long distance calls.

To use DISA, the employee:
• Dials the telephone number that rings the DISA trunk
• Waits for the DISA trunk to automatically answer with a unique dial tone
• Dials the 6-digit DISA password (access code). The password can be optional
• Waits for a second unique dial tone (only if password is enabled)
• Accesses a system trunk, uses a selected feature or dials a system extension DISA calls ring system extensions like other outside calls. If an extension has a line key for the DISA trunk, the call rings that key. If the extension does not have a line key, the call rings an idle CALL key.

You can set DISA operation differently for each Night Service mode. For example, a trunk can be a normal trunk during the day and a DISA trunk at night. You can also set the routing for DISA trunks when the caller dials a busy or unanswered extension, dials incorrectly or forgets to dial.

DISA Class of Service

DISA Class of Service provides features and dialing restrictions for DISA callers. This allows you to control the capabilities of the DISA callers dialing into your system. When a DISA caller first accesses the system, they will need to enter a DISA password before proceeding. The system associates the password entered with a specific user number, which in turn has a Class of Service. If the Class of Service allows the action (such as making outgoing trunk calls), the call goes through. If the DISA Class of Service doesn't allow the action, the system prevents the call. The DISA Class of Service options are:

• Delete First Digit Received
If necessary the system can ignore the first DTMF digit received, the remaining digits will be used to access the DISA options.

• Trunk Group Routing/F-Route access
When a DISA caller dials into the system, they may be able to dial 9 and place outside calls. Any toll charges are incurred by the system. The call follows the system's Trunk Group Access or Automatic Route Selection - whichever is enabled.

• Trunk Group Access
DISA callers may be able to access a specific trunk group for outgoing calls through the system.
To access a Trunk Group, the user dials Service Code 804 followed by the Trunk Group number (e.g., 1). This allows the DISA caller to place an outgoing call over the selected group. Trunk Group Access bypasses the system's Trunk Group Routing/F-Route/Trunk Access Maps. As with dial 9 access, any toll charges are incurred by the system. Also see Direct Trunk Access below.

• Common Abbreviated Dialing
The system's Common Abbreviated Dialing bins may be available to DISA callers. This could save the DISA caller time when dialing.

• Operator Calling
A DISA caller may be able to dial 0 for the system's operator.

• Paging
Internal and External Paging may be available to DISA callers. This allows co-workers in adjacent facilities, for example, to broadcast announcements to each other.

• Direct Trunk Access
DISA callers may be able to select a specific trunk for outgoing calls through the system. To directly access a trunk, the user dials Service Code 805 followed by the trunk's number (e.g., 001). This allows the DISA caller to place an outgoing call over the selected trunk. Direct Trunk Access bypasses the system's Trunk Group Routing/ARS/Trunk Access Maps. As with dial 9 access, any toll charges are incurred by the system. Also see Trunk Group Access above.

• Set Call Forward for an Extension via DISA
DISA callers can set/cancel call forward for an extension. This is useful when users have off premise call forward set, they can control the call forward via a DISA call without the need to visit the office.

• DISA/Tie Trunk Barge In
The DISA/Tie Trunk Barge In option allows a tie line caller to break into another extension's established call. This sets up a three-way conversation between the intruding party and the two parties on the initial call.

DISA Toll Restriction

The digits a DISA caller dials for an outgoing call may be subject to the system's Toll Restriction. For example, Toll Restriction can prevent users from dialing a 1-900 service. When an incoming DISA caller tries to use system trunks to dial 1-900, Toll Restriction will deny the call.

DISA Operating Modes

The DISA Operating Modes determine what happens when a DISA caller forgets to dial, calls a busy or unanswered extension or dials incorrectly. The system can either drop the call or send it to a preset Ring Group (called a the DISA Transfer Destination).

Department Calling with Overflow Message

If a DISA caller dials a busy Department Calling Group, the system can periodically play the voice prompt, "Please hold on. All lines are busy. Your call will be answered when a line becomes free." while the caller waits. The interval between the voice prompts is the DISA Overflow Message Time. When an extension in the Department Group becomes available, the call automatically goes through. If the Department Calling Group remains busy past the DISA No Answer Time, the DISA call routes to the overflow destination or disconnects. (What happens to the unanswered call is set by the DISA Operating Mode). The Overflow Message requires a VRS.

Warning Tone for Long DISA Calls

You can set up the system to provide a warning tone to DISA callers that have been on a call too long. The warning tone can be just a reminder (which the caller can ignore) or can be followed by a forced disconnect of the call. When the DISA caller hears the warning tone, they have the option of dialing a code to continue the conversation or disconnect.

DISA Dial in Mode
You can select the method that the digits dialled by a DISA caller are routed by the system.
You can select Intercom mode where the received digits are used to route the DISA caller or you can select Convert mode where the received digits are passed to the DDI Routing Table.
When you select the Convert mode the operation of the DISA call is the same as a DDI call, as follows:
The trunk port that the DISA call is received on will define the trunk group number.
The trunk group number will define the DDI Table Area Target for each night mode, The DDI Routing table area can now be identified.
The quantity of (DDI) digits that the DISA caller must dial is 4 digits at default.
The digits received are then compared to the receive dial entries of the DDI Routing table, if an exact match is found the call will be routed to the DDI target number.
If a match is not found the DISA call will be disconnected.
If the DDI target number is invalid the call will not use the DISA option for Wrong Dial. If the DDI target is invalid the options for the DDI will be used.
If the DDI target number is busy/no answer the call will not use the DISA option for Busy/RNA.
If the DDI target is busy/RNA the options for the DDI will be used.

DISA Password/User ID
You can enable/disable the DISA password per trunk port. When enabled all DISA callers must enter a valid 6 digit password before the call can proceed. The DISA password is used to assign a DISA Class of Service.
If the DISA password is disabled the incoming caller does not need to enter any password to access all of the DISA options.
Care should be taken when the password is disabled as any DISA caller can access outgoing trunks and paging zones etc.

Conditions
The DISA caller must use a DTMF telephone. Analogue DISA trunks must be ground start or disconnect supervision for loop start.

Default Setting
No DISA trunks are assigned.

Related Features
F-Route: When a DISA caller dials 9 for an outside call (if allowed), the system routes the call via F-Route.
In a system with F-Route disabled: When a DISA caller dials 9 for an outside call (if allowed), the system uses the routes programmed for Trunk Group Routing.
Central Office Calls, Answering: When a call is transferred because of Call Forwarding No Answer, Call Forwarding Busy, or DND, the Reason for Transfer option can display to the transferred extension why the call is ringing to their phone.
Direct Inward Dialing (DID) / Direct Inward Line (DIL) / Voice Response System (VRS)
These features also allow outside callers to directly access system extensions.
Long Conversation Cutoff: Long conversation cutoff is controlled separately for DISA and tie lines.
Voice Response System (VRS): Department Calling with Overflow Message requires a DSP daughter board for VRS.

Operation
To place a DISA call into the system (from any DTMF type telephone):
1. Dial the telephone number that rings the DISA trunk.
2. Wait for the DISA trunk to automatically answer with a unique dial tone.
3. Dial the 6-digit DISA password (access code), optional.
4. Wait for a second unique dial tone, optional.
5. Dial an extension.
OR
Dial 9 for Trunk Group Routing or ARS.
OR
Dial Alternate Trunk Route Access Code (if enabled).
OR
Dial 804 + a trunk group number (1-25) for an outside call.
OR
Dial 805 + a trunk number (1-51) for an outside call.
OR
Dial 813 + Common Abbreviated Dialing bin number.
OR
Dial 0 for the operator.
OR
Dial 801 + an Internal Paging Zone number (0, 1-9, 00, 01-32).
OR
Dial 803 + an External Paging Zone number (1-8 or 0 for All Call).
OR
Dial 810 + a busy extension number to barge in to a call.

OR
Dial 848, 842, 843, 845 + extension number + 1 + forwarding destination number to set call forward for an extension
Dial 848, 842, 843, 845 + extension number + 0 to cancel a call forward set at an extension.

DUD/DISA Extension Call

Description
DUD/DISA Extension Call is available to Auto Attendant (DUD/DISA) callers when they have dialled an extension that is either busy or does not answer.
The system can play a DSPDB message to the incoming caller who can then have further single digit options available if assigned in Auto Attendant Single Digit Options.
The single digit options allow the incoming caller to select one of the following destinations:
- Leave a message in the called extension's DSPDB mail box.
- Route to the Incoming Ring Group IRG, this could be used to route to an operator position.
- Dial another extension number.

The feature can be enabled for all extensions on the system or by individual extensions.
The system setting will take priority over the extension setting.
When enabled for the system in Extension Call - System Message the feature is available automatically when a DUD/DISA caller dials a busy or ring no answer extension. The DSPDB message that will be played to the incoming caller can be defined independently for busy or ring no answer.
When set by individual extensions each user can select the operation for either busy or ring no answer or both via a service code or programmable function key. The DSPDB message that will be played to the incoming caller can be defined independently for busy or ring no answer for each extension (up to a maximum of 48 DSPDB messages).

Conditions
The DSPDB card must be installed.
The feature is not available if IntraMail is enabled for the DSPDB card.

Default Setting
No DISA trunks are assigned.
No Messages are assigned for DUD/DISA Extension call.
There are no service codes assigned for busy or ring no answer DUD/DISA Extension Call.

Related Features
DISA (Direct Inward System Access)

Operation
System Setting
The operation is automatic, extension users do not need to make any feature settings.
System setting is enabled/disabled in Extension Call - System Message.

Extension Operation
Only available if the system setting is disabled in Extension Call - System Message.
By using service code:
1. Press SPK or lift the handset.
2. Dial to service code for DUD/DISA Extension Call. There are separate service codes for busy and no-answer.
   Dial 1 to enable
   Dial 0 to cancel

You can set both features simultaneously if required.

By using the programmable function key:
1. Press the function key for DUD/DISA Extension Call (SC 851: 94 'Call Attendant').
   Each time the key is pressed you will select the next mode:
   Off - lamp off
   Busy - lamp flashes slowly
   No-Answer - lamp flashes fast
   Busy and No-Answer - lamp in

Do Not Disturb

Description
Do Not Disturb blocks incoming calls and Paging announcements. DND permits an extension user to work by the phone undisturbed by incoming calls and announcements. The user can activate DND while their phone is idle or while on a call. Once activated, incoming trunk calls still flash the line keys.
The user may use the phone in the normal manner for placing and processing calls.
There are five Do Not Disturb options available at each extension:
1 = Incoming trunk calls blocked
2 = Paging, incoming Intercom, Call Forwards and transferred trunk calls blocked
3 = All calls blocked
4 = Incoming Call Forwards blocked
0 = Do Not Disturbed canceled

Setting DND while a call is ringing
For system phones DND can also be set while a call is ringing at your extension. Do not go off hook, press the DND key and select the option. Once set the DND will apply to further calls, until the DND is cancelled.
If an internal call is ringing then you can set DND 2 or 3. The internal call will be released and the caller will receive DND busy tone. The same operation can be used even if you have already got DND set and the caller has selected DND override.
If an ISDN trunk call is ringing then you can set DND 1 or 3. The trunk call will be released, the caller will hear a supervisory tone/ announcement dependent on the network supplier. The system will send Disconnect user busy to the network.
This feature is not available for analogue trunks ringing at the system phone, the analogue trunk will continue to ring.

Conditions
If there is no Call Forwarding programmable function key, the DND key will blink when the extension is forwarded.

Default Settings
'DND/Call Forward Override (Bypass Call)' is on in Class of Service.
'Do Not Disturb' setting is on in Class of Service.

Related Features
Call Forwarding: If an extension user activates DND option 4, the system prevents other extensions from forwarding calls to them. If an extension already receiving forwarded calls activates DND option 4, callers to the forwarded extension hear DND tone.
Call Forwarding/Do Not Disturb Override: An extension user can override Call Forwarding or Do Not Disturb at another extension.
Distinctive Ringing, Tones and Flash Patterns: Set up DND flash patterns for DSS and Hotline keys.

Operation
To activate or deactivate Do Not Disturb while your extension is idle:
System Phone
1. Do not lift the handset.
2. Press DND key.
OR
Press idle CALL key and dial 847.
3. Dial the DND option code.
0 = Cancel DND
1 = Incoming trunk calls blocked
2 = Paging, incoming Intercom, Call Forwards and transferred trunk calls blocked
3 = All calls blocked
4 = Call Forwards blocked

Single Line Telephone
1. Lift handset.
2. Dial 847.
3. Dial the DND option code.
0 = Cancel DND
1 = Incoming trunk calls blocked
2 = Paging, incoming Intercom, Call Forwards and transferred trunk calls blocked
3 = All calls blocked
4 = Call Forwards blocked

Door Box

Description
The Door Box is a self-contained Intercom unit typically used to monitor an entrance door. A visitor at the door can press the Door Box call button (like a door bell). The Door Box then sends chime tones to all extensions programmed to receive chimes. To answer the chime, the called extension user just lifts the handset. This lets the extension user talk to the visitor at the Door Box.
The Door Box is convenient to have at a delivery entrance, for example. It is not necessary to have company personnel monitor the delivery entrance; they just answer the Door Box chimes instead.
Any number of system extensions can receive Door Box chime tones.
Each Door Box has a pair of normally open relay contacts that can connect to an electric door lock. Use these contacts to remotely control the entrance door. After answering the Door Box chimes, a system phone user can press FLASH to activate the Door Box contacts. This in turn releases the electric lock on the entrance door.

The system can have up to eight Door Boxes.

Conditions
Each channel in the 2PGDU has a jumper which must be set for Door Box operation when the 2PGDU is first installed.

Default Setting
No Door Boxes are installed.
Door Boxes are setup in Doorphones.

Related Features
Paging, External: If a 2PGDU has a Door Box connected, you cannot use that port for External Paging.
Single Line Telephones, Analogue Single Line Sets: Door Boxes can ring single line phones if allowed in programmed.

Operation
To call a Door Box:
System Phone
1. Press idle CALL key.
2. Dial 802.
3. Dial Door Box Number (1-6).
Single Line Telephone
1. Lift handset.
2. Dial 802.
3. Dial Door Box Number (1-6).

To activate the Door Box lock contacts:
System Phone
1. While talking to the Door Box, press the Flash key.
Single Line Telephone - Analogue Set
1. While talking to the Door Box, hook flash (Recall).
To answer a Door Box chime:
1. Lift handset.

To Forward the Doorbox to an External Destination
System Phone
1. Press idle CALL key
2. Dial 822
3. Dial Door Box Number (1-6) that is set to ring the phone
4. Dial Abdial location to be forward to
Single Line Telephone
1. Lift handset.
2. Dial 802.
3. Dial Door Box Number (1-6) that is set to ring the phone.

To Cancel the forward
System Phone
1. Press idle CALL key
2. Dial 822
3. Dial 0
Single Line Telephone
1. Lift handset.

2. Dial 802.
3. Dial 0.

DSS Console

Description

The DSS Consoles (64-Button, 24-Button) gives a system phone user a Busy Lamp Field (BLF) and one-button access to extensions, trunks and system features. This saves time for users that do a lot of call processing (e.g., operators or dispatchers).
The DSS Console simplifies:

.. Calling extensions and Door Boxes
.. Placing, answering and transferring outside calls
.. Making an External or Internal Page
.. Switching the Night Service mode

You can also program the DSS Console keys to store Function codes. This provides the DSS Console user with many of the features available on Programmable Feature Keys.
The maximum number of 64 button DSS Consoles allowed on the system is 9. Each unit can have 3 64-button consoles.

An extension can have one 64-button DSS console assigned.
An extension (display system phone only) can have one 24-button DSS console connected.
An extension can have both a 24-button DSS and 64-button DSS console installed.

Installation of the Consoles
The 64-button console plugs into a main/expansion unit or a 308/008 card. It must be connected to port 8 of each unit/card. Each console will take a port on the system. The console is assigned to a system phone in Program 30-02. The keys are set in Program 30-03, each key has a function number and optional additional data.
The 24-button console plugs into any system phone with a display via the adaptor socket in the base of the phone, it does not take a port on the system. The keys are set either via Program 15-07 or by service codes 851/852 from the system phone it is plugged into. Each key has a function number and optional additional data.

Conditions

A) Changing flash patterns for DSS Consoles will also change them for Hotline keys.

B) When installing a DSS, the system must auto-detect the console in order for the LEDS to function correctly. When connecting the DSS to an extension previously defined with another circuit type, undefine the circuit type (enter 00 in Program 10-03 for the extension number), then connect the DSS Console.

C) To program the keys on a 24-Button DLS in Program 15-07, use the extension number to which the DLS is installed and, regardless of the type of system phone connected, *start programming the DLS keys at key number*.

Default Setting

No 64-button DSS Consoles assigned (in Program 30-02).
All 64-button DSS Console key ranges are DSS/One Touch for extension numbers 200-263.
Once a 64-button DSS Console is enabled, the console's keys are DSS/One Touch keys (Program 30-03).
Once a 24-button console is attached the keys are Not Defined (Program 15-07).

Programming

10-03-01 : Terminal Type
10-03-09 : PCB Setup
15-07-01 : Programming Function Keys
20-02-03 : System Options for System Telephones - BLF Control and

20-06-01 : Class of Service for Extensions
30-01-01 : DSS Console Operating Mode
30-02-01 : DSS Console Extension Assignment
30-03-01 : DSS Console Key Assignment

Related Features
Central Office Calls, Answering and Placing
Door Box
Night Service
Paging, External and Internal
Programmable Function Keys

Operation

Calling an extension from your DSS Console:
Optional for 64-Button Consoles.
1. Press DSS Console key.
If the call voice-announces, you can make it ring by dialling 1.
If you don't have Handsfree, you must lift handset to speak.

Placing a trunk call from your 24-Button DSS Console:
1. Press DSS Console key assigned to trunk.
2. Dial outside number.
If you don't have Handsfree, you must lift the handset to speak.

Answering a trunk call from your DSS Console:
1. Press flashing DSS Console key assigned to trunk.
If you don't have Handsfree, you must lift the handset to speak.

Calling a Door Box from your 64-button DSS Console:
1. Press DOOR1 or DOOR2.
If you don't have Handsfree, you must lift the handset to talk to the Door Box.

Transferring a call using your DSS Console:
1. Place or answer call.
If you are on an Intercom call, press HOLD before going to the next step.
2. Press DSS key for extension that will receive transfer.
You cannot Transfer to an extension that is busy or in Do Not Disturb.
3. (Optional) Announce call.
If called party doesn't want the call, press flashing line or CALL key to retrieve it.
4. Press SPK to hang up.

Making a External Page using your 64-button DSS Console:
1. Press DSS Console External Page zone key (EZ1, EZ2 or EAZ).
If the zone you want is busy, try again later.
If you don't have Handsfree, lift the handset to make your announcement.

Making an Internal Page using your 64-button DSS Console:
1. Press DSS Console Internal Page zone key (IZ1 – IZ4).
If the zone you want is busy, try again later.
If you don't have Handsfree, lift the handset to make your announcement.
Switching the Night Service mode from your 64-button DSS Console:
1. Press Night Service (NT) key.
The NT key will flip the system between DAY1 and NIGHT1 mode.
Using a DSS Console key as a Programmable Function Key:
You can store Function codes under DSS Console keys.
1. Press DSS Console key for function.
For example, you can Forward your calls by pressing DSS Key + 1 + destination. Your DSS key must have been previously programmed for the Call Forward feature.

Edit Caller ID & Redial

Description

The 'Edit Caller ID & redial' feature enables the ability to edit the caller ID of a call from the Call History list and redial the edited number.

Conditions
A Call History function key (key 08) is required.

Default Setting
Storage of Caller ID for answered call is enabled in Keyphone options.
Call History key requires setting in Function key programming.

Operation
From an idle display Dterm.

Delete Dial
1. Press Call History key
2. Press HOLD key
3. Press 'UP' volume key - cursor moves one position to right
4. Press 'DOWN' volume key - cursor moves one position to left
5. Press [DEL] soft key - flashing position is deleted
6. Press HOLD key - the edit is saved, or press [CNCL] soft key - the edit is cancelled.
7. To dial the number, go off hook.

Edit Dial
1. Press Call History key
2. Press HOLD key
3. Press 'UP' volume key - cursor moves one position to right
4. Press 'DOWN' volume key - cursor moves one position to left
5. Press any numeric key - digit is inserted behind flashing position
6. Press HOLD key - the edit is saved, or press [CNCL] soft key - the edit is cancelled.
7. To dial the number, go off hook.

External Paging

Description
With External Paging, a user can broadcast announcements over paging equipment connected to external Paging zones. Like Internal Paging, External Paging allows a user to locate another employee or make an announcement without calling each extension individually.
The system allows up to nine External Paging zones. There is one zone on the Main Unit. All other zones require a port on a 2PGDU card, with a maximum of two external paging circuits per module. You must have four 2PGDU card to get the eight external zones. In addition, each external zone has an associated relay contact. When a user pages to a zone, the corresponding contact activates (closes). This provides for Paging amplifier control.

Combined Paging
Use Combined Paging when you want to simultaneously Page into an internal and corresponding external zone. Combined Paging is available for Paging zones 1-6 and All Call. Refer to 'Paging, Internal' for more on setting up Combined Paging. In addition, you can program a Function Key as a Combined Paging key. Using the External Page Function Key, when an All Call External Page Function Key is programmed, it will include both the external zones and the assigned internal zone(s). If the internal page zone is busy or there are no extensions in a page group, the announcement will be made on the external zones only.

Conditions
For more than one external page zone (as provided on the Main Unit), External Paging requires 2PGDU card and customer-provided Paging equipment.
With Combined Paging, the system will allow a page over just the external page zone if the internal zone is busy or if there are no extensions in a page group.
It is not possible to prevent an extension user from making an external paging call.

Default Setting
No External Paging defined.

Related Features
Door Box: If a 2PGDU card has a Door Box connected, you cannot use that port for External Paging.
Night Service (Universal Night Answer): To have outside calls ring External Paging Zones at night, refer to the Night Service feature.
Paging, Internal: Internal Paging broadcasts announcements to extensions in programmed Internal Paging Zones.
Programmable Function Keys: Function keys simplify External Paging operation.

Operation
To Page into an external zone:
1. Press External Paging key (SC 851: 19 + zone for External Paging zones or 20 for External All Call Paging).
2. Make Announcement.
OR
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 803 and the External Paging Zone code (1-6 or 0 for All Call).
OR
Dial 751 and the Combined Paging Zone code 1-6 (for Internal/External Zones 1-6) or 0 (for Internal/External All Call).
Display indicates the Combined Paging as an External Page.
If the Internal Page Zone is busy or if there are no extensions in a page group, the page may be announced as an External Page only.
3. Make Announcement.

F-Route

Description
Flexible Routing (F-Route) provides call routing and digit translation based on the digits a user dials. F-Route consists of look up tables that compare the dialled digits and decide which translation table should be used. The decision can optionally be dependent on time and day of the week. The translation table will then delete and/or add digits select the trunk group and seize the outgoing trunk.

Conditions
Line keys, outgoing loop keys, outgoing trunk group keys, dialing 804+trunk group, dialing 805+trunk number, and abbreviated dial numbers assigned to a certain trunk group can all be used to by-pass F-Route.

Default Setting
F-Route is not programmed.

Flash

Description
Flash allows an extension user to access certain CO and PBX features by interrupting trunk loop current. Flash lets an extension user take full advantage of whatever features the connected network or PBX offers. You must set the Flash parameters for compatibility with the connected network or PBX.

Conditions
The system does not provide a ground flash.

Default Setting
The Flash type is set to 'Disconnect' in Analogue Trunk Data Setup.
Single Line Telephones can use Flashing in SLT Basic Setup.
Flash type 1 'Flash' is set to 800mS in Analogue Trunk Initial Data Setup.
Flash type 2 'Disconnect' is set to 2496mS in Analogue Trunk Initial Data Setup.

Related Features
PBX Compatibility: If the system is behind a PBX, Flash normally gives the extension user access to many PBX features.
Toll Restriction: The system applies Toll Restriction (if applicable) to the number a user dials after flashing a trunk.

Operation
To flash the trunk you are on:
System Phone and Digital Single Line
1. Press FLASH.
Single Line Set
1. Hook flash.
2. Dial 806.

Flexible Ringing by Caller ID

Description
The system can route incoming calls by using the Caller ID received from the network.
The following diagram shows an overview of Caller ID routing.
By using the Incoming Ringing by VRS channel feature detailed later, the incoming call can be matched to a VRS message to be be played instead of ring tone.

Caller ID Received with the incoming trunk call — No — Route the caller as defined in Incoming Service Type Setup
|
Yes
|
Caller ID entered within Abbreviated Dial table — No — Route the caller as defined in Incoming Service Type Setup
|
Yes
|
Use the routing defined within the Abbreviated Dial table

The system can route the call to either a defined number or to an 'Incoming Ring Group', there is also an option to reject the incoming call.

Routing to a pre-defined number
The system can route the call to a number defined within the Abbreviated Dial table, this number can be an extension on the system, the DSPDB voice mail access number, Department Group pilot number or an F-Route number.

Routing to an Incoming Ring Group
The system can route the call to an Incoming Ring, the call will follow the settings defined in Incoming Ring Group Setup.

Rejecting Incoming Calls
The system can reject incoming ISDN calls based on the Callers ID.
Since the call is rejected the incoming caller may hear a Network tone/announcement indicating that the number is not obtainable.
The system can also use this option for incoming calls on an analogue trunk; the incoming call will be ignored (the incoming caller will hear ring-back tone) since it is not possible to reject a call on an analogue trunk.
The system can also play a pre-recorded DSPDB message instead of rejecting the incoming call. The message will be repeated twice and the call will be disconnected.

Caller ID not received or identified as Private.
If the incoming call does not have any Caller ID or it is identified as Private the system will route the call as defined in Incoming Service Type Setup.

Selecting a unique ring tone
The system can use a different ringing tone for calls routed by the Caller ID. You can select from any of the standard ringing tones available on the system.

Conditions
- Caller ID must be sent by the Network Supplier for this feature to operate.
- Caller ID routing will take priority over Normal/DDI routing.
- All calls with a matching Caller ID number will be routed, it is not possible to bypass caller ID routing.
- Incoming calls that are routed via the caller ID will not use any other system routing options, for example 'Step on if not answered'.
- The Abbreviated Dial store is used to route calls based on the incoming callers ID number, ensure that only the System Administrator has the ability to edit the entries in Class of Service (Common/Group Abbreviated Dial Reg).

Default Setting
Caller ID routing is available when the Abbreviated Dials are entered.
Rejecting of calls based on Caller ID is disabled in Trunk Basic Data Setup (Caller ID Refuse Setup).
No DSPDB message is assigned to Caller ID Refuse in VRS Options.
No Service codes are assigned to 'Entry Caller ID Refuse' or 'Set Caller ID Refuse' in 3 Digit Codes.
No function keys are assigned for 'Set Caller ID Refuse' in Function Key Programming.
'Set / Cancel Caller ID Refuse' is disabled in Class of Service.

No Group Abbreviated dial area is assigned in Caller ID Setup for Caller ID Refuse.

Related Features
Abbreviated Dial.
Caller ID.

Operation

Caller ID Routing
Caller ID Routing is automatic once the Abbreviated Dial entries are allocated.

Caller ID Reject
Rejecting calls based on their Caller ID will require the following system configuration setup:
The trunk must be enabled in the 'Refuse Caller ID Setup' option in Trunk Basic Data Setup.
A Group Abbreviated dial area must be assigned in Caller ID Setup.
The Group Abbreviated dial area must have an entry that matches the callers number.
An extension must be enabled in the 'Set / Cancel Caller ID Refuse' option in Class of Service.
The user must have a function key assigned to 'Refuse Caller ID' (code 87) or a service code must be assigned to 'Set Caller ID Refuse' in 3 Digit Codes.
The DSPDB message is optional but if this is not assigned the rejected callers may hear a network tone/announcement indicating that the number called is not obtainable.

Caller ID Refuse can then be switched on/off by a the user with either the service code or function key.

Edit the Caller ID Refuse Group Abbreviated Dial Entries
A user can search and edit the Group Abbreviated Dial area assigned to Caller ID Refuse.
Note - a user that is also assigned to this Group Abbreviated dial could also edit the entries. To prevent accidental editing ensure that no extensions are assigned to this Abbreviated Dial group in Abbreviated Dial Group Assignment for Extensions.

The user must have a keyphone with a display and the service code for 'Entry Caller ID Refuse' must be assigned in 3 Digit Codes.

1. Press SPK
2. Dial the service code for 'Entry Caller ID Refuse'
3. The display will show the options to Set or Search against the soft keys.
To add a new entry press the Set key, to search and delete an entry press the Search key.

To Search the entries
1. Press the Search key at step 3 above.
2. Use the Up and Down soft keys to scroll through the entries
3. While the entry is displayed press the DEL soft key to delete the entry.

To Add a New Entry
1. Press the Set key at step 3 above.
2. If there are no spare entries the display will show NO MEMORY, you must delete an existing entry berore you can add a new one.
3. If there are spare entries enter the callers number that you want to reject.
4. Press the SET soft key to confirm the entry.

Entering Caller Numbers.
The system will match the leading digits of the callers number with the leading digits entered in the Abbreviated Dial Group.
For example.

Caller ID	Abbreviated Dial Entry	Result
2039261111	2039261111	Matched
2039261111	20392611119	Matched
2039261111	203	Matched
9261111	2039261111	Unmatched
2039261111	9261111	Unmatched

Incoming Ringing by VRS

This feature enhances the flexible ringing by called ID to play a VRS message instead of the ring tone.

Entries are either set by programming within the Flexible Ringing by Caller ID are or can be added by handset.

Entering new entries by service code.
This can also be set from the Call History from the OPAC key.

1. Press SPK or lift handset
2. Dial VRS Setting service code (default 878)
3. Dial '1' to set VRS message for Caller ID
4. Enter caller ID and press hold
5. Caller ID is added to next available abdial area
6. Press hold and enter name to be displayed for caller id
7. Press hold
8. Enter 0 for not define, 1 for internal Dial, 2 for IRG
9. Enter destination (extension number or IRG 1-100)
10. Set ring pattern
11. Set VRS message (0-48) if 0 entered VRS message is not played.
12. Press hold - confirmation tone is heard.

Setting VRS message as ring tone

1. Press SPK or lift handset
2. Dial VRS Setting service code (default 878)
3. Dial '2' to set VRS message at own terminal
4. Input VRS message number
5. Press Hold

Flexible System Numbering

Description
Flexible System Numbering lets you reassign the system’s port-to-extension assignments. This allows an employee to retain their extension number if they move to a different office. In addition, factory technicians can make comprehensive changes to your system’s number plan. You can have
factory technicians:
• Set the number of digits in internal (Intercom) functions. For example, extension numbers can be up to eight digits long.
• Change your system’s Service Code numbers
• Assign single digit access to selected Service Codes

You can also use Flexible System Numbering to change the system’s Trunk Group Routing code.
Although the default code of 9 is suitable for most applications, you can alter the code if you have to.

Conditions
Programming follows a telephone’s extension number, not the port number in most cases. If you relocate a phone, you may need to change additional programming.
If the extension numbering plan is changed from ‘2xx’ to ‘1xx’, and you would like to consecutively
press two DSS keys without toggling the hook switch, 1 Digit Codes must be removed. If not, pressing the second DSS key will actually change voice/ringing call to the first extension.
Since making changes in Program 11-01 does not automatically make any other changes in any other program, changing the number plan after the system is in operation may cause problems in other programs:
Any feature which requires dialing a code or extension number can be affected by changes to the system numbering plan.

Default Setting
Refer to System Numbering Plan for details.

Related Features
None.

Operation
None.

Forced Trunk Disconnect

Description
Forced Trunk Disconnect allows an extension user to disconnect (release) another extension’s active outside call. The user can then place a call on the released trunk.
This can happen if a trunk does not properly disconnect when the outside party hangs up.
Forced Trunk Disconnect lets a user access a busy trunk in an emergency, when no other trunks are available.

Conditions
None

Default Setting
'Force Trunk Disconnect' is off in Class of Service

Related Features
Central Office Calls, Placing: A user can use Forced Trunk Disconnect only for trunks to which it would normally have access.

Operation
To disconnect a busy trunk:
System Phone
1. Press line key for trunk.
OR
Dial trunk access code (805 + trunk number).
You hear busy tone. Trunk numbers are 01-51.
2. Dial 724.
You hear confirmation beeps as the system disconnects the trunk.
You can now place a call on the free trunk.
3. Press line key for the trunk disconnected in Step 2.
OR
Dial the trunk access code (805 + trunk number) for the trunk disconnected in Step 2.
Single Line Telephone
1. Dial trunk access code (805 + trunk number).
You hear busy tone. Trunk numbers are 01-51.
2. Dial 724.
You hear confirmation beeps as the system disconnects the line.
3. Hook flash.
You can now place a call on the free line.
4. Dial the trunk access code (805 + trunk number) for the trunk disconnected in Step 2.

Handsfree Answerback

Description
Handsfree Answerback permits an extension user (system phone or digital single line) to respond to a voice-announced Intercom call by speaking toward the phone, without lifting the handset. Like Handsfree, this is a convenience for workers who don’t have a free hand to pick up the handset.

Conditions
Handsfree Answerback is not available at a analogue single line telephone.

Default Setting
'Incoming Call form Extension Call' is set to Voice Calling mode in System Options for Keyphones.
20-08-10 'Signal/Voice Call Switching' is on in Class of Service, allows a user to change between modes while calling another extension.
20-09-05 'Signal/Voice Call Switching' is on in Class of Service, allows a user to set their phone to voice or signal calling mode.

Related Features
Handsfree and Monitor: A system phone user can process calls using the speaker and microphone in the telephone (instead of the handset).
Microphone Cutoff: With Microphone Cutoff enabled, Handsfree Answerback callers to an extension hear a single beep (instead of

two).
Single Line Telephones: Incoming Intercom calls always ring single line telephones.

Operation
To enable Handsfree Answerback for your incoming Intercom calls:
1. Press idle CALL key.
2. Dial 821.
3. Press SPK to hang up.
This disables Forced Intercom Ringing.

To enable Forced Intercom Ringing for your incoming Intercom calls:
1. Press idle CALL key.
2. Dial 823.
3. Press SPK to hang up.
This disables Handsfree Answerback.

To change the way your Intercom call signals the extension you are calling:
1. Dial 1.
If ringing, your call voice-announces. If voice-announced, your call starts to ring the destination. This option is also available at single line telephones.

Handsfree

Description
Handsfree Answerback permits a system phone user to respond to a voice-announced Intercom call by speaking toward the phone, without lifting the handset. Like Handsfree, this is a convenience for workers who don't have a free hand to pick up the handset.
Conditions
Handsfree Answerback is not available at an analogue single line telephone.

Default Setting
Enabled.

Related Features
Central Office Calls, Answering / Central Office Calls, Placing: Extensions should be programmed for incoming and outgoing access, ringing, etc.
Handsfree Answerback: Answer Intercom calls without lifting the handset - just speak toward the phone.
Microphone Cutoff: For privacy, mute the phones microphone while on a call.
Single Line Telephones: Handsfree and Monitor are not available to single line telephones.
Prime Line Selection: Prime Line Selection affects how incoming and outgoing calls are handled and thus determines what happens when the user presses the SPK key.

Operation
To talk Handsfree:
1. Press SPK, CALL key or line key.
2. Place call.
3. Speak toward phone when called party answers.

To change a handset call into a Handsfree call:
1. Press SPK.
2. Press SPK to hang up.
To change a Handsfree call into a handset call:
1. Lift handset.

Headset

Description
A system phone user can utilize a customer-provided headset in place of the handset. Like using Handsfree, using the headset frees

up the user’s hands for other work.
An extension in the headset mode has two options for when it appears busy to incoming callers.
The headset extension can be:
• Busy to incoming callers when only one extension appearance is busy (i.e., Off-Hook Signaling prevented)
OR
• Busy to incoming callers only when both extension appearances are busy (i.e., Off Hook Signaling allowed)
As the headset plugs into a separate jack on the bottom of the phone, the handset can still be connected to the phone. This provides you with the option to use the handset, headset or the speaker phone for calls.

Conditions
• While in the headset mode, the Headset function key becomes a release (disconnect) key and no dial tone is heard from the speaker or handset.
• While in the headset mode, the hook switch and SPK button is not functional.
• Headset operation is not available at a Basic system phone.

Default Setting
No headset keys are defined in Function Key Programming.
'Busy Call in Headset Mode' is disabled in System Options for Keyphones. When disabled the system phone is busy to incoming callers when only one CALL key is in use.
When enabled then both CALL keys must be in use for the keyphone to be busy.
'Headset Ear-Piece Ringing' for SLT's is off inClass of Service.
'Headset Ringing Start Time for SLT' is set to 5 seconds in SLT Options.
Service code for SLT headset mode is 788.

Related Features
Handsfree Answerback/Forced Intercom Ringing: An extension with a headset can still receive voice-announced Intercom calls and respond Handsfree.
Programmable Function Keys: A Headset Function key is required to answer or place a call in headset mode.
Single Line Telephones: Single line telephones can also use the Headset feature.

Operation
To enable the headset at a keyphone:
1. Plug in the headset into the headset jack on the back of the phone.
2. Program a Headset key (SC 851: 05).
You hear a confirmation beep.

To use the headset at a keyphone:
The Headset key lights when you’re on a call. To disconnect, press the Headset key again.
You can still use the handset for calls or respond to voice-announced Intercom calls
with the headset plugged in. The headset only activates when the Headset key is pressed.
• Answer a ringing call by pressing the Headset key.
OR
• Press the Headset key and then a line key to make a trunk call.
OR
• Press the Headset key to get Intercom dial tone
OR
• If on a call, press the Headset key to hang up.

To enable headset at an SLT:
1. Lift the handset and dial 788. You hear a single beep.
2. Replace the handset.
3. Lift the handset and remain off hook. Dial tone will stop after 5 seconds and the SLT is now in headset mode.

To cancel headset mode at an SLT:
1. Lift the handset and dial 788. You hear two beeps.
2. Replace the handset.

To answer a call at an SLT during headset mode:
1. You will hear a double beep when a call is 'ringing' at the SLT.
2. Press RECALL to answer the call.
3. To end the call go on hook.
4. Lift the handset to return to headset mode.

Hold

Description
Hold lets an extension user put a call in a temporary waiting state. The caller on Hold hears silence or Music on Hold. While the call waits on Hold, the extension user may process calls or use a system feature. Calls left on Hold too long recall the extension that placed them on Hold.
There are four types of Hold:
• System Hold
An outside call a user places on Hold flashes the line key (if programmed) at all other keyphones.
Any system phone user with the flashing line key can pick up the call.
• Exclusive Hold
When a user places a call on Exclusive Hold, only that user can pick up the call from Hold.
The trunk appears busy to all other keyphones that have a key for the trunk. Exclusive hold is important if a user doesn't want a co-worker picking up their call on Hold.
• Group Hold
If a user places a call on Group Hold, another user in the Department Group can dial a code to pick up the call. This lets members of a department easily pick up each other's calls.
• Intercom Hold
A user can place an Intercom call on Hold. The Intercom call on Hold does not indicate at any other extension.

Conditions
None.

Default Setting
Enabled.

Related Features
Music on Hold: Callers on Hold hear Music on Hold, if programmed.
Programmable Function Keys: An extension can have function keys for System Hold and Exclusive Hold.
Single Line Telephones: Analogue single line telephones can only use Exclusive Hold and Group Hold. Digital single line phones can use System Hold as well.

Operation
System Hold
To place an outside call on System Hold (System Phone only):
1. Press HOLD.
A line/loop/CALL key flashes slowly while on Hold; flashes fast when recalling.
OR
1. If you know the specific line number, dial 772 + Line number (01-51).
To pick up an outside call on System Hold:
System Phone
1. Press flashing line/loop/CALL key.
OR
1. If you know the specific line number, dial 772 + Line number (01-51).

Exclusive Hold
To place an outside call on Exclusive Hold:
System Phone
1. Press Exclusive Hold key (SC 851: 45).
A line/loop/CALL key flashes slowly while on Hold, flashes fast when recalling.

To pick up an outside call on Exclusive Hold:
System Phone
1. Press flashing line/loop/CALL key.

Group Hold
To place a call on Hold so anyone in your extension group can pick it up:
System Phone
1. Press HOLD.
2. Dial 832.
3. Press SPK to hang up.
Single Line Telephone
1. Hook flash.

2. Dial 832.
3. Hang up.

To pick up a call on Group Hold:
System Phone
1. Press idle CALL key.
2. Dial 862.
Single Line Telephone
1. Lift handset.
2. Dial 862.
Intercom Hold

To place an Intercom call on Intercom Hold:
1. Press HOLD.
The CALL key flashes.
2. Press SPK to hang up.
To pick up an Intercom call on Intercom Hold:
1. Press SPK.
2. Press flashing CALL key.

Hotel

Description
The system can provide hotel services in addition to the features normally available to business users.
Hotel features include:
• Do Not Disturb
A guest can enable and disable Do Not Disturb for their room telephone. In addition, a hotel receptionist with a system phone can enable and disable Do Not Disturb for a specific room telephone.
• Message Waiting
A hotel receptionist with a system phone can send a Message Waiting to a room telephone.
The message lamp on the room telephone flashes until the guest answers the Message Waiting.
• Room Telephone Status
To better manage room usage, a hotel receptionist with a system phone can change the status of a room telephone, including:
• Room Available
• Room Occupied
• Room Ready to be Cleaned
• Room to Room Call Restriction
To control inter-room guest calling, a hotel receptionist with a system phone can enable and disable room- to-room calling.
• Room Status with Printout
A DSS Console can indicate the status of the hotel rooms. Optionally, a printer connected to the EXIFU card can print out room status reports:
• Room Status (occupied, available, ready and to be cleaned)
• Room Telephone Call and Toll Restriction Information
• Do Not Disturb and Clean Up Extension List
• Message Waiting Report
• Wake-up Call No-Answer Report
• Single Digit Extension Access
To simplify guest calling, room telephones can have single digit access to selected extensions.
For example, this allows guests to dial 1 for the front desk, 2 for house cleaning etc.
• Toll Restriction Changing
An employee can change the Toll Restriction for a guest's telephone. For example, the receptionist can enable long distance calling for each room telephone as the guests check in.
• Wake-up Call
A guest can set or cancel a wake-up call request. A hotel receptionist with a system phone can also set or cancel a wake-up call for a room telephone.

Conditions
Hotel print outs require an EXIFU card.

Related Features
Room Monitor: Room Monitor allows both system phone users and single line telephone users to monitor the activity at another extension.
Toll Restriction: Hotel room extensions can have different toll restriction setting when checked in/out.

Operation

Message Waiting

Key Telephone
To set the Message Waiting:
1. Call to busy or unanswered extension.
2. Dial 841( ).
OR
Press Massage Waiting Key (SC851: 38).
OR
Dial One Digit Service Code for Message Waiting.
3. Press SPK to hang up.
OR
1. Press idle CALL key.
2. Dial 726( )
3. Dial extension number.
4. Press SPK key to hang up.

To cancel the Message Waiting:
1. Press idle CALL key.
2. Dial 871( )
3. Dial extension number.
4. Press SPK key to hang up.
To answer the Message Waiting:
1. Press idle CALL key.
2. Dial 841( )

Single Line Telephone
To set the Message Waiting:
1. Call to busy or unanswered extension.
2. Dial 841( ).
OR
Dial One Digit Service Code for Message Waiting.
3. On hook.
OR
1. Off hook
2. Dial 726( )
3. Dial extension number.
4. On hook.

To cancel the Message Waiting:
1. Off hook.
2. Dial 871( )
3. Dial extension number.
4. On hook.

To answer the Message Waiting:
1. Off hook.
2. Dial 841( )

Do Not Disturb (DND)

Key Telephone
To set DND for own extension:
1. Press idle CALL key.
2. Dial 727( )
3. Press SPK key to hang up.
To cancel DND for own extension:
1. Press idle CALL key.
2. Dial 728( )
3. Press SPK key to hang up.

To set DND for other extension:
1. Press idle CALL key.
2. Dial 729( )
3. Press SPK key to hang up.

To cancel DND for other extension:
1. Press idle CALL key.

2. Dial 730( )
3. Press SPK key to hang up.

Single Line Telephone
To set DND for own extension:
1. Off hook.
2. Dial 727( )
3. On hook.

To cancel DND for own extension:
1. Off hook.
2. Dial 728( )
3. On hook.

To set DND for other extension:
1. Off hook.
2. Dial 729( )
3. On hook.

To cancel DND for other extension:
1. Off hook.
2. Dial 730( )
3. On hook.

Wake Up Call
Key Telephone
To set Wake Up Call for own extension:
1. Press idle CALL key.
2. Dial 731( )
3. Dial time (24 hours).
For example, dial 1540 for 3:40PM.
4. Press SPK key to hang up.

To cancel Wake Up Call for own extension:
1. Press idle CALL key.
2. Dial 732( )
3. Press SPK key to hang up.

To set Wake Up Call for other extension:
1. Press idle CALL key.
2. Dial 733( )
3. Dial extension number.
4. Dial time (24 hours).
For example, dial 1540 for 3:40PM.
5. Press SPK key to hang up.

To cancel Wake Up Call for other extension:
1. Press idle CALL key.
2. Dial 734( )
3. Dial extension number.
4. Press SPK key to hang up.

Single Line Telephone
To set Wake Up Call for own extension:
1. Off hook.
2. Dial 731( )
3. Dial time (24 hours).
For example, dial 1540 for 3:40PM.
4. On hook.

To cancel Wake Up Call for own extension:
1. Off hook.
2. Dial 732( )
3. On hook.

To set Wake Up Call for other extension:
1. Off hook.

2. Dial 733( )
3. Dial time (24 hours).
For example, dial 1540 for 3:40PM.
4. On hook.

To cancel Wake Up Call for other extension:
1. Off hook.
2. Dial 734( )
3. On hook.

Room to Room Barring
Key Telephone
To set Room to Room Barring:
1. Press idle CALL key.
2. Dial 735( )
3. Dial extension number.
4. Press SPK key to hang up.

To cancel Room to Room Barring:
1. Press idle CALL key.
2. Dial 736( )
3. Dial extension number.
4. Press SPK key to hang up.

Single Line Telephone
To set Room to Room Barring:
1. Off hook.
2. Dial 735( )
3. Dial extension number.
4. On hook.

To cancel Room to Room Barring:
1. Off hook.
2. Dial 736( )
3. Dial extension number.
4. On hook.

Toll Restriction Change
Key Telephone
To change Toll Restriction Class:
1. Press idle CALL key.
2. Dial 737( )
3. Dial extension number.
4. Dial Toll Restriction Class (01-15).
5. Press SPK key to hang up.

Single Line Telephone
To change Toll Restriction Class:
1. Off hook.
2. Dial 737( )
3. Dial extension number.
4. Dial Toll Restriction Class (01-15).
5. On hook.

Check-In/Out, Clean Up
Key Telephone
To set Check-in:
1. Press idle CALL key.
2. Dial 738( )
3. Dial extension number.
4. Press SPK key to hang up.
To set Check-out:
1. Press idle CALL key.
2. Dial 739( )
3. Dial extension number.
4. Press SPK key to hang up.

To set Additional Room Status for own extension:
1. Press idle CALL key.
2. Dial 740( )
3. Dial an Additional Room Status Code (1-4)
1: Room is clean
2: Maid Required
3: Maid in Room
4: Inspection Required
4. Press SPK key to hang up.

To set Additional Room Status for other extension:
1. Press idle CALL key.
2. Dial 741( )
3. Dial an Additional Room Status Code (1-4)
1: Room is clean
2: Maid Required
3: Maid in Room
4: Inspection Required
4. Dial extension number.
5. Press SPK key to hang up.

Single Line Telephone
To set Check-in:
1. Off hook.
2. Dial 738( )
3. Dial extension number.
4. On hook.
To set Check-out:
1. Off hook.
2. Dial 739( )
3. Dial extension number.
4. On hook.

To set Additional Room Status for own extension:
1. Off hook.
2. Dial 740( )
3. Dial an Additional Room Status Code (1-4)
1: Room is clean
2: Maid Required
3: Maid in Room
4: Inspection Required
4. On hook.

To set Additional Room Status for other extension:
1. Off hook.
2. Dial 741( )
3. Dial extension number.
4. Dial an Additional Room Status Code (1-4)
1: Room is clean
2: Maid Required
3: Maid in Room
4: Inspection Required
5. On hook.

One Digit Extension Access
Key Telephone
To place 1-Digit Extension Access call:
1. Press idle CALL key.
2. Dial One Digit Access code. (0-9, *, #)

Single Line Telephone
To place 1-Digit Extension Access call:
1. Off hook.
2. Dial One Digit Access code. (0-9, *, #)

Room Status Print Out
Key Telephone

To set Print out:
1. Press idle CALL key.
2. Dial 742( )
3. Dial 1 or 2 or 3 or 4 or 5 for the following enquiry
0: All items
1: Room Status List (Check-in and House Cleaning Status)
2: Call (Toll) Restriction List
3: Do Not Disturb and Room Clean List
4: Message Waiting List
5: Wake Up Call List
4. Press SPK key to hang up.

Single Line Telephone
To set Print out:
1. Off hook.
2. Dial 742( )
3. Dial 1 or 2 or 3 or 4 or 5 for the following enquiry
0: All items
1: Room Status List (Check-in and House Cleaning Status)
2: Call (Toll) Restriction List
3: Do Not Disturb and Room Clean List
4: Message Waiting List
5: Wake Up Call List
4. On hook.

Hotel Application DSS Console
DSS Console
To indicate Message Waiting Status:
1. Press IZ1 key.
ON : Set
OFF: Not Set

To indicate Wake Up Call Status:
1. Press IZ2 key.
ON: Set
OFF : Not Wake up Call
FLASH : Wake Up Call Missed

To indicate Room Status:
1. Press IZ3 key.
ON: Checked In and Clean
OFF : Checked Out (Clean and Available)
Slow FLASH : Maid Required
Slow Wink : Maid in Room
Fast FLASH : Inspection Required

Hot Keypad

Description
The hot keypad feature enables the user to make a call without pressing the speaker key when the Dterm is in idle status. This is achieved by automatically taking the Dterm off hook when any keypad digit or one touch key is pressed.

Conditions
When both hot keypad and VRS fixed messages are enabled for a Dterm, VRS messages do not function.
When both Hot Keypad and Hotline are enabled for a Dterm, Hot Keypad takes priority.

Default Setting
Hot Keypad is disabled in Class of Service.

Configuration
Enter the class of service number for the Dterm in Class of service per night modes.
Enable Hot Keypad in Class of service options

Hotline

Description
Hotline gives a system phone user one-button calling and Transfer to another extension (the Hotline partner). Hotline helps co-workers that work closely together. The Hotline partners can call or Transfer calls to each other just by pressing a single key.
A hotline key is also referred to as a DSS key.
In addition, the Hotline key shows the status of the partner's extension.

Conditions
An extension user cannot use Hotline to pick up a call ringing their partner's extension.

Default Setting
No Hotline (DSS) keys are assigned in Function Key Programming.
Lamp patterns are setup in DSS Lamp Table.

Related Features
Distinctive Ringing, Tones and Flash Patterns: Set up flash patterns for DSS and Hotline keys.
Do Not Disturb: Hotline does not override Do Not Disturb.
Handsfree Answerback/Forced Intercom Ringing: Hotline always follows the Handsfree Answerback/Forced Intercom Ringing mode set at the called extension. The Hotline caller can override the setting, if desired.
Hotline, External: External Hotline (Ringdown Extension) will automatically dial a telephone number or Common Abbreviated Dialing number when the handset is lifted.
Multiple Directory/Call Coverage: A call coverage key can also be set to DSS mode in Keyphone Options to allow similar operation to a hotline key.
Off Hook Signaling: If the partner's extension is busy, Hotline does not automatically activate Off Hook Signaling.
Programmable Function Keys: A Hotline is a uniquely programmed function key.

Operation
To place a call to your Hotline partner:
1. Press Hotline key (SC 851: 01 + partner's extension number)
You can optionally lift handset after this step for privacy.

To transfer your outside call to your Hotline partner:
1. Press Hotline key.
2. Announce call and hang up.
OR
Hang up to have the call wait at your Hotline partner un-announced.
If unanswered, the call recalls like a regular transferred call.

To answer a call from your Hotline partner:
1. If you hear two beeps, speak toward phone.
OR
1. If your telephone rings, lift handset.

Intercom

Description
Intercom gives extension users access to other extensions. This provides the system with complete internal calling capability.

Handsfree Answerback/Forced Intercom Ringing.
Handsfree Answerback permits an extension user to respond to a voice-announced Intercom call by speaking toward the phone, without lifting the handset. Like Handsfree, this is a convenience for workers who don't have a free hand to pick up the handset.

Conditions
Preventing ICM calls does not affect dialing other service codes, including Emergency calls.

Default Setting
20-08-01 'Intercom Call' is on in Class of Service.
'Incoming Call from Extension Mode' is set to Voice calling in System Options for Keyphones.

20-08-10 'Signal/Voice Call Switching' is on in Class of Service (Change from voice/signal while calling an extension).
20-09-05 'Signal/Voice Call Switching is on n Class of Service (Selecting voice/signal for calls to your phone).

Related Features
Handsfree Answerback/Forced Intercom Ringing: Intercom calls can ring or be voice-announced at the called extension.
Intercom Abandoned Call Display: Intercom Abandoned Call Display remembers the last five Intercom calls to an extension.
Line Preference: Ringing Line Preference can automatically answer ringing Intercom or trunk calls when the user lifts the handset.
Name Storing: An extension can have a name assigned that identifies the extension to callers.
Busy Status Display: A system phone with a display can see the status of busy co-worker when placing a call.

Operation
To place an Intercom call:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial extension number (or 0 for your operator).
Your call may voice-announce or ring the called extension. Dial 1 to change the way your call alerts the called extension.
If the extension you call is busy or doesn't answer, you can dial another extension without hanging up.

To answer an Intercom call:
1. If you hear two beeps, speak toward phone.
Your telephone picks up your voice.
OR
If your telephone rings, lift handset.

To check your extension's data (System Phone Only):
1. Press CHECK.
2. Press CALL.
You display shows your telephone's extension number, port number and extension/ Department Group.
3. Press CLEAR to return the normal time/date display.

To change how Intercom calls ring your extension:
1. Press the CALL key or lift the handset.
2. Dial 823 to have calls ring your extension
OR
3. Dial 821 to have calls voice announce to your extension.

Internal Paging

Description
Internal Paging lets extension users broadcast announcements to other idle system phone users. When a user makes a Zone Paging announcement, the announcement broadcasts to all idle extensions in the zone dialed. With All Call Paging, the announcement broadcasts to all idle extensions programmed to receive All Call Paging. An extension can be a member of only one Internal Paging Zone. Like External Paging, Internal Paging allows a user to locate another employee or make an announcement without calling each extension individually.

Combined Paging
Use Combined Paging when you want to simultaneously Page into an internal and corresponding external zone. For example, you can Page your company's warehouse and outside loading dock at the same time. Combined Paging is available for Paging zones 1-6 and All Call. Optionally, you can change the Combined Paging assignments. For example, you can associate External Paging Zone 1 with Internal Paging Zone 4. You can be able to program a Function Key as a Combined Paging key. When an All Call External Page Function Key is programmed, it will include both the external zones and the assigned internal zone(s). If the internal page zone is busy or there are no extensions in a page group, the announcement will be made on the external zones only.

Remove Paging Information from Display Phones
A Class of Service option is available in system programming to prevent display telephones from showing incoming internal paging information. This allows the system to save processor time and speed up system operation for large groups.

Conditions
Internal Paging does not require a 2PGDU module.
You can assign any number of extensions to an Internal or All Call Paging Zone. But, as a practical limit you should not assign more than 30 extensions to a paging group.
A system must have at least one extension port idle in order to make an Internal Page. If no extension port is idle, the extension

performing the Page will hear a busy signal.
With Combined Paging, the system allows a page over just the external page zone if the internal zone is busy or if there are no extensions in a page group.
It is not possible to prevent an extension user from making an internal page call.

Default Setting
The extensions are not assigned to any paging group in Extension Basic setup.
Internal paging group 1 is assigned to the combined page for each external group in Combined Internal/External Paging.

Related Features
Meet Me Paging / Meet Me Paging Transfer: Preventing access to internal page also prevents these features.
Paging, External: An extension user can broadcast an announcement over an External Paging Zone.
Programmable Function Keys: Function keys simplify Internal Paging operation.

Operation
To make an Internal Page announcement:
System Phone
1. Press the zone’s Internal Paging key (SC 851: 21 + 1-6 or 01-32 for zones, 22 for All Call).
OR
1. Press idle CALL key.
2. Dial 801 and the Paging Zone number (0-6 or 00-32).
Dialing 0 or 00 calls All Call Internal Paging.
OR
Dial 751 and the Combined Paging Zone code 1-6 (for Internal/External Zones 1-6) or 0 (for Internal/External All Call).
Display indicates the Combined Paging as an External Page.
If the Internal Page Zone is busy or if there are no extensions in a page group, the page will be announced as an External Page only.
3. Make announcement.
4. Press SPK to hang up.

Single Line Telephone
1. Lift handset.
2. Dial 801 and the Paging Zone number (0-6 or 00-32).
Dialing 0 or 00 calls All Call Internal Paging.
Dial 751 and the Combined Paging Zone code 1-6 (for Internal/External Zones 1-6) or 0 (for Internal/External All Call).
3. Make announcement.
4. Hang up.

ISDN Compatibility

Description
SMDR Includes Dialed Number
The SMDR report can optionally print the trunk’s name (entered in system programming) or the number the incoming caller dialed (i. e., the ISDN DDI digits). This gives you the option of analysing the SMDR report based on the number your callers dial.
When using the SMDR reports for ISDN, all incoming BRI calls will be displayed under the CLASS column as “IVIN”.

Display Shows Why Caller ID is Not Available
With Caller ID enabled, the system will provide information for ISDN calls that do not contain the Caller ID information. If the Caller ID information is restricted, the telephone display will show “PRIVATE”. If the system is not able to provide Caller ID information because the network information is not available, then the display will show “OUT OF AREA”.

Calling Party Number Notification
The system can provide calling party number notification for outgoing ISDN calls. When a call is made on an ISDN line by an extension, the system will send the identification for the extension placing the call, if it’s programmed. If there is no Extension Calling Number assigned, the system will send the calling number for the ISDN trunk. If both the extension and trunk information is programmed, the extension information will be sent as it takes priority.
When the option for calling party sub address is on, the extension number will be sent as the sub address information. Both the calling party number and calling party sub address are sent in a SETUP message as the calling party information element and a calling party sub address information element.

Basic Rate Interface (BRI)
Your system also provides compatibility with ISDN Basic Rate (BRI) services, including:
• Point-to-Point BRI Terminal Connection (no daisy-chaining)
• Multi point BRI Terminal Connection (daisy-chaining)
• S-Bus (allows BRI PCB's to be used as either a trunk or station interface)

BRI services require the installation of BRI Interface PCBs.

ISDN S-Point
The ISDN BRI circuit can be set to S-point in Cards. It is possible to select the mode of each BRI circuit independently.
In this mode the circuit will present as extension ports on the system.
The following features are available to a device connected to an ISDN S-point:
• Outgoing Feature
Intercom call
Outgoing call dial 9
Outgoing call TRG access dial 804
Outgoing call Specific trunk access dial 805
Group hunt call dial pilot number
Intercom call voice/signal switch dial 812 or 1
Abb dial common dial 813
Abb dial group dial 814
LCR (standard feature on trunk digits)
F-Route (standard feature on dialled digits)
• Incoming Feature
IRG (member of)
DUD (destination = S0 extn number)
DISA (destination = S0 extn number)
DDI (destination = S0 extn number)
Group hunt call (if department group contains ONLY S0 devices). A group that contains a mix of telephones and S0 devices will only ring the telephones, the S0 devices are ignored.
Call pickup - Specified extension dial 856
Call pickup - Group dial 867
Call pickup - specified group dial 868
Call pickup - other group dial 869
Call pickup - Direct extension dial 715
• Hold/Transfer
Hold (S0 device can be held by a telephone and can place a call on hold
Transfer by on hook
Park hold pickup dial 861 (S0 device can not place a call onto park hold)
Group hold pickup dial 862 (S0 device can not place a call onto group hold)
Call forward - dual ring dial 842 to an S0 device (S0 device can not set call forward)
Call forward - busy dial 843 to an S0 device (S0 device can not set call forward)
Call forward - no answer dial 845 to an S0 device (S0 device can not set call forward)
Call forward - all dial 848 to an S0 device (S0 device can not set call forward)
• Paging/Doorphone
Internal paging dial 801
External paging dial 803
Combine page dial 751
Meet me paging answer - same page group dial 863
Meet me paging answer - other page group dial 864
Meet me paging answer - external page dial 865
Door phone call dial 802 (S0 device will not ring as part of the doorphone ring assignment)
• Other features
Camp on to extension dial 850 (a telephone can not camp on to an S0 device)
Camp on to trunk dial 850

Conditions
None.

Default Setting
No ISDN BRI cards are installed.

Related Features
Direct Inward Dialling (DID) / Direct Dial In (DDI).
Central Office Calls- Answering

ISDN Emergency Override - Forced Disconnect

Description

ISDN Emergency Override - Forced Disconnect gives an extension the ability to dial a defined emergency services number even if all the ISDN CO lines are busy.
The system will first check the availability of a free CO line and in the event of all channels being busy it will perform a forced disconnect of a channel and use that channel to dial the number.

Configuration
Enter the number, including trunk access code, in ISDN Emergency override - forced disconnect.
E.g. if the emergency number is 999 and the trunk access code is 9 enter 9999 or if the emergency number is 112 and the trunk access code is 9 enter 9112.

Last Number Redial

Description
Last Number Redial allows an extension user to quickly redial the last number dialed. For example, a user may quickly recall a busy or unanswered number without manually dialing the digits.
Last Number Redial saves in system memory the last 24 digits a user dials. The number can be any combination of digits 0-9, # and *. The system remembers the digits regardless of whether the call was answered, unanswered or busy. The system normally uses the same trunk group as for the initial call. However, the extension user can preselect a specific trunk if desired.

Redial List
The system allows display telephones to have a Redial List. Up to 10 dialed numbers (both external and internal destinations) are automatically stored in the Redial List. The user can display and select one of the stored numbers and then redial it. If more than 10 destination numbers are dialed, the oldest number is automatically erased to make room for the new number.

Conditions
Redial List requires the use of a display telephone. Non-display and single line phones can not use this feature.

Default Setting
'Call List' saves trunk calls in Keyphone Options.

Related Features
Automatic Route Selection: When using Automatic Route Selection, ARS selects the trunk for the call unless the user preselects.
Repeat Redial: The system can periodically redial an unanswered trunk call.
Save: A user can save the number of an outgoing call to be accessed at a later time.

Operation
To redial your last call:
1. Without lifting the handset, press LND.
The last dialed number is displayed.
2. To redial the last number, press #.
OR
Search for the desired number from the Redial List by pressing the LND or VOLUME keys.
3. Lift the handset or press SPK to place the call.
The system automatically selects a trunk from the same group as your original call and dials the last number dialed.
OR
1. At system phone, press idle line key (optional).
The system automatically selects a trunk from the same group as your original call.
2. Press LND.
OR
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 816.
The system automatically selects a trunk from the same group as your original call and dials the last number dialed.

To check the number saved for Last Number Redial:
1. Press LND.
The stored number displays for six seconds. The stored number dials out if you:
- Lift the handset,
- Press an idle line key,
- Press an idle CALL key, or

- Press SPK
2. Press CLEAR.

To erase the stored number:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 876.

Least cost routing

Description
Least cost routing allows the system to automatically select the indirect carrier defined by routing tables within the system.
An indirect carrier is accessed via the PSTN lines connected to the system (these are the direct carrier lines), a special access code is used to select the indirect carrier, all dialled digits are passed to the indirect carrier for routing of the call to the destination.
The routing tables list the leading digits of numbers dialled by the users and the associated indirect carrier access code. It is possible to route calls to more than one indirect carrier.

Conditions
The PSTN numbers accepted by the indirect carrier may vary, consult the carrier for details.
Special attention must be given to Emergency calls (Police/Fire/Ambulance etc.), if you route emergency calls to an indirect carrier you must confirm that they will accept this type of call.
It is normal practice to have an 'override code' that the users can dial to route the call to a chosen carrier (direct or indirect) in the event of faults with the carrier.
F-Route/ARS operation takes place on the digits dialled by the user, before the trunk is seized. LCR will use the digits sent to line i.e after any translation by F-Route/ARS.
Toll Restriction takes place on the digits dialled by the user. Toll restriction check will take place after any F-Route/ARS operation and before the LCR operation.
Local area calls can not be routed via an indirect carrier that also has Cost Centre Codes enabled. This is due to the order that the digits are dialled out by the system.
Local calls can be routed via an indirect carrier if Cost Centre Codes are not required.

Default Setting
No trunks are set to use LCR in LCR Dial Data.

Related Features
Cost Centre Codes: Send a cost center code for outgoing calls.

Operation
Change to DTMF Operation
In LCR Dial Options the 'LCR Code' can have the @ symbol, this will have different operation for Analogue trunks or ISDN trunks.
Analogue Trunks:
At the point in the dialled digits where the @ appears the system will revert to DTMF dialling. This is only required when the analogue trunk is set the Loop Disconnect dialling in Analogue Trunk Data Setup>.
ISDN Trunks:
At the point in the dialled digits where the @ appears the system will stop dialling and wait for the CONNECT from the indirect carrier. The system will then continue to dial DTMF in the B-channel.
The DTMF digits will be received by the indirect carrier for routing.
This is only required for indirect carriers that have a two stage setup process where the Access code is dialled in the D-channel to the direct carrier and all other digits are dialled as DTMF in the B-channel to the indirect carrier.

Order of LCR Routing Digits
When a user dials a number that is routed by LCR the individual LCR elements will be dialled to line in the order shown below.
Start [*Access Code*] then [*Authorisation Code*] then [*CCC*] then [*Delete leading digits*] then [*Dialled digits*] end.

The system will process the Least Cost Routing operation as shown below:
1. Check the setting of 'LCR On/Off' in LCR Dial Options.
There are three options:- LCR Off, LCR On or Cost Centre Code only.
LCR Off
The digits dialled by the user are sent directly to line, LCR process is complete.
LCR ON
The digits will be compared to the 'Manual Exemption Table' in LCR Options.
Cost Centre Code only

The system will insert the Cost Centre Code assigned to the extension in Cost Centre Codes and then dial the digits directly to line, LCR process is complete.

2. Manual Exemption Table
This table contains emergency numbers.
If the dialled digits correspond to an entry the system will dial the digits directly to line, LCR is complete.
If the dialled digits do not correspond the system will compare the digits to the 'Dial Data' table in LCR Dial Data.

3. LCR Dial Data Table
This table contains the area codes that should be routed via an indirect carrier and any override codes that a user can dial to manually select the indirect carrier.
Enter the leading digits of the area code and specify the Carrier Table number that will be used to specify the LCR digit translations.
If the dialled digits do not correspond to an entry the system will dial the digits directly to line ONLY IF the first digit is a 0. For all other digits the system will use the settings of Carrier Table 01.

4. Carrier Table
This table specifies the digit translation required to route the calls via the indirect carrier.
Specify the quantity of leading digits to be deleted, the LCR Access Code for the indirect carrier, The option Authorisation Code table number and the optional Cost Centre Codes.
Note that Carrier Table 01 is the default table used for numbers that do not correspond to an entry in the LCR Dial Data Table in step 3.

5. Authorisation Code Table
Specify the Authorisation Code that may be required by the indirect carrier to identify the customer for billing purposes.

Line Preference

Description
Line Preference determines how a system phone user places and answers calls. There are two types of Line Preference: Incoming Line Preference and Outgoing Line Preference.

Incoming Line Preference
Incoming Line Preference establishes how a system phone user answers calls. When a call rings the system phone, lifting the handset answers either the ringing call (for Ringing Line Preference) or seizes an idle line (for Idle Line Preference). The idle line can provide either Intercom or trunk dial tone (see Outgoing Line Preference below). Ringing Line Preference helps users whose primary function is to answer calls (such as a receptionist). Idle Line Preference is an aid to users whose primary function is to place calls (such as a telemarketer).

Outgoing Line Preference
Outgoing Line Preference sets how a system phone user places calls. If a system phone has Outgoing Intercom Line Preference, the user hears Intercom dial tone when they lift the handset. If a system phone has Outgoing Trunk Line Preference, the user hears trunk dial tone when they lift the handset. Outgoing Line Preference also determines what happens at extensions with Idle Line Preference. The user hears either trunk (“dial 9”) or Intercom dial tone.

Auto-Answer of Non-Ringing Lines
With Auto-Answer of Non-Ringing Lines, an extension user can automatically answer trunk calls that ring other extensions (not their own). This would help a user that has to answer calls for co-workers that are away from their desks. When the user lifts the handset, they automatically answer the ringing calls based on Trunk Group Routing programming. The extension user’s own ringing calls, however, always have priority over calls ringing other co-worker’s extensions.

Conditions
If a system phone extension has more than one call ringing its line keys, Ringing Line Preference answers the calls on a first-in first-answered basis.

Default Setting
Automatic answer for trunk and intercom calls is on in Keyphone Options. This option is used for both Incoming and Outgoing Line Preference for the keyphone.
'Priority for Incoming Call' is set to trunk in Incoming Call.
'Auto Trunk Seize' is off in Extension Basic Setup.

Operation

If Incoming or Outgoing Line Preference is set:
1. Lift the handset or press SPK to answer a ringing call (or hear dial tone if no calls are ringing).

If Incoming or Outgoing Line Preference is not set:
1. Lifting the handset will not answer a ringing call, you must press the line key or CALL key to answer the call (or hear Intercom/Trunk dial tone).

Line Reversal

Description
Line reversal is supplied by the network provider to indicate when the called party has answered. It is therefore only available for outgoing calls made from the system.
Line reversal is only available for analogue trunks connected to CO ports.
When the line reversal is detected by the system the call timer (at the system phone with a display) will start.
The call duration of the SMDR will also start when the line reversal is detected, this gives an accurate indication of call durations.
Without line reversal the call timer and SMDR call duration will start after the 'External Call Inter-digit time' in Timers.

Conditions
Line reversal must be supplied by the network provider.

Default Setting
Polarity reverse and timeout is selected in Analogue Trunk Data Setup.
'External Call Inter-digit time' is set to 10 seconds in Timers.

Long Conversation Cut-Off

Description
For incoming and outgoing central office calls, each trunk can be programmed to disconnect after a defined length of time. The timer begins when the trunk is seized and disconnects the call after the timer expires.
When used with the Warning Tone for Long Conversation feature, the system can provide a warning tone on outgoing trunks calls before the call is disconnected. This tone is not available to analogue single line telephone users nor is it available for incoming calls.

Conditions
None

Default Setting
'Long Conversation Cut-Off' is disabled in Trunk Basic Data Setup.
'Long Conversation Alarm before Cut-Off' is disabled in Trunk Basic Data Setup.
'Incoming Long Conversation Cut-Off' is off in Class of Service.
'Outgoing Long Conversation Cut-Off' is off in Class of Service.
'Incoming Long Conversation ' is on in Class of Service.
'Long Conversation Alarm 1 (Until sending 1st Alarm)' is set to 170 seconds in Timers
'Long Conversation Alarm 2 (Until sending next Alarm)' is set to 180 seconds in Timers
'Long Conversation Cut-off for Incoming' is set to 0 seconds in Timers
'Long Conversation Cut-off for Outgoing' is set to 0 seconds in Timers

Related Features
Central Office Calls, Answering/Central Office Calls, Placing: Long Conversation Cutoff can disconnect incoming and outgoing CO calls after a set time period.
Direct Inward System Access (DISA): Long conversation cutoff is controlled separately for DISA.
Warning Tone for Long Conversation: Using the Warning Tone for Long Conversation feature allows users on outgoing calls to hear a warning tone prior to the call disconnecting.

Operation
Feature is automatic once programmed.

Loop keys

Description
Loop keys are uniquely programmed function keys that simplify placing and answering trunk calls.
There are three types of loop keys: Incoming Only, Outgoing Only and Both Ways.
• Incoming Only Loop Keys
Incoming Only loop keys are for answering trunk calls. An extension can have an incoming loop key for a specific trunk group (fixed) or a “catch all” loop key for any trunk group (switched). Fixed loop keys allow an extension user to tell the type of call by the ringing key. Switched loop keys are ideal for an extension with a large number of feature keys. In addition, switched loop keys are a destination for any trunk not on a line key or fixed loop key. Without a switched loop key, calls not appearing on a line key or fixed loop key will ring only the CALL key. Incoming Only loop keys also receive Transferred trunk calls.

• Outgoing Only Loop Keys
Outgoing Only loop keys are for placing trunk calls. An extension can have outgoing loop keys for a specific trunk group or for F-Route access. When a user presses the loop key, they get dial tone from the first available trunk in the group (or from F-Route if programmed). Outgoing Only loop keys help ensure that an extension will always have a key available for placing calls.

• Both Ways Loop Keys
Both Ways loop keys combine the functions of both Incoming Only and Outgoing Only loop keys. Both Ways loop keys work well for extension users that handle a moderate amount of calls and don’t separate keys for incoming and outgoing calls. Both Ways loop keys also receive Transferred trunk calls.

An extension can have many loop keys - of any type. You can program an operator, for example, with four loop keys for incoming calls and four for outgoing calls.
Once a loop key call is set up, the user can handle it like any other trunk call. For example, the user can place the call on Hold, Transfer it to a co-worker or send it to a Park Orbit.
An incoming call will ring the first available loop key, beginning with the lowest numbered key. If keys 1-3 are loop keys, for example, the first incoming call rings key 1. If key 1 is busy, the next call rings key 2. If keys 1 and 2 are busy, the next call rings key 3. If all three keys are busy, additional incoming calls queue for the first available key. The telephone display will show “WAITING - LOOP KEY” if the user presses a loop key when there are additional calls waiting.

Conditions
None

Default Setting
No Loop keys are assigned in Function Key Programming.
No trunk group is specified for the loop keys in Loop Keys.

Related Features
F-Route / Central Office Calls, Answering / Central Office Calls, Placing: Program incoming and outgoing access and routing options.
Off Hook Signaling: If enabled, a user hears Call Waiting beeps if additional calls are waiting behind a loop key.
Programmable Function Keys: If you have a line and loop key for the same trunk, the line key has precedence. An incoming call rings the line key, not the loop key. When you press the loop key for an outgoing call, the line key lights.
Ring Groups: Trunks ring telephones according to their Ring Group assignments.
Direct Inward Dialing (DID) / Direct Inward Line (DIL) / Direct Inward System Access (DISA) / Tie Lines: Transferred DID, DIL, DISA calls do not require ring group programming.

Operation
To place a call on a loop key:
1. Press outgoing or both ways loop key.
You hear dial tone and the key lights green.
2. Dial number.
To answer a call on a loop key:
Listen for ringing a look for a flashing loop key.
1. Press loop key.
The key lights green and you connect to the call.

If there are additional calls waiting to be answered, your display shows:
WAITING - LOOP KEY
To program a loop key:
1. Press the SPK key.
2. Dial 852.
3. Press the key you want to program as a loop key.

4. Dial *05.
5. Dial the loop key type:
• 0 = Incoming only
• 1 = Outgoing only
• 2 = Both ways (incoming and outgoing)
6. Dial the loop key routing option for incoming, outgoing, or incoming and outgoing calls:
• 000 = Trunk Group Routing or F-Route
• 01-25 = Trunk Groups
If you selected option 2 in step 5 above, enter the incoming Trunk Group followed by the outgoing Trunk Group.
7. Press SPK to hang up.

Meet Me Conference

Description
With Meet Me Conference, an extension user can set up a Conference with their current call and up to 32 other internal or external parties. Each party joins the Conference by dialing a Meet Me Conference code. Meet Me Conference lets extension users have a telephone meeting without leaving the office.

Conditions
None.

Default Setting
'Meet Me Answer' is on in Class of Service.
'Privacy Release Time' is set to 90 seconds in Paging Options.

Related Features
Conference: An extension user can also use other types of Conferences to join callers together.
Meet Me Paging: An extension user can have a telephone meeting with a co-worker on a Page zone.
Programmable Function Keys: Meet Me Conference requires a Conference key. In addition, Internal and External Paging keys simplify Meet Me Conference operation.

Operation
Meet Me External Conference
To make a Meet Me External Conference:
System Phone
1. While on a call, press Conference key (SC 851: 07).
2. Dial 803 and the External Paging Zone code (1-6 or 0 for All Call)
OR
Dial 751 and the Combined Paging Zone code 1-6 (for Internal/External Zones 1-6) or 0 (for Internal/External All Call).
OR
Press Page key (SC 851: 19 + zone & 20).
3. Announce the zone.
4. When co-worker answers your page, press the Conference key twice.
5. Repeat steps 1-4 for each co-worker you want to add.

Single Line Telephone
1. Single Line Telephone:
While on a call, hook flash and dial 826.
2. Dial 803 and the External Paging zone code (1-6 or 0 for All Call).
OR
Dial 751 and the Combined Paging Zone code 1-6 (for Internal/External Zones 1-6) or 0 (for Internal/External All Call).
3. Announce the zone.
4. When co-worker answers your page:
Single Line Telephone:
Press the hook flash twice.
5. Repeat steps 1-4 for each co-worker you want to add.

To join a Meet Me External Conference:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 865.

3. Dial the announced External Paging Zone code (0-6).
You connect to the other parties.

Meet Me Internal Conference:
To make a Meet Me Internal Conference:
System Phone
1. While on a call, press Conference key (SC 851: 07).
2. Dial 801 and the Internal Paging Zone code (0-9 or 00-32).
OR
Dial 751 and the Combined Paging Zone code 1-6 (for Internal/External Zones 1-6) or 0 (for Internal/External All Call).
3. Announce the zone.
4. When co-worker answers your page, press the Conference key twice.
5. Repeat steps 1-4 for each co-worker you want to add.

Single Line Telephone
1. Single Line Telephone:
While on a call, hook flash and dial 826.
2. Dial 801 and the Internal Paging Zone code (0-9 or 00-32).
OR
Dial 751 and the Combined Paging Zone code 1-6 (for Internal/External Zones 1-6) or 0 (for Internal/External All Call).
3. Announce the zone.
4. When co-worker answers your page:
Single Line Telephone:
Press the hook flash twice.
5. Repeat steps 1-4 for each co-worker you want to add.

To join a Meet Me Internal Conference:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 863 (if your extension is in the zone called).
OR
Dial 864 and the zone number (if your extension is not in the zone called).
OR
Press the Meet Me Conference/Paging Pickup key (SC 851: 23) if your extension is in the zone called.

Meet Me Paging Transfer

Description
If a user wants to Transfer a call to a co-worker but they don't know where the co-worker is, they can use Meet Me Paging Transfer. With Meet Me Paging Transfer, the user can Page the co-worker and have the call automatically Transfer when the co-worker answers the Page. Since Meet Me Paging Transfer works with both Internal and External Paging, a call can be quickly extended to a co-worker anywhere in the facility.

Conditions
External paging requires a 2PGDU card be installed in the system if more than one zone is required. There is one zone available on the Main Unit.

Default Setting
'Meet Me Answer' is on in Class of Service.

Related Features
Meet Me Conference: An extension user can set up a Conference with their current call and up to 31 other inside parties.
Meet Me Paging: An extension user can Page a co-worker and meet with them on a Page zone.
Paging, External: With External Paging, an extension user can broadcast an announcement over Paging equipment connected to external Paging zones.
Paging, Internal: Internal Paging lets extension users broadcast announcements to other keyphones.
Programmable Function Keys: Function keys simplify Meet Me Paging Transfer operation.

Operation
Meet Me External Paging Transfer
To make a Meet Me External Paging Transfer:

1. At system phone, while on a call, press HOLD.
OR
At single line telephone, while on a call, hook flash.
2. Press the External Paging Zone key (SC 851: 19 + zone & 20).
OR
Dial 803 and the External Paging Zone code (1-6 or 0 for All Call).
OR
Dial 751 and the Combined Paging Zone code 1-6 (for Internal/External Zones 1-6) or 0 (for Internal/External All Call).
3. Announce the call.
4. When Paged party answers, hang up to Transfer the call to them.

To join a Meet Me External Paging Transfer:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 865.
3. Dial the announced External Paging Zone (0-6).
You connect to the Paging party.
4. Stay on the line.
After the Paging party hangs up, you connect to the transferred call.

To make a Meet Me Internal Paging Transfer:
1. At system phone, while on a call, press HOLD.
OR
At single line telephone, while on a call, hook flash.
2. Press Internal Paging Zone key (SC 851: 20 + zone).
OR
Dial 801 and the Internal Paging Zone code (0-9 or 00-32).
OR
Dial 751 and the Combined Paging Zone code 1-6 (for Internal/External Zones 1-6) or 0 (for Internal/External All Call).
3. Announce the call.
4. When Paged party answers, hang up to Transfer the call to them.
The answering party connects to the trunk call when you hang up.

To join a Meet Me Internal Paging Transfer:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 863 (if your extension is in the zone called).
OR
Dial 864 and the zone number (if your extension is not in the zone called).
OR
Press the Meet Me Conference/Paging Pickup key (SC 851: 23) if your extension is in the zone called.
3. Stay on the line.
After the Paging party hangs up, you connect to the transferred call.

Meet Me Paging

Description
Meet Me Paging allows an extension user to Page a co-worker and privately meet with them on a Page zone. The Paging zone is busy to other users while the meeting takes place. While the co-workers meet on the zone, no one else can hear the conversation, join in or make an announcement using that zone. Meet Me Paging is a good way to talk to a co-worker when their location is unknown. If the co-worker can hear the Page, they can join in the conversation.

Conditions
External paging requires a 2PGDU be installed in the system if more than one zone is required. There is one zone available on the NTCPU.

Default Setting
'Meet Me Answer' is on in Class of Service.

Related Features
Meet Me Conference: An extension user can set up a Conference with their current call and up to 31 other inside parties.

Meet Me Paging Transfer: With Meet Me Paging Transfer, a user can page a co-worker and have the call automatically transfer when the co-worker answers the page.
Paging, Internal / Paging, External: An extension's access to internal and external page zones affects the Meet Me Paging feature.
Programmable Function Keys: Internal and External Paging keys simplify Meet Me Paging operation.

Operation
Meet Me External Page
To make a Meet Me External Page:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 803 and the External Paging Zone code (1-8 or 0 for All Call).
OR
Dial 751 and the Combined Paging Zone code 1-8 (for Internal/External Zones 1-8) or 0 (for Internal/External All Call).
3. Announce the zone.
OR
1. At system phone, press the External Paging Zone key (SC 851: 19 + zone & 20).
2. Announce the zone.

To join a Meet Me External Page:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 865.
3. Dial the announced External Paging Zone (0-8).
You connect to the other party.

Meet Me Internal Page
To make a Meet Me Internal Page:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 801 and dial the Internal Paging Zone code (0-9, 00-32 or 00-64).
OR
Dial 751 and the Combined Paging Zone code 1-8 (for Internal/External Zones 1-8) or 0 (for Internal/External All Call).
3. Announce the zone.
OR
1. At system phone, press the External Paging Zone key (SC 851: 19 + zone & 20).
2. Announce the zone.

To join a Meet Me Internal Page:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 863 (if your extension is in the zone called).
OR
Dial 864 and the zone number (if your extension is not in the zone called).
OR
Press the Meet Me Conference/Paging Pickup key (SC 851: 23) if your extension is in the zone called.

Memo Dial

Description
While on an outside call, Memo Dial lets a display system phone user store an important number for easy re-dialing later on. The telephone can be like a note pad. For example, a user could dial Directory Assistance and ask for a client's telephone number. When Directory Assistance plays back the requested number, the caller can use Memo Dial to jot the number down in the telephone's memory. They can quickly call the Memo Dial number after hanging up.
When a user enters a Memo Dial number, the dialed digits do not output over the trunk. Dialing Memo Dial digits does not interfere with a call in progress.

Conditions
When Memo Dial calls out, it out dials the entire stored number. Memo Dial does not automatically strip out trunk or PBX access codes if entered as part of the stored number.

Default Setting
No Memo Dial keys are assigned in Function Key Programming

Related Features
Central Office Calls, Placing: A user's outgoing dialing options affect how a Memo Dial call is placed.
Last Number Redial: Quickly redial the last outside number dialed.
Save Number Dialed: Save the last outside number dialed.
Single Line Telephones: Memo Dial is not available at single line telephones.

Operation
To store a number while you are on a call:
1. While on a call, press Memo Dial key (SC 851: 31).
2. Dial number you want to store.
3. Press Memo Dial key again and continue with conversation.

To call a stored Memo Dial number:
1. Do not lift the handset.
2. Press Memo Dial key (SC 851: 31).
3. Press idle CALL key.
The stored number dials out only if you store a trunk access code before the number.
OR
Press line key.
The stored number dials out.

To check to see the stored Memo Dial number:
1. Do not lift handset.
2. Press Memo Dial key (SC 851: 31).
The stored number displays.

To cancel (erase) a stored Memo Dial number:
1. Press idle CALL key.
2. Press Memo Dial key (SC 851: 31).

Message Waiting

Description
An extension user can leave a Message Waiting indication at a busy or unanswered extension requesting a return call. The indication is a flashing MW lamp at the called extension and a steadily lit MW lamp on the calling extension. Answering the Message Waiting automatically calls the extension which left the indication. Message Waiting ensures that a user will not have to recall an unanswered extension. It also ensures that a user will not miss calls when their extension is busy or unattended. Additionally, Message Waiting lets extension users:
• View and selectively answer messages left at their extension (display system phone only)
• Cancel all messages left at their extension
• Cancel messages they left at other extensions
An extension user can leave Messages Waiting at any number of extensions. Also, any number of extensions can leave a Message Waiting at the same extension. A periodic VRS announcement may remind users that they have Messages Waiting.

Message Waiting Available for Single Line Telephones
This feature provides message waiting indication to single line telephones without message waiting lamps.
1. Special dial tone

Note the following items for this feature:
• All of the message waiting indication methods may be active at any SLT.
• If the special dial tone is not set, then the user will hear regular dial tone. Note: The regular dial tone may be different based on the phone's status (call forward, DND, etc.).
• The Message Wait dial tone will be a system tone '3-Special Dial Tone' and can be changed in Service Tones.

Conditions
Message Waiting lamp at an SLT must be compatible with the 'voltage increase' method of the system.

Default Setting
The MSG key of the keyphone is set to 'Message Waiting' in Keyphone Options.
'Message Waiting' in on in Class of Service.

Related Features
Handsfree Answerback/Forced Intercom Ringing: When a user responds to a Message Waiting, the system does not cancel the Message Waiting indication if the called party uses Handsfree Answerback. The system cancels the indication only if the called party lifts the handset or presses SPK.
Hotel/Motel: With the Hotel/Motel set up, an employee with a system phone can send a Message Waiting to a room telephone if allowed in system programming.
Programmable Function Key: A Message Waiting key simplifies this feature's operation.
Single Line Telephones: The intermittent ringing or stutter dial tone options may be used to indicate voice mail messages for single line sets which do not have a Message Waiting lamp.
Telephone-to-telephone Message Waiting works when the voice mail is installed.
The MW LED may be used to indicate voice mail messages if there is no extension number assigned to the voice mail key in system programming.
The MSG key at system phone can also be set as the Voice Mail Message key.

Operation
To leave a Message Waiting:
1. Call busy or unanswered extension.
2. Press Message Waiting key (SC 851: 38)
3. Hang up.With system phone phones, the MW LED lights.

To answer a Message Waiting:
When you have a message, your MW LED flashes fast for keyphones.
1. At a system phone, press idle CALL key and dial 841.
OR
Press Message Waiting key (SC 851: 38).
OR
At single line telephones, lift the handset and dial 841.
If the called extension doesn't answer, dial 0 or press your Message Waiting key to automatically leave them a message.
Normally, your MW LED goes out. If it continues to flash, you have new messages in your "Voice Mail" mailbox or a new "General Message". Go to "To check your messages" below.

To cancel all your Messages Waiting:
This includes messages you have left for other extensions and messages other extension
have left for you.
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 873.
3. Hang up.

To cancel the Messages Waiting you have left at a specific extension:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 871.
3. Dial number of extension you don't want to have your messages.
4. Hang up.

To check your messages:
1. Press CHECK
2. Dial 841.
You can have any combination of the message types in the table below on your phone.
3. Press VOL up or VOL down to scroll through your display.
4. When you find the message you want to answer, press CALL. You'll either:
• Go to your Voice Mail mailbox.
• Listen to the new General Message.
• Automatically call the extension that left you a Message Waiting.

Single Line Telephones Without Message Waiting Lamps:
With special dial tone:
1. Lift handset.
The system provides the special Message Wait dial tone immediately.

Microphone Cutoff

Description
Microphone Cutoff lets a system phone user turn off their phone’s hands free or handset microphone at any time. When activated, Microphone Mute prevents the caller from hearing conversations in the user’s work area. The user may turn off the microphone while their telephone is idle, busy on a call or ringing. The microphone stays off until the user turns it back on.

Conditions
It is not possible to prevent an extension user from using the microphone cut off function.
Handset microphone cutoff requires a 'Handset Transmission Cutoff (40)' programmable function key.
Not supported at an SLT, the SLT would have provide this feature.

Default Setting
Enabled (using MIC key).

Related Features
Handsfree Answerback/Forced Intercom Ringing: Microphone Cutoff does not operate if the user calls another extension and the called extension responds without lifting the handset or pressing SPK.
With Microphone Cutoff enabled, Handsfree Answerback callers to an extension hear a single beep (instead of two).

Operation
To mute your telephone’s handset or Handsfree microphone while on a call:
1. Press MIC.
This only turns off the Handsfree microphone.
OR
Press Handset Microphone Cutoff key (SC 851: 40).
This turns off both the handset and Handsfree microphone.

To turn your telephone’s microphone back on:
1. Press MIC.
Use MIC only if you pressed it initially to turn off your Handsfree microphone.
OR
Press Microphone Cutoff key (SC 851: 40).
Use the Microphone Cutoff key only if you pressed it initially to turn off your handset or Handsfree microphone.

Mobile Extension

Description
A mobile extension is an external telephone (preferably a mobile phone) linked to the PBX, via a Proxy Port, in order to operate
with the featureset of an internal SLT extension.
Note: - a mobile extension cannot be used as a Voice Mail port.
The Mobile extension's internal extension number (Proxy Port) is linked to an abbreviated dial location to provide integration.
Note: if all external trunks are busy when a call is made to the mobile extension, ringback tone is presented giving the impression the phone is ringing.
A DDI is directed to the Mobile Extension's internal extension number (Proxy Port) and in order to provide internal dial tone to the Mobile Extension the incoming CLI of the Mobile Extension must match the number in the abbreviated dial location. On presentation of internal dial tone, operation is similar to an SLT lifting its handset.

In the absence of DDI's, DUD (VRS) can be used to transfer the Mobile Extension call to the Mobile Extension extension number, this will provide internal dial tone providing the CLI is presented and matches the number in the associated abbreviated dial location.
Alternatively, if CLI routing is enabled the relevant abbreviated dial location could be 'transferred' to the Mobile

Extension proxy port which would then provide internal dial tone.

The number of Mobile Extensions per system is limited by the following rules: -

Number of Mobile extension ports = 25% of physical ports (8 port extension card allows 2 Mobile extension entries).

In the event of the limit being reached, the Mobile Extensions will be able to be added in programming but will give the indication of a invalid dial entry when called.

The features available from a Mobile Extension are listed below, as the Mobile Extension is based on an SLT port the service codes used are as per an SLT port.
Any feature not listed should be assumed to be not supported: -

Hold and transfer
Incoming Ring Group membership
Department group membership
DDI
Toll restriction
Class of Service
DSS keys are available for the Mobile Extension but, obviously, cannot give an exact indication of busy status if, for example, the Mobile Extension is active on a call not linked to the PBX.

The following service codes are supported

Feature Code	Set By Mobile Extension	Set to Mobile Extension
(848)- Call Forward - Immediate	Yes	Yes
(843)- Call Forward - Busy	Yes	Yes
(845)- Call Forward - No Answer	Yes	Yes
(844)- Call Forward - Busy/No Answer	Yes	Yes
(842)- Call Forward - Both Ring	Yes	Yes
(888)- Call Forward - (United Method) - Answer Machine Emulation not available	Yes	-
(846)- Disable/Enable Follow-me	Yes	Yes
(847)- DND	Yes	-
(841)- Message Waiting	Yes	-
(873)- Cancel All Messages Waiting	Yes	-
(790)- Automated Attendant (DSPDB)	Yes	-
(807)- DND/FWD Override Call (Bypass Call)	Yes	Yes
(826)- Conference	Yes	Yes
(809)- Override Off-hook Signalling	Yes	-
(850)- Enable Camp-on	Yes	Yes
(870)- Disable Camp-on	Yes	Yes
(808)- Step Call	Yes	Yes
(813)- Common/Station Speed Dialling	Yes	-
(814)- Group Speed Dialling	Yes	-
(804)- Trunk Group Access	Yes	-
(805)- Specified Trunk Access	Yes	-
(866)- Trunk Access via Networking	Yes	-
(720)- Common Cancel Service Code	Yes	-
(884)- Voice Mail Centre Access	Yes	-
(891)- Account Code	Yes	-
(717)- Call own Mailbox (In-skin VM)	Yes	-
(855)- One Touch Dial Number Entry (dial *0 to hang up)	Yes	-
(812)- Voice Call & Signal Call Switching	Yes	-
(801)- Internal Group Paging (Mobile Extension	Yes	-

cannot be a member of a paging group)		
(803)- External Paging	Yes	-
(864)- Meet-me Answer to specified Internal Paging Group	Yes	-
(865)- Meet-me Answer to External Paging	Yes	-
(863)- Meet-me Answer in same Paging Group (although Mobile extension cannot be paged)	Yes	Yes
(751)- Combine Paging	Yes	-
(831)- Park Hold	Yes	-
(861)- Answer Park Hold	Yes	-
(832)- Group Hold	Yes	-
(862)- Answer Group Hold	Yes	-
(773)- Station Park Hold	Yes	-
(802)- Doorphone Access (Doorphone can also ring ME. *# operates relay)	Yes	-
(849)- Enable SLT On-hook when Holding	Yes	-
(859)- Answer SLT On-hook when Holding	Yes	-
(894)- Call Waiting Answer/Split Answer for SLT	Yes	-
(882)- ANI/DNIS Routing to VAU	Yes	-
(753)- Tandem Trunking	Yes	-
( )- Transfer into Conference	Yes	-
(718)- Night Mode Switching (other group)	Yes	-
(702)- Automatic Transfer Setup per Extension Group	Yes	-
(703)- Automatic Transfer Cancellation per Extension Group	Yes	-
(705)- Delayed Transfer per Extension Group	Yes	-
(706)- Delayed Transfer Cancellation per Extension Group	Yes	-
(707)- DND Setup per Extension Group	Yes	-
(708)- DND Cancellation per Extension Group	Yes	-
(750)- Pilot Group Withdrawing<	Yes	-
(713)- VAU/Off-premise Call Forwarding (common cancelling code (720) required to cancel)	Yes	-
(810)- Barge-in	Yes	Yes
(780)- Enable Extension Group to all ring	Yes	-
(883)- General Purpose Indication	Yes	-
(761)- Personal Speed Dial	Yes	-
(890)- Voice Over	Yes	-
(806)- Flash on Trunk lines	Yes	-
(880)- General Purpose Relay	Yes	-
(754)- SLT Live Recording	Yes	-
(729)- Enable DND for other extensions	Yes	Yes
(730)- Disable DND for other extensions	Yes	Yes
(733)- Enable Wake-up Call for other extensions	Yes	Yes
(734)- Disable Wake-up Call for other extensions	Yes	Yes
(735)- Enable Room to Room Call Restriction	Yes	Yes
(736)- Disable Room to Room Call Restriction (Hotel)	Yes	Yes
(737)- Set Toll Restriction Class for other extensions	Yes	Yes
(738)- Check-in	Yes	Yes

(739)- Check-out	Yes	Yes
(741)- Set Room Status for other extensions	Yes	Yes
(818)- Day/Night Mode Switching	Yes	-
(856)- Direct Call Pickup own group	Yes	Yes
(868)- Call Pickup for specified group	Yes	Yes
(867)- Call Pickup	Yes	Yes
(869)- Call Pickup for another group	Yes	Yes
(715)- Direct Extension Call Pickup	Yes	Yes
(731)- Enable Wake-up Call for own extension	Yes	-
(732)- Disable Wake-up Call for own extension	Yes	-
(740)- Set Room Status for own extension	Yes	-
(742)- Room Status Output	Yes	-
(770)- Hotel Room Monitor	Yes	Yes

Although some features may be available to the Mobile Extension it may be advisable to disable them in Class of service. There are also features that should be disabled in any case.

The features to be disabled for Mobile Extension include: -
ACD
TAPI
SIP/H323 Trunks
Port Swap
Hotline
General Message
Message Waiting
Headset Mode for SLT
Virtual Loopback/Flexible Transfer
Tandem Ringing
Virtual extension key as call coverage key for mobile extension
Automatic conversation record for trunks

Default setting
No Mobile Extensions are configured.

Related Features
Abbreviated dial
DDI
Extension numbering
System Tones

Operation
The Mobile Extension port must be an unequipped extension port on the PBX.
This extension port is directed to an abdial location.
Since the CLI of the Mobile Extension must match the number in the abdial location, in order to ascertain it is the 'owner' of the ddi, it is recommended that Caller ID Edit Mode be enabled.

It is also necessary to adjust the DUD/DISA dial tone (tone 44)in Service Tones to be as follows: -

Repeat Count = 250								
Unit Number	1	2	3	4	5	6	7	8

Basic tone	1	2	1	2	1	2	1	2
Duration	2	1	2	1	2	1	2	1
Level	26	26	26	26	26	26	26	26

As the Mobile Extension can be a GSM phone it is necessary to be certain a person and not e.g. a GSM voicemail has answered the call, this is achieved by returning Music on Hold tone to the Mobile Extension on answer after which the Mobile Extension presses * to connect the call.
This setting is enabled/disabled by setiing the connection method to either Always, Analogue or none.
Always = answer confimation by * always necessary.
Analogue = answer confirmation by * only required when outgoing call to Mobile Extension made on alalogue CO lines.
None = answer confirmation by * not required.

Whilst in conversation a hook flash is returned by dialling *# from the Mobile Extension.
Hang up followed by internal dial tone is achieved by dialling *0.

*Note: - Analogue lines can be used for integration with the Mobile Extension using either DIL's or VRS Auto Attendant to access the Mobile Extension Proxy Port, however, it must be noted that the *0 Hang Up code must be used prior to terminating any call e.g. transfer, hang up etc.*

SMDR Integration
The following basic call scenarios explain the integration of the Mobile Extension with the SMDR output.
Note: - Mobile Extension has not been verified with MyCalls

EXT 400 = Mobile Extension
07979378016 = Mobile Extension external number

Mobile Extension to Internal Extension
01 IVIN 05:54 003 00:00:13 EXT 400 07979378016 0:00

Internal Extension to Mobile Extension
02 IVOT 05:54 003 00:00:10 EXT 400 07979378016 0

Mobile Extension makes an outgoing trunk call
03 IVOT 05:56 001 00:00:08 EXT 400 643100 0
04 IVIN 05:55 003 00:00:44 EXT 400 07979378016 0:00

Internal Dual Ring to Mobile Extension answered on ME
05 IVOT 05:57 003 00:00:06 EXT 400 07979378016 0

Incoming DDI Call to Mobile Extension
08 IVOT 06:07 003 00:00:06 EXT 400 07979378016 0
09 IVIN 06:07 009 00:00:05 EXT 400 01509643100 0:17

Incoming DDI call to Mobile Extension answered on ME
10 IVOT 06:09 003 00:00:08 EXT 400 07979378016 0

11 IVIN 06:09 009 00:00:06 EXT 400 01509643100 0:15

Multiple Directory Numbers

Description
Multiple Directory Numbers let a system phone have more than one extension number. Calls can route to the system phone’s

installed number or to the system phone's "virtual extension" Multiple Directory Number key. This helps users identify incoming calls. For example, an extension installed at 304 (Sales) could have a virtual extension for 460 (Service). Calls to 304 ring the extension normally. Calls to 460 ring the Multiple Directory Number key. This lets the user at extension 304 differentiate Sales calls from Service calls.

Call Coverage

A system phone can have Multiple Directory Number keys set up as Call Coverage keys for co-worker's extensions. The Call Coverage key lights when the co-worker's extension is busy and flashes slowly when the co-worker has an incoming call. The Call Coverage key can ring immediately when a call comes into the covered extension, ring after a delay or not ring at all. In addition, the system phone user can press the Call Coverage key to intercept their co-worker's incoming call.
The user can also go off hook and press the Call Coverage key to call the covered extension.
If the covered extension is busy and they receive a second call, the covering extension's Call Coverage key will flash. The user just presses the flashing key to pick up the call.
The Call Coverage keys follow the extension's Do Not Disturb and Off-Hook Signaling programming.
These keys do not, however, indicate the lamp indication for extensions in DND. If this is required, a Hotline (DSS) key can be used instead.

Place and Receive Calls on Call Coverage/Multiple Directory Number Keys

Multiple Directory Number keys/Call Coverage keys can be used three separate ways, depending on how the key is defined in 'Virtual Mode' in Keyphone Options.
• a DSS key to the extension and for receiving incoming calls
• answering incoming calls with the ability to place outgoing ICM or CO calls
OR
• just for receiving incoming calls

A system phone can have Multiple Directory Number/Call Coverage keys for many different extensions and virtual extensions. In addition, co-workers can share the same Multiple Directory Numbers.
For example, everyone in the Service Department could have a key for the Sales Department's virtual extension.

Auto Off-Hook Answer and Ringing Line Preference for Call Coverage Keys

An extension's Call Coverage Keys can be programmed to allow the user to simply pick up the handset to answer a ringing call. So as not to interfere with ringing trunk or Intercom calls, the system automatically assigns Call Coverage Key ringing with the lowest answering priority. If multiple Call Coverage Keys are ringing, answering priority is set first by the assigned ring pattern and then by the key position.

Virtual Extension vs. Ring Groups

As the system does not allow voice mail calls to ring Ring Groups, a virtual extension can be created which will allow you to direct calls to more than one extension. When you program a Call Coverage Key for that extension to ring, the next call can then be answered.
This could allow a voice mail caller to dial the virtual extension and have all the extensions with the same virtual extension key ring.
The system can be programmed as follows:
•Virtual Extension Basic Setup: Assign a virtual extension number (example: extension 5400).
• Function Key Programming: Assign a Call Coverage key (*03) to an extension for the virtual extension number assigned.
The end user can then simply transfer the call to the virtual extension number (example: 5400).

To allow trunk calls to be queued at the virtual you must place the virtual within a Department Group and have the user transfer calls to the Department Group pilot number, the call is in placed in a queue and will be answered in turn as soon as the virtual extension is available.

Conditions

• More than one extension can share the same Multiple Directory Number.
• An extension can have more than one Multiple Directory Number (limited only by the number of available function keys).
• Calls can not be queued at a virtual extension, when it is ringing the virtual is busy. Place the virtual in a Department Group to make use of Department Group queueing.

Default Setting

No Virtual Extensions (Call Coverage keys) are assigned in Function Key Programming.
No Virtual extension numbers are assigned in Virtual Extension Basic Setup.
'Virtual Mode' is set to 'Ignore Key' in Keyphone Options
'Auto Off Hook Answer for Virtual Extension' is off in Class of Service.

Related Features

Class of Service: Class of Service options apply to Multiple Extension Appearances.
Department Calling: Multiple Extension Appearances can be in Department Calling Groups.
Do Not Disturb / Off-Hook Signaling: A Call Coverage Key follows DND and Off-Hook Signaling programming for an extension.
Group Call Pickup: Multiple Extension Appearances can be in Call Pickup Groups.
Line Preference: An extension user can answer an outside call on a Call Coverage Key just by lifting the handset.

Programmable Function Keys: This feature requires programmed function keys.

Operation
To answer a call ringing a Multiple Directory Number:
1. Press flashing Multiple Directory Number key (SC 852: *03 + ext.).

To place a call to a Multiple Directory Number (including a Call Coverage key):
1. Press idle CALL key.
2. Dial Multiple Directory Number number or press Multiple Directory Number key.

To place a call from a Multiple Directory Number (including a Call Coverage key):
1. Press the Multiple Directory Number key.
ICM dial tone is heard.
2. Place an intercom call or dial a trunk access code to seize an outside line and place your call.

To set up a Call Coverage Key:
1. Press idle CALL key or SPK key.
2. Dial 852.
3. Press the programmable key you want to program.
The previously programmed entry displays.
4. Dial *03.
5. Dial the number of the extension you want to cover.
6. Press HOLD once for Immediate Ring
To set for Delayed Ring, skip to Step 8.
7. Dial the Mode number(s) in which the key will be used.
1=Day 1
2=Night 1
3=Midnight 1
4=Rest 1
5=Day 2
6=Night 2
7=Midnight 2
8=Rest 2
8. Press HOLD to set up Delayed Ring
OR
Skip to Step 10.
9. Dial the Mode number(s) in which the key will be used.
1=Day 1
2=Night 1
3=Midnight 1
4=Rest 1
5=Day 2
6=Night 2
7=Midnight 2
8=Rest 2
10. Press SPK to hang up.

Music on Hold (MOH)

Description
Music on Hold (MOH) sends music to calls on Hold and parked calls. The music lets the caller know that their call is waiting, not forgotten. Without Music on Hold, the system provides silence to these types of calls. The Music on Hold source can be internal (synthesized) or from a customer-provided music source (i.e., tape deck, receiver, etc.). The customer-provided source can connect to a PGDU analogue port or to a connector on the Main Unit.

Note: In accordance with copyright law, a license may be required if radio, television broadcasts or music other than material not in the public domain are transmitted through the Music on Hold feature of telecommunications systems.

Music on Hold Source
There are 3 options available.
Internal Music Tune - The tune is set by Music On Hold Setup.
External Source - Via the Main Unit's EXMOH input or ACI input via a PGDU card.
Silence - Callers on hold hear silence.

Music on Hold per DDI Number
The music on hold source can be selected for individual DDI numbers in DDI Routing Table.
There are 3 options available:
0 - Use the music source set by Music On Hold Setup.
1 - Back Ground Music input.
2 - ACI input via a PGDU card.
The music source will be used for incoming DDI calls only.

Music on Hold for Internal calls
The music source is set by Music On Hold Setup.
It is not possible to have an input via a PGDU card for internal calls, the external input must be via the Main Unit (EXMOH input).

Music on Hold for non-DDI Trunk calls
The music on hold source is set per trunk port by Outgoing MOH Source.
There are 3 options available:
0 - Use the music source set by Music On Hold Setup.
1 - Back Ground Music input.
2 - ACI input via a PGDU card.
The music source will be used for outgoing trunk calls or incoming non-DDI calls only.

Options available for the Internal Music Tune
With system software V5.20 the internal music tune can be played from a file on the compact flash card installed on the DSPDB card.
This requires the optional DSPDB card is installed in the system.
The music source can be selected in Music On Hold Setup.
The music file must have the following properties:
- Type - WAV (see WAV File Attributes below)
- Format - CCITT A-Law
- Bit Rate - 64kbps
- Audio sample size - 8 bit
- Channels - 1 (mono)
- Audio sample rate - 8kHz
- Maximum duration - There is no limit other than available size on the compact flash card. The system will automatically loop the file.

The music files must be stored in the following folder on the compact flash card inserted in the DSPDB card - VM\OGM\3\1\9
The files must be named in the range G00.wav to G47.wav, these correspond to VRS message numbers 01 to 48 on the system.

WAV File Attributes
The system identifies the length of the WAV file by checking bytes 54 to 57 of the file, if these bytes do not contain the length data the system will play silence.
The file length data will be in the correct position if the WAV file has the 'fact' chunk included.

Conditions
MOH type per DDI or per trunk may be overridden by the MOH type for other features.
Department Group and Ring Group Queue Announcements specify the tone between messages, if this is set to MOH then it will override the setting of the DDI or trunk.
Single line telephones cannot change the Music on Hold tone.

Default Setting
'Changing of MOH tone' by Service Code 881 is disabled in Class of Service.
Internal music on hold tune 1 is set in Music On Hold Setup.
DDI calls use the system's MOH tone in DDI Routing Table.
Outgoing trunk calls us the system's MOH tone in Outgoing MOH Setup.

Related Features
Analogue Communications Interface (ACI): Analogue input of the 2PGDU card used for external music on hold sources.
2PGDU card: Refer to the Hardware Manual for installation of the 2PGDU card for external music sources.
Main Unit: Refer to the Hardware Manual for installation of external music sources to the Main Unit.

Operation
To change the Music on Hold tone:
1. Press idle CALL key.
2. Dial 881.
3. Dial Music on Hold tone code:
00 No Tone
01 Tune 1
02 Tune 2

4. Press SPK to hang up.
This service code setting will change the music on hold tone for any call that is set to use the system's MOH Tone as the music on hold source.
If the system's MOH tone is not set as the MOH source then the music on hold tone played to the held caller will not be changed by this service code.

Music On Hold Setup setting	Type of call placed on hold	MOH set by 881=00	MOH set by 881=01	MOH set by 881=02
Set to Internal	Internal call	Silence	Tune 1	Tune 2
	DDI call uses system MOH	Silence	Tune 1	Tune 2
	DDI call uses BGM input	BGM input	BGM input	BGM input
	DDI call uses ACI input	ACI input	ACI input	ACI input
External	Internal Call	Silence	EXMOH input of Main Unit	EXMOH input of Main Unit
	DDI Call uses system MOH	Silence	EXMOH input of Main Unit	EXMOH input of Main Ubit
	DDI call uses BGM input	BGM input	BGM input	BGM input
	DDI call uses ACI input	ACI input	ACI input	ACI input
System Tone 64	Internal Call	System Tone	System Tone	System Tone
	DDI Call uses system MOH	System Tone	System Tone	System Tone
	DDI call uses BGM input	BGM input	BGM input	BGM input
	DDI call uses ACI input	ACI input	ACI input	ACI input

Name Storing

Description
Extensions and trunks can have names instead of just circuit numbers. These names show on a system phone’s display when the user places or answers calls. Extension and trunk names make it easier to identify callers. The user does not have to refer to a directory when processing calls. A name can be up to 12 digits long, consisting of alphanumeric characters, punctuation marks and spaces.

Conditions
Display keyphone is required.

Default Setting
Trunks are named in Trunk Basic Data Setup.
Extensions are named in Extension Basic Setup.
'Caller ID Name Display' is on in Class of Service.
'Program Extension Name' is on in Class of Service.
No 'Extn Name Changed (55)' Programmable function keys are assigned.

Related Features
Directory Dialing.

Operation
To program an extension's name:
1. Press idle CALL key.
2. Dial 800
OR
Press Extension Name Change key (SC 851: 55).
3. Enter the extension number to be named.
4. Enter name (see below).
Your name can be up to 12 digits maximum.

keypad digit . . .	When you want to. . .
1	Enter characters: 1 @ [¥] ^ _ ' { | } <- ->
2	Enter characters: A B C a b c 2
3	Enter characters: D E F d e f 3
4	Enter characters: G H I g h i 4
5	Enter characters: J K L j k l 5
6	Enter characters: M N O m n o 6
7	Enter characters: P Q R S p q r s 7
8	Enter characters: T U V t u v 8
9	Enter characters: W X Y Z w x y z 9
0	Enter characters: 0 ! 2 # $ % & ' ( )
*	Enter characters: * + , - . / : ; < = > ?
#	Enter characters: Accept an entry when two characters are the same
CONF	Enter characters: Clear one character
FLASH	Enter characters: Clear all characters

5. Press HOLD.
6. Press SPK to hang up.

Night Service

Description
Night Service lets system users activate one of the Night Service modes. Night Service re directs calls to their night mode destination.
A user typically activates Night Service after normal working hours, when most employees are unavailable to answer calls.
The system also provides external contacts to enable Night Service.

There are eight Night Service modes, for example:
• Day 1 / Day 2 Modes - for normal working hours
• Night 1 / Night 2 Modes - after hours (usually evening)
• Midnight 1 / Midnight 2 Modes - late at night to early in the morning

• Rest 1 / Rest 2 Modes - interval usually used for lunch

Assigned Night Answer (ANA)
With Assigned Night Answer, Night Service has calls ring extensions directly. Assigned Night Answer provides an answering point for Night Service calls. For certain applications, this may be more appropriate than Universal Night Answer. For example, you could program trunks to ring the security station telephone during off hours.

Universal Night Answer (UNA)
Universal Night Answer makes incoming calls ring over the External Paging speakers. With UNA, an employee can go to a telephone and press the flashing line key or use “Universal Answer” to pick up the call. For more on setting up Universal Answer, turn to the “Central Office Calls, Answering” feature.
You may also be able to use Transfer to UNA. An extension user can Transfer their call to UNA (i.e., External Paging at night). Once transferred, the call will ring the External Paging speakers like any other UNA call and can be picked up at any extension.

Automatic Night Service
The system will allow or deny Automatic Night Service based on the extension’s class of service programming.
If allowed, the calls will then route according to the service patterns programmed.

Manual Night Service
Users can manually select the night mode either by a service code or function key.

Conditions
Almost all features are affected by Night Mode except for the following:
• Dial Tone Detection
• External Alarm Sensors
• Flexible System Numbering
• Pulse to Tone conversion
• SMDR
• Volume Control

Default Setting
Service code for 'Transfer to Incoming Ring Group' is not assigned in3 Digit Codes.
Manual night service is enabled on the system in Night Service Options.
Automatic night service is enabled on the system in Night Service Options.
Automatic Night is set to use Pattern 1 between 8:00 and 17:00 Monday to Friday and Pattern for all other times/days.
No Holiday patterns are set in Holiday Night Service Switching.
'Manual Night service Switching is on for class 1 in Class of Service.
Universal Night Answer is not assigned in Trunk Ringing on External Speaker.

Related Features
Central Office Calls, Answering and Placing/Ring Groups: There are separate Access Map and Ring Group programming entries for each Night Service mode (Day 1, Night 1, Midnight 1, Rest 1, Day 2, Night 2, Midnight 2, Rest 2). Also, “Universal Answer” allows an extension user to pick up a UNA call.
Paging, External: With Universal Night Answer, outside calls can ring External Paging Zones.
Programmable Function Keys: Function keys simplify activating Night Service.
Voice Response System (VRS): Using the service code for 'Transfer to Incoming Ring Group', a caller listening to the VRS message can have the ability to transfer their call and have it ring the external page.

Operation
To activate Night Service by dialing codes:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 818.
3. Dial Night Service mode number:
1 Day 1 mode
2 Night 1 mode
3 Midnight 1 mode
4 Rest 1 mode
5 Day 2 mode
6 Night 2 mode
7 Midnight 2 mode
8 Rest 2 mode
4. Press SPK to hang up.

To activate Night Service by using programmable keys:
1. Press Night Service key (SC 851: 09) + Mode code number (below).

0 One Button night mode key (range set by PRG 12-08-01). Press the key to select the next night mode within the range.
1 Day 1 mode
2 Night 1 mode
3 Midnight 1 mode
4 Rest 1 mode
5 Day 2 mode
6 Night 2 mode
7 Midnight 2 mode
8 Rest 2 mode

To transfer a call to the Universal Answer External Page zones:
1. Place the CO call on hold and dial the Transfer to Trunk Ring Group code.
You will hear a confirmation tone.
2. Hang up.
The call rings over the External Paging, enabling anyone to answer the call.

Resilience Mode (back up) for System Interdependence

There are an increasing number of switch installations where correct operation of the switch relies to some extent on a separate piece of equipment connected to it.
This equipment will typically be an external Voicemail system.
In all cases where any external equipment is controlling incoming calls it is very important that a "Resilience Mode" or back-up exists, and that the customer knows how to initiate the Resilience Mode in the case of failure of the ancillary equipment.
This is achieved simply by programming a spare Time Mode on the switch so that incoming calls
will by-pass the Voicemail Auto Attendant Application and ring directly to extensions or groups.
A spare Time Mode is not usually a problem.
Even in the cases where calls "fall-over" from extensions to Voicemail or a TAM in hard programming, a Resilience Mode is desirable so callers will not experience Ring Tone No Reply at the faulty equipment.

Off Hook Signalling

Description
When a user calls an extension busy on a call, they can send an off hook signal through the handset and through the telephone's speaker indicating they are trying to get through. The signal is an off hook ringing over the idle (second) line appearance. Off Hook Signaling helps important callers get through, without waiting in line for the called extension to become free.
The system provides the following Off Hook Signaling options:
• Automatic Signaling
Calling a busy extension automatically initiates Off Hook Signaling. This option is useful to receptionists, operators and others that must quickly process calls. This is set in the called extension's Class of Service 'KST Automatic Override/SLT Call Waiting'.
• Manual Signaling
After reaching a busy extension, manual signaling gives the caller the choice of using Off Hook Signaling or activating other features. Extension's without automatic signaling can have manual signaling. The users can dial a service code or press a Programmable Function Key to send Off Hook Signaling to the called phone. This is set in the calling extension's Class of Service 'Intercom Off-Hook Signalling'.

• Selectable Off Hook Signaling Mode
The Off Hook Signal can be muted ringing, no off hook ringing, a beep in the speaker, or a beep in the handset - based on the caller's programming. This is set in the called extension's Keyphone Options.
• DID Call Waiting
An extension can optionally have DID calls camp on without Off Hook signaling. This is set in the 'Call Waiting' option of the DDI Routing Table.
• Block Manual Off Hook Signals
This Class of Service option enables/disables a busy extension's ability to block off hook signals manually sent from a co-worker. If disabled (not blocked), callers can dial the service code at busy or busy/ring to signal the extension. If enabled (blocked), nothing happens when the caller dials the service code for off hook signal.

Conditions
Single Line Telephones: Single line telephones can only send Off Hook Signals, they can receive Call Waiting tone.
Intercom: You cannot send Off Hook Signals to an extension that is already receiving a voice announcement.
Handsfree and Monitor: You cannot send Off Hook Signals to an extension busy on a Handsfree (Speaker phone) call. The called extension's idle CALL key flashes fast, with no ringing.

Default Setting

'Off Hook Signal' is set to muted off hook ring in Keyphone Options.
'Intercom Off Hook Signalling' is on in Class of Service (Ability to request off hook signaling to a busy extension call).
'KST Automatic Override/SLT Call Waiting' is on in Class of Service (Automatically receive off hook signalling while you are busy).
'Interval of Call Waiting Tone' is set to 10 seconds in Timers.

Related Features
Call Waiting/Camp On and Callback: An extension user cannot Camp On to a busy extension or leave a Callback if Off Hook Signaling has already gone through. Off Hook Signaling allows an extension to block a caller's ability to Camp On.
Direct Inward Dialing (DID): DDI calls have a 'Call Waiting' option in the DDI Routing Table.
Hotline/Reverse Voice Over: The setting of 'KST Automatic Override/SLT Call Waiting' is on in Class of Service affects the BLF display for Hotline and Reverse Voice Over.
One-Touch Calling: An extension user can store the Off Hook Signaling Service Code (809) under a One-Touch Key to provide quick Off Hook Signaling access.
Programmable Function Keys: Function keys simplify sending Off Hook Signals.

Operation
To send Off Hook Signals to an extension busy on a call:
Your extension may send Off Hook Signals automatically.
1. Place a call to the busy extension, you will hear busy tone.
1. Press Off Hook Signaling key (SC 851: 33).
You hear special ring back tone.

Park

Description
Park places a call in a waiting state (called a Park Orbit) so that an extension user may pick it up.
There are two types of Park: System and Personal. Use System Park when you want to have the call wait in a system orbit. Personal Park allows a user to Park a call at their extension so a co-worker can pick it up. After parking a call in orbit, a user can Page the person receiving the call and hang up. The paged party dials a code or presses a programmed Park key to pick up the call. With Park, it is not necessary to locate a person to handle their calls. A call parked for too long will recall the extension that initially parked it., however the call remains in the park orbit until it's answered. There are 64 Park Orbits (1-64) available for use.

Splitting Between Parked Calls
A system phone user can retrieve two calls from Park Orbit (for which they don't have line appearances) and easily split (alternate) between them. The split operation brings the calls to the user's telephone and frees up the Park Orbits.

Extended Park
An extension's Class of Service determines whether it will use the normal Park Orbit Recall time or the Extended Park Orbit Recall time. The timers are set up in system programming. When an extension with Extended Park Recall Class of Service option parks a call, it recalls after the Extended Park Orbit Recall time. When an extension with the Normal Park Orbit Recall Class of Service option parks a call, it recalls after the normal Park Orbit Recall time, however the call remains in the park orbit until it's answered.

Call Park Searching
An extension's Class of Service determines whether an extension has to select the park orbit location or allow the system to automatically assign a free orbit.
The automatic method is recommended for display Dterms only as the park orbit location assigned is displayed on the display.

Conditions
An extension can park a call in any Park Orbit. However, an extension can only pick up a call Parked by a member of its own Park group.
If an extension is not allowed access to trunks in the Access Maps, calls in Park and on Hold can be blocked.

Default Setting
All extensions are assigned to Park Hold Group 01 in Extension Basic Options.
'Normal/Extended Park Hold' is off in Class of Service (off is Normal, on is Extended).
The recall time for 'Normal' park hold is 90 seconds, set by 'Park Hold Time' in Timers.
The recall time for 'Extended' park hold is 300 seconds, set by 'Park Hold Recall Time Extended' in Timers.
Parked calls will ring back at the extension for 30 seconds, set by the 'Normal Hold Call-back time' in Timers.
'Call Park Searching' is off in Class of Service (off is manual selection, on is automatic selection).

Related Features
Hold: A user can place a call in a temporary waiting state without putting it in orbit.
Programmable Function Keys: Function keys simplify Park operation.

Operation

To Park a call in a system orbit:

You can Park Intercom or trunk calls.
1. Press Park key (SC 852: *04 + orbit).
The Park key LED lights.
If you hear busy tone, the orbit is busy. Try another orbit.
2. Use Paging to announce call.
3. Press SPK to hang up.
If not picked up, the call will recall to you.
OR
1. At system phone press HOLD.
OR
At a single line telephone, hook flash.
2. Dial 831 and the Park orbit (01-64).
If you hear busy tone, the orbit is busy. Try another orbit.
3. Use Paging to announce call.
4. Press SPK to hang up.
If not picked up, the call will recall to you.

To Park a call in a system orbit on a display phone using Call Park Searching (if enabled):

You can Park Intercom or trunk calls.
1. Press Hold key.
2. Dial 831 and *.
The LCD display will indicate the Park Orbit Location.
If you hear busy tone, all orbits are busy.
3. Use Paging to announce call.
4. Press SPK to hang up.
If not picked up, the call will recall to you.

To pick up a parked call:

1. Lift handset.
2. Press Park key (SC 852: *04 + orbit).
OR
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 861 and the Park orbit (01-64).

To park a call at your extension:

1. Do not hang up.
2. Press HOLD and dial 773.
At a SLT, hook flash instead of pressing HOLD.
3. Page your co-worker to pick up the call.
4. Press SPK to hang up (or hang up at SLT).
If not picked up, the call will recall to you.

To park a call at your extension - while you are calling another extension:

1. While you are on a call, do not hang up.
2. Press HOLD and dial the co-workers extension.
At a SLT, hook flash instead of pressing HOLD.
3. If the extension does not answer press the single digit code assigned to Station Park Hold.
4. You will hear dial tone again, you can make a page call or place another call.
Or
5. Press SPK to hang up (or hang up at SLT.
If not picked up, the call will recall to you.

To pick up a call parked at an extension (yours or a co-worker's):

1. If parked at your extension:
Press idle CALL key and dial 773.
At an SLT, skip pressing CALL.
OR
If parked at a co-worker's extension
Press idle CALL key dial 715 plus the co-worker's extension number.
At an SLT, skip pressing CALL.

To split between two parked calls (System Phone Only):

You must have Park Orbit keys for the parked in calls. In addition, your system phone cannot have line keys defined for the parked calls.
1. Press CALL1.
2. Press Park Orbit key (SC 852: *04 + orbit) to retrieve first parked call.
Call1 lights steadily. This moves the first parked call to your phone.
3. Press HOLD and press SPK.
CALL1 flashes.
4. Press another Park Orbit key (SC 851: *04 + orbit) to retrieve the second parked call.
CALL2 lights steadily. This moves the second parked call to your phone.
5. To switch between the two parked calls, press HOLD then the flashing CALL key.
You can only split between two active calls. To retrieve and split with a new call, you must first hang up one of the initial calls.

Prime Line

Description
Prime Line Selection allows an extension user to place or answer a call over a specific trunk by just lifting the handset. The user does not have to first press keys or dial codes. This simplifies handling calls on a frequently used trunk.
Prime Line operation is achieved by assigning exclusive access of a trunk to an extension and then enabling automatic seize/answer for the extension. This can be achieved by either placing the trunk port into a unique trunk group or using the Trunk Access Maps to give access to one trunk.
Prime Line Selection has the following two modes of operation:
• Outgoing Prime Line Preference
Lifting the handset seizes the Prime Line. (Outgoing Prime Line Preference may be affected by Incoming Prime Line Preference.)
• Incoming Prime Line Preference
When the Prime Line rings the extension, lifting the handset answers the call. (Incoming Prime Line Preference can optionally seize an idle line appearance.)

Conditions
The DECT cordless handset does not support Prime Line Preference.

Default Setting
No trunks are assigned exclusively to an extension:
All trunks are assigned to trunk group 01 in Trunk Group.
Trunk Group 01 is assigned to Trunk Route 01 in Trunk Group Routing.
All extensions use Trunk Route 01 in Trunk Group Routing for Extensions.
All extensions use trunk access map 01 in Trunk Access Map for Extensions.
All trunks are available in trunk access map 01 in Trunk Access Map Setup.
'Auto Trunk Seize' is off in Extension Basic Setup (Outgoing Prime Line).
'Auto Answer for Trunk Call' is on in Keyphone Options (Incoming Prime Line).

Related Features
Central Office Calls, Placing: Other programmed options for outgoing calls also affect a Prime Line.
Direct Inward Lines/Direct Inward System Access: DILs and DISA calls also ring extensions directly, even if not allowed in ring group programming.
Line Preference: Prime Line Selection directly interacts with Line Preference.

Operation
To place a call on your Prime Line:
1. Lift handset.
You hear dial tone on your Prime Line.

To answer a call on your Prime Line:
1. Lift handset.
Depending on your Line Preference programming, you’ll either answer the Prime Line or get dial tone on the idle line appearance.

Private Line

Description

A Private Line is a trunk reserved for a system phone for placing and answering calls. Private Line operation is achieved by assigning exclusive access of a trunk to an extension, the keyphone usually has a programmable function key for the line.
A user with a Private Line always knows when important calls are for them. Additionally, the user has their own trunk for placing calls that is not available to others in the system.
• Incoming only
The system phone has a Private Line only for incoming calls. The user cannot place calls on the Private Line.
• Outgoing only
The system phone has a Private Line only for outgoing calls. The Private Line does not ring for incoming calls.
• Both ways
The system phone has a Private Line for both incoming and outgoing calls.

Default Setting
No trunks are assigned exclusively to an extension:
All trunks are assigned to trunk group 01 in Trunk Group.
Trunk Group 01 is assigned to Trunk Route 01 in Trunk Group Routing.
All extensions use Trunk Route 01 in Trunk Group Routing for Extensions.
All extensions use trunk access map 01 in Trunk Access Map for Extensions.
All trunks are available in trunk access map 01 in Trunk Access Map Setup.
'Auto Trunk Seize' is off in Extension Basic Setup (Outgoing Prime Line).
'Auto Answer for Trunk Call' is on in Keyphone Options (Incoming Prime Line).
No lines are assigned to one keyphone only in Function Key Programming.

Related Features
Central Office Calls, Placing: Other programmed options for outgoing calls also affect a Prime Line.
Direct Inward Lines/Direct Inward System Access: DILs and DISA calls also ring extensions directly, even if not allowed in ring group programming.
Call Forwarding: Private Lines do not follow Call Forwarding.
Line Preference: An extension user can have Line Preference options applied to their Private Line.
Prime Line Selection: A Private Line can also be a Prime Line.
Programmable Function Keys: You should always program a line key for each Private Line.
Single Line Telephones: Private Lines are not available on single line telephones.
Toll Restriction: Private Lines follow normal Toll Restriction.
Transfer: An extension user can Transfer their Private Line. Since other users have hold access, the destination can answer the transferred Private Line and place it on Hold.

Operation
To place a call on your Private Line:
1. Press Private Line key.
2. Dial number.

To answer a call on your Private Line:
1. Press Private Line key or lift the handset.

Programmable Function Keys

Description
Each system phone has Programmable Function Keys. Programmable Function Keys simplify placing calls, answering calls and using certain features. You can customise the function of a system phone’s programmable keys within system configuration or the extension user can do it themselves. Depending on your telephone you can have different quantity of Programmable Function Keys.

Conditions
When a key is programmed using service code 852, that key cannot be programmed with a function using the 851 code until the key is undefined (852+000). For example with a Park Key programmed by dialing 852 + *04 must be undefined by dialing 852 + key + 000 before it can be programmed as a Voice Over key by dialing 851 + key + 48.
All keyphones can have 24 function keys configured regardless of the actual quantity of keys on the keyphone.
The same applies to the DLS Key Programming; a keyphone can have these keys configured even if the DLS console is not installed.
Service codes 851 and 852 can also be used to program Function Keys and DLS Console keys.
Certain keys can not be duplicated (for example you can not have two line keys for the same line).
When re-arranging the function key layout (for example moving a function to a new key) you must first set the original key to undefined before programming the new key.

Default Setting
The first 12 keys on a telephone are line keys (e.g., key 1 = line 001). The remaining keys are unassigned.
'Function Key Programming' is on in Class of Service.

Related Features
Abbreviated Dialing/One-Touch Calling: Abbreviated Dialing and One-Touch Calling also offer quick access to calls and features.
Direct Station Selection (DSS) Console: Programming a 64-button console requires separate programming.

Operation
To change the function of a General Function programmable key:
1. Press idle CALL key.
2. Dial 851.
3. Press the key you want to program.
4. Enter the 2-digit key function, any additional information needed for the key and press HOLD.
Available functions are 00-99 (refer to chart) and line keys 001-200.
To un-define a key, enter 00.

To change the function of an Appearance Function programmable key:
1. Press idle CALL key.
2. Dial 852.
3. Press the key you want to program.
4. Enter the 3-digit key function and any additional information needed for the key.
Available functions are *00-*99 (refer to chart) and line keys 001-200.
To un-define a key, enter 000.
When a key is programmed using service code 852, that key cannot be programmed with a function using the 851 code until the key is undefined (000).

DSS Function programmable key (851+01):
The DSS Function key will allow the entry of up to 24 digits, these can be any digits e.g extension numbers, Service Codes or outside numbers.
If the digits entered match an extension number then the function key will be a DSS key for the extension and show busy lamp information.
If the digits entered match a Service Code the function key will access the feature.
If the digits entered begin with a trunk access code the system will seize a trunk and dial the remaining digits to line. For trunk calls the digits can contain the following special functions:
• Pause - Entered as a P with the CONF or MIC key at the system phone.
Will wait for approximately 3 seconds. Will only function on an ISDN line if it is preceded by the wait for answer symbol, this is because the ISDN line can not pause during D-channel dialling.
• Wait for Answer - Entered as a @ with the LND key at a system phone.
Used for ISDN lines to stop D-channel dialling and wait for the called destination to answer. After answer the system will dial the remaining digits as DTMF in the B-channel.
• Flash - Entered as a R with the FLASH key at a system phone.
Used for COIU trunks to send a Flash (line break) to line.

To check the function of a programmable key:
1. Press CHECK.
2. Press the programmable key.
The programmed function displays.

Function Number	Function	Additional Data
000	Not defined	-
01	DSS/One-Touch	Extension number or any numbers(Max. 24 digits)
02	Microphone Key (ON/OFF)	-
03	DND Key	-
04	BGM(ON/OFF)	-
05	Headset	-
06	Transfer Key	-
07	Conference Key	-
08	Incoming Caller-ID List	-
09	Night Mode Switch	Mode number(0 or 1-8)

10	Call Forward-Immediate	-
11	Call Forward-Busy	-
12	Call Forward-No Answer	-
13	Call Forward-Busy/No Answer	-
14	Call Forward-Both Ring	-
15	Call Forward - Follow Me	-
16	Call Forward to Station	-
17	Call Forward to Device	-
18	Text Message Setup	Message No.(00-20)
19	External Group Paging	External Paging Zone No. (1-6)
20	External All Call Paging	-
21	Internal Group Paging	Internal Paging No.
22	Internal All Call Paging	-
23	Meet-Me Answer to Internal Paging	-
24	Call Pickup for Own Group	-
25	Call Pickup for Another Group	-
26	Call Pickup for Specified Group	Call Pickup Group Number
27	Abbreviated Dial- Common	Abbreviated dial No.
28	Abbreviated Dial-Group	Abbreviated dial No.
29	Repeat Dial	-
30	Saved Number Redial	-
31	Memo Dial	-
32	Meet-Me Conference	-
33	Off-Hook Signaling	-
34	Break-In	-
35	Camp-On, Call-Back	-
36	Department Step Call	-
37	DND/FWD Override Call	-
38	Message Waiting	-
39	Room Monitor	-
40	Handset Transmission Cut-off	-
41	Secretary(Buzzer) Call	Extension No. (Max.4 digits)
42	Boss-Secretary Call	Extension No. (Max.4 digits)
43	Series Call	-
44	Common Hold	-
45	Exclusive Hold	-
46	Department Group Log Out	-

47	-Not Used-	-
48	-Not Used-	-
49	Call Redirect	Extension Number or Voice Mail Number
50	Account Code	-
51	-Not Used-	-
52	Incoming Call Queuing Message Setup	Incoming Ring Group No.
53	Queuing Message Starting	-
54	External Call Forward by Doorphone Box	-
55	Extension Name Edit	-
56	Presence Display Operation	1-100
57	Presence Display Indication	1-100
58	Department Incoming Call- Immediate	Extension Group No.
59	Department Incoming Call- Delay	Extension Group No.
60	Department Incoming Call- DND	Extension Group No.
63	Outgoing Call Without Caller-ID (ISDN)	-
64	-Not Used-	-
65	-Not Used-	-
66	CTI Communication	-
67	Mail Box (DSPDBU)	Extension No. or Department Group No.
68	Voice Mail Service (DSPDBU)	0 : Skip, 1 : Back Skip, 2 : Monitor
69	Conversation Recording Service (DSPDBU)	0 : Conversation recording, 1 : Delete, Re-recording, 2 : Delete, 3 : Immediate delivery
70	Automated Attendant for Extension(DSPDBU)	Extension No. or Department Group No.
71	Change Attendant Message (DSPDBU)	Extension Number or Pilot Number
72	-Not Used-	-
73	-Not Used-	-
74	-Not Used-	-
75	-Not Used-	-
76	Toll Restriction in Credit	Extension Number
77	-Not Used-	-
78	-Not Used-	-

79	-Not Used-	-
80	Tandem Ring Setup Key	(Max.4 digits)
81	Automatic Transfer to Transfer Key	Trunk Line No.
85	Directory Dialling	-
*00	Not used	-
*01	Trunk Key	Trunk Number
*02	Trunk Group/ Loop Key	Trunk Group Number
*03	Virtual Extension Key	Extension Number. or Department Group Number (Max.4 digits)
*04	Park Hold Key	Park Number
*05	Hybrid Operation Key(Loop key)	0 : Incoming, 1 : Outgoing, 2 : Both

Pulse to Tone Conversion

Description An extension can use Pulse to Tone Conversion on trunk calls. Pulse to Tone Conversion lets a user change their extension's dialing mode while placing a call. For systems in a Dial Pulse area, this permits users to access dial-up services from their Dial Pulse line.
The user can, for example:
• Place a call over a Dial Pulse trunk.
• Depending on programming:
Manually implement Pulse to Tone Conversion
OR
Wait 10 seconds.
• The system dials the digits after the conversion as DTMF.

Conditions
Pulse to Tone Conversion is only valid for Dial Pulse trunks.

Default Setting
'Change DP to DTMF Dial' is set to manual in Analogue Trunk Data Setup.

Related Features
Central Office Calls, Placing: Other programmed options for outgoing calls can affect how a call is placed.

Operation
To manually convert your phone's dialing to tone after placing your call on a pulse line:
1. Place call over pulse line.
2. Dial # to switch the DP trunk to DTMF dialing.

Reason of Transfer Display

Description
The display at a system phone can show the reason that call is ringing when it has been diverted/ forwarded to them.
The reason can be call forward, DND, Absence message for example at the original target and the call is then route to an alternate extension. The alternate extension can then answer the call and explain why the call has been diverted.

Conditions
This feature is available at a display system phone.

Default Setting
'Display Reason of transfer' is off in Class of Service.

Related Features
Call Forward: Set call forward to an alternate destination, the type of call forward will be displayed as the reason of transfer.
Text Message: The absence message will be displayed as the reason of transfer.
Do Not Disturb - DND: DND will be displayed as the reason of transfer.

Operation
The operation is automatic.
The display at the alternate extension will show the reason that the call is ringing at their extension as shown below.

Original extension has this set	At the alternate extension the display will show
Call Forward Immediate	TRANSFER < Mike
Call Forward Busy	TRANSFER BUSY< Mike
Call Forward No Answer	TRANSFER NO ANSWER< Mike
DND	TRANSFER DND< Mike
Absence Message (IN METTING UNTIL 10:00)	IN MEETING UNTIL 10:00< Mike

Remote Conference

Description
Users can join a conference by dialling a pilot code and 4 digit password.
There are four remote conference circuits available, each has a separate pilot code and password.
The system allows internal, ISDN, DUD/DISA and System Feature Network calls to join the conference.
The pilot codes, password, maximum quantity of parties and maximum duration can be specified.
When the maximum duration is reached the conference will be disconnected, a warning tone can be played to alert all parties that the maximum duration has been reached.
The system supports a maximum of 8 parties within any conference.

Internal Calls.
The user dials the pilot code of the remote conference and will hear a prompt from the DSPDB card indicating them to enter the password.

ISDN Trunk - DDI calls.
Route to DDI call to the pilot code of the remote conference circuit.

Analogue Trunks.
Set the trunk as Direct Inward Line (DIL) and route to the pilot code.

ISDN - S-point call
The user dials the pilot code of the remote conference.

Conditions
The system must have the DSPDB card installed and set to PVMU mode.
Two or more analogue trunks can not join a conference.
A held trunk call can not be transferred into a remote conference.
A user can not use the hold feature during the conference.
The maximum quantity of conference parties may be limited by other conference calls, conversation recording or Break In.

Default Setting

No pilot codes are assigned.
Passwords (1111, 2222, 3333, 4444) are assigned to each conference.
The maximum quantity of parties is set to 8.
The maximum duration is set to 7200 seconds (120 minutes).
A warning tone will be played 300 seconds (5 minutes) before the conference is disconnected.
Remote Conference is enabled in Class of Service.

Related Features
Conference: Start a conference while in conversation with a caller.

Operation
To join the conference.
1. Dial the pilot code of the conference you want to join into. If you have a display keyphone the name of the conference will be displayed.
2. You will hear a message "Password Please".
3. Enter the 4 digit password for the conference.
If the password is correct you will hear a confirmation tone and you will join the conference.
If the password is incorrect you will hear busy tone.

To exit from the conference.
1. Replace the handset.
All other parties within the conference will continue.

Repeat Redial

Description If a system phone user places a trunk call that is busy or unanswered, they can have Repeat Redial try it again later on. Repeat Redial automatically retries it until the called party answers (the number of retries is based on system programming).

Conditions
Lifting the handset will cancel Repeat Redial.
Single line telephones cannot use Repeat Redial.

Default Setting
'Automatic Repeat Dial' is on in Class of Service.
'Time of Redial' is set to 3 in Timers (the amount of times the system will redial).
'Interval of Redial' is set to 60 seconds in Timer (interval between redial attempts).
'Redial Calling Time' is set to 30 seconds in Timers (redial duration).

Related Features
F-Route: F-Route configuration selects the trunk for the Repeat Redial call.
Central Office Calls, Placing: Other programmed options for outgoing calls can affect how a call is placed.
Last Number Redial/Save Number Dialed: An extension user can quickly redial their last call.

Operation
To use Repeat Redial (if the outside party you call is unavailable or busy):
1. Place trunk call.
Listen for busy tone or ring-no-answer.
2. Press DIAL + LND.
OR
Press Repeat Redial Key (SC 851: 29).
Your Repeat Redial key flashes while you wait for the system to redial.
3. Press SPK to hang up.
The system periodically re dials the call.
4. Lift handset when called party answers.

To cancel Repeat Redial:
1. Press DIAL.
2. Press LND.
OR
1. Press Repeat Redial Key (SC 851: 29).
See also Last Number Redial.

Ring Groups

Description
Ring Groups are used to allow trunks to ring at one or more extensions simultaneously.
For example, to make a trunk ring an extension:
• Assign the trunk and the extension to the same Ring Group.

Any number of extensions and trunks can be in a specific group.
If an extension has a line key for the trunk, Ring Group calls ring the line key. If the extension doesn't have a line key, the trunk rings the line appearance key. If an extension has a key for a trunk that is not in its ring group the lamp will flash if the user has access to the trunk in trunk access map configuration.

Conditions
DIL trunks do not use ring group programming until DIL overflow.

Default Setting
All trunks are in Ring Group 1 in IRG Assignment (Normal).
Telephone 200 is a member of Ring Group 1 in Incoming Ring Group Setup.

Related Features
Direct Inward Dial (DID) / Direct Inward System Access (DISA): DID, DISA overflow options can be to a Ring Group.
Direct Inward Line (DIL): DILs ring extensions without being in a Ring Group.
Night Service: Ring Group programming can be different for each Night Service mode.
Programmable Function Keys: Function keys simplify answering incoming calls.
Transfer: Transferring calls to a Ring Group using a service code forwards the call to the group defined in Incoming Ring Group Setup.

Operation
Refer to Central Office Calls, Answering

Ringdown Extension

Description
With a Ringdown Extension (Hotline), a user can call another extension, outside number, or Abbreviated Dialing number by just lifting the handset. The call automatically goes through - there is no need for the user to dial digits or press additional keys.
After the Ringdown Extension user lifts the handset, ringdown occurs after a programmable interval.
Depending on the setting of this interval, the extension user may be able to place other calls before the ringdown goes through.
The system allows each extension in the system to have a Ringdown Extension. All extensions can share the same dialing number, if desired.

Conditions
Ringdown extension has no effect on an extension's current (active) call.
The Ringdown Extension user must lift the handset for ringdown to work.

Default Setting
'Hotline is off in Class of Service.
No hotline destinations are assigned in Hotline Setup.
The hotline start time is set to 5 seconds in Hotline Setup.

Related Features
Abbreviated Dialing: Ringdown Extension can use Abbreviated Dialing numbers (and follow their trunk routing) as the destination number.
Call Forwarding: Ringdown Extension follows Call Forwarding. For example, the ringdown destination can forward their calls. When the Ringdown Extension user lifts the handset, ringdown automatically calls the extension to which calls are forwarded.
Call Waiting/Camp On, Callback and Off Hook Signaling: If the Ringdown Extension user hears busy tone when they lift the handset, they can Camp On to the destination, leave a Callback or activate Off Hook Signaling.
Do Not Disturb: The ringdown destination user can activate Do Not Disturb. When the Ringdown Extension user lifts the handset, they hear DND. If enabled, the Ringdown Extension user can override the destination's DND.
Handsfree Answerback/Forced Intercom Ringing:

If the destination extension has Handsfree Answerback enabled, the call will voice-announce.
If the destination extension has Forced Intercom Ringing enabled, the call will ring.
Multiple Directory Numbers/Call Coverage Keys: A Multiple Directory Number key can be a ringdown destination. This would allow a ‘front door’ key to be programmed on every extension.

Operation
To place a call if your extension has ringdown programmed:
1. Lift handset.
If you want to place a trunk call, press a line key before lifting the handset.
Depending on the setting of your ringdown timer, you may be able to dial an Intercom call before your ringdown goes through.
If the destination has Handsfree Answerback enabled, your call will voice announce.
If the destination has Forced Intercom Ringing enabled, your call will ring.

To bypass ringdown (if enabled for your system phone):
1. Do not lift handset.
2. Press CALL.
3. Place Intercom or trunk call.

To answer a call if you are another extension’s ringdown destination:
1. Speak toward phone to answer incoming voice-announcement.
OR
Lift handset to answer ringing Intercom call.

Room Monitor

Description
Room Monitor lets an extension user listen to the sounds in a co-workers area. To use Room Monitor, the initiating extension and the receiving extension must activate it.
When using keyphones for monitoring, an extension user can only Monitor one extension at a time.
However, many extensions can Monitor the same extension at the same time.
For example system phone 200 sets Monitored. System phone 210 can monitor system phone 200, at the same time system phone 211 can also monitor system phone 200.
With single line phones, multiple SLTs can be programmed to be monitored by the same SLT. However, an SLT can monitor only one SLT at a time.
For example SLT 410 sets Monitored by SLT 400. SLT 420 also sets Monitored by SLT 400.
SLT 400 can now monitor either SLT 410 OR SLT 420, but can not monitor both at the same time.
This is normally used in Hotel applications where SLT 400 would be at Reception and SLT 410 and 420 are guest rooms. The guest room would set room monitor for baby listening service.

Room Monitor for Single Lines/System Phones
This option enables you to monitor the room status through your single line telephone. This can be used with the Hotel/Motel feature as well. Between keyphones, the monitored room status is picked up by the phone’s microphone and the activity is heard through the speaker of the monitoring system phone. Between single line phones, a user goes off hook on the monitored phone and, from another single line phone, dials a service code and the extension number. The activity of the area where the monitored phone is placed can then be heard at the monitoring phone. This service is available until the handset of the monitored telephone is placed on hook.

Conditions
Room Monitor is for listening only. It does not allow for conversation between the monitoring and monitored extensions.
An extension user cannot monitor an Attendant.
A system phone user cannot monitor a single line phone and a single line phone cannot monitor a system phone.

Default Setting
No 'Room Monitoring' function keys are assigned in Function Key Programming.
'Room Monitor' is off in Class of Service (for the extension monitoring).
'Room Monitored' is off in Class of Service (for the extension being monitored).
'SLT Room Monitor' is off in Hotel Class of Service Setup.

Related Features
Programmable Function Keys: Room Monitor requires uniquely programmed function keys for system phones.

Operation
You must activate Room Monitor at the extension initiating the monitor and at the extension you want to monitor. You can only listen to

one extension at a time.

Keyphones:
To activate Room Monitor (at the extension to be monitored):
1. Go to the extension you want to monitor.
2. Do not lift handset or press SPK.
3. Press Room Monitor key (SC 851: 39).
4. Dial the number of the extension you are at.
For example, if you are at extension 306, dial 306.
You can place and answer other calls while Room Monitor is active.

To activate Room Monitor (at the extension that will be monitoring):
1. Do not lift handset or press SPK.
2. Press Room Monitor key (SC 851: 39).
3. Dial number of extension you want to monitor.
You can place and answer other calls while Room Monitor is active.
You can activate room monitor at another system phone by repeating the same operation as more than one system phone can monitor the same extension.

To cancel Room Monitor (at either extension):
1. Press Room Monitor key at both the initiating extension and the monitored extension.

Single Line Telephones:
To activate Room Monitor (at the extension to be monitored e.g. the Hotel guest room):
1. Go to the extension you want to monitor.
2. Lift handset at the phone to be monitored.
3. Dial 770.
4. Dial 1.
5. Dial number of extension number which will be monitoring the phone.
Only the extension number entered can monitor your phone.
6. Place the handset on the desk, placing the handset's transmitter towards the room.
You cannot place or answer other calls while Room Monitor is active.

To activate Room Monitor (at the extension that will be monitoring e.g. the Reception phone):
1. Lift handset at the phone which will be monitoring another phone.
2. Dial 770.
3. Dial 2.
4. Dial number of extension number which will be monitored.
You cannot place or answer other calls while Room Monitor is active.
To monitor another phone you must go on hook to cancel the current room monitoring
and then repeat the steps above for the new extension to be monitored.

To cancel Room Monitor (at either extension):
1. Hang up the handsets for both the monitored and the monitoring phones.

Save Number Dialed

Description
Save Number Dialed permits an extension user to save their last outside number and easily redial it later on. For example, an extension user can recall a busy or unanswered number without manually dialing the digits. The system retains the saved number until the user stores a new one in its place.
Save Number Dialed saves in system memory a dialed number up to 24 digits. The number can be any combination of digits 0-9, # and *. The system remembers the digits regardless of whether the call was answered, unanswered or busy. The system normally uses the same trunk group as for the initial call. However, the extension user can preselect a specific trunk if desired.

Conditions
None.

Default Setting
Enabled.

Related Features
Automatic Route Selection

For systems with F-Route: F-Route selects the trunk for the call unless the user preselects.
Central Office Calls, Placing: Other programmed options for outgoing calls can affect how a call is placed.
Dial Tone Detection: Refer to this feature on how the system handles Dial Tone Detection.
Last Number Redial: An extension user can quickly redial the last number placed.
Programmable Function Keys: Function keys simplify Save Number Dialed operation.
Repeat Redial: The system can automatically retry a trunk call that was unanswered or busy.

Operation
To save the outside number you just dialed (up to 24 digits):
Use this feature before hanging up.
System Phone
1. Press Save Number Dialed key (SC 851: 30).
Single Line Telephone
1. Hook flash.
2. Dial 815.

To redial a saved number:
System Phone
1. (Optional) Press line key.
This selects a specific trunk for the call.
2. Press Save Number Dialed key (851: 30).
The stored number dials out.
OR
1. Press idle CALL key
2. Dial 815.
OR
Press Save Number Dialed key (SC 851: 30).
Save Number Dialed automatically selects a trunk from the same group as your original call.
The stored number dials out.
Single Line Telephone
1. Hook flash.
2. Dial 815.

To check to see the number you have saved:
1. Press Save Number Dialed key (SC 851: 30).
The stored number displays for ten seconds.
The stored number dials out if you:
- Lift the handset,
- Press an idle line key,
- Press an idle CALL key, or
- Press SPK
2. Press CLEAR.

To clear your saved number:
System Phone
1. Press idle CALL key.
2. Dial 885.
3. Press SPK to hang up.
Single Line Telephone
1. Lift handset and dial 885.
2. Hang up.

Secretary Call Pickup

Description
Secretary Call Pickup lets a system phone user easily reroute calls to their extension that were routed to a co-worker.
By pressing a Secretary Call Pickup key, the user can have all calls to a co-worker's phone ring or voice-announce theirs instead.
Secretary Call Pickup is a simplified type of Call Forward with Follow Me for employees that work closely together.
An extension can have Secretary Call Pickup keys for any number of extensions, limited only by the available number of programmable keys.

Conditions
Secretary Call Pickup is not available to single line telephone users.

Default Setting
No 'Secretary Call Pickup' keys are assigned in Function Key Programming.

Related Features
Call Forwarding with Follow Me: An extension user can also have Call Forwarding with Follow Me reroute a co-worker’s calls to themselves.
Programmable Function Keys: Secretary Call pickup requires a uniquely programmed function key.
Secretary Call: Co-workers can alert each other without disturbing their work.
Single Line Telephones: A system phone can have a Secretary Call Pickup key for a single line telephone.

Operation
To activate Secretary Call Pickup:
1. Press your Secretary Call Pickup key (SC 851: 42 + boss extension.).
Your Secretary Call Pickup key lights and the Boss’s telephone display shows “BOSS FWD>>”.
Calls intended for covered extension ring your phone instead.

To cancel Secretary Call Pickup:
1. Press your lit Secretary Call Pickup key (SC 851: 42 + boss extension.).
To check a key’s Secretary Call Pickup assignment:
1. Press CHECK.
2. Press your Secretary Call Coverage key (SC 851: 42 + boss extension.).
3. Press CLEAR.

Secretary Call

Description
Secretary Call lets two co-workers alert each other without disturbing their work. To have Secretary Call, both co-workers must have keyphones with Secretary Call buzzer keys. When a user presses their buzzer key, the system alerts the called extension by ringing the phone and flashing the called extension’s buzzer key. The called user can respond by placing an Intercom call to the calling party. The called extension’s buzzer key continues to flash and the phone continues to ring until either user cancels the Secretary Call. A secretary could use this feature, for example, to get a message through to the boss who is already on a phone call. After being alerted, the boss could call the secretary when it’s most convenient.
An extension can have Secretary Call keys for any number of extensions, limited only by the available number of programmable keys.
Secretary buzzer is not prevented by Do Not Disturb set at the destination telephone.

Conditions
Secretary Call is not available to single line telephone users.
Secretary Call does not set up an Intercom call.
When assigning Secretary Call, a user enters the associated extension numbers, not port numbers.

Default Setting
No 'Secretary Call' keys are assigned in Function Key Programming.
Both co-workers must have buzzer keys for each other.

Related Features
Programmable Function Keys: Secretary Call requires a uniquely programmed function key.
Secretary Call Pickup: Have your co-workers calls re-routed to your extension.

Operation
To buzz your secretary or boss:
1. Do not lift handset.
2. Press buzzer key (SC 851: 41 + sec. extension.).
Your boss or secretary hears ringing (if the phone is idle).
Your buzzer key lights steadily.
Your boss’s or secretary’s buzzer key flashes fast.

To check to see who left you a Secretary Call:
1. Do not lift handset.
2. Press CHECK.
3. Press flashing Secretary Call key.
4. Press CLEAR.

To answer your Secretary Call indication:
1. Place an Intercom call to the extension that called you.

To cancel a Secretary Call you left at another extension:
1. Press your lit Secretary Call key.

To cancel a Secretary Call left at your extension:
1. Do not lift handset.
2. Press flashing Secretary Call key.

Selectable Display Messaging

Description
Display system phone callers see the selected message when they call the user's extension. Selectable Display Messaging provides personalized messaging. For example, an extension user could select the message "GONE FOR THE DAY". Any display system phone user calling the extension would see the message.
An extension user can add digits for date, time or phone number after messages 1-8 and 10 (up to 24 characters). For example, an extension user could select the message "ON VACATION UNTIL" and then enter the date. Callers see the original message followed by the appended date.
The default messages are:

Number	Message
1	IN MEETING UNTIL ##:##
2	MEETING ROOM - ########
3	COME BACK ##:##
4	PLEASE CALL ###########
5	BUSY CALL AFTER ##:##
6	OUT FOR LUNCH BACK ##:##
7	BUSINESS TRIP BACK ##/##
8	BUSINESS TRIP ##########
9	GONE FOR THE DAY
10	ON VACATION UNTIL ##/##
11-20	MESSAGE 11-20

Conditions
The system allows for a maximum of 50 phones using the Selectable Display Messaging feature at the same time.

Default Setting
'Date and Time Display Mode' is set to type 4 in System Options for Keyphones (Type 4 is 12 hour clock).
'Text Message' is on in Class of Service.
Text messages are defined in Text Messages.
'Text message Mode' is set to Calling in Text Messages.

Related Features
Do Not Disturb: The DND key blinks when an extension is forwarded and it does not have Programmable Function Key programmed for 16 or 17.
Programmable Function Keys: Function keys simplify Selectable Display Messaging operation.
Assign a function key for Call Forwarding (Device) (code 17) or Text Message (code 18). The Call Forwarding Device key allows the user to select a message each time they activate the feature, while the Text Message key automatically selects the message used when programming the key.

Operation
To select a message:

1. Press idle CALL key + dial 713.
OR
Press Call Forward (Device) key (SC 851: 17).
OR
Press idle CALL key + press Text Message key (SC 851: 18) + enter digits to append, if needed + SPK to hang up.
2. Dial 3 + Message number (01-20).
Use VOL Up or VOL Down to scroll through the messages.
3. (Optional for messages 1-8 and 10)
Dial the digits you want to append to the message.
You can append messages 1-8 and 10 with digits (e.g., the time when you will be back).
You enter the time in 24-hour format, but it will display in 12-hour format is set on the system.
4. Press SPK to hang up.

To cancel a message:
1. Press idle CALL key + dial 713.
OR
Press Call Forward (Device) key (SC 851: 17).
OR
Press idle CALL key + press Text Message key (SC 851: 18) + SPK to hang up.
2. Dial 3.
3. Press SPK to hang up.

Serial Call

Description
Serial Call is a method of transferring a call to a co-worker and have it automatically return to the transferring extension when the co-worker clears down.

Conditions
The transferring extension can remain off-hook to auto-receive the call back or hang up and it will ring back to them.
Serial Call requires a uniquely programmed function key.
Serial Call is not available to single line telephones.
The held caller must not clear down or the Serial Call is cancelled.

Default Setting
No 'Series Call' keys are assigned in Function Key Programming.

Related Features
Programmable Function Keys: Serial Call requires a uniquely programmed function key.
Transfer: An extension user can extend (send) a call to a co-worker.

Operation
To place a Serial Call to a co-worker:
1. Place or answer a call.
2. Press HOLD.
3. Dial co-worker’s extension number.
Co-worker must lift handset to respond to your announcement.
4. Press Serial Call key (SC 851: 43) but do not hang up.
When your co-worker hangs up the call, the system makes an automatic transfer back to your extension.

Single Line Telephones

Description
The system is compatible with Dial Pulse and DTMF analogue single line telephones (SLTs). You can install single line telephones as On-Premise or Off-Premise extensions. Single line telephone users can dial codes to access many of the features available to system phone users. With Single Line Telephones, you can have your system simulate PBX type operation.

Feature Description	SLT Port	
Ringing to the SLT	Yes	
DTMF Dialling	Yes	
DTMF Resource Required	Yes	
Dial Pulse Dialling	Yes	
Timed Break Recall	Yes	
Generate Line Reversal on Answer (Outgoing Calls)	No	
Generate Disconnect Clear	Yes	
Generate Message Waiting Lamp	Yes *1	
Voice Mail Connection	Yes	
Caller ID to SLT	Yes	
Loop Current	20mA	
Maximum cable length	1500 metres at 24AWG	

*1 Message Waiting Lamp voltage is -24V to -100V at 500mS intervals.

Conditions
• Dial Pulse single line telephones cannot access any features that require the user to dial # or *.
• When Caller ID is enabled for an SLT port the ring pattern will be fixed at 2 Sec ON/4 Sec OFF for both internal and external calls. The caller ID is sent after the first ring pulse.

Default Setting
SLT are setup in Single Line telephones.

Related Features
Single line telephone users have access to the following features:
Data Communications
APA and APR modules can be used with system phones to provide an analogue port for connection of a modem.
Power Fail for Analogue Trunks: Refer to the Getting Started Guide, for installation instructions to use single line telephones as power fail extensions.

Operation
Single Line Telephones have access to the following system features.

Abbreviated Dialing	Department Step Calling	Message Waiting
Account Codes	Directed Call Pickup	Night Service
Alarm	Do Not Disturb	Off Hook Signaling
Automatic Route Selection	Door Box	Paging
Barge In	Flash	PBX Compatibility
Call Forwarding	Forced Trunk Disconnect	Pulse to Tone Conversion
Call Forwarding with Follow Me	Group Call Pickup	Ringdown Extension
Call Forwarding/DND Override	Hold	Save Number Dialed
Call Waiting/Camp On with Split	Intercom	Selectable Display Messages
Callback	Handsfree Answerback/Forced Intercom Ringing	Toll Restriction

Central Office Calls, Answering (Not at the APA adapter)	Last Number Redial	Transfer
Central Office Calls, Placing	Line Preference	Trunk Queuing and Camp On
Conference	Meet Me Conference	Voice Mail
Department Calling	Meet Me Paging	Voice Over
	Meet Me Paging Transfer	Warning Tone for Long Conversation

SIP Extension Features

Description

The following is a list of features supported by the system when using SIP extension.

Outgoing

- Enbloc sending
- Service code (see 'Service code list' below)
- Account Code
- Abbreviated dial

Service code list

- Night mode switching (own group)
- Record/erase VRS Message
- General Message Playback (VRS)
- Record and erase general message (VRS)
- Call Forward - Immediate/Busy/No answer/Busy-No answer/Both ring/United method
- Dial block
- Tempory Toll restriction override
- Pilot group withdraw
- Walking Toll restriction
- VRS/Off premise call forwarding
- Transfer dial setting for out of range
- DND/FWD overrife call (Bypass call)
- Conference
- Call waiting
- Barge in
- Last number dialling
- Saved number dialling
- Clear LND
- Clear SND
- Specifed trunk answer
- Park a call in orbit
- Place a call on group hold
- Station park hold
- Common cancelling code
- Personal speed dial
- Call own mailbox (in skin voice mail)
- Live recording at SLT (in skin voice mail)
- Tandem trunking (unsupervised conference)

Incoming

- Extension
- Normal
- VRS/DISA
- DID
- DIL
- Leased Line
- Door phone (doorlock not supported)
- CLIP display

Hold/Transfer

- Normal hold
- Park Hold
- Group hold
- Station Park hold

Type of transfer service

- Call forward - immediate
- Call forward - Both ring
- Call forward - No answer
- Follow me
- Call forward - Busy
- Call forward - Busy/no answer
- Fixed call forward
- Fixed call forward off premise
- Call forward to device
- Automated attendant

Transfer operation

- ISDN Normal transfer
- ISDN Blind transfer
- SLT Normal tarnsfer
- SLT blind transfer
- DSPDB VM normal transfer
- DSPDB VM blind transfer

Others

- Can belong to an Incoming Ring Group
- Peer-to-Peer Mode (allows RTP to be sent directly between SIP endpoints)
- Voicemail Message Waiting
- Assign as a virtual extension
- Conference/barge-in/monitor

Conditions

The list above is the list of features supported by the system.
The SIP extension certificate confirms which features are supported in Peer to Peer mode and non peer to peer mode.

Default Setting

All default feature settings are as per the normal class of service settings for the individual feature

Related Features

None

Operation

The operation for each feature is as described in each specific feature section.

Station Message Detail Recording (SMDR)

Description

Station Message Detail Recording (SMDR) provides a record of the system’s trunk calls. Typically, the record outputs to a customer-provided printer, terminal or SMDR data collection device.
SMDR allows you to monitor the usage at each extension and trunk. This makes charge-back and traffic management easier.
SMDR provides the following options:
• Abandoned Call Reporting
The SMDR report includes calls that rang into the system but were unanswered (i.e., abandoned).
SMDR can include all abandoned calls or only those abandoned calls that rang longer
than the specified duration. The Abandoned Call Report helps you keep track of lost business.
• Blocked Call Reporting
When Toll Restriction blocks a call, you can have SMDR print the blocked call information.
Or, you can have SMDR exclude these types of calls. With Blocked Call Reporting, you can better customize Toll Restriction for the site’s application.
• Customized Date Format
The SMDR header can show the report date in one of three formats: American, European or Japanese. Set the format for your

preference.

• Transferred Call Tracking

SMDR shows each extension's share of a transferred call. If an outside call is transferred among four extensions, SMDR shows how long each of the callers stayed on the call.

• Digit Counting

With Digit Counting, SMDR can selectively keep track of toll calls. For example, if the digit count is nine, SMDR won't include toll calls within the home area code. Digit Counting permits SMDR to include only the types of calls you want to monitor.

• Digit Masking

Digit Masking lets you "X" out portions of the number dialed on the SMDR report. Digit Masking makes it easier to keep track of calling patterns, without having to interpret each individual number. You can also use Digit Masking to block out access and security codes.

• Duration Monitoring

SMDR can include calls of any duration, or only those that last longer than the interval you specify. If you want to keep track of all trunk activity, use a short duration. To keep track of only significant usage, use a longer duration.

• Extension Exclusion

You can selectively exclude extensions from the SMDR report. This ensures privacy for high profile callers..

• LAN Communication

The system can output SMDR information via the Ethernet port of the EXIFU-A1 card.

• PBX Call Reporting

If you system is behind a PBX, you can have SMDR monitor all traffic into the PBX or just calls placed over PBX trunks. The SMDR record can include all PBX calls (including calls to PBX extensions) or just calls that include the PBX trunk access code.

• Serial and USB SMDR Communication

The system is compatible with both serial and USB SMDR devices. This gives you many SMDR output options. For example, you can output the SMDR report to a high speed printer or send it to disk through a PC's serial or USB port.

• Trunk Exclusion

Use Trunk Exclusion to exclude certain trunks not subject to per-call charges from the SMDR report.

• Usage Summaries

SMDR can automatically print daily, weekly and monthly call activity summaries. Each summary includes the total number of regular trunk calls and ISDN trunk calls. The daily report prints every day at midnight. The weekly report prints every Sunday night at midnight. The monthly report prints at midnight on the last day of the month.

• Extension Name or Number

The SMDR report can include an extension's name or extension number. Choose the method that makes it easier for you to track call usage.

Sample SMDR Report (European format 2)

```
01/01/2002 PAGE 003
   CLASS TIME LINE DURATION STATION DIALLED No./CLI RD/COST ACCOUNT
01 IVOT 05:39 003 00:00:00 EXT 200                  0
02 IVOT 05:39 004 00:00:00 EXT 200                  0
03 IVOT 05:39 003 00:00:00 EXT 200                  0
04 IVIN 05:38 003                   200            0:00 NO ANSWER
20 IVOT 05:38 003 00:00:00 EXT 200  20X             0
21 IVIN 05:38 004 00:00:00 EXT 200                 0:00
22 IVOT 05:38 003 00:00:00 EXT 200                  0
23 IVOT 05:38 004 00:00:00 EXT 200                  0
```

Call Record Number	SMDR record number (consecutive)
CLASS	Type of call (see Class Definitions below)
TIME	Time call placed or answered. (For Transferred calls, shows time user picked up Transfer.)
LINE	Trunk number used for call
DURATION	How long call lasted. (For Transferred calls, shows how long user was on call after answering the Transfer.)
STATION	Extension number of call "owner" (i.e., extension that first placed or answered call) (For Transferred calls, there can be

	more than one owner - depending on how many extensions shared the call.)
DIALLED No./CLI	For outgoing calls, the number dialed or, for incoming calls, the Caller ID information
RD/COST	Ring duration for incoming calls. COST is not used
ACCOUNT	Account Code number entered by extension user

Class Definitions

PIN	Incoming trunk calls
POT	Outgoing trunk call
POTW	Outgoing trunk call placed using Toll Restriction Override
PTRS	Transferred call
ALB	All lines in group are busy (group number follows TIME field)
BRD	Call blocked due to Toll Restriction
IVIN	ISDN incoming trunk call
IVOT	ISDN outgoing trunk call
IVOTW	ISDN Outgoing trunk call placed using Toll Restriction Override
ITRS	ISDN transferred call

Conditions

The SMDR report does not include Intercom calls.

The SMDR call buffer stores up to 250 call records. The buffer stores calls when the SMDR device is unavailable. When the buffer fills, each new call is not recorded. The alarm display telephone shows “SMDR n Full,” indicating that the buffer is full. To clear the buffer, the SMDR information must be printed out.

The system will not buffer SMDR call records when output via the LAN port.

When SMDR reports are enabled using the same port as the Traffic Management Reporting feature (example: 147), the SMDR block the Traffic Management reports. Unplugging the cable and plugging in back in again will allow Traffic Management reports to print.

SMDR requires a connection to the EXIFU via a LAN or serial port.

If two applications require the same SMDR output from the the system then use of a software SMDR splitter should be considered. This must be sourced and installed by the installer/customer and must often be licensed to the customer.

Default Setting

SMDR is not installed.

Related Features

PBX Compatibility: To use the PBX Call Reporting option, program system for behind PBX operation.

Traffic Management Report (TMS): Traffic Management Reports and SMDR should not use the same data port.

Line Reversal - CO Trunks: Line reversal will give accurate call duration for calls made on analogue trunks.

Operation

Once installed and programmed, SMDR operation is automatic.

Swap extension

Description
Swap extension allows the settings of two extensions to be swapped over.
Swap extension can be performed with the 'Extension Data Swap' service code.
The following settings will be swapped: (Program numbers shown for keyphone programming reference only).
11-02 : Extension Number
12-05 : Night mode group
13-03 : Abbreviated dial group
15-01 : Extension basic setup
15-02 : Multi-Line basic setup
15-03 : Single line basic setup
15-06 : Trunk access map for extensions
15-07 : Programmable function keys
15-08 : Virtual ring tone setup
15-09 : Virtual ring assignment
15-10 : Virtual ring tone order
15-11 : Virtual extension delayed ring assignment
15-12 : Conversation recording destination
15-13 : Loop key data
15-14 : Programmable one touch keys
16-02 : Department group assignment
20-06 : Class of service
21-02 : Trunk group routing for extensions
21-04 : Toll restriction class
21-07 : Toll restriction override password setup
21-10 : Dial block restriction class
21-11 : Hotline assignment
21-13 : ISDN CLIP setup
21-15 : Individual trunk group routing for extensions
21-18 : H.323 CLIP setup
21-19 : SIP CLIP setup
21-20 : SIP trunk call discernment setup
22-04 : Incoming ring group setup
22-06 : Normal incoming ring mode
23-02 : Pickup groups
23-03 : Ringing line preference
23-04 : Ringing line preference for virtuals
24-03 : Park hold group assignment
24-06 : Fixed call forward
24-07 : Fixed call forward off premise
31-02 : Internal paging group assignment
42-02 : Hotel basic setup
92-05 : Password setup for port swap

Conditions
The swap will not be performed if the target extension is in use (off hook).
When using the service code the extension performing the swap must have a password setup in Program 92-05.
When the swap is performed the following features will be canceled/cleared:
Camp On
Common cancel (720)
Last number redial
Saved number redial
Caller List
Swap can not be performed with a Virtual extension number (Multiple Directory number).

Default setting
The service code for 'Extension Data Swap' is not defined in 3 Digit Codes.
There are no passwords assigned in 92-05.

Operation
Using Service Code

The service code is used at one of the extensions to be swapped (the originating extension). You will be prompted to enter the password assigned to the extension and the target extension number. The swap will then be performed between originating extension and the target extension.

To swap your extension settings with those of the target extension:
1. At your extension go off hook or press SPK.
2. Dial the Service Code for Extension Data Swap.
3. Enter your password (4 digits).
The display will show Enter Password-
4. Enter the target extension number.
The display will show ICM Dial-
If the swap is successful you will hear a confirmation beep and the display will show COMPLETED.
If the swap was not successful you will hear busy tone.
5. Go on hook or press SPK.

Tandem Trunking

Description
Tandem Trunking allows an extension user to join two or more outside callers in a trunk-to-trunk Conference. The extension user can then drop out of the call, leaving the trunks in an Unsupervised Conference. The extension user that established the Conference is not part of the conversation. The Conference continues until either outside party hangs up. In addition, the extension user that set up the Conference can end the tandem call at any time.
The number of Conference calls is limited by the number of Conference circuits in the system. Due to this fact, the maximum number of Conference calls cannot exceed two blocks of 32 parties.

There are two methods for Enhanced Tandem Trunking:
• Method A - Set Up From Conference
An extension user can set up Tandem Trunking (Unsupervised Conference) by using the CONF key. This option uses a uniquely programmed Transfer key to set up a tandem call.
• Method B - Tandem Trunking with Transfer Key
This method allows an extension user to easily set up an Unsupervised Conference with a call they have placed on Hold. It uses a uniquely programmed Transfer key to set up a tandem call.

Conditions
Tandem Trunking requires analogue loop start trunks with disconnect supervision, ground start trunks or ISDN trunks.
The maximum number of trunk-to-trunk conferences allowed is determined by the Conference feature setup.

Default Setting
Tandem Trunking Method A - Tandem Trunking from Conference
Service code for 'Tandem Trunking' is 753.
Transmit and receive gain in Conference and Transfer mode is set to -5dB in Trunk Basic data Setup.
'Trunk to Trunk Transfer' is disabled in Trunk Basic data Setup.
'Forced Trunk Disconnect' is off in Class of Service. This allows the extension to disconnect an Unsupervised Conference in progress.
'Conference' is on in Class of Service.
'Break in Monitor Mode' is off in Class of Service.
'Trunk to Trunk Transfer Disconnect time' is set to 0 seconds in System Timers.

Tandem Trunking Method B - Tandem Trunking with Transfer Key
Transmit and receive gain in Conference and Transfer mode is set to -5dB in Trunk Basic data Setup.
'Trunk to Trunk Transfer' is disabled in Trunk Basic data Setup.
'Forced Trunk Disconnect' is off in Class of Service. This allows the extension to disconnect an Unsupervised Conference in progress.
'Trunk to Trunk Transfer Restriction' is off (Trunk to Trunk Transfer is allowed) in Class of Service.
Break in Monitor Mode' is off in Class of Service.
'Trunk to Trunk Transfer Disconnect time' is set to 0 seconds in System Timers.

Related Features
Central Office Calls, Answering / Central Office Calls, Placing: Other programmed options for incoming and outgoing calls can affect how calls are handled.
Conference, Voice Call: Set up a Conference with a co-worker in your immediate work area.
Meet Me Conference: Meet Me Conference lets an extension user set up a Conference via Paging.
Meet Me Paging: Meet Me Paging lets an extension user set up a two-party meeting via Paging.

Operation
Method A - Tandem Trunking from Conference
To set up a Multiple Trunk Conference Call:
1. Place or answer first trunk call.
2. Press CONF key.
3. Place or answer second trunk call.
When adding an answered call, the call must first be answered and placed on hold. A call ringing in cannot be added.
4. Press CONF key.
5. Place or answer third trunk call.
When adding an answered call, the call must first be answered and placed on hold. A call ringing in cannot be added.
6. To set up the conference call, press CONF key twice. This sets up a Conference between you and the outside parties.
7. Press HOLD and dial 753.
This sets up the Multiple Trunk Conference between the outside parties.
The line keys for the trunks blink green as long as the Unsupervised Conference continues.

To end the Multiple Trunk Conference Call:
1. Press any of the flashing line keys.
The line keys light steadily (green). You can listen (i.e., monitor) to the call or rejoin the conversation, based on the setting of 'Break in Monitor Mode' in Class of Service.
Only trunk keys can be used for re-entering the call - trunk group keys and loop keys will not work.
2. Press SPK or Hang up.
If 'Break in Monitor Mode' is set to off: the Conference ends and the line keys go out.
If 'Break in Monitor Mode' is set to on: the Tandem Trunk call continues.
To manually disconnect the Conference use Forced Trunk Disconnect (i.e., Press the line key + 724 - analogue trunks only), must also be used by an extension other than the originating extension.

Method B - Tandem Trunking on Hang up
To set up a Multiple Trunk Conference Call:
1. Place or answer first trunk call.
2. Press CONF key.
3. Place or answer second trunk call.
When adding an answered call, the call must first be answered and placed on hold. A call ringing in cannot be added.
4. Press CONF key.
5. Place or answer third trunk call.
When adding an answered call, the call must first be answered and placed on hold. A call ringing in cannot be added.
6. To set up the conference call, press CONF key twice.
This sets up a Conference between you and the outside parties.
7. Press Transfer key (SC 851: 06).
This sets up the Multiple Trunk Conference between the outside parties.
8. The line keys for the trunks stutter flash green.
To disconnect the Conference, Forced Trunk Disconnect (i.e., Press the line key + *3) must be used by an extension other than the originating extension.

Single Line Telephone
To set up a Tandem Call:
1. Place or answer first trunk call.
2. Press Recall and dial 826.
3. Place or answer second trunk call.
4. To set up the tandem call, press Recall and dial 753.
5. Hang up.
This sets up a Conference between both outside parties.

Time and Date

Description
The system uses Time and Date for:
• Central Office Calls (Access Maps)
• Class of Service (Class)
• Direct Inward Lines
• Display Telephones
• Night Service (Automatic)
• Programmable Trunk Parameters
• Ring Groups

• Station Message Detail Recording
• System Reports
• Toll Restriction (Class)
• Trunk Group Routing
• Voice Response System

Conditions
The system retains the Time and Date after a power failure or system reset.

Default Setting
'Time Setting' is on in Class of Service.
'Date-Time Display Mode' is set to 12Hour format in System Options for Keyphones.

Related Features
Class of Service: Changing the time may change the current COS service depending on the COS mode setup.
Single Line Telephones: Single line telephones cannot set the Time and Date.

Operation
The date must be set in system programming or via PCPro.
To set the system Time:
1. Press idle CALL key.
2. Dial 828.
3. Dial two digits for the hour (24 hour clock, 13 = 1:00 PM).
4. Dial two digits for the minutes (00-60).
5. Press SPK to hang up.

Toll Restriction Dial Block

Description
Toll Restriction Dial Block lets a user temporarily block an extension's Toll Restriction. This helps a user block his or her phone from being used by another person while they are away from their desk. A user would need to enter a 4-digit personal code to enable/ disable this feature.
Dial Block can also be set by the system administrator. If Dial Block has already been set by an extension user, the supervisor can not release it. Additionally, if Dial Block has been set by the supervisor, and extension user can not release it.
Important: This function works by password and Class of Service control (the supervisor is not an assigned extension). If Dial Block is available for all Classes of Service, everyone may become a supervisor if they know the Dial Block password.

Conditions
If the system is reset by a cold start, the Dial Block feature is cleared.
This feature is not available for ISDN S-Bus extensions.
Both the 'Common Dial Block Toll Table' and extension 'Dial Block Table' can be set at the same time. The system gives priority to the setting in the extensions Dial Block Table.

Default Setting
'Dial Blocking' is off in Class of Service.
Supervisor password is not set in Dial Block.
'Common Dial Block Toll Table' is set to 1 in Dial Block.
Extension 'Dial Block Table' is set to 0 in Dial Block.

Related Features
Toll Restriction: Dial Block can temporarily block an extension's Toll Restriction setting.

Operation
To set Dial Block:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 700.
3. Dial the 4-digit Dial Block code (any 4 digits).
4. Dial 1.
A confirmation tone is heard.
5. Press SPK or replace the handset to hang up.

To release Dial Block:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 700.
3. Dial the 4-digit Dial Block code.
4. Dial 0.
A confirmation tone is heard.
5. Press SPK or replace the handset to hang up.

Note: In the event of the password being forgotten, PRG90-19-01 can be used to release the Dial Blocked extension. Navigate to required extension number and enter '1' followed by HOLD key to release.

To set Dial Block from another extension:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 701.
3. Dial the 4-digit Dial Block code (Supervisor password).
4. Dial the extension number to be blocked.
5. Dial 1.
A confirmation tone is heard.
6. Press SPK or replace the handset to hang up.

To release Dial Block from another extension:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 701.
3. Dial the 4-digit Dial Block code (Supervisor password).
4. Dial the extension number to be released from Dial Block.
5. Dial 0.
A confirmation tone is heard.
6. Press SPK or replace the handset to hang up.

Note: In the event of the password being forgotten, PRG90-19-01 can be used to release the Dial Blocked extension. Navigate to required extension number and enter '1' followed by HOLD key to release.

Toll Restriction, Outgoing Disable on Incoming Call

Description
Toll Restriction, Outgoing Disable on Incoming Call allows the system to restrict call made on an analogue trunk where the user has answered an incoming call just after the call has abandoned.
This can occur in the following instance:
1. An outside call rings the system, but the caller hangs up before the call is answered.
2. The Network detects ‘end of call’ and returns to idle.
3. The system continues to ring ('Time Ringing Signal Detection Time' in Analogue Trunk Initial Data Setup) for up to 3 seconds.
4. A user seizes the trunk before the ringing stops and is able to dial out un-restricted.

The system only provides Toll Restriction for outgoing calls.
To prevent this problem the system can be set to detect DTMF digits for incoming calls.
The operation can be enabled/disabled on the system by 'Outgoing Disable on Incoming Call' in Outgoing Call. Individual extensions can be exempt from the operation by 'Outgoing Disable' in Extension Basic Setup.
This feature requires the system's DTMF resource.
The quantity of DTMF digits that can be dialled by a user is set by 'Outgoing Disable on Incoming Line - Digits' in Outgoing Call. When the quantity of digits has been dialled the trunk will be disconnected, the dialled digits are not checked against any Toll Restriction tables.
The quantity of digits dialled set by 'Outgoing Disable on Incoming Line - Digits' in Outgoing Call is useful if users answer calls from automated device and are required to enter DTMF digits, for example by message notification systems.
The duration that the system will detect the DTMF digits can also be adjusted by 'Outgoing Disable on Incoming Line - Timer' in Outgoing Call. This timer must be set greater than the duration that the network will provide dial tone.
This feature will not be used for calls answered by a voice mail system.

Conditions

DTMF detection requires the system's resources.
Only operates for incoming calls on analogue trunks.

Default Setting
'Outgoing Disable' is enabled in Extension Basic Setup
'Outgoing Disable on Incoming Call' is off in Outgoing Call
'Outgoing Disable on Incoming Line - Digits' is set to 4 digits in Outgoing Call
Outgoing Disable on Incoming Line - Timer' is set to 20 seconds in Outgoing Call

Related Features
None, the digits dialled are not checked against any toll restriction tables.

Operation
The operation is automatic for all incoming calls.

Toll Restriction Override

Description
Toll Restriction Override lets a user temporarily bypass an extension's Toll Restriction. This helps a user that must place an important call that Toll Restriction normally prevents. For example, you could set up Toll Restriction to block long distance calls and then provide a Toll Restriction Override code to your attendant and executives.
There are two methods to bypass an extension's Toll Restriction:
Toll Restriction Override
Overrides the toll restriction of the phone; allows the caller to place one call with no toll restriction applied.
Requires a pre-defined 4 digit password that is assigned to each extension. The caller must use the password assigned to the extension they are placing the from.
The SMDR will report the call against the extension that the call was made from. The CLASS will be modified for analogue trunks only (POTA).
Walking Toll restriction
Overrides the toll restriction of a phone; allows the caller to place one call using the toll restriction assigned to the password.
Requires a pre-defined 6 digit password; the password also defines the toll restriction class. The caller can use the same password at any phone.
The SMDR will report the call against the extension that the call was made from with the CLASS modified (POTW or IVOW) and the walking toll restriction number identified in the ACCOUNT field.

Conditions
None

Default Setting
No 'Override Passwords' are assigned in Toll Restriction Override Passwords.
'Toll Restriction Override Release Timer' is set to 10 seconds in Toll Restriction Override Passwords.
No Walking Toll restriction passwords are assigned in Walking Toll Passwords.
The Toll class for Walking Toll Restriction is set to class 15 in Walking Toll Passwords.

Related Features
Station Message Detail Recording: In the Class heading in the SMDR report, POTA indicates that the call was placed using Toll Restriction Override.
In the Class heading in the SMDR report, POTW or IVOW indicates that the call was placed using Walking Toll Restriction. The Account heading the walking toll restriction password is identified by W/xxx (xxx is the password entry 001-500).
Voice Response System (VRS): If the system has VRS, users hear, "Your call cannot go through. Please call the operator" when they dial a number that Toll Restriction prevents.

Operation
Toll Restriction Override:
You can override restriction for only one call at a time.
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 875.
3. Dial the 4-digit Toll Restriction Override password assigned to the phone you are using. If you wait too long before going to the next step, you may have to repeat the procedure. You'll hear error tone if you dial your password incorrectly.
4. Press idle line key or dial trunk access code.
5. Dial number with no toll restriction applied.

Walking Toll Restriction:
You can override restriction for only one call at a time.
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 763.
3. Dial the 6-digit Walking Toll Restriction password. If you wait too long before going to the next step, you may have to repeat the procedure. You'll hear error tone if you dial your password incorrectly.
4. Press idle line key or dial trunk access code.
5. Dial number with toll restriction applied (defined by the Walking Toll restriction password).

Toll Restriction In Credit

Description
Toll Restriction In Credit allows an administrator to set a limit to the cost of outgoing calls calls made by individual extensions.
When the cost limit is reached no further calls can be made from the extension until the restriction is reset by the administrator. It is possible to allow the extension to make Toll Free calls if they are entered in the Toll Restriction tables specified by the Dial Block class in Toll Restriction Dial Block.

This feature requires Advice of Charge (AOC) information to be provided by the Network supplier, there are two types of AOC that can be sent by the network:

- AOC-D
  Sent during the call by the network supplier.
  With this type of AOC the system will disconnect the outgoing call as soon as the cost limit is reached and prevent further calls being made.
- AOC-E
  Sent at the end of the call by the network supplier.
  With this type of AOC the system will not disconnect the call as soon as the limit is reached. The user will be prevented making further calls once the limit is reached.

Conditions
Not supported on analogue lines.
Advice of Charge must be provided by the network.
AOC-E will not disconnect the active call when the limit is reached.

Default Setting
No credit limits are set for any extension.
The Service code to set the credit limit is 774.
'Charging Cost Display' is disabled in Class of Service, this option must be enabled for the administrator telephone only.

Related Features
Toll Restriction.

Operation
At the Administrator Telephone.

Set the credit limit for an extension.
Dial the service code for Toll Restriction in Credit.
Dial the extension number of the telephone you want to set/check.
Dial 1 to set the credit limit.
Enter the amount and press HOLD.

Check the balance for an extension.
Dial the service code for Toll Restriction in Credit.
Dial the extension number of the telephone you want to set/check.
Dial 2 to check the balance (the credit remaining for the extension).
If 0:00 is displayed the extension is not installed on the system.
If the credit limit is not set the display will show '- - - - - - - '.

Clear the credit limit for an extension.
Dial the service code for Toll Restriction in Credit.
Dial the extension number of the telephone you want to clear.

Dial 1 or 2 to set or check.
Press CLEAR.

Toll Restriction

Description

Toll Restriction limits the numbers an extension user may dial. By allowing extensions to place only certain types of calls, you can better control long distance costs. The system applies Toll Restriction according to an extension's Toll Restriction Class. The system allows for up to 15 Toll Restriction Classes.

Toll Restriction offers the following capabilities:

• Common Permit Code Table

Use the Common Permit Code Table when you have numbers you want all Toll Restriction Classes to dial. To let all users dial 911, for example, put 911 in the Common Permit Code Table. The Common Permit Code Table overrides the Restrict Code and Common Restrict Code Tables. The system provides 10 tables, with 10 entries in each table. Each code is 4 digits max., using 0-9, #, * and FLASH (as a wild card).

• Common Restrict Code Table

The Common Restrict Code Table lets you globally restrict certain numbers for all Toll Restriction Classes. To prevent all users from dialing directory assistance (411), for example, put 411 in the Common Restrict Code Table. Be sure you don't allow the codes you want to restrict in the Permit Code Table or the Common Permit Code Table. The system provides 10 tables, with 10 entries in each table. Each code is 4 digits max., using 0-9, #, * and FLASH (as a wild card).

• Restrict Code Table

When you want Toll Restriction to allow most calls and restrict only selected calls, use the Restrict Code Table. To block only 1-900 calls, for example, enter 1900 in the Restrict Code Table. (If the same Toll Restriction Class has both Permit and Restrict Code Tables, the system restricts calls that you enter only in the Restrict Code Table. Calls entered in both tables are not restricted.) The system provides 4 tables, with 60 entries (restricted codes) in each table. A restricted code is 12 digits maximum, using 0-9, #, * and FLASH (as a wild card).

• Permit Code Table

The Permit Code Table lets you set up Toll Restriction so that users can dial only selected (permitted) telephone numbers. Use this table when you want to restrict most calls.

To allow all users to dial only area code 1203, for example, enter 1203 in the Permit Code Table. 1203 + nnnn are the only numbers users can dial. (If the same Toll Restriction Class has both Permit and Restrict Code Tables, the system restricts calls that you enter only in the Restrict Code Table. Calls entered in both tables are not restricted.) The system provides 4 tables, with 60 entries (permitted codes) in each table. A permitted code is 12 digits maximum, using 0-9, #, * and FLASH (as a wild card).

• International Call Restriction

International Call Restriction lets you limit the international calls an extension user may dial.

You can build a restrict table to prevent only certain calls, or you can build a permit table to allow only certain calls. To allow most international calls, use the International Call Restrict Table. To prevent most international calls, use the International Call Allow Table. The system provides 10 International Call Restrict tables with up to 4 digits in each table entry and 20 International Call Allow tables, with up to 6 digits in each table entry. Valid entries are 0-9, #,* and FLASH (for a wild card).

• Toll Restriction for Abbreviated Dialing

Abbreviated Dialing can bypass or follow Toll Restriction. If you allow many users to program Abbreviated Dialing, consider Toll Restricting the numbers they dial. If only administrators can program Abbreviated Dialing, Toll Restriction may not be necessary. You can separately restrict Group and Common Abbreviated Dialing.

• Call Digit Counting

Use Call Digit Counting to limit the number of digits local callers can dial. You can use this option to prevent users from accessing local dial-up services. The system provides 4 tables in which you can make entries for this option. The range is 4-30 digits.

• Toll Call Digit Counting

With Toll Call Digit Counting, you can limit the number of digits long distance callers can dial. This lets you prevent callers from dialing extensively into long distance dial-up services. You can make four entries (4-30 digits).

• Toll Free Trunks

Certain trunks can be completely unrestricted.

• PBX Call Restriction

Toll Restriction programming lets you enable/disable PBX Call Restriction and enter PBX access codes. You only need to do this if your system is behind a PBX and you have trunks programmed for behind PBX operation. Refer to PBX Compatibility feature for the specifics.

• Restrict Transfer for incomplete dial

Toll restriction can prevent a user transferring an outgoing trunk call before sufficient digits have been dialled to complete toll restriction.

• Restrict Common hold for incomplete dial

Toll restriction can prevent a user placing an outgoing call on hold before sufficient digits have been dialled to complete toll restriction.

• Toll Restriction Class for Virtual Extension
A toll restriction class can be assigned for calls made from a virtual extension key.
• Toll Restriction Class for trunks
A toll restriction class can be assigned to individual trunk circuits. The higher of the Toll Restriction classes, extension or trunk or incoming trunk or outgoing trunk, in the case of trunk to trunk transfers, is applied to the outgoing call.
• Select Toll Restriction Class (VE or real extension)
When an outgoing call is made from a virtual extension key, it is possible to define whether the virtual extension's or physical extension's toll restriction class is applied.

Conditions
If a Toll Restriction Class has the same entries in both a permit and restriction table, the system does not restrict the call.
Toll Call Digit counting may prevent users from taking advantage of long distance automated services like ACD and automated Technical Service.
Toll Restriction is applied when accessing F-Route.

Default Setting
The Toll Restriction tables are not assigned.
'Toll Restriction' is enabled in Trunk Basic Data Setup.
Extensions and trunks are assigned to Toll Restriction Class 2 in Toll Restriction per Night Mode.

Related Features
Direct Inward System Access (DISA) / Tie Lines
When using DISA or tie lines, additional programming is required for Toll Restriction (DISA = DISA Toll + COS; tie lines = Tie Line Toll Restriction Class).
Toll Restriction, Dial Block: A user can temporarily block their extension’s Toll Restriction access, preventing unwanted calls from being placed on their phone while they are away from their desk.
Toll Restriction, Outgoing Disable on Incoming Call: The system can restrict calls made on analogue trunk ports when the user answers an incoming call after the call has been abandoned.
Toll Restriction Override: A user can temporarily override an extension’s Toll Restriction.

Operation
To place a trunk call if your system is Toll Restricted:
1. Place call normally.
If your Toll Restriction Class does not allow the number you dial, your call will be cut off.

Transfer Call into Conference

Description
Transfer Call into Conference permits an extension user to send (i.e., extend) an active Intercom or outside call to any existing call. This call can be a 2-party call or an existing conference call.
This feature will allow, for example, the receptionist to place calls to co-workers and add them into an existing telephone call.
This feature is also useful with boss/secretary working. The boss may be on a call and want to add other people into the call. Rather than the boss calling each party and creating a conference the boss can ask the secretary to place the calls and then add them into the boss’s call to create the conference.
The user that adds the call to the conference can not listen or speak to any of the conference callers while setting up the conference. In the above example, the secretary would not be able to listen or speak to the boss’s call.
When each call is added, all callers in the conference will hear a warning tone to alert them to a new caller being added.

Conditions
Intercom, analogue trunk, ISDN trunk and IP trunk calls can be added to a conference.
S-Bus calls can not be added to a conference. A call can not be added to an existing S-Bus call.
Barge in must be enabled to allow the call to be transferred into a conference.

Default Setting
'Barge in' is on Class of Service.
Service code for 'Transfer into Conference' is not set in 3 Digit Codes.

'Barge in Monitor Mode' is off in Class of Service.
'Break In' is on in Class of Service.
'Broken In' (the ability for another extension to Break In to your extension) is on in Class of Service.
'Intrusion Tone' is on in Class of Service.
'Intrusion Tone' repeat time is set to 0 (not repeated) in Timers.

Related Features
Transfer: Sending a call to a co-worker.
Barge In: Breaking into an existing call.

Operation
Transferring a call into a conference
To transfer a call to a co-worker already busy on a call:
1. While on a call.
2. At system phone, press HOLD.
OR
At single line telephone, hook flash.
3. Dial service code for Transfer into Conference (e.g. 709).
The display will show ‘Transfer to Conf. ICM dial’.
If ‘Barge in initiate’ is disabled you will hear busy tone.
4. Dial the busy extension to receive your call. The held call will immediately be added to the conference.
If ‘Barge in receive’ is disabled you will hear busy tone.
5. Hang up.

Transfer to Ring Group

Description
Transfer to Ring Group allows a user to transfer a trunk call to the ring group defined in system programming.
The ring group is defined for each trunk by IRG Assignment (Normal). If the transferred call is not answered it can overflow to the overflow ring group defined by 'IRG Second Target' in IRG Assignment (Normal). The ring no answer time is defined by 'Normal DIL Incoming Call No Answer Time' in Incoming Call.
The transferred call will not recall at the transferring extension.
The transfer operation is also available to Auto Attendant (DUD/DISA) callers by entering the service code for Transfer to Ring Group or a single digit option.
The transferred call can also ring over an external page zone if set in Trunk Ring on External Speaker.

Conditions
Only trunk calls can be transferred to a ring group.
The trunk should have a ring group defined in IRG Assignment (Normal).
The target ring group should have extensions assigned Incoming Ring Group Setup.

Default Setting
No service code assigned for 'Transfer to Incoming Ring Group' in 3 Digit Service Codes.
All trunks are in Incoming Ring Group 1 in IRG Assignment (Normal).
Extension 200 is in ring group 1 in Incoming Ring Group Setup.
No IRG Second Target is setup in IRG Assignment (Normal).
'Normal DIL Incoming Call No Answer Time' is set to 30 seconds in Incoming Call.
No trunks ring over the external paging speakers in Trunk Ring on External Speaker.

Operation
To transfer a trunk call to the Ring Group:
1. While on a trunk call press HOLD.
OR
At single line telephone press Recall.
2. Dial the service code for Transfer to Ring Group.
You will hear confirmation tone.
The trunk call will ring at the ring group defined in IRG Assignment (Normal).
3. Hang Up.

Transfer

Description
Transfer permits an extension user to send (i.e., extend) an active Intercom or outside call to any other extension in the system. With Transfer, any extension user can quickly send a call to the desired co-worker. A call a user transfers automatically recalls if not picked up at the destination extension. This assures that users do not lose or inadvertently abandon their transfers. While a transferred call

is ringing an extension the system can optionally play ring back tone or Music on Hold to the caller.
The system allows the following types of transfers:
• Screened Transfer
The transferring user announces the call to the destination before hanging up
• Unscreened Transfer
The transferring party extends the call without an announcement.
• Extension (Department) Groups Transfer
The Transferring party sends the call to a Department instead of an extension.
• Transfer Without Holding
A user presses a busy line key and waits for the call to complete. The system automatically sends them the call when the internal caller hangs up.
Automatic On-Hook Transfer Operation
With Automatic On-Hook Transfer, a Transfer goes through as soon as the transferring user hangs up. For example, extension 304 can answer a trunk, press HOLD, dial 305 and hang up. The system extends the call to extension 305. Without Automatic On-Hook Transfer, the call would stay on Hold at extension 304 when the user hangs up. To extend the call, the user at extension 304 would have to press CONF or a Transfer function key before hanging up.

Each method has advantages. Automatic On-Hook Transfer makes transferring calls easier. However, users have to be more aware of how they handle their calls on Hold. Without Automatic On-Hook Transfer, extending a call becomes a two-step operation - but separate from placing calls on Hold.

Prevent Recall of Transferred Call
Class of Service option 'No Recall' will prevent a Transferred call from recalling the originating extension if the call is not answered.

Conditions
None

Default Setting
No Function Keys are assigned as Transfer in Function Key Programming.
'Retrieve Line After Transfer' is off in System Options for Keyphones.
'Call Waiting Answer Method (SLT only)' is set to Hooking in SLT Options.
'Ringback Tone to Transferred Calls' is set to Hold Tone in Hold and Transfer.
'transfer to Busy Extension' is disabled in Hold and Transfer.
'Ring Inward Transfer' is on in Class of Service.
'Transfer Without Holding' is off in Class of Service.
'Transfer Information Display' is on in Class of Service.
'Automatic On Hook Transfer' is on in Class of Service.
'No Recall' is off in Class of Service.
'Ring Inward Recall Time' is set to 30 seconds in Hold and Transfer.

Related Features
Caller ID: Unscreened Transfers from voice mail will show pre-answer Caller ID information.
Call Forwarding: With Transfer to Busy Extensions enabled, Call Forwarding with Both Ringing offers a unique option. A transferred call will wait for either the forwarding or destination extension to become free. The call goes through to whichever extension becomes available first. If neither extension becomes free within the Transfer Recall Time, the call recalls the transferring extension.
Meet Me Paging Transfer: Page a co-worker and have the call automatically Transfer when the co-worker answers the Page.
One-Touch Calling: When transferring, an extension user can press a One-Touch Key instead of dialing the extension number.
Serial Call: Serial Call is a method of transferring a call so it automatically returns to the transferring extension.
Trunk to Trunk Transfer: Transfer two trunk calls together.

Operation
Transferring Trunk Calls
To Transfer a trunk call to a co-worker’s extension:
1. At system phone, press HOLD.
OR
At single line telephone, hook flash.
You hear Transfer dial tone.
2. Dial co-worker’s extension number.
If the extension is busy or doesn’t answer, you can dial another extension number or press the line key to return to the call. In addition, you may be able to hang up and have the call Camp-On.
SLT users can retrieve the call by pressing hook flash. If a call has been transferred and the single line user has hung up the handset, the call be can retrieved by dialing 715 and the extension number to which it had been transferred.
3. Announce call and hang up.
If you don’t have Automatic On Hook Transfer, you must press CONF (TRF) or your Transfer Programmable Function Key to Transfer the call.
If your co-worker doesn’t want the call, press the flashing line key to return to the call.
SLT users can retrieve the call by pressing hook flash. If a call has been transferred and the single line user has hung up the

handset, the call be can retrieved by dialing 715 and the extension number to which it had been transferred.
If you don't want to screen the call, hang up without making an announcement.

To answer a call transferred to your extension:
1. Lift the handset when a co-worker announces the call.

Transferring Without Holding
To Transfer without holding (system phone only):
1. Lift handset.
2. Press busy line key.
3. When original caller hangs up, you are connected.

Transferring Intercom Calls
To Transfer your Intercom call:
1. At system phone, press HOLD.
OR
At single line telephone, hook flash.
2. Dial extension to receive your call.
If the extension is busy, doesn't answer or does not want the call, you can dial another extension number or press the lit CALL key to return to the call. In addition, you may be able to hang up and have the call Camp-On.
SLT users can retrieve the call by pressing hook flash. If a call has been transferred and the single line user has hung up the handset, the call be can retrieved by dialing 715 and the extension number to which it had been transferred.
3. Announce your call and hang up.

With Automatic On Hook Transfer
When you hang up, the call is automatically transferred.
Without Automatic On Hook Transfer
You must press your Transfer Programmable Function Key to Transfer the call.
If your co-worker just speaks toward their phone to answer, the Intercom call being transferred disconnects when you hang up.
To Transfer the call unscreened, press your Transfer Programmable Function Key and hang up without making an announcement.

Trunk Group Routing

Description
Trunk Group Routing sets out bound call routing options for users that dial the Trunk Group Routing code for trunk calls. Trunk Group Routing routes calls in the order specified by system programming.
If a user dials the trunk access code and all trunks in the first group are busy, the system may route the call to another group.

Conditions
None

Default Setting
All trunks are in Group 1 in Trunk Group.
Extensions use Route 1 in Trunk Group Routing for Extensions.
Route one uses Trunk Group 1 in Trunk Group Routing.
Trunk Access code is defined in 1 Digit Codes.

Related Features
F-Route: F-Route can overriding trunk group routing.
Central Office Calls, Placing: Instead of using Trunk Group Routing, an extension user can place a trunk call by:
• Pressing a line key
• Dialing a trunk service code
• Pressing a trunk group key (refer to the Trunk Group feature)
• Dialing a trunk group service code (refer to the Trunk Group feature).
Programmable Function Keys: Programmable Function Keys simplify placing calls using Trunk Group Routing.
Ringing Line Preference: The system uses Trunk Group Routing when setting up Ringing Line Preference.
Trunk Groups: Use trunk group programming to set the order in which users access trunks within a specific trunk group.

Operation
To place a call using Trunk Group Routing:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.

2. Dial 9.
3. Dial number.
OR
1. At system phone, press Trunk Group Routing key (SC 852: *05).
2. Dial number.

Trunk Groups

Description
Trunk Groups let you optimize trunk usage for incoming and outgoing calls. With Trunk Groups, users can have loop (rotary) keys for trunk calls. Incoming trunk group calls ring these loop keys.
For outgoing calls, the user presses a loop key to access the first available trunk within the group. You set the access order in trunk group programming.
Loop keys give an extension user more available function keys, since the user doesn’t need a separate line key for each trunk. The user only needs one loop key for each trunk group. This simplifies placing and answering calls.

Conditions
None.

Default Setting
All trunks are in group 1 in Trunk Group.
'Outgoing Skip on No Dial Tone' is disabled in Analogue Trunk Data Setup
There are no Trunk Group or Loop keys assigned in Function Key Programming.
'Trunk Group Access Key Operation Mode' is set to incoming/outgoing in System Options for Keyphones

Related Features
Abbreviated Dialing: Unless a user preselects a trunk, Trunk Group programming selects the trunk Abbreviated Dialing uses for trunk calls.
F-Route: F-Route can bypass trunk group routing.
Central Office Calls, Placing: Instead of using Trunk Groups, an extension user can place a trunk call by:
• Pressing a line key
• Dialing a trunk access code
• Dialing a Trunk Group Routing code (9) - refer to the Trunk Group Routing feature
Direct Inward Dialing (DID): All DID trunks of the same type should be placed in the same trunk group. These trunk groups must then be assigned to a 'DDI Table Area Target'.
Loop Keys: Program a function key as a Loop Key to allow an extension user to answer incoming trunks within a trunk group.
Programmable Function Keys: Function keys simplify placing and answering trunk group calls.
Trunk Group Routing: Trunk Group Routing sets out bound call routing options for users that dial the Trunk Group Routing code for trunk calls.

Operation
To place a call over a trunk group:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 804.
3. Dial trunk group number (1-9 or 01-25).
4. Dial number.
OR
1. Press trunk group key (SC 852: *02 + group)
2. Dial number.

To answer an incoming trunk group call:
1. Lift handset.
2. Press flashing trunk group key.

Trunk Queuing

Description

Trunk Queuing permits an extension user to queue (wait in line) on hook for a busy trunk or trunk group to become free. The system recalls the queued extension as soon as the trunk is available. The user does not have to manually retry the trunk later. Trunk Queuing lets the caller know when the call can go through. If the extension user does not answer the Trunk Queuing ring, the system cancels the queue request.
With Trunk Camp On, an extension user can queue (wait in line) off hook for a busy trunk or trunk group to become free. The caller connects to the trunk when the trunk becomes free. As with Trunk Queuing, the user does not have to manually retry the trunk later.
Any number of extensions may simultaneously queue or Camp On for the same trunk or trunk group. When a trunk becomes free, the system connects the extensions in the order that the requests were left.

Conditions
None

Default Setting
No function keys are assigned as Trunk Camp On in Function Key Programming.
'Camp On Trunk Call Back Time' is set to 15 seconds in System Timer.

Related Features
Central Office Calls, Placing: Other programmed options for outgoing calls can affect how a call is placed.
Programmable Function Keys: Function keys simplify Trunk Queuing operation.

Operation
To queue for a busy trunk:
1. Try to access busy trunk.
2. Press Trunk Queuing/Camp On key (SC 851: 35).
3. Hang up to leave a Trunk Queuing request.
OR
Wait off hook to Camp On to the trunk.

To answer when Trunk Queuing calls you back:
1. Lift handset.
To cancel a Trunk Queueing/Camp On request:
1. At system phone, press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 870.
3. At system phone, press SPK to hang up.
OR
At a single line telephone, go on hook.

Trunk to trunk Forwarding

Description
Trunk to trunk Forwarding refers to the forwarding of incoming trunk calls off premise. The forward can be set for each trunk and each trunk can have its own destination number that the calls are forwarded to.
The feature can be set for trunk ports that are set as Normal (Type 0 in Incoming Service Type Setup) and can be used for COIU and ISDN trunk ports.
The call forward is controlled by service codes. The destination can also be changed by the user for each night mode (1-8).
When call forward is set all incoming calls to the trunk will be forwarded immediately.
The destination of the call forward is saved in an Abbreviated Dial bin, the bin is defined by Automatic Trunk To Trunk Transfer Target Setup. If a different call forward destination is required for each night mode then a different Abbreviated Dial bin must be defined for each night mode.
A call that is forwarded will be disconnected after the timers in Trunk to Trunk Options.
The outgoing trunk route is defined by Outgoing Route Setup. A free trunk within this route will be used when a call is forwarded.
Trunk to trunk transfer must be enabled for each trunk in Trunk Basic Data Setup. If this item is not enabled the call forward can not be set.

Step Transfer
The system can try up to 8 different off premise numbers by selecting the Step Transfer option of 'Automatic Trunk to Trunk Transfer Mode in Trunk Basic Data Setup for the incoming trunks.
With Step Transfer not selected the system will make one off premise call to the Abbreviated Dial number for the current night mode.
With Step Transfer the system will ignore the night mode and place a call to the Abbreviated Dial entered for Mode 1 in Automatic Trunk to Trunk Transfer Target Setup. If this call is not answered before the 'No Answer Time for Step Transfer' in Trunk to Trunk Options the system will attempt to call to the Abbreviated Dial entered for Mode 2. This will repeat for each entry for Modes 1 to 8; the call will continue to ring the last entry, it will not go back to Mode 1 and try again.

Step Transfer is available for Normal and DDI type trunks by setting the Trunk to Trunk Forwarding feature.
Note that for DDI trunks you must also enable step on for the DDI and enter the first Abbreviated dial in 'Target 2' or 'Target3' within the DDI Routing Table , if this off premise call is not answered before the 'DID to Trunk to Trunk No Answer Time' in Incoming Call the system will place a calls to the Abbreviated Dial entered in Mode 2 to 8 of Automatic Trunk to Trunk Transfer Target Setup.
You must also set the Trunk to Trunk Forwarding feature for ALL trunks that the DDI may be received on for Step Transfer to operate.

Conditions
Analogue trunks must have disconnect clear enabled to ensure the lines are cleared when a call is disconnected.
Trunk to trunk call forward can not be set from an analogue SLT.

Default Setting
The service code to enable trunk to trunk forwarding is 833 ('Enable Automatic Transfer per Trunk' in 3 Digit Codes).
The service code to disable trunk to trunk forwarding is 834('Disable Automatic Transfer per Trunk' in 3 Digit Codes).
The service code to set/change the destination number is 835 ('Set Automatic Transfer to Transfer Destination' in 3 Digit Codes).
Abbreviated Dial bin 1999 is used as the call forward destination for all night modes in Automatic Trunk To Trunk Transfer Target Setup.
The disconnect timers are set to give a warning tone after 30 seconds and then disconnect after another 15 seconds in Trunk to Trunk Options.
The outgoing trunk route is not defined in Outgoing Route Setup.
Trunk to trunk transfer is disabled in Trunk Basic Data Setup.
Disconnect clear is not set in Analogue Trunk Data Setup.
'Set/Cancel Automatic Trunk to trunk Transfer' is enabled in Class of Service.
Step Transfer is not selected in Trunk Basic Data Setup.

Related Features
Trunk to Trunk Transfer: Other methods of forwarding an incoming call off premise.
Call Forward Off Premise: Other methods of forwarding an incoming call off premise.
Call Forward to Abbreviated Dial Bin: Call forward of incoming calls to an Abbreviated Dial bin.
Department Group - Call Forwarding: Call forward for calls routed to department groups.

Operation
To set the destination number for each trunk port:
1. Press SPK to go off hook.
2. Dial 835.
3. Dial the trunk port number (01 to 51).
4. Dial the night mode number (1 to 8).
5. Dial the off premise destination number (do not include any trunk access digits).
6. Press HOLD to set the destination for other night modes.
7. Press SPK to hang up.

Note that at step 5 the destination number will be saved to the Abbreviated Dial bin number specified by Automatic Trunk To Trunk Transfer Target Setup for the chosen night mode.
If the same Abbreviated Dial bin is also used for other night modes then you do not need to set them separately. For example at default bin number 1999 is used for all night modes so when a night mode destination number is set for any night mode it will be used for all night modes.

To set the call forward for incoming calls to a trunk:
1. Press SPK to go off hook.
2. Dial trunk to trunk call forward service code - 833.
3. Dial the trunk port number (01 to 51).
4. Press SPK to hang up.
5. Repeat steps 1 to 4 for further trunk ports.

To cancel the call forward for incoming calls to a trunk:
1. Press SPK to go off hook.
2. Dial trunk to trunk call forward cancel service code - 834.
3. Dial the trunk port number (01 to 51).
4. Press SPK to hang up.
5. Repeat steps 1 to 4 for further trunk ports.

Trunk to trunk Transfer

Description
Trunk to trunk Transfer refers to the connection of any two trunk ports together.

Hold and Transfer.
An extension user places one trunk call on hold and then either makes an outgoing call, answers an incoming call or retrieves a call from hold and transfers the two trunk calls together.

Withdraw from Conference.
An extension user sets up a conference call with two (or more) trunk calls and then withdraws from the conference, leaving the trunk callers connected.

Extension User sets Call Forward Off Premise.
An incoming trunk call is routed to an extension with call forward off premise. The user can set call forward off premise by setting call forward to an Abbreviated Dial bin or by Service Code 713+6+Trunk Access Code+off premise number.

A Trunk is routed directly Off Premise.
The system is configured to route the incoming DDI trunk directly to an off premise number. This is set in the DDI Routing Table by entering the trunk access code followed by the off premise extension number as the target for the DDI number e.g 90123 654321.

Warning Tone and Disconnect Timers.
Depending on the method that was used to setup the trunk to trunk transfer there are timers available to play a warning tone to the callers and also disconnect the calls.

Continue/Disconnect Digit
You can assign a DTMF digit that the caller can press to continue the trunk to trunk call after the warning tone is played.
There is also the option to assign a DTMF digit that will disconnect the trunk to trunk call.
These digits would have to be known by the caller prior to setting up the trunk to trunk transfer.

Conditions
Two outgoing analogue trunks can not be connected together, this is because a disconnect clear signal is not generated by the PSTN for calls originated by the system.
The operation of the CONF key at System Phones is set to Conference at default, change to Transfer with Keyphone Options per extension.
The following tables show the settings required to allow the various trunk to trunk options.
For each call type the programming required is listed.
A - Trunk to Trunk Transfer in Trunk Basic Data Setup.
B - Network Disconnect Clear Signal for Analogue trunks in Analogue Trunk Data Setup.

Trunk call type	Trunk call type	Required Settings
Incoming Analogue	Incoming Analogue	A B
	Outgoing Analogue	A B
	Outgoing ISDN	A
Outgoing Analogue	Outgoing Analogue	Not available
	Outgoing ISDN	A
Any ISDN	Any other type	A

Disconnect timers for trunk to trunk calls.
'Trunk to Trunk transfer warning tone timer' in Trunk to Trunk Options, will give a warning tone when an analogue trunk is included as one of the trunks. The default setting is 1800 Seconds (30 minutes).
'Trunk to Trunk transfer disconnect timer' in Trunk to Trunk Options, will disconnect the trunk to trunk transfer when an analogue trunk is included as one of the trunks. The default setting is 0 Seconds (no disconnect).
'DISA COnversation Warning Tone Time' and 'DISA Conversation Disconnect Time', will disconnect calls routed automatically by the system. This includes calls forwarded off premise and trunks routed directly off premise.

Trunk Routing for calls routed directly off premise.
An ISDN DDI routed directly off premise must have the Route Option enabled for DISA Class 1 in DISA Class of Service Options.
An ISDN DDI routed directly off premise will need the trunk route defining for each incoming ISDN trunk port in Outgoing Route Setup.

'Trunk to Trunk transfer restriction' in Class of Service must be set to off (Not restricted) to allow an extension to perform Hold and Transfer of two trunk calls.

Trunk to Trunk transferred call is not answered by the called party

Default Setting
'Trunk to Trunk Transfer' is disabled in Trunk Basic Data Setup.
'Detect Network Disconnect Signal' is disabled in Analogue Trunk Data Setup.
'External Call Forward (Off Premise) is disabled in Class of Service.
'Trunk to Trunk Transfer Warning Tone' timer is set to 0 Seconds in Trunk to Trunk Options.
'Trunk to Trunk Disconnect' timer is set to 0 Seconds in Trunk to Trunk Options.
DISA conversation (Trunk to Trunk conversation) timer is set to 45 Seconds in System Timer for DUD/DISA Service
Route option for DISA class 1 is disabled in DISA Class of Service Options.
Trunk route for all trunk ports is set to 0 (None) in Outgoing Route Setup.
Trunk to Trunk Transfer restriction is set to off (not restricted) in Class of Service.
'Continue/Disconnect Trunk to Trunk Conversation' is set to disconnect in Trunk Basic Data Setup.
Conversation continue/disconnect codes are not set in Trunk to Trunk Options.
'Conversation Extend time' not set in Trunk to Trunk Options.

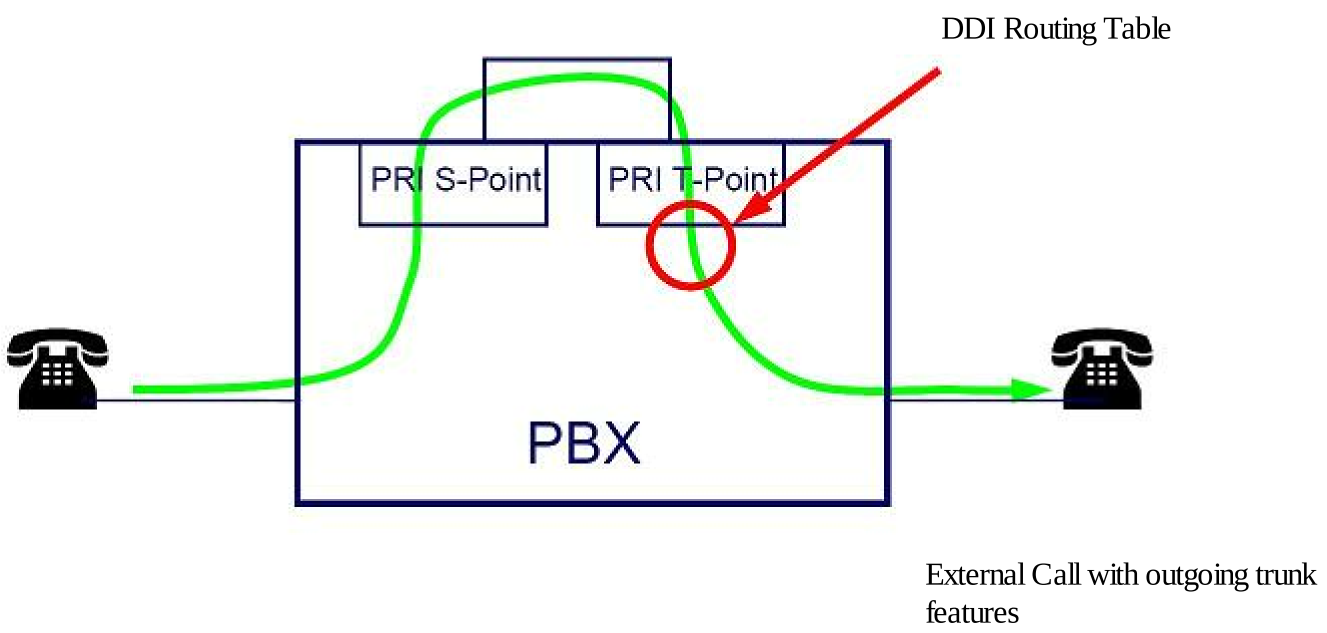
Related Features
Hold: Placing Calls on Hold.
Transfer: Sending your call to another user.
Call Forward to Abbreviated Dial: Divert your calls off premise.
Call Forward Off Premise: Divert your calls off premise.
Trunk to Trunk Forwarding: Diverts incoming calls on selected trunks off premise.

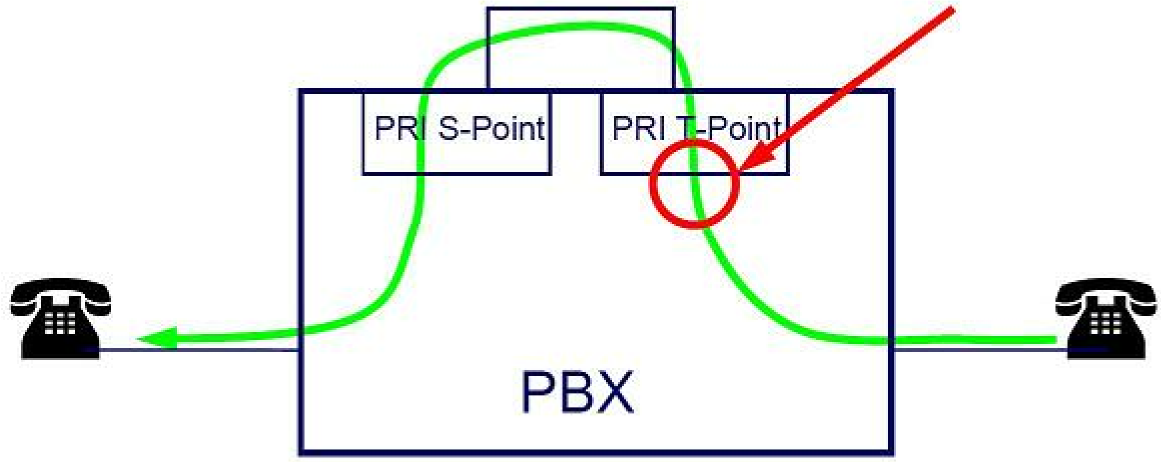
Operation
Refer to each Related Feature above.

Virtual Loopback

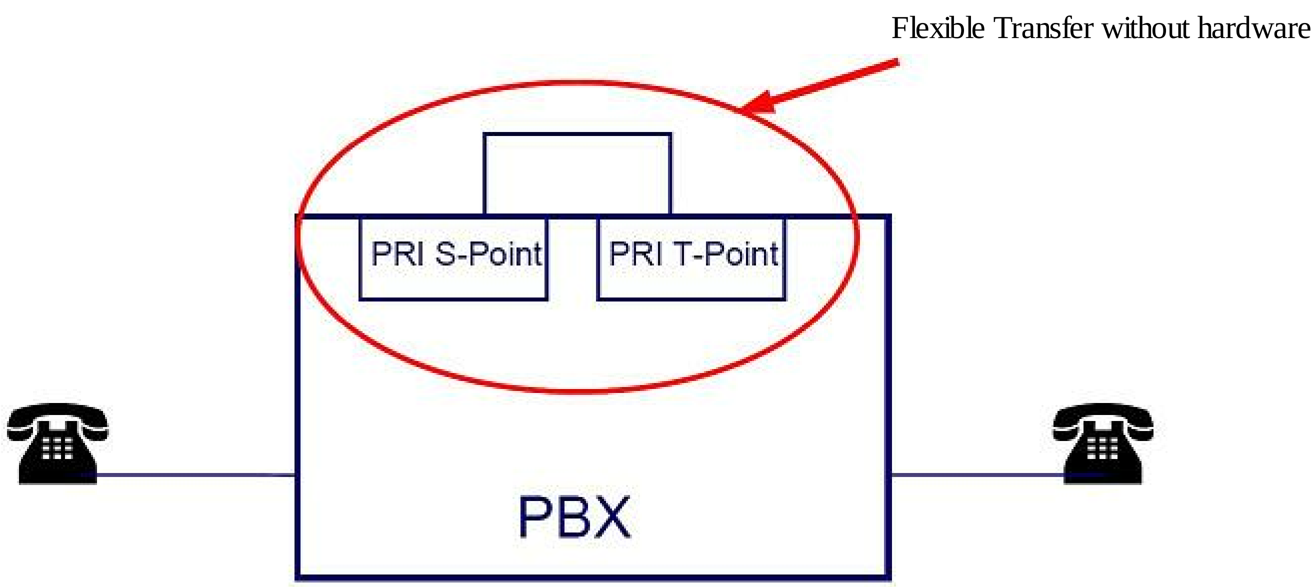
Description
The Virtual Loopback feature gives users the ability to dial, or route, calls to the DDI routing tables by creating DDI calls internally.
This can give additional flexibility in routing when transferring manually or 'automatically' using, for example, single digit options from DSPDB and voice mail Auto Attendants.
Additionally, if the system is only equipped with non-ddi CO lines with CLI enabled (analogue or ISDN) it is possible to create a DDI for the received CLI by transferring it to a loopback port.
Note: Virtual Loopback requires the configuration of Department Groups and Direct Inward Dialling (DDI) for maximum flexibility.

Overview of Virtual Operation
Previously it was necessary to use physical cards to achieve the loop back as per the example below:





With the Virtual LoopBack it is now possible to achieve exactly the same flexibility but minus the additional hardware.



The Virtual Loopback creates up to 30 extension ports and 30 trunk ports for use as Loopback ports.
Each trunk created by the virtual loopback must be set to DID mode. The trunks are then placed into a Trunk Group and routed to a DID Translation Table Area.
You can specify a different area for each night mode.
The DID Translation Table Area contains a specified quantity of DID Translation Entries, the received DID number is compared to each entry to find a match.
The DID Entry will then specify the destination extension number, Incoming Ring Group, Department Group or Voice Mail.
Each DID Entry has three destinations available, the incoming call can be made to fall over to the next destination if the call is busy or not answered.

Configuration
Enter the numbers of loopback ports required, using Easy Edit this is configurable in a drop down list in 'Cards, Virtual Loopback Setup, Virtual Loopback Assignment' and clicking the 'Virtual Port Setting' button.
The start ports for the trunks and extensions will be automatically configured after pressing 'OK'.

Conditions
Only one Virtual Loopback can be configured per system.

Default Setting
No Virtual Loopback ports are configured by default
Trunks are set to Normal incoming type, for DDI operation change in each trunk to DID type (Cards, Virtual Loopback Setup, Virtual Loopback Routing, Virtual Loopback Incoming Service Type Setup).
All trunks are in Trunk Group 01 (Cards, Virtual Loopback Setup, Virtual Loopback Routing, Virtual Loopback Trunk

Group).
All Trunk Groups are routed to DID Translation Tale Area 01 in all night modes (Cards, Virtual Loopback Setup, Virtual Loopback Routing, DDI Table Area Target).
DID Translation Table Area 1 has received DID numbers 00 to 99 routed to extensions 200 to 299, DID Fall Over disabled, Call Waiting disabled, Queue Limit is unlimited (Cards, Virtual Loopback Setup, Virtual Loopback Routing, Virtual Loopback DDI Routing Table.).

Example
By default it is not possible to directly dial an Incoming Ring Group (IRG) but by configuring the Virtual Loopback it is possible to dial a 'Pilot Number' for this purpose.

1. Enter required number of loopback ports
   - Easy Edit reference = Cards, Virtual Loopback Setup, Virtual Loopback Assignment
2. Make a note of automatically configured trunk and extensions ports (e.g. trunk = 33, extn = 17)
3. Enter the virtual loopback extension ports into a free department group e.g. 50
   - Easy Edit reference = Cards, Virtual Loopback Setup, Virtual Loopback Port Numbering.
4. Give the department group a pilot number (e.g. 6 - re-configuration of system numbering will be required for this.)
   - Easy Edit references =
   - Cards, Virtual Loopback Setup, Virtual Loopback Department group
   - Cards, Virtual Loopback Setup, Numbering Plan.
5. Allocate additional digits required e.g. 2
   - Easy Edit reference = Cards, Virtual Loopback Setup, Virtual Loopback Basic Setup, *Flexible Digits for DDI Routing*.
6. Place virtual loopback trunk ports into a trunk group
   - Easy Edit reference = Cards, Virtual Loopback Setup, Virtual Loopback Routing, Virtual Loopback Trunk Group
7. Configure DDI routing for trunk group and virtuals ddi's in this example 600 - 699
   - Easy Edit references =
   - Cards, Virtual Loopback Setup, Virtual Loopback Routing, Virtual loopback DDI Table Area Setup
   - Cards, Virtual Loopback Setup, Virtual Loopback Routing, DDI Table Area Target,
   - Cards, Virtual Loopback Setup, Virtual Loopback Routing, Virtual Loopback Incoming Service Type Setup
   - Cards, Virtual Loopback Setup, Virtual Loopback Routing, Virtual Loopback DDI Routing Table.
8. To configure, for example 600 to IRG 1 enter receive digits as 600 in a DDI table and enter target 2 with IRG 1.

It should now be possible to dial 600 internally and ring IRG 1.
The same scenario could be used for dialling across System Feature Networking.

Voice Mail - External

Description
The system is compatible with analogue Voice Mail/ Automated Attendant Systems connected to SLT ports. These systems provide telephone users with comprehensive Voice Mail and Automated
Attendant features.
Automated Attendant automatically answers the system’s incoming calls. After listening to a customised message, an outside caller can dial a system extension or use Voice Mail.

Integrated Voice Mail enhances the telephone system with the following features:
• Call Forwarding to Voice Mail
An extension user can forward their calls to Voice Mail. Once forwarded, calls to the extension connect to that extension’s mailbox. The caller can leave a message in the mailbox instead of calling back later. Forwarding can occur for all calls immediately, for unanswered calls or only when the extension is busy. When a user transfers a call to an extension forwarded to Voice Mail, the call waits for the Delayed Call Forwarding time before routing to the called extension’s mailbox. This gives the transferring party the option of retrieving the call instead of having it go directly to the mailbox.
• Leaving a Message
Voice Mail lets a system phone extension user easily leave a message at an extension that is unanswered, busy or in Do Not Disturb. The caller just presses their Voice Mail key to leave a message in the called extension’s mailbox. There is no need to call

back later.
• Transferring to Voice Mail
By using Transfer to Voice Mail, a system phone extension user can Transfer a call to the user's own or a co- worker's mailbox. After the Transfer goes through, the caller can leave a message in the mailbox.
• Conversation Record
While on a call, an extension user can have Voice Mail record the conversation. The system phone user just presses the Voice Mail Record key; the SLT user dials a code. Once recorded, the Voice Messaging System stores the conversation as a new message in the user's mailbox. After calling their mailbox, a user can save, edit or delete the recorded conversation.
• Personal Answering Machine Emulation
A system phone user can have their idle extension emulate a personal answering machine. This lets Voice Mail screen their calls, just like their answering machine at home. If activated, the extension's incoming calls route to the user's subscriber mailbox. Once the mailbox answers, the user hears the caller's incoming message.
The system phone user can then:
- Let the call go through to their mailbox
- Intercept the call before it goes to their mailbox
- Reject the call before it goes to their mailbox
• Voice Mail Overflow
If Voice Mail automatically answers trunks, Voice Mail Overflow can reroute those trunks to other extensions when all Voice Mail ports do not answer or, with certain software, are busy.
During periods of high traffic, this prevents the outside calls from ringing Voice Mail for an inordinate amount of time. There are two types of Voice Mail Overflow: Immediate and Delayed. With immediate overflow, calls immediately reroute to other extensions when all Voice Mail ports do not answer or, with certain software, are busy. With delayed overflow, calls reroute after a preset interval.
Without any type of overflow, the outside calls ring Voice Mail until a port becomes available or the outside caller hangs up.
• Voice Mail Caller ID
Voice Mail can use Caller ID information to identify the outside caller that left a message in a user's mailbox.
• Voice Mail Queuing
When accessing the voice mail, the system provides a voice mail queue. If all the voice mail ports are busy, any calls trying to get to the voice mail will be placed in queue. As the voice mail ports become available, the calls will be connected to the voice mail in the order in which they were received.
As the Voice Mail Queue follows Department Hunting programming, the queue can hold a maximum of 10 calls. If the queue is full or if the voice mail ports are not assigned to a Department Group, the calls will be handled as though there were no voice mail queuing feature enabled. The calls will either access voice mail if a port is available or they will receive a busy signal.
The Voice Mail Queuing feature does not work with the Conversation Record feature.
• Voice Mail - MSG key will operate as Voice Mail Key
The MSG key at system phones can be set to call the voice mail. The MSG has the same operation as the Voice Mail function key (851 +77) except that the MSG does not have a lamp.
The MSG key can be used to retrieve messages, send calls to voice mail and access voice mail.
The mode of the MSG key is set by Keyphone Options for each extension.
The Basic system phone does not have a MSG key.
The MSG key will not access a Centralised Voice Mail system, it must be a local Voice Mail.

Conditions
• Ring Group calls do not follow extension call forwarding to voice mail.
• The SLIU Timed Break Recall timers must be set correctly to allow the voice mail to transfer
calls. For example, set SLT Data Setup as follows.
Minimum Hook Flash time = 13 (65mS).
Maximum Hook Flash time = 25 (125mS).
• Smart Office Compact II: configuration changes to ensure correct answer
In order to ensure the Smart Office Compact II answers internal and external calls with the correct announcement the Aspire configuration must be changed from default.
The programming command 82-04-10 must be changed from default 21 to at least 60 and in some cases up to 120
This change ensures that there is a delay of 300mS or more after the Smart Office Compact answers before the integration digits are sent to Smart Office.
Therefore the Smart Office Compact has stabilized and is in the state to accept the digits.
If a value less than 60 is used the Smart Office Compact may answer with an inappropriate (default) greeting.

Default Setting
No External Voice mail system is installed:
There is no Department Group assigned to voice mail in Voice Mail External.
The SLT ports connected to the external voice mail must be placed into their own Department Group in Department Group Assignment, the Department Ring Order is not valid for voice mail SLT ports.
The Department Group pilot number is used as a method of placing calls to the external voice mail system. Assign the pilot number for the voice mail department group in Department Group Options.
The SLT ports must be set to Special in the Terminal Type option of SLT Basic Setup.
Caller ID can also be sent to the voice mail system if the SLT ports are enabled in SLT Basic Setup. You must also ensure the voice mail system is configured to receive the caller ID information.
Ensure that the system is setup to accept the correct Timed Break Recall signal duration in SLT Data Setup, the voice mail system will use recall to transfer calls.

Related Features

Call Forwarding, Fixed: Fixed Call Forwarding can be used to transfer a user's unanswered calls to their voice mail. Call Forwarding does not have to be programmed manually by every user.
Caller ID: Caller ID information will be passed from the Voice Mail to an extension for pre-answer display on an unscreened transfer from Voice Mail.
Direct Inward Line: To have the Voice Mail Automated Attendant answer a trunk, program the trunk as a DIL to the Voice Mail pilot number.
Message Waiting: Message Waiting functions normally with Voice Mail installed.
One-Touch Calling: An extension can have a One-Touch Key for the Voice Mail Master Number.
Programmable Function Keys: Function keys simplify calling the Voice Mail system.
- Voice Mail key (851+77+Mail Box Number)
- Conversation Record (851+78)
- Automated Attendant key (851+79+Extension Number)
- Call Forward (Station) key (851+16)

Voice Response System (VRS): There is a separate voice mail system available on he DSPDB daughter card.

Routing incoming trunk calls to the External Voice Mail system:
The External Voice Mail system has a unique IRG number (102) that can be used to route incoming trunk calls.
IRG 102 can be used in the following options:
DDI Routing Table, targets 2 and 3.
IRG Assignment (Normal), First and Second targets.
DIL Step On Target Assignment.

The Voice Mail Pilot number can also be used to route trunk calls in the following options:
DDI Routing Table, target 1.
DIL Target Assignment.

Operation

CALLING YOUR MAILBOX

To call your mailbox:

With a system phone, or Voice Mail key flashes green and your Message Center keys flash red when they have messages waiting. If you don't have a Voice Mail key, your MW LED flashes instead.
System Phone
1. Press your Voice Mail key (SC 851: 77).
OR
Press idle CALL key and dial the Voice Mail Master Number. After Voice Mail Answers, dial your mailbox number.
Your mailbox number is normally the same as your extension number. You may optionally dial a co-worker's mailbox - or use this procedure to call your mailbox from a co-worker's phone.
OR
Press idle CALL key and dial 717.
2. If requested by Voice Mail, enter your security code.
Ask your Voice Mail system administrator for your security code.
Normally, your MW LED goes out (if applicable). If it continues to flash, you have unanswered “Message Waiting” requests or a new “General Message”. Go to “To check your messages” below.

Single Line Telephone
1. Lift handset and dial 717.
If you are at a co-worker's phone, you can dial the Voice Mail master number and your mailbox number instead. You can also use this procedure from your own phone to call a co-worker's mailbox.
2. If requested by Voice Mail, enter your security code.

LEAVING A MESSAGE (System Phone Only)

To leave a message in the mailbox of an unanswered extension:

The extension you call can be busy, in DND or unanswered.
1. Press Voice Mail key (SC 851: code 77)
The Voice Mail system will prompt you to leave a message.

FORWARDING CALLS TO YOUR MAILBOX

To activate or cancel Call Forwarding:

1. Press idle CALL key (or lift handset at SLT) and dial 888.
OR
Press your Call Forward (Station) key (SC 851: code 16).
2. Dial Call Forwarding condition:
2 = Busy or not answered
4 = Immediate
6 = Not answered

0 = Cancel
3. Dial Voice Mail master number or press Voice Mail key.
4. Dial Call Forwarding type:
2 = All calls
3 = Outside calls only
4 = Intercom calls only
5. Press SPK to hang up (or hang up at SLT) if you dialed 888 in step 1.
Your DND or Call Forwarding (Station) key flashes when Call Forwarding is activated.

TRANSFERRING CALLS TO A MAILBOX
To Transfer your active call to a mailbox:
System Phone
1. Press HOLD.
2. Press Voice Mail key (SC 851: code 77).
3. Dial number of mailbox to receive Transfer.
This number can be your mailbox number or a co-worker’s mailbox number.
OR
Press DSS Console or One Touch key for extension who’s mailbox will receive the Transfer.
If the Transfer destination is an extension forwarded to Voice Mail, the call waits before routing the called user’s mailbox. This gives you the option of retrieving the call instead of having it picked up by Voice Mail.
4. Hang up.
Voice Mail will prompt your caller to leave a message in the mailbox you selected.
OR
1. Dial extension number or press a DSS Console key for extension who’s mailbox will receive the Transfer.
2. Press Voice Mail key (SC 851: code 77).
3. Hang up.
Voice Mail will prompt your caller to leave a message in the mailbox you selected.

Single Line Telephone
1. Hook flash (Recall) to place your call on hold.
2. Dial Voice Mail master number followed by destination mailbox.
If the Transfer destination is an extension forwarded to Voice Mail, the call waits before routing the called user’s mailbox. This gives you the option of retrieving the call instead of having it picked up by Voice Mail.
3. Hang up.

RECORDING YOUR CALL
To record your active call in your mailbox:
2. Press Conversation Record key (SC 851: code 78).
To stop recording, press the Conversation Record key. You can restart and stop recording as required.

Single Line Telephone
1. Hook Flash (Recall).
2. Dial 754.
The system automatically reconnects you to your call.
To stop recording, hook flash twice. You can restart and stop recording as required.

PERSONAL ANSWERING MACHINE EMULATION (System Phone Only)
To enable or cancel Personal Answering Machine Emulation:
1. Press idle CALL key (or lift handset at SLT) and dial 888.
OR
Press your Call Forward (Station) key (SC 851: code 16).
2. Dial 1 and the Call Forwarding type:
2 = All calls
3 = Outside calls only
4 = Intercom calls only
3. Press SPK to hang up if you dialed 888 in step 1.
Your DND or Call Forwarding (Station) key flashes when Call Forwarding is activated.

When Personal Answering Machine Emulation broadcasts your caller’s message, you can:
Your telephone must be idle (not on a call).
1. Do nothing.
The message is automatically being recorded in your mailbox. The broadcast stops when your caller hangs up.
OR
1. (Optional) Lift the handset.
2. Press the flashing CALL key to intercept the call.
You connect to the caller. The system records the first part of the message in your mailbox.

The line key changes from red to green.
OR
1. (Optional) Lift the handset.
2. Press a line key or idle CALL key for a new call.
The message is recorded in your mailbox.
OR
1. (If you have Automatic Handsfree) Press a line key or idle CALL key for a new call.
The message is recorded in your mailbox.
OR
1. Press SPK to cut off the message broadcast and send the call to your mailbox.
Voice Mail records the entire message in your mailbox.

CHECKING YOUR MESSAGES (System Phone Only)
To check your messages:
1. Press CHECK
2. Dial 841.
You can have any combination of the message types in the table below on your phone.

If you see...	You have...
VOICE MESSAGE n MESSAGES	New messages in your Voice Mail mailbox
CHECK MESSAGE VRS GENERAL MESSAGE	Not listened to the current General Message
CHECK MESSAGE (name)	Message Waiting requests left at your phone by (name)

3. Press VOL Up or VOL Down to scroll through your display.
4. When you find the message you want to answer, press CALL1. You'll either:
- Go to your Voice Mail mailbox.
- Listen to the new General Message.
- Automatically call the extension that left you a Message Waiting.

Voice Response System

Description
The DSP daughter board (installed onto the Main Unit) provides the option for Voice Response System (VRS) which gives the system voice recording and playback capability.
This enhances the system with:
• VRS Messages - are 48 system messages used for the General Message, Automated Attendant greetings and Preamble.
• General Message - provides a pre-recorded message to which any user can listen
• Personal Greeting - lets an extension user record a message and forward their calls. Callers to the extension hear the recorded message and are then redirected.
• Park and Page - parks a call at an extension and automatically pages the user to pick it up
• Automated Attendant (Operator Assistance) - answers incoming calls, plays a greeting to the caller and then lets the caller directly dial a system extension
• Transfer to the VRS - Any extension user can Transfer their outside call to the VRS.
• Voice Prompting Messages - plays call and feature status messages to users
• Preamble - alerts callers using 900 lines of the cost and features of the "pay-per-call" service
• Time, Date and Station Number Check - lets a system phone extension user quickly hear a recording for the time, date, or the extension's number.
• Queue Announcements - For trunk callers to an IRG or Department group. The VRS can play up to 2 different announcements to the callers.

VRS Messages
The VRS allows you to record up to 48 VRS messages. You allocate these messages for Automated Attendant greetings, the General Message and the 900 Preamble message.The maximum duration of any one VRS message is 2 minutes. VRS messages are stored permanently in the event of a power failure.
Any on-premise extension can listen, record and erase VRS Messages (unless restricted in programming).
DISA and DID callers can listen and record VRS Messages (unless restricted in programming), except that they must additionally enter a VRS password.

General Message

A General Message is a pre-recorded message available to all callers. A General Message typically contains important company information that all employees should hear. To hear the General Message, an employee can go to any system phone and press 4 (for General Message). You can restrict the ability to record the General Message in an extension's Class of Service. This allows you to give recording capability to the System Administrator or Communications Manager, for example, but not any employee.The MW LED at each telephone flashes when a new General Message is recorded. Once the extension user listens to the message, the MW LED goes out.

Personal Greeting

Personal Greeting allows an extension user to record a message and forward their calls. Callers to the extension hear the recorded message and are then forwarded to the new destination. The message can be up to 2 minutes long. With Personal Greeting, an extension user can add a personal touch to their Call Forwards.

After they record their Personal Greeting, the extension user chooses the condition that will activate Personal Greeting. Personal Greeting will activate for:

- Calls to the extension when it is busy or not answered
- All calls immediately
- Calls to the extension that are unanswered

The extension user then selects the destination for their calls. The choices are:

- A co-worker's extension
- Personal Greeting only (without forwarding)
- The extension user's own subscriber mailbox (if Voice Mail (DSP) is installed)
- Off-Premise via Common Abbreviated Dialing

In addition, the user can have Personal Greeting activate automatically for all calls, just trunk calls or just Intercom calls. When the user implements Personal Greeting for all calls, the system plays the greeting and reroutes:

- Calls transferred from the Automated Attendant (OPA)
- DISA calls ringing the extension
- DID calls ringing the extension
- Direct Inward Lines (DILs) ringing the extension
- Intercom calls

With Personal Greeting for only trunk calls, the system reroutes all of the calls listed above except Intercom calls.

Personal Greetings are stored permanently. If there is a commercial power failure or if the system resets, any recorded Personal Greetings are kept.

Unique Personal Greeting Conditions

If a call comes into the extension when there are no VRS ports available to play the Personal Greeting, the system forwards the call without playing the recorded message to the caller.

If an extension has Personal Greeting (RNA) enabled, Intercom calls that voice announce are not subject to Personal Greeting rerouting.

Personal Greeting does not reroute normal Ring Group calls. Calls transferred from a co-worker or Voice Mail route to the forwarding destination without listening to the Personal Greeting.

Park and Page

When an extension user is away from their phone, Park and Page can let them know when they have a call waiting to be answered. To enable Park and Page, the user records a Personal Greeting along with an additional Paging announcement. Park and Page will then answer an incoming call and play the Personal Greeting to the caller. The caller then listens to Music on Hold (if available) while the system broadcasts the pre-recorded Paging announcement. When the extension user hears the Page, they can go to any telephone and use Directed Call Pickup to intercept the call.

Park and Page follows the rules for Personal Greeting for All Calls, immediately rerouted. This means that Park and Page will activate for ringing Intercom calls, DID calls and DISA calls. It will also activate for calls transferred from the Automated Attendant. Additionally, calls from the Automated Attendant follow Automatic Overflow routing if not picked up. Park and Page will activate for transferred outside calls but not play the Personal Greeting to the caller. If a call comes in when the specified Page zone is busy, the system broadcasts the announcement when the zone becomes free.

Automated Attendant (Operator Assistance)

Automated Attendant automatically answers outside calls, plays a pre-recorded greeting and then lets the outside callers directly dial system extensions, Department Calling Groups and Voice Mail.

Automated Attendant provides immediate answering and routing of outside calls without the need for an operator or dispatcher.

Automated Attendant provides:

- Single Digit Dialing

Single Digit Dialing allows Automated Attendant callers to dial extensions, Department Calling Groups, and Voice Mail by pressing a single digit.

- Simultaneous Call Answering

With VRS installed, the Automated Attendant can answer up to 8 calls simultaneously.

- Flexible Routing

The outside caller can directly dial any system extension, Department Calling Group or Voice Mail. If the caller dials a busy extension, Automated Attendant allows them to dial another extension or wait for the busy extension to become free.

• Automatic Overflow
Automatic Overflow can automatically redirect a call if it can’t go through. This can happen if all VRS ports are busy, if the called extension doesn’t answer, or if the caller mis-dials or waits too long to dial. (This would occur if the caller is using a dial pulse telephone.) When the call overflows, it rings a designated Ring Group or the Voice Mail system.
• Programmable Automated Attendant Greetings
You can record a different greeting for each trunk answered by the Automated Attendant. The greetings can be different in the day, at night or on holidays or weekends. You can also have a special greeting if the caller mis-dials. You record the greetings just the way you want. When assigning and recording Automated Attendant greetings, you can choose among the 48 VRS messages.

Transfer to the VRS
Any extension user can Transfer their outside call to the VRS. This lets their caller take advantage of the Automated Attendant’s extensive routing capabilities. To Transfer the call, the user simply places the call on Hold, dials the unique VRS service code and hangs up.

Voice Prompting Messages
The VRS feature provides the system with Voice Prompting Messages. These Voice Prompting Messages tell the extension user the status or progress of their call. For example, if a user calls extension 300 when it is busy, they hear, “Station 300 is busy. For Callback, dial 850.”

Preamble
If the system has trunks that are part of a caller paid service, the VRS can automatically play a pre-recorded message when a user answers the call. This pre-recorded message should describe the service features and cost. The Preamble ensures that the caller is always aware that they have accessed a “pay- per-call” service. A system user cannot converse with the caller until the preamble message ends. If the caller waits for the message to end, they can talk to a system user . The system will answer as many calls as there are available VRS ports. If a call comes in when all VRS ports are busy, the call will not appear on an extension until a VRS port is available.

Time, Date and Station Number Check
If the system has a DSP daughter board installed for VRS, any system phone user can find out the time, date or the extension’s number while their phone is idle (on hook). The time and date check saves the user time since they don’t have to look for a clock or calendar. Hearing the extension number conveniently identifies non-display keyphones. To find out their extension number, the user presses 6 (for Number). To listen to the time and date, the user presses 8 (for Time).

Queue Announcements
The VRS can play announcements to incoming trunk callers that are waiting at either an Incoming Ring Group (IRG) or Department Group. Callers listening to the queue announcement can not dial an option to escape from the announcement.

Incoming Ring Group Queue Announcements
Calls to the IRG can hear a VRS announcements while the call is ringing at the group. Up to two different announcements can be played. The interval between each message and the number of times the message repeats can be set per IRG. Between messages the system can play either simulated ring back tone, MOH or BGM tone.

Department Group Queue Announcements
Trunk calls waiting at a busy department group can hear VRS announcements until a member of the group becomes free.
Up to two different announcements can be played. The interval between each message and the number of times the message repeats can be set per department group.
Between messages the system can play either simulated ring back tone, MOH or BGM tone.

VRS Multi Language
With a Ver2 DSPDB compact flash card it is possible to select different languages for the VRS announcements.
There are ten languages on the Ver2 compact flash card installed on the DSPDBU card.
The languages are:
0 - Japanese
1 - English
2 - German
3 - Norwegian
4 - Dutch
5 - French
6 - Italian
7 - Spanish
8 - Chinese
9 - Flexible
The flexible language can be loaded onto the Ver2 compact flash card.
The flexible languages available are:
Portuguese
Swedish

Greek
Flemish

You can select the language for the system's VRS Messages with Default Menu Language. This will also change the system prompts within the VRS Voice Mail (but not the prompts within individual mail boxes which are set with Mail Box Setup.

Conditions
DSPDB daughter card must be installed
Park and Page announcements will only repeat once.
VRS Record time is fixed at 2 minutes per message this cannot be increased.

Default Setting
Service code for Operation for VRS Message' is 716 in 3 Digit Codes.
Refer to Queue Messages in PCPro for the default settings related to Queue Messages, Preamble and VRS options.

Related Features
Voice Mail
Voice Response Unit (VRS) - VRS Voice Mail
Voice mail system available on the DSPDB PCB installed onto the Main Unit.

Operation
VRS MESSAGES
To record a VRS message:
1. Press idle CALL key.
OR
At a single line telephone, lift handset.

2. Dial 716.
3. Dial 7 (Record).
4. Dial the VRS message number you want to record (01-48).
5. When you hear, "Please start recording" followed by a beep, record your message.
Normally, your message cannot exceed 2 minutes. If you hear, "Recording finished," you have exceeded the allowed message length.
6. Press # to end recording
OR
Hang up to save the message.

To listen to a previously recorded VRS message:
1. Press idle CALL key.
OR
At a single line telephone, lift handset.
2. Dial 716.
3. Dial 5 (Listen).
4. Dial the VRS message number to which you want to listen (01-48).
You'll hear the previously recorded message. If you hear a beep instead, there is no previous message recorded.
5. Press # to hear the message again.
OR
To hear another message, press 5 and then enter the message number (01-48).
OR
Hang up.

To erase a previously recorded VRS message:
1. Press idle CALL key.
OR
At a single line telephone, lift handset.
2. Dial 716.
3. Dial 3 (Erase).
4. Dial the number of the VRS message you want to erase (01-48).
5. Press HOLD (system phone only) to cancel the procedure without erasing (and return to step 3).
OR
Hang up to erase the message.

To record, listen to or erase a VRS message if you call in using DISA:
1. Place call to the system.
2. After the system answers, dial the DISA password (normally 000000).
3. Dial 716 and the VRS password.
4. Dial the function you want.

7 = Record
5 = Listen
3 = Erase
5. Dial the message number (01-48), record the message and press # to end recording.
If you dialed 7 to record, you can dial # to listen to the message you just recorded.
If you dialed 5 to listen, you can dial 5 and the message number to hear it again or if you want to Record, listen to or erase another message, go back to step 4.

GENERAL MESSAGE
To listen to the General Message:
System Phone Only
Your MW LED flashes when there is a new General Message. A voice message periodically reminds you.
1. Do not lift the handset or press CALL.
2. Dial 4 (General).
OR
1. At lift the handset and dial 711.
You will hear the General Message.
Normally, your MW LED goes out. If it continues to flash, you have unanswered “Message Waiting” requests or new messages in your “Voice Mail” mailbox.

To record, listen to or erase the General Message:
1. Press idle CALL key.
OR
At single line telephone, lift handset.
2. Dial 712.
3. Dial the function you want.
7 = Record
5 = Listen
3 = Erase
If you dialed 7 to record, press # to end the recording.
If you dialed 5 to listen, you can dial 5 to listen to the message again.
To Record the General Message again, go back to step 1.
If you dialed 3 to erase the General Message, you must go to step 4 (hang up). To cancel without erasing on a system phone, press HOLD instead and go back to step 1.
4. Hang up when you are done.

PERSONAL GREETING
To enable a Personal Greeting:
1. Press idle CALL key (or lift handset at SLT) and dial 713.
OR
Press Call Forwarding (Device) key (SC 851: 54).
2. Dial 7 + When you hear, “Please start recording,” record your Personal Greeting.
If you already have Personal Greeting or Park and Page set up, you can dial:
7 to re-record
5 to listen (then # to listen again)
3 to erase (then optionally HOLD to cancel the erase)
3. Dial # + Personal Greeting condition:
2 = Busy or not answered
4 = Immediate
6 = Not answered
4. Dial the destination to receive your calls. The destination can be:
- A co-worker’s extension
- Your Voice Mailbox (by dialing the Voice Mail master number)
- Off-premise via Common Abbreviated Dialing (by entering 813 + bin)
- Greeting without forwarding so caller hears busy (by entering your extension number)
You cannot forward to a Department Group pilot number.
5. Dial Personal Greeting type:
2 = All calls
3 = Outside calls only
4 = Intercom calls only
6. Press SPK to hang up (or hang up at SLT).
Your DND or Call Forwarding (Device) Programmable Function Key flashes when Call Forwarding is activated.

To cancel your Personal Greeting:
1. Press idle CALL key (or lift handset at SLT).
2. Dial 713 7 + 3.
3. Press SPK to hang up (or hang up at SLT).

PARK AND PAGE

To have the system Page you when you have a call:

1. Press idle CALL key (or lift handset at SLT) and dial 713.
OR
Press Call Forwarding (Device) key (SC 851: 17).
2. Dial 7 + When you hear, “Please start recording,” record you Personal Greeting.
If you already have Park and Page or Personal Greeting set up, you can dial:
7 to re-record
5 to listen (then # again to listen again)
3 to erase (the optionally HOLD to cancel the erase)
3. Dial #7.
4. When you hear, “Please start recording,” record your Page.
5. Dial # + Dial the Page Zone that should broadcast your announcement.
For example, for Internal Zone 1 dial 801 + 1. Or, for Combined Paging Zone 1 dial 751 + 1.
6. Dial Park and Page type:
2 = All calls
3 = Outside calls only
7. Press SPK to hang up (or hang up at SLT).
Your DND or Call Forwarding (Device) Programmable Function Key flashes when Call Forwarding is activated.

To pick up your Park and Page:

1. Press idle CALL key (or lift handset at SLT).
2. Dial 715 + your extension number.

To cancel your Park and Page:

1. Press idle CALL key (or lift handset at SLT).
2. Dial 713 7 + 3.
3. Press SPK to hang up (or hang up at SLT).

TIME, DATE AND STATION NUMBER CHECK

To check the extension number of any system phone:

1. Do not lift the handset or press idle CALL key.
2. Dial 6 for extension Number.

To check the system time and date from any system phone extension:

1. Do not lift the handset or press idle CALL key.
2. Dial 8 for Time and date.

PREAMBLE

To answer a 900 Preamble call:

1. Answer the ringing call.
The line key turns solid red as the system plays the preamble to the caller.
2. When you hear two beeps and the line key turns green, converse with the caller.

SETTING VRS MESSAGE AS RING TONE

1. Press SPK or lift handset
2. Dial VRS Setting service code (default 878)
3. Dial '2' to set VRS message at own terminal
4. Input VRS message number
5. Press Hold

Volume Control

Description

Each system phone user can control the volume of incoming ringing, splash tone, Paging, Background Music, Handsfree and your handset. Keyphones consolidate all adjustments into the volume buttons. Pressing the VOLUME UP or VOLUME DOWN will adjust the volume level for whichever feature is active (outside call, ICM, ICM ringing, paging, etc.). Pressing these keys when the phone is idle will adjust the contrast level of the telephone’s display. The users should set the volumes for their most comfortable levels.

Conditions

The contrast is not adjustable when the phone has background music enabled.

Default Setting
Enabled.

Related Features
None

Operation
To adjust the volume of incoming ringing and splash tone:
1. If the phone is idle, dial 829. If the phone is ringing, skip to Step 2.
2. Press VOLUME Up or VOLUME Down.

To adjust the volume of incoming Paging announcements, Handsfree, the handset or Background Music:
1. Press VOLUME Up or VOLUME Down
The feature must be active to change the volume.
Pressing the volume keys when the phone is idle will adjust the display's contrast.

VRS 1 Digit Translations

The Service Codes and Option Codes used while accessing the VRS voice mail can be either fixed or flexible.
Select the method to be used in Voice Mail Basic Setup. The DSPDB card must be installed to provide the VRS Voice Mail.

The flexible codes available are shown in the following tables.
The single digit # can not be used as a 1 Digit Code.

1 Digit Code	DSPDB Voice Mail Service Code	DSPDB Voice Mail Option Code
1		
2		
3		
4		
5		
6		
7		
8		
9		
0		
*		

- The 1 Digit Code is dialled by the user when accessing the VRS voice mail.
- The VRS Voice Mail Service Codes and Option Codes are shown in the table below.
  Note - Some codes can not be changed, for example Service Code 9# to End Recording of a Message can not be assigned to a 1 Digit Code, you must always dial 9# to end recording.
- A 1 digit code can have both a Service Code and Option Code if required.
- You will not be able to assign a 1 digit code to all Service Codes and Option Codes as there are more service/option codes than 1 digit codes.

VRS Voice Mail Fixed Service Codes and Option Codes Table.
The service / option codes shown in bold are fixed and can not be changed.

Function	Service Code	Optional Function	Option Code
Help	0#		
Play Message	1#	Replay Message	1#
		Pause/Restart the playback	4#
		Erase message and play next message (during playback)	7#
		Save message and play next message	9#
		Copy the message	2#
		Skip playback forward 8 seconds	3#

		Skip playback backward 8 seconds	6#
		Broadcast the message	28#
		Exit	*#
Erase ALL messages	7#	**Confirm Erase** **Exit**	**0#** ***#**
End recording of message (This Service code can not be assigned in the 1 Digit Code translation table)	**9#**		
Broadcast message to a multiple address group	2#		
Greeting Message 1	Play	31#	
	Record	32#	
	Erase	37#	
Greeting Message 2	Play	35#	
	Record	33#	
	Erase	38#	
Greeting Message 3	Play	36#	
	Record	34#	
	Erase	39#	
Message Notification	61#	**Notify to an extension**	**1#**
		Notify to an external number	**2#**
		Cancel notification	**0#**
		Exit	***#**
Set Automated Attendant	62#	**Confirm** **Exit**	**0#** ***#**
Play messages 'First in First out'	63#		
Play messages 'Last in First out'	64#		
Password setting	65#	**Confirm** **Exit**	**0#** ***#**
Turn On/Off message recording (including Conversation Recording)	66#	Voice Mail key flashes when recording is off.	
Exit	*#		
Escape from mail box (for incoming trunk callers)	50#		

VRS Voice Mail

Description

The VRS Voice Mail included within the Voice Response System (VRS) is a basic voice mail system capable of the following features:
• Personal greeting/mailbox
• Conversation record
• General trunk answering guidance greeting

VRS Multi Language
With a Ver2 DSPDB compact flash card it is possible to select different languages for the system and mail box prompts announcements.
There are ten languages on the Ver2 compact flash card installed on the DSPDBU card.
The languages are:
0 - Japanese
1 - English
2 - German

3 - Norwegian
4 - Dutch
5 - French
6 - Italian
7 - Spanish
8 - Chinese
9 - Flexible
The flexible language can be loaded onto the Ver2 compact flash card.
The flexible languages available are:
Portuguese
Swedish
Greek
Flemish

You can select the language for the system's Voice Mail prompts with Default Menu Language (this will also change the language for the VRS messages on the system).
The system prompts within the individual mail boxes which are set with Mail Box Setup.

Conditions
300 mailboxes maximum.
Storage time is limited to DSPDB compact flash size (64MB = 1hour, 512MB=17hour approximately).
Please note that it is not possible to try and increase the recording time by fitting a larger memory card. This is because the DSPDB simply will not work, a factory - applied license key is required for the DSPDB to operate
100 messages per mailbox max.
3 Selectable personal greetings
Multi Languages available with the Ver2 compact flash card installed on the DSPDBU card.

Default
There are no mail boxes setup in Mail Box Setup.
There are no function keys assigned for VRS Voice Mail services in Function Key Programming.

Related features
Programmable Function Keys:
- Mail Box key (851+67+Mail Box Number)
- Voice Mail Service - Skip message forward 8 seconds (851+68+0)
- Voice Mail Service - Skip message backward 8 seconds (851+68+1)
- Conversation Record - Start/stop (851+69=0)
- Conversation Record - Delete and re-record (851+69=1)
- Conversation Record - Cancel and delete (851+69=2)
- Automated Attendant key (851+70+Mail Box Number)
- Greeting Message selection (851+71+Mail Box Number)

Operation
Service/Option code list
In order to use the various services provided by voice mail, operation codes must be input.
There are two types of operation codes:
1. Service code: following the voice guidance, a code to select the service.
2. Option code: a number to select optional settings or operations of a service.

Function	**Service Code**	**Optional Function**	**Option Code**
Help	0#		
Play Message	1#	Replay Message	1#
		Pause/Restart the playback	4#
		Erase message and play next message (during playback)	7#
		Save message and play next message	9#

		Copy the message	2#
		Skip playback forward 8 seconds	3#
		Skip playback backward 8 seconds	6#
		Broadcast the message	28#
		Exit	*#
Erase ALL messages	7#	Confirm Erase	0#
End recording of message	9#		
Broadcast message to a multiple address group	2#		
Greeting Message 1	Play	31#	
	Record	32#	
	Erase	37#	
Greeting Message 2	Play	35#	
	Record	33#	
	Erase	38#	
Greeting Message 3	Play	36#	
	Record	34#	
	Erase	39#	
Message Notification	61#	Notify to an extension	1#
		Notify to an external number	2#
		Cancel notification	0#
		Exit	*#
Set Automated Attendant	63#		
Play messages 'First in First out'	63#		
Play messages 'Last in First out'	64#		
Password setting	65#		
Turn On/Off message recording (including Conversation Recording)	66#	Voice Mail key flashes when recording is off.	
Exit	*#		

Forwarding/Cancelling forward to voice mail from a system phone
1. If the system phone has an Auto Attendant key (code 70 + mailbox), this key acts as a toggle for call forwarding for the extension/ mailbox assigned to the key.
The key toggles through Forward All, No Answer, Busy, No Answer/Busy, Cancel.
2. If the system phone does not have an Auto Attendant key it is possible to forward their calls to voice mail by dialling the service code for 'Automated Attendant (DSPDB) in <u>3 Digit Codes</u> (default= 790). Dial the Service code then dial one of the following options:
1 - Forward All Calls
2 - Forward on No Answer
3 - Forward on Busy
4 - Forward on Busy/No Answer
0 - Cancel Auto Attendant
3. If the system phone does not have an Auto Attendant key it is also possible to ‘forward all calls’ to the voice mail, this is achieved by dialling the service code for 'Voice Mail Centre Access' (default =884), entering your mailbox number (and password if enabled) and dialling option code 62#, this is a toggle between forward all and cancel.

Forwarding/Cancelling forward to voice mail from an SLT
1. An SLT user can forward their calls to voice mail by dialling the service code for 'Automated Attendant (DSPDB) (default= 790). Dial the Service code then dial one of the following options:
1 - Forward All Calls
2 - Forward on No Answer
3 - Forward on Busy
4 - Forward on Busy/No Answer
0 - Cancel Auto Attendant
2. Dial the Voice Mail Centre Access service code (default - 884) dial option code 62#, this acts as a toggle between forward and cancel.

Setting greeting messages
1. If calling from a system phone with Voice Mail Access key for the mailbox (code 67 + mailbox) press the key, if calling from an SLT or a System Phone without Voice Mail access key dial the Voice Mail Center Access code (default - 884) and enter your mailbox number.
2. There are 3 greeting messages available for each mailbox. Service codes are available to record, play or erase each individual message.
The codes are :-
Message Number Function Service code
Message 1 Play 31#
Message 1 Record 32#
Message 1 Erase 37#
Message 2 Play 35#
Message 2 Record 33#
Message 2 Erase 38#
Message 3 Play 36#
Message 3 Record 34#
Message 3 Erase 39#

Selecting message to play to incoming callers
1. If calling from a System Phone with ‘Change Attendant Message’ key for the mailbox (code 71 + mailbox) press the key, if calling from an SLT or a System Phone without ‘Change Attendant Message’ key the answering message cannot be changed.
2. The ‘Change Attendant Message’ key toggles to select the answering message and the lamp indicates which message is selected :-
Message 1 = Lamp extinguished
Message 2 = lamp steady
Message 3 = lamp flashing

If 'Automatic Busy response Message' feature has been set at installation in <u>Message Recording Setup</u>, answering message 3 will play when the extension is busy, therefore messages 1 & 2 should only be used for normal answering.

Setting Password
You can set a four digit password to your mailbox.
1. If calling from a system phone with Voice Mail Access key for the mailbox (code 67 + mailbox) press the key, if calling from an SLT or a system phone without Voice Mail access key dial the Voice Mail Centre Access code (default - 884) and enter your mailbox number (and password if enabled).
2. Dial service code 65#
3. Enter your new four digit password followed by #.
4. Your password will be spoken back to you followed by the prompt ”Please dial 0# for yes, or dial 1# for no”.
5. Enter 0 then # to confirm.

6. You will be prompted “your password has been registered. Service code please”
7. Hang up.

Deleting your password

1. If calling from a system phone with Voice Mail Access key for the mailbox (code 67 + mailbox) press the key, if calling from an SLT or a System Phone without Voice Mail access key dial the Voice Mail Centre Access code (default = 884) and enter your mailbox number (and password if enabled).
2. Dial service code 65#.
3. At password prompt dial ‘9999’ followed by #.
4. You will be prompted “your password will be erased, dial 0# for yes, or dial 1# for no”.
5. Dial 0 then #.
6. You will be prompted that you password has been erased.

Listening to messages

1. If calling from a system phone with Voice Mail Access key for the mailbox (code 67 + mailbox) press the key, if calling from an SLT or a System Phone without Voice Mail access key dial the Voice Mail Centre Access code (default = 884) and enter your mailbox number followed by #. You may also be prompted to enter your password, press # after entering the password.
2. Upon entry from System Phone using Voice Mail Access key the messages are automatically played back, if entering using the service code dial Service code 1# to listen to messages.

Record a message to send to a mailbox

1. Lift handset and dial Voice Mail Centre Access code (default = 884).
2. Upon answer the voice mail will respond with the prompt “This is Voice service centre, The mail box number please”
3. Dial *, the mailbox number you wish to send a message to, then #.
4. Record message
5. Hang up

Record a message to send to a Broadcast List.

It is possible to send a message to up to 100 mailboxes simultaneously. This feature is set up at installation and can create up to 10 Broadcast lists containing up to 100 mailboxes in each using Voice Mail Broadcast List.

1. If calling from a system phone with Voice Mail Access key for the mailbox (code 67 + mailbox) press the key, if calling from an SLT or a System Phone without Voice Mail access key dial the Voice Mail Centre Access code (default = 884) and enter your mailbox number (and password if enabled).
2. Dial service code 2 followed by #.
3. Dial abbreviated Broadcast list number.
4. Record Message.
5. Hang up.

Setting Message Notification

It is possible to set notification if a message is left in your mailbox. The notification can call an extension or an external number.

1. If calling from a system phone with Voice Mail Access key for the mailbox (code 67 + mailbox) press the key, if calling from an SLT or a System Phone without Voice Mail access key dial the Voice Mail Centre Access code (default = 884) and enter your mailbox number (and password if enabled).
2. Dial Service Code 61 followed by #.
3. You will be prompted “Dial 1# for an extension call, 2# for an outside call, or 0# to cancel.
4. If either 1# or 2# is selected enter number to be dialled followed by # (it is not necessary to enter the trunk access code for an external number).
5. The dialled number will be read out. Press 0# to confirm.

Cancelling Message Notification

1. If calling from a system phone with Voice Mail Access key for the mailbox (code 67 + mailbox) press the key, if calling from an SLT or a System Phone without Voice Mail access key dial the Voice Mail Centre Access code (default = 884) and enter your mailbox number (and password if enabled).
2. Dial service code 61 followed by #.
3. You will be prompted “Dial 1# for an extension call, 2# for an outside call, or 0# to cancel.
4. Dial 0 then #.
5. Dial 0 then # to accept.
6. You will then be prompted “the setting has been cancelled”.

Conversation Recording

Conversation record is set up at install with the option of either Automatic or manual recording.

Automatic conversation record automatically records the call upon answer whereas manual conversation recording requires the depression of a function key to evoke the service.

To perform manual conversation record a key must be programmed on the System Phone. The Conversation Record Start/Stop key is code 69+0.

In order to start manual conversation recording simply press the Conversation Record key during a telephone conversation.

Conversation recording can also be configured at installation with Conversation recording Destination for Trunk or Conversation recording Destination for Extensionto either record to your mailbox or to define the mailbox after the call has finished via a ‘call back’. If the call back option is specified the voice mail will call you back after the conversation has ended and you to specify a mailbox for storage, after specify a valid mailbox followed by # the voice mail will confirm the message has been saved. If you do not answer the call back the recorded conversation will be deleted, if you do not specify the mailbox number during the call back the recorded conversation will be deleted.

Voice Mail access from external
It is possible to ‘log in’ to your mailbox from an external location.
Your extension must have call forward to voice mail set.
1. Dial into the system to a line that will route to your extension. You will hear your personal greeting.
2. Listen to your greeting.
3. After your greeting has played two short beeps will be heard, enter your password followed by #.
4. You will then have access to the service code options for your mailbox.

To record a trunk answering Guidance message:
1. Press idle CALL key.
OR
At a single line telephone, lift handset.
2. Dial 716.
3. Dial 7 (Record).
4. Dial the Guidance message number you want to record (01-48).
Ensure that the message number selected for the Guidance is not used for other VRS functions e.g. queue messages etc.
5. When you hear, “Please start recording” followed by a beep, record your message.
Normally, your message cannot exceed 16 seconds. If you hear, “Recording finished,” you have exceeded the allowed message length.
6. Press # to end recording
OR
Hang up to save the message.

To listen to a previously record trunk answering Guidance message:
1. Press idle CALL key.
OR
At a single line telephone, lift handset.
2. Dial 716.
3. Dial 5 (Listen).
4. Dial the VRS message number to which you want to listen (01-48).
You’ll hear the previously recorded message. If you hear a beep instead, there is no previous message recorded.
5. Press # to hear the message again.
OR
To hear another message, press 5 and then enter the message number (01-48).
OR
Hang up.

To erase a previously recorded trunk answering Guidance message:
1. Press idle CALL key.
OR
At a single line telephone, lift handset.
2. Dial 716.
3. Dial 3 (Erase).
4. Dial the number of the VRS message you want to erase (01-48).
5. Press HOLD (system phone only) to cancel the procedure without erasing (and return to step 3).
OR
Hang up to erase the message.

Warning Tone for Long Conversation

Description
The system can broadcast warning tones to a trunk caller warning them that they have been on the call too long. The tones are just a reminder, the user can disregard the tones and continue talking if they choose. The outside caller does not hear the warning tones. In addition, warning tones do not occur for Intercom calls and most incoming trunk calls. DISA trunks can also have warning tones. Warning tones are not available to analogue single line telephone (SLT) users.

There are two types of warning tones: Alarm Tone 1 and Alarm Tone 2.

Alarm Tone 1 is the first set of tones that occur after the user initially places a trunk call.
Alarm Tone 2 broadcasts periodically after Alarm Tone 1 as a continued reminder.
Each alarm tone consists of three short beeps.

Warning Tone for DISA Callers
For DISA callers, with this feature enabled, the warning tone timer begins when an incoming DISA call places an outgoing call and either the inter-digit timer expires or the outgoing call is answered.
With this feature enabled, the warning tone timer begins when an incoming DISA call places an outgoing call and either the inter-digit timer expires or the outgoing call is answered.

If an outside call is transferred or forwarded off-premise using an outside trunk, the warning tone timer begins immediately. This will occur only if either trunk involved in the call is programmed for this feature ('Trunk to Trunk Long Conversation Alarm' in Trunk Basic Data Setup)

Conditions
Warning Tone for Long Conversation does not occur for Intercom calls.
Warning tones are not available to single line telephone (SLT) users.

Default Setting
'Long Conversation Alarm Before Cut-off' is disabled in Trunk Basic Data Setup.
'Trunk to Trunk Long Conversation Alarm' is disabled in Trunk Basic Data Setup.
'Long Conversation Alarm' is on in Class of Service.
'Long Conversation Alarm 1' is set to 170 seconds in System Timers.
'Long Conversation Alarm 2' is set to 180 seconds in System Timers.
'DISA Conversation Warning tone time is set to 30 seconds in System Timer for DUD/DISA Service.
'Continue Code for DISA Trunk to Trunk' is not assigned in DISA Service Options.
'Disconnect Code for DISA Trunk to Trunk' is not assigned in DISA Service Options.

Related Features
Central Office Calls, Answering: Warning Tone for Long Conversation does not occur for incoming trunk calls.
Central Office Calls, Placing / Toll Restriction: Warning Tone for Long Conversation occurs for all outgoing trunk calls, regardless of how they are placed or other outgoing restrictions.
Direct Inward System Access (DISA): Warning Tone for Long Conversation can be enabled for DISA calls.
Long Conversation Cutoff: Warning Tone for Long Conversation can be used with the Long Conversation Cutoff feature for outgoing calls.

Operation
Warning Tone for Long Conversation is automatic if programmed.

Warning Tone for Long Conversation for DISA Callers:
1. A DISA caller dials into the system and places a call.
2. Once the Warning Tone is heard,
To continue the call, the DISA caller presses the Continue Code.
OR
To disconnect the call, the DISA caller presses the Disconnect Code.

XN120

Getting Started Guide
Rev 1.5 (May 2006)

This guide explains the installation, configuration and operation of the XN120 Telephone System including the exchange line and telephone connections.

System Specifications

Maximum System Capacities

System Parts	Description	Maximum Capacities	Notes
XN120 Main Unit		1	
XN120 Expansion Unit		2	
DSPDB Card	Voice Mail and Announcement card	1	1 card in the Main unit
008/308 Card	008 = 8 ST ports 308 = 3 CO ports + 8 ST ports	6	2 of either card per Main or Expansion unit
PGDU Card	2 External Paging ports 2 Door phone ports 2 Audio ports	3	1 card per Main or Expansion unit
EXIFU Card		1	1 card in the Main unit
2OPBox	Option box for BRIU and VoIPU cards	3	1 box per Main or Expansion unit
BRIU Card	Basic Rate ISDN	6	2 cards per 2OPBox
VOIPU Card	VoIPU resource	6	2 cards per 2OPBox

Station Interface	Description	Maximum Capacities	Notes
ST port	Hybrid extension port	72	
BRIU S0	Basic Rate ISDN S-bus	48 channels (24 circuits)	2 channels per BRI circuit
SIP Telephone	VoIP Telephones	24	
Virtual Extension		50	

Trunk Interface	Description	Maximum Capacities	Notes
CO port	Analogue trunk	27	
BRIU	Basic Rate ISDN	40	2 channels per BRI circuit
VoIP Trunk	VoIP trunk resource	40	

Terminals	Description	Maximum Capacities	Notes

System telephones		72	
Single Line telephones		72	
64 Button DSS console		9	3 per Main/Expansion unit
24 Button Console		72	1 per Display phone

Cabling Requirements

Terminal	Max loop resistance	Maximum Cable Length	Notes
System telephones	54 Ohms	300 Metres	2-pair cable (0.5mm 24 AWG)
Single Line telephones	267 Ohms	1500 Metres	1-pair cable (0.5mm 24 AWG)
64 Button DSS console	54 Ohms	300 Metres	1-pair cable (0.5mm 24 AWG)
Door phone unit	27 Ohms	150 Metres	1-pair cable (0.5mm 24 AWG)
EXIFU Ethernet port		100 Metres	2-pair Ethernet cable
EXIFU Serial port		15 Metres	Serial cross cable
VOIPU Ethernet port		100 Metres	2-pair Ethernet cable
ISDN Terminals	18 Ohms 54 Ohms 90 Ohms	100 Metres P-MP Short bus 300 Metres P-MP Long bus 500 Metres P-P	2-pair cable (0.5mm 24 AWG)
XN120 Expansion Unit		1.25 Metre	Cable supplied with

Electrical Specifications

Unit		Notes
XN120 Main Unit And XN120 Expansion Unit	Input Voltage: 85 to 264 VAC Frequency: 50 / 60 Hz Power requirement: 1.8A @ 100VAC (180 VA) 1.0A @ 240VAC (216 VA) Power consumption: 120 W Phase: Single Conductors: 2-Wire Grounding: 14AWG Copper wire (2.5mm^2)	Separate AC supplies are required for each Main and Expansion unit.
Battery Backup charging voltage	Output Voltage: 27 VDC Output current: 0 to 0.2A	
Fuse replacement	Main / Expansion units: There are no installer replacable fuses. Battery box: 4A 20mm	Normal blow glass tube fuse.
Battery Types	XN120 Main unit: 1 x 3V Lithium for memory back up. Battery Box: 2 x Valve regulated Lead Acid for AC power failure backup.	CR2450 or CR2032 depending on revision of unit. 12V 2.6Ah (Yuasa NP2.6-12) 134x67x64mm

Environmental Requirements

Unit		Notes
XN120 Main Unit, XN120 Expansion Unit. Battery Box. Cards. XN120 Telephones.	Temperature: 0 to +40 °C (32 to 104 °F) Humidity: 10 to 90% (non-condensing)	
Doorphone unit	Temperature: -20 to +60 °C (-4 to 140 °F) Humidity: 20 to 80% (non-condensing)	

Weights and Dimensions

Unit	Width (mm)	Depth (mm)	Height (mm)	Weight (Kg)
Main/Expansion unit	360	90	275	2.8 (fully equipped)
2OPBox	146	86	279	1.1 (fully equipped)
Battery box	384	106	99	5.2 (fully equipped)
Display telephone	178	219	84 (low) 115 (high)	0.83
System telephone	178	219	84 (low) 115 (high)	0.82
64 Button DSS console	315	177	59	0.48
24 Button console	212	60	59	0.2
Door phone unit	100	35	132	0.2

External Equipment Interfaces

Unit		Notes
Doorphone unit	Output Impedance: 270 Ohm Output level: Nominal 250mV (-10dBm) Maximum Output: 400mV RMS	
PGDU relay contacts	Relay configuration: Normally open Contact rating: 24VDC, 0.5A and 120VAC, 0,25A	
External paging	Impedance: 600 Ohm @ 1kHz Nominal output level: 250mV (-10dBm) Maximum Output: 400mV RMS	
External Music on Hold	Impedance: 600 Ohm @ 1kHz Nominal input level: 250mV (-10dBm) Maximum input: 1V RMS	

Analogue Trunk and Station port Specification

Unit		Notes
Analogue trunk	Caller ID Receiver:	EN 300659-1 V1.3.1 (2001-

	Caller ID will be detected by either FSK or DTMF. Single Message Format (number) or Multiple Message Format (number and name) are supported. FSK or DTMF detection can be selected per CO port.	01) Clause 6.1.1. YD/T 1277.1-2003 5th June 2003
	Dialled Digits: Dial pulse 10pps or 20pps DTMF (1-9, * and #)	
Analogue station	Caller ID Sender: Caller ID will be sent after a single ring burst. Single Message Format (number) or Multiple Message Format (number and name) are supported. FSK or DTMF sending can be selected per ST port.	EN 300659-1 V1.3.1 (2001-01) Clause 6.1.1 YD/T 1277.1-2003 5th June 2003
	Idle state voltage: 24 VDC Loop current: 20mA	
	Message Waiting Indication: Voltage increase 24 to 100 VDC @ 500mS intervals.	
	Digit detection: Dial pulse: 10pps or 20pps DTMF (1-9, * and #)	

TCP/UDP Port List

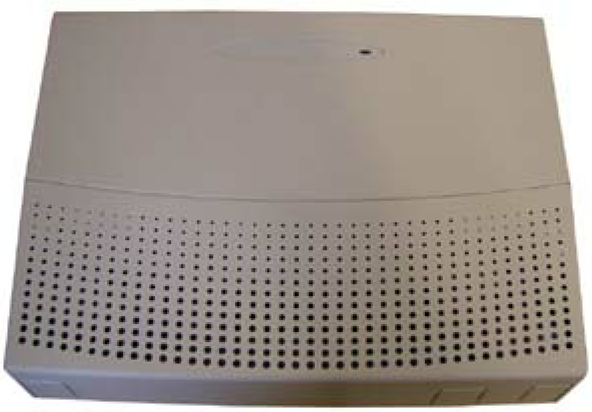
Application	**Type**	**Port**	**System Data**	**Comments**
ACD-MIS	TCP	4000	10-20-02 10-20-01 Device 2	Default value is 0
AspireNet Network Listener	TCP	30000	10-20-02 10-20-01 Device 4	
CTI Server	TCP	8181	10-20-02 10-20-01 Device 1	Default value is 0
DHCP Server	TCP	67	Fixed	
DIM	TCP	5963	Fixed	For NECi use
H.323 Extension GRQ	TCP	1718	Fixed	
H.323 Extension RAS	TCP	1719	Fixed	
H.323 Extension Signalling	TCP	1730	84-02-33	
H.323 H.245	TCP	5600	Fixed	
H.323 Trunk RAS	TCP	20001	84-02-36	
H.323 Trunk Signalling	TCP	1720	Fixed	
IP Phone DRS (Registration)	UDP	3458	84-03-02	

IP Phone H.245	TDP	10100	84-06-03	Increments per concurrent call
IP Phone Ready/Answer	UDP	4000	84-06-11	
IP Phone Signalling	UDP	3456	84-02-01 84-03-02	
PCPro (PC Programming Application)	TCP	8000	Fixed	
RTCP	UDP	10021	84-06-02	Increments per concurrent call
RTP	UDP	10020	84-06-01	Increments per concurrent call
SIP Extension	UDP	5070	84-20-01	
SIP Trunk	UDP	5060	84-14-06	
SMDR	TCP	4001	10-20-02 10-20-01 Device 5	Default value is 0
WebPro (HTML Based Programming)	TCP	80	Fixed	
SIP Sever Registrar	TCP	5060	10-29-07	
DNS Server	TCP	53	10-29-10	
SMTP host	TCP	25	90-11-07	

Main Unit

Top

XN120 Main Unit



Caution:

Only suitably qualified engineers should install and/or maintain this system

The XN120 system consists of a main unit with a base board pre-installed.

The XN120 main unit will allow the connection of up to three analogue exchange lines plus eight telephones. The eight telephones can be either XN120 system telephones or normal telephones.

The XN120 main unit can have an external music source connected that can be played to callers placed on hold.

There is also a connection for an external paging system.

The main unit can also have optional cards to increase the quantity of analogue lines or telephones.

There are option cards for Voice Mail/Queue Announcements, Door phones/Door locks, additional external music devices and external paging systems.

You can connect an optional unit that will allow the installation of ISDN BRI cards or VoIP cards.
Further more, there are additional expansion units available to increase the number of exchange lines and telephones that can be connected.
All equipment will operate when the XN120 is installed as shown in this guide.
With the default settings:

1. • Each telephone will function and is assigned an extension number.
2. • Calls received on the exchange lines will ring at telephone number 200.
3. • Each telephone can make outside calls by dialling 9.
4. • Each exchange line is presented at a Function Key with busy lamp indication.

System Capacities

The maximum capacities of the XN120 system are shown below.
Other factors may limit the maximum quantity.

Description	Maximum Quantity	Comments
XN120 Main unit	1	
XN120 Expansion unit	2	
Exchange lines	51	27 Analogue lines 40 BRI lines (20 circuits)
Telephones	72	System phones, normal phones and 64 button consoles
24 Button Add On console	72	Connects to a 22 button display phone
64 Button console	9	Connects to port 8 of each unit/card
Door phone unit	6	2 per 2PGDU card
External paging zones	6	2 connections per 2PGDU card and 1 on the XN120 main unit

Analogue lines

You can connect up to nine analogue exchanges lines to the XN120 main unit.

- The exchange lines must be loop start type.
- The XN120 will also detect Caller ID sent by the Network Provider. The Caller ID must be the ring alert type with FSK or DTMF signalling.
- Each line is connected via an RJ11 6/4 way socket.

XN120 System Telephones

There are four types of XN120 System phones available.

Feature	**Available on the XN120 22-button or 16-button system phone**		**Available on the XN120 22-button or 16-button display phone**	
Programmable Keys	Yes	12 with lamps + 10 without lamps 6 with lamps + 10 without lamps	Yes	12 + 10 with lamps 6 + 10 with lamps
LCD Display	No		Yes	2 lines x 20 character
Hands free	No	Talkback only	Yes	
Accept 24 button console	No		Yes	
Wall mount kit	Yes	Built in	Yes	Built in

You can connect the XN120 system telephones to any of the station connections labelled ST. (If you need more telephones you can install optional cards). The XN120 system telephones have feature keys and illuminated function keys that can be tailored to your own requirements.

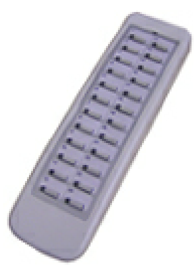
The following headsets are the only supported headsets for the XN 120 System telephones.

Compatible Headsets
Plantronics S12 (Adapter and headset)
Plantronics M12E Vista Base with any H-series headset.
Other headsets may not operate correctly

The XN120 system telephones with an LCD display will show information about who is calling you, the call you are on or the feature you are using.

- Each telephone is connected to the XN120 via an RJ11 6/4 way socket.

There are also two types of XN120 Consoles available.

Feature	Available on the 24 Button add on console		Available on the 64 Button DSS console	
Programmable Keys	Yes	24 with lamps	Yes	64 with lamps
Fixed feature keys	No		Yes	14 keys
Connection method	Connects into any XN120 display phone.		Connects into a hybrid extension port via an RJ11 cable.	
Wall mount kit	No		No	
				

XN 120 Tel 2 System Telephones

XN 120 System phones have been enhanced with the introduction of the Tel 2 variant.

The enhancements are:

- Group Listening - by pressing the SPK key whilst using the handset co-workers can listen to the conversation (dependant on setting of PRG 20-13-26),
- Dual Colour LED's - the block of 10 programmable function keys are dual colour on the display telephone,
- Multiple Incoming Ring Patterns - The ability to select from 7 ring patterns,

- "Month" addition to idle display indication,
- Electret Microphone Headset support - a wider range of headsets are now supported.

N.B. the 24 button add-on console is connected to the TEL 2 variant System Phone via an RJ11 style connector. Ensure that the correct add-on console is available prior to attempting to connect it. Do not attempt to modify the connector to fit an old style console to a new style system phone or vice versa.

- The part number can determine a new or older type, as the new items were given a different part number as indicated in the tables below.

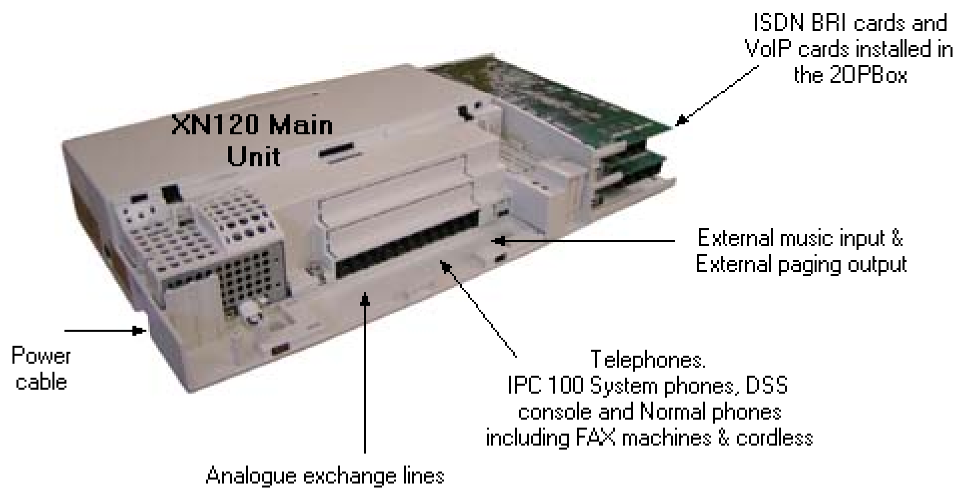
TEL Part Numbers - Pre April 2006 NO RJ11 Connectors for 24DLS / 60DSS		TEL2 Part Numbers – Post April 2006 WITH RJ11 Connectors for 24DLS / 60DSS	
006310-5	IP2AT-6TD TEL (WH)	006341-5	IP2AT-6TD TEL2 (WH)
006316-5	IP2AT-6TD TEL (BK)	006342-5	IP2AT-6TD TEL2 (BK)
006311-5	IP2AT-6TXD TEL (WH)	006343-5	IP2AT-6TXD TEL2 (WH)
006317-5	IP2AT-6TXD TEL (BK)	006344-5	IP2AT-6TXD TEL2 (BK)
006312-5	IP2AT-12TD TEL (WH)	006345-5	IP2AT-12TD TEL2 (WH)
006318-5	IP2AT-12TD TEL (BK)	006346-5	IP2AT-12TD TEL2 (BK)
006313-5	IP2AT-12TXD TEL (WH)	006347-5	IP2AT-12TXD TEL2 (WH)
006319-5	IP2AT-12TXD TEL (BK)	006348-5	IP2AT-12TXD TEL2 (BK)
006314-5	IP2AT-64D DSS CONSOLE (WH)	006349-5	IP2AT-64D DSS CONSOLE2 (WH)
006320-5	IP2AT-64D DSS CONSOLE (BK)	006350-5	IP2AT-64D DSS CONSOLE2 (BK)
006315-5	IP2AT-24DL DLS CONSOLE (WH)	006351-5	IP2AT-24DL DLS CONSOLE2 (WH)
006321-5	IP2AT-24DL DLS CONSOLE (BK)	006352-5	IP2AT-24DL DLS CONSOLE2 (BK)

Normal Telephones

You can connect normal telephones or cordless phones to any of the station connections.

- The telephones can be dial pulse or DTMF dialling.
- They can have Hook Flash or Timed Break Recall.
- The XN120 can send Caller ID to the normal telephone.

System Connection Diagram



Optional Items	
008 Expansion Card 8 Extension ports Installed onto the main board in the XN120 unit. **308 Expansion card** 3 Exchange lines + 8 Extension ports Installed onto the main board in the XN120 unit. **2PGDU Card** Installed onto the main board in the XN120 unit. For 2 door unit and 2 external page amplifiers, external MOH inputs, or audio input/output. 2 Relay contacts available that can be assigned to any of the door units or audio ports. **DSPDB** Voice Mail and Voice response unit (Queue Announcements). **EXIFU Card** For adding expansion cabinets, Call Logging, Ethernet connection and CF card slot. There is also an EXIFU card with only Call Logging. Installed onto the main board in the XN120 unit.	**2OPBOX** For ISDN BRIU and/or VOIPU cards. 2 slots available per unit. Plugs onto the right side of the XN120 unit. **BRIU Card** ISDN Basic Rate cards (2 or 4 circuits available) can be installed in the 2OPBOX. **XN120 Expansion cabinet** For more lines, stations and BRI. Provides the same card capacity as the main cabinet. Each expansion cabinet requires a dedicated power cable. Connected to the EXIFU card in the Main XN120 unit via an RJ45 cable. **64 Button DSS console** Provides 64 programmable keys and 14 fixed feature keys. Gives operator type functionality. **24 Button Add On console** Provides 24 additional function keys for the XN120 display phones.

Unpack the XN120 Main Unit.

1 x XN120 system
1 x Wall mounting template
1 x Power cord
4 x Fixing screws

Plastic Spacer (Required when EXIFU is installed)
Getting Started Guide

Additional Items Required:

- Cross head screwdriver.
- 4 Wall fixing plugs suitable for the type of wall.
- Solid wire for extending telephone cabling:
  Recommended cable type: Twisted pair (CW1308 or similar specification)
  Conductor diameter: 0.4 to 0.6 mm
  Maximum cable length: (with 0.5 mm diameter cable)
    XN120 system telephone – 300 metres
    Normal telephone – 1500 metres

If you need to extend the exchange line cables:

- Solid wire for exchange line cables:
- Recommended cable type: Twisted pair (CW1308 or similar specification)
- Conductor diameter: 0.4 to 0.6 mm

Replacing an Existing Telephone System

If you are replacing your existing telephone system with the XN120 we recommend that you check the following.

- Do not disconnect all of the lines or extensions from your existing telephone system. If for any reason you have problems installing the XN120 you will need your old system in working order to continue your business.
- If you plan to use existing telephone cabling within your building check:
  - The cable is similar to CW1308 specification (see the Glossary for help).
  - There are 4 wires (2 pairs) available to each XN120 system phone location.
  - You will need an RJ11 socket for each XN120 system phone and 64 Button DSS console.
- Move the exchange lines / normal telephones one at time and test each one before moving over the next.

Wall Mount the XN120 system

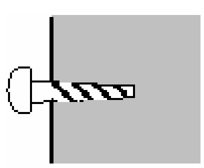
Installation Considerations:

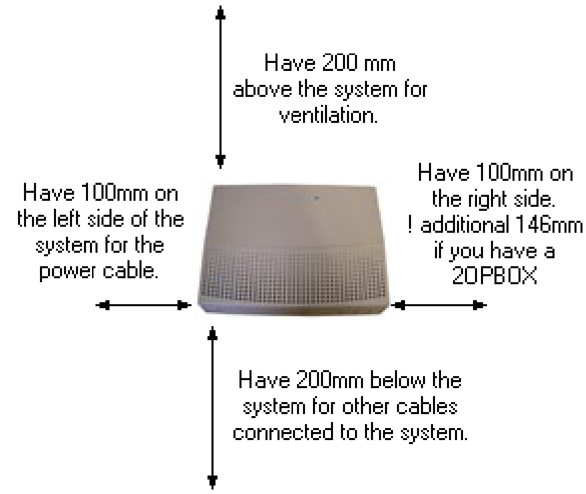
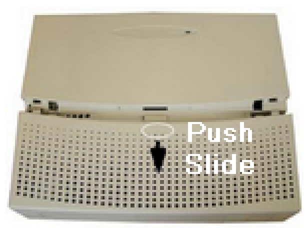
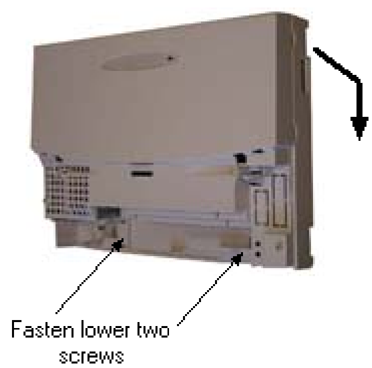
- To avoid electric shock or damage do not plug in or turn on the system power before completing the installation.
- Avoid working with the system during electrical storms.
- Use the power cord supplied with the product.
- Do not bundle power cords together, the cords may overheat.
- Ensure the system has a suitable Earth Ground connection.

Environmental Considerations – Be sure the system is not:

- In direct sunlight or in hot, cold or humid places.
- In dusty areas or in areas where sulphuric gasses are produced.
- In places where shocks or vibrations are frequent or strong.
- In places where water or other fluids may come into contact with the equipment.
- In areas near electric welders or machines that emit high frequency radiation.
- Near computers, microwaves, air conditioners etc.
- Near radio antennas (including shortwave).

- If you are installing the optional expansion box or expansion cabinets ensure there is sufficient wall space and ventilation. Refer to the wall mounting diagrams below.

1	You will need a minimum of 560mm x 680mm (W x H) wall space for the XN120 system. The system is 360mm x 279mm (W x H)	Use the wall mount template supplied to mark the four screw locations. Ensure that you use the correct wall plugs for the type of wall. Leave approximately 3~5mm of the screw protruding from the wall. 

	! You will need more space if you are installing the optional units: • 2OPBox • Expansion cabinets	
2	Remove the sub cover of the XN120. Hook the XN120 onto the wall mount screws. Tighten the lower screws to secure the unit to the wall.	 
3	Earth the XN120 system. ***Caution:*** If this cable is not installed or requires disconnection, then the telecommunication network connection(s), CO and/or ISDN, must be disconnected <u>first</u> from the XN120.	***Important.*** The system <u>must</u> have a permanent Earth Ground connection to a verified Earth point using a minimum of 2.5mm^2 green/yellow cable. The Earth connection must have no other purpose than connecting to the XN120 unit. The Earth point is located on the left side under the sub cover. 

Connect the Telephones

Precautions for Cabling:

- Do not run the cable with a power cable, computer cable etc.
- Do not run the cable near any high frequency generating equipment.
- Use cable protectors if the cables are run on the floor.
- Aerial distribution wiring is not allowed.

Connecting the XN120 System Phones.

Connecting the XN120 System Phones

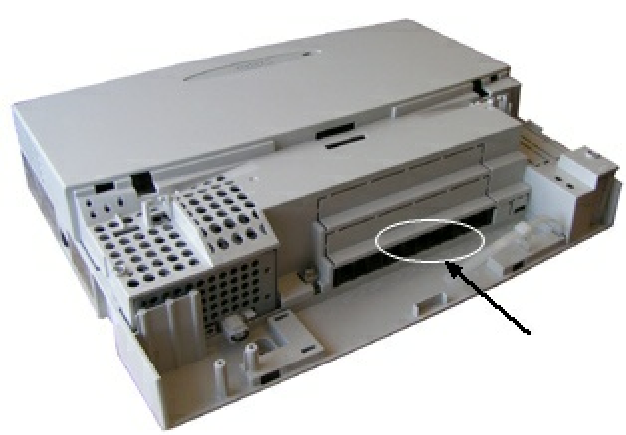
The XN120 system phones can be connected to any of the eight RJ11 telephone sockets labelled ST in the XN120 unit.

! Check Power Fail Operation later in this guide, ST8 can be used to provide a working telephone should the power be disconnected from the XN120.

<table>
<tr><td></td><td>Use the line cord supplied with the system telephone.</td><td>The line cord supplied with the XN120 system telephones can be plugged directly into the RJ11 sockets labelled ST in the XN120 unit.</td></tr>
<tr><td></td><td>OR</td><td></td></tr>
<tr><td></td><td>Use RJ11 to bared wire cables and RJ11 sockets.

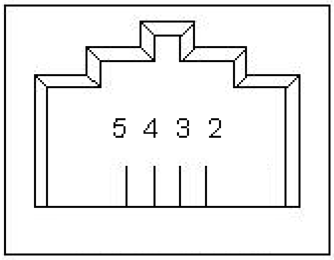
Connections:
<table>
<tr><th>XN120
ST socket</th><th>RJ11 socket</th></tr>
<tr><td>Pin 2</td><td>Pin 2</td></tr>
<tr><td>Pin 3</td><td>Pin 3</td></tr>
<tr><td>Pin 4</td><td>Pin 4</td></tr>
<tr><td>Pin 5</td><td>Pin 5</td></tr>
</table>
! The connections are polarity sensitive - you must connect as shown.

! Ensure that you use a separate twisted pair for the speech and data:
Pins 3 & 4 (speech)
Pins 2 & 5 (data)</td>
<td>If you need to extend the telephone cables:
Fit an RJ11 socket at the location the telephone is required, run a telephone cable back to the distribution frame. Connect to one of the sockets labelled ST using an RJ11 plug.
• Ensure that you connect the 4 wires as shown otherwise the telephone will not initialize.
• Use CW1308 or similar telephone cable.
• Maximum cable length is 300 metres (with 0.5 mm diameter conductor).


XN120 RJ11 socket (ST)</td></tr>
</table>

This will ensure that you can reach the maximum cable length of 300 metres.

! Identify each RJ11 socket with its connection number (e.g. ST1). This will help you configure the system later.



Route the cables into the lower side of the system. Use cable ties to secure the cables.

Remove the breakouts of the sub cover before re-fitting onto the system.

OR

Use RJ11 to bared wire cables and RJ45 sockets.

Connections:

XN120 ST socket	RJ45 socket
Pin 2	White/green
Pin 3	Blue/white
Pin 4	White/blue
Pin 5	Green/white

! The connections are polarity sensitive - you must connect as shown.

! Ensure that you use a separate twisted pair for the speech and data:
Pins 3 & 4 (speech)
Pins 2 & 5 (data)
This will ensure that you can reach the maximum cable length of 300 metres.

! Identify each RJ45 socket with its connection number (ST

Refer to the diagram above for the pin numbering of the XN120 RJ11 ST sockets.

	number). This will help you configure the system later.	

Use of BT LJU with XN120 Key Telephone

Although we do not recommend or support a connection using BT 431A or 631A plug/socket on the XN120 (and therefore will not provide such line cords) - we have KTL test reports that indicate the XN120 Key Telephones can be connected in this manner as regards the safety issue specified below.
This report covers the user safety issue as regards subsequent connection of such a Key Telephone to the public network. This is, the Key Telephone will not work or may be damaged, but the XN120 key telephone to public network connection will not be unsafe.
The BT 431A or 631A plug/sockets pin-outs are shown on later pages of the publication.

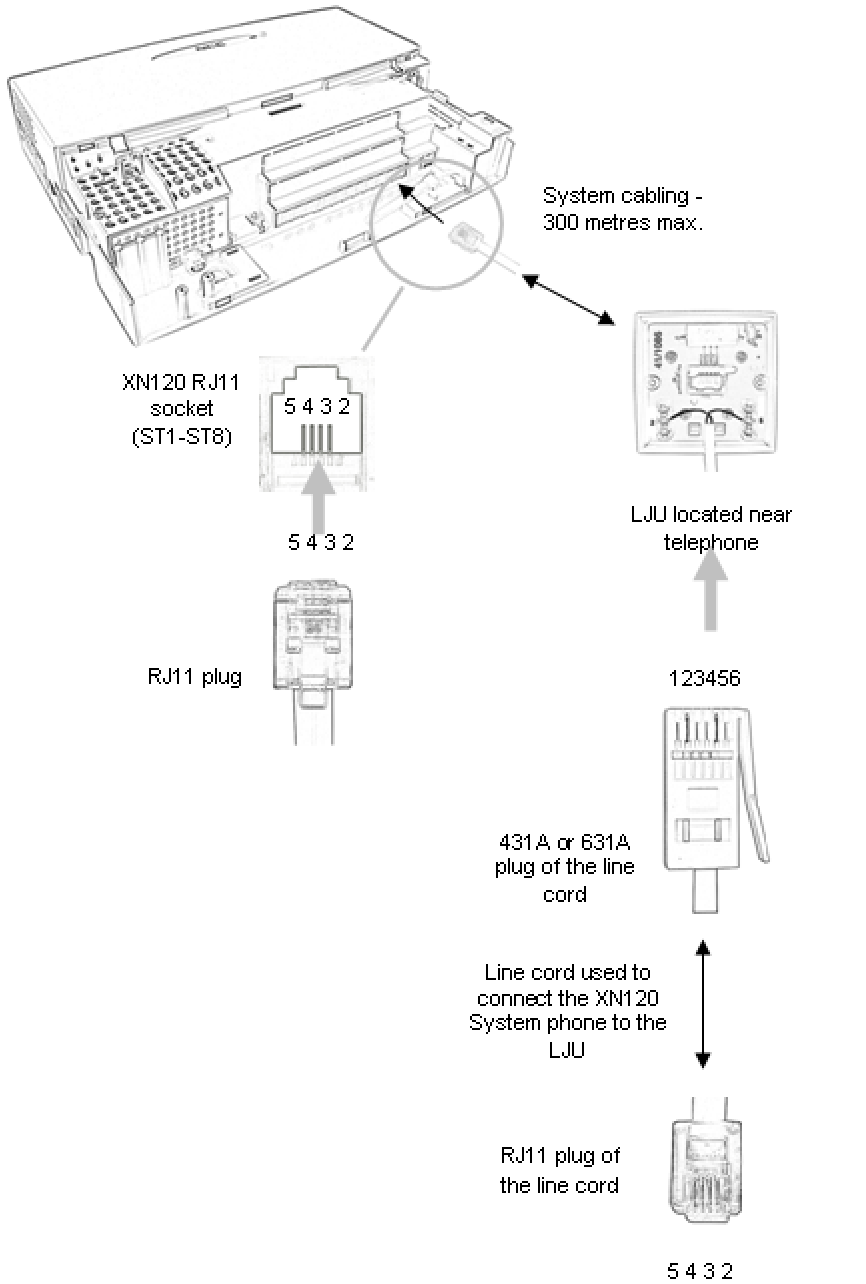
Similarly an XN120 Key Telephone will not work or may be damaged if connected to an analogue extension port of a PABX
Please note that unauthorised modification of any NEC Infrontia equipment may invalidate the standard warranty.

	Pin connections for the XN120 System phones when using a Master/Slave Line Jack Unit. Connections: The pin connections between the RJ11 socket of the ST ports on the XN120 and the LJU depend on the pin connections of the line cord used to connect the XN120 system phone. See below for options. ***!*** The connections are polarity sensitive - you must connect as shown. ***!*** Ensure that you use a separate twisted pair for the speech and data: ST Pins 3 & 4 (speech) ST Pins 2 & 5 (data) This will ensure that you can reach the maximum cable length of 300 metres. ***!*** Identify each LJU with its connection number (ST number). Line Cord Type: Some line cords (Analogue SLT	• Ensure that you connect the 4 wires as shown otherwise the telephone will not initialize. • Use CW1308 or similar telephone cable. • Maximum cable length is 300 metres (with 0.5 mm diameter conductor).

type) will use pins 3 and 4 within the LJU; if this is the case you must ensure the LJU is not a Master type (with bell capacitor). The bell capacitor is intended for analogue SLT's and will cause noise on the speech path of the XN120 System phone.

! We recommend that you use a line cord of the type intended for system phones.

4 Wire Line Cords:
Some line cords may only have 2 wires; these are not compatible with the XN120 System phones as 4 wires are required.



The pin connections of the line cord may vary, below are the connections for some of the typical line cords available.

Line cord (System phone type)	
RJ11 Plug	431A / 631A plug
2	6
3	2
4	5
5	1

XN120 ST socket (RJ11)	LJU socket
Pin 2 black	6
Pin 3 red	2
Pin 4 green	5
Pin 5 Yellow	1

Line cord (Analogue SLT type)	
RJ11 Plug	431A / 631A plug
2	3
3	2
4	5

5	4

XN120 ST socket (RJ11)	LJU socket
Pin 2 black	3
Pin 3 red	2
Pin 4 green	5
Pin 5 Yellow	4

If you are unsure that a Master type LJU is not used with an analogue SLT type line cord, fit a slave type LJU.

Connecting the Normal Phones

The normal phones can be connected to any of the eight RJ11 telephone sockets labelled ST on the XN120.

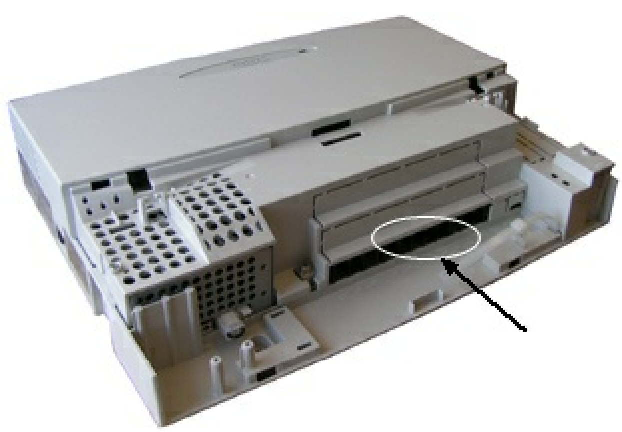
The XN120 is supplied with RJ11 to bared wire cables for connection to optional distribution frames.

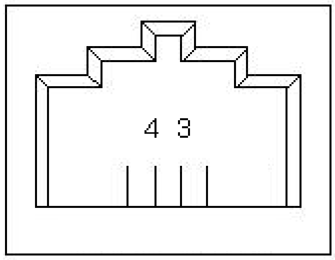
! Check Power Fail Operation later in this guide, ST8 can be used to provide a working telephone should the power be disconnected from the XN120.

<table>
<tr><td></td><td>Use the line cord supplied.</td><td>If the line cord supplied with the telephone has an RJ11 plug it can be plugged directly into the RJ11 sockets labelled ST in the XN120 unit.</td></tr>
<tr><td></td><td>OR</td><td></td></tr>
<tr><td></td><td>Use RJ11 to bared wire cables and Line Jack Units.

Connections:
<table><tr><td>XN120
ST socket</td><td>LJU socket</td></tr><tr><td>3</td><td>2</td></tr><tr><td>4</td><td>5</td></tr></table>! The connections are not polarity sensitive.

! Ensure that you use a twisted pair for the speech:
Pins 3 & 4 (speech)
This will ensure that you can reach the maximum cable length of 1500 metres.</td><td>If you need to extend the telephone cables:
Fit a Master type Line Jack Unit (LJU) at the location the telephone is required, run a telephone cable back to the distribution frame. Connect to one of the sockets labelled ST using an RJ11 plug.
• Ensure that you connect the 2 wires as shown otherwise the telephone will not operate correctly.
• Use CW1308 or similar telephone cable.
• Maximum cable length is 1500 metres (with 0.5 mm diameter conductor).


XN120 RJ11 socket (ST)</td></tr>
</table>

	! Identify each line jack unit with its connection number (ST number). This will help you configure the system later.	 Connect the normal telephone to pins 3 and 4 only, do not make connections to other pins of the ST socket.
	Route the cables into the lower side of the system. Use cable ties to secure the cables.	

Connecting the 64 Button Consoles

The 64 Button Consoles can be connected to the RJ11 telephone socket labelled ST8 on the XN120 (refer to Appendix A for further connection options). The console must be used in conjunction with an XN120 System Telephone (a display phone is recommended).
The console provides 64 buttons with lamps that can be programmed to give quick access to extensions on the system – for operator type working. Buttons can have outside numbers assigned to give the operator quick access to frequently called numbers.
There are also 14 fixed feature keys to give access to system features such as page zones, door phones and changing the Day/Night mode.

! Check Power Fail Operation later in this guide, ST8 can be used to provide a working telephone should the power be disconnected from the XN120. You cannot use socket ST8 for both power fail and a 64 Button DSS console; ST8 can have one function only.
If you want to use the power fail operation and have a 64 Button Console connected you can either install an additional 008/308 expansion card or connect the power fail telephone directly to the exchange line. See Power Fail Options later in this guide.

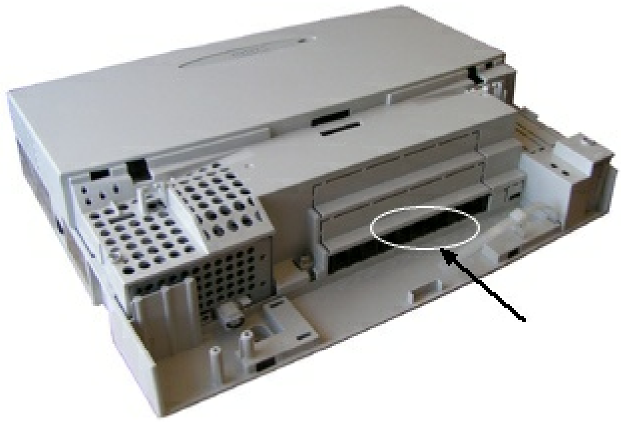
	Use the line cord supplied.	The line cord supplied with the console can be plugged directly into the RJ11 socket labelled ST8 in the XN120 unit.
	OR	
	Use RJ11 to bared wire cables and RJ11 sockets. Connections: <table><tr><td>XN120 ST8 socket</td><td>RJ11 socket</td></tr><tr><td>Pin 2</td><td>Pin 2</td></tr></table>	If you need to extend the telephone cables: Fit an RJ11 socket at the location the console is required, run a telephone cable back to the distribution frame. Connect to the socket labelled ST8 using an RJ11 plug. • Ensure that you connect the 2 wires as shown otherwise the telephone will not operate correctly. • Use CW1308 or similar telephone cable. • Maximum cable length is 300 metres (with 0.5 mm diameter conductor).

Pin 5	Pin 5

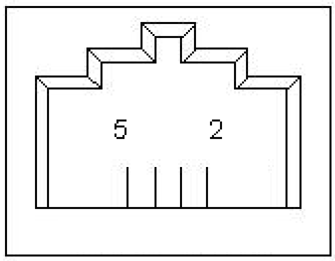
! The connections are polarity sensitive - you must connect as shown.

! Ensure that you use a twisted pair for the console:
Pins 2 & 5 (data)
This will ensure that you can reach the maximum cable length of 300 metres.

! Identify the RJ11 socket with its connection number (ST8) and that it will have the 64 Button Console connected.
Also identify the connection number of the XN120 system phone that the console will be used in conjunction with. This will help you configure the system later.



XN120 RJ11 socket



OR

Use RJ11 to bared wire cables and RJ45 sockets.

Connections:

XN120 ST socket	RJ45 socket
Pin 2	White/green
Pin 5	Green/white

! The connections are polarity sensitive - you must connect as shown.

! Ensure that you use a twisted pair for the console:
Pins 2 & 5 (data)
This will ensure that you can reach the maximum cable length of 300 metres.

! Identify the RJ45 socket with its connection number (ST8) and that it will have the 64 Button Console connected.
Also identify the connection number of the XN120 system phone that the console will be used in conjunction with. This will help you configure the system later.

Refer to the diagram above for the pin numbering of the XN120 RJ11 socket (ST8).

	Note. The 64 Button console uses pins 2 and 5 (data pair) on the XN120. You will not cause any damage if the console is plugged into an RJ11 socket that also has the speech pair connected (pins 3 and 4 of the RJ11 socket).	
	Refer to the separate section within this guide for all other information related to the 64 Button Consoles.	Refer to Appendix A – 64 Button Consoles

Connect the Analogue Exchange Lines

The analogue exchange lines can be connected to any of the three RJ11 sockets labelled CO.

Make a note of each exchange line number (i.e. the number dialled to ring the line) and its CO connection to the XN120. You will need this when you configure the XN120.

! Check Power Fail Operation later in this guide, CO1 can be used to provide a working telephone should the power be disconnected from the XN120.

Precautions for Cabling:

- Do not run the cable with a power cable, computer cable etc.
- Do not run the cable near any high frequency generating equipment.
- Use cable protectors if the cables are run on the floor.
- Aerial distribution wiring is not allowed.

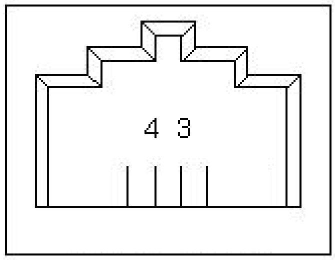
Use lightning protectors for each exchange line.

<table>
<tr>
<td></td>
<td>Use pre-formed telephone cables (not supplied).

Connections:
<table>
<tr><td>XN120
CO socket</td><td>Exchange line connection</td></tr>
<tr><td>3</td><td>T</td></tr>
<tr><td>4</td><td>R</td></tr>
</table>
! The connections are not polarity sensitive.</td>
<td>You can use a separate pre-formed cable to connect each line to a CO socket of the XN120 unit.

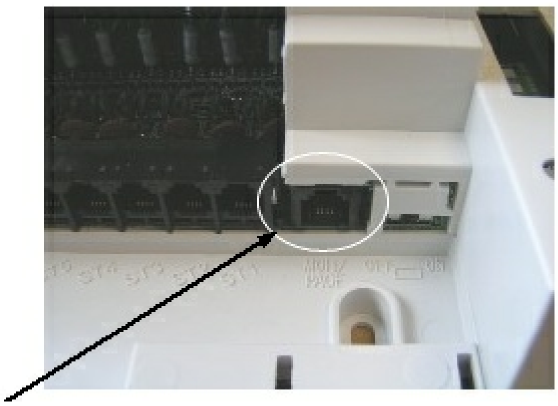
Ensure that you connect the 2 wires as shown otherwise the line will not operate correctly.
Use CW1308 or similar telephone cable.
Fit lightning protectors to each line.


XN120 RJ11 socket (CO)</td>
</tr>
</table>

Connecting the External Music on Hold Device

The external music device is connected to the RJ11 socket labelled MOH/PAGE located under the sub cover to the right of the ST1-ST8 sockets.
The external music device is not supplied with the XN120.



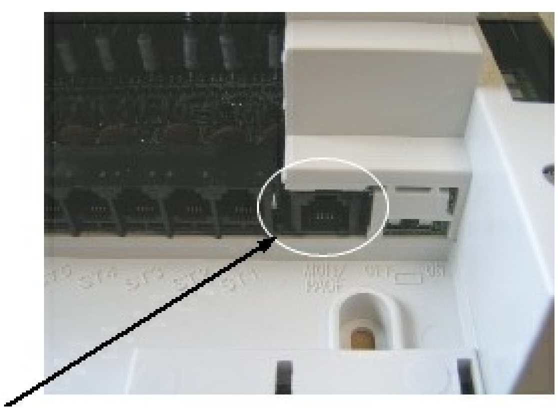
The connections of the RJ11 plug:

XN120 MOH/PAGE socket	Description	Specification
3 4	External MOH input	Impedance: 600 Ohm @1KHz Nominal input level: 250mV (-10dBm) Maximum input level: 1V RMS

! The connections are not polarity sensitive.

Connecting the XN120 to an External Paging System.

The external paging system is connected to the RJ11 socket labelled MOH/PAGE located under the sub cover to the right of the ST1-ST8 sockets.
The external paging system is not supplied with the XN120.



The connections of the RJ11 plug:

XN120 MOH/PAGE socket	Description	Specification
2 5	External Paging output	Impedance: 600 Ohm @1KHz Nominal output level: 250mV (-10dBm) Maximum output level: 400mV RMS

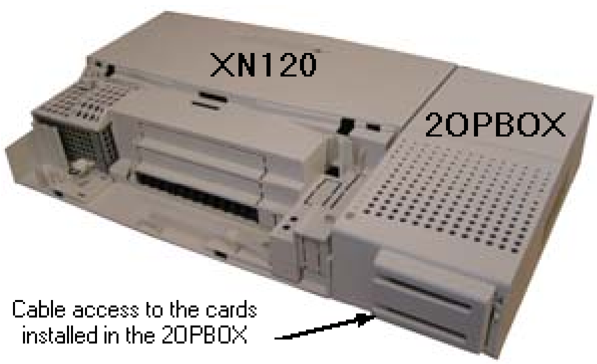
! The connections are not polarity sensitive.

2OPBox

Top

2OPBOX

- The 2OPBOX is an optional unit that connects to the side of either a XN120 main unit or expansion unit.
- One box can be connected to each of the XN120 main/expansion units.
- The 2OPBOX provides two universal slots for the installation of Basic Rate ISDN cards or Voice over IP resource cards.



Installation Procedure

1	**Unpack the 2OPBOX**	
2	**Power off the XN120 system**	
3	**Fit the 2OPBOX**	

4	**Install the BRI/VoIP cards into the 2OPBOX**	
5	**Power on the XN120**	

Unpack the 2OPBOX

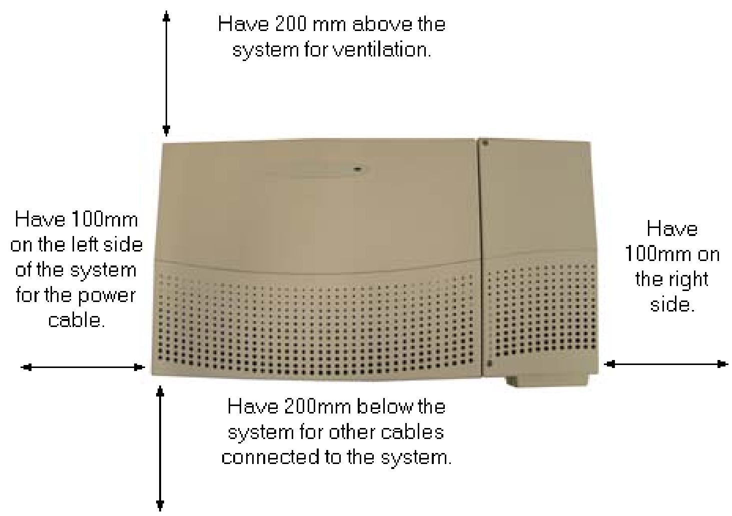
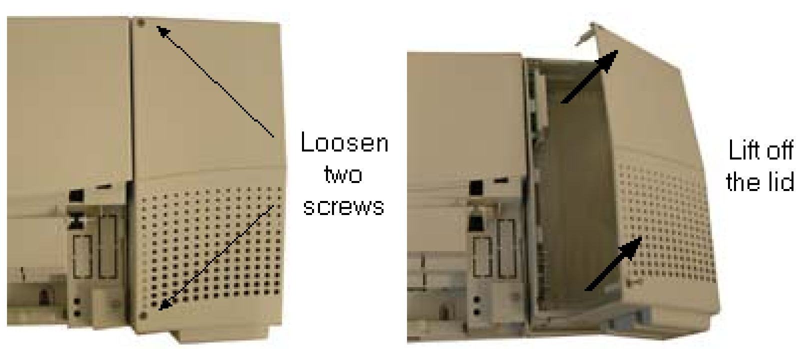
The 2OPBOX is supplied with 6 screws to fix the unit to the XN120 main/expansion cabinet.

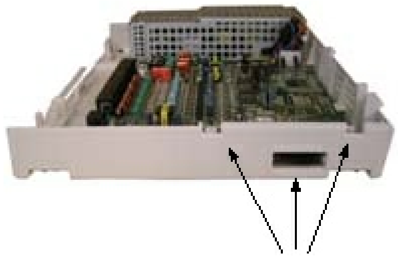
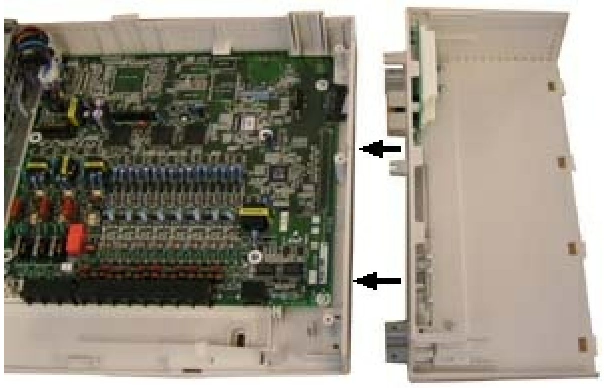
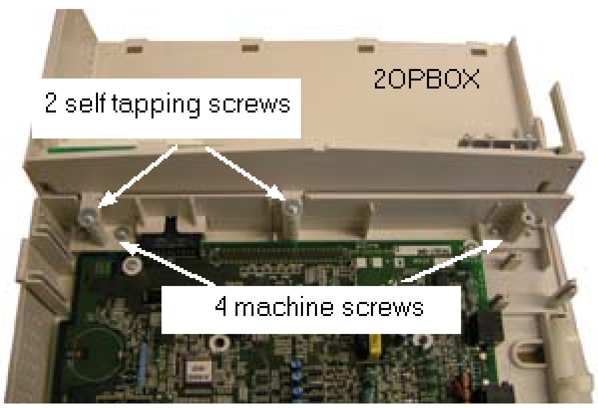
<u>Additional Items Required:</u>
Cross head screwdriver.
Small cutters/pliers.

<u>Optional Items</u>
The 2OPBOX can have BRI cards or VoIP cards installed.

Fitting the 2OPBOX

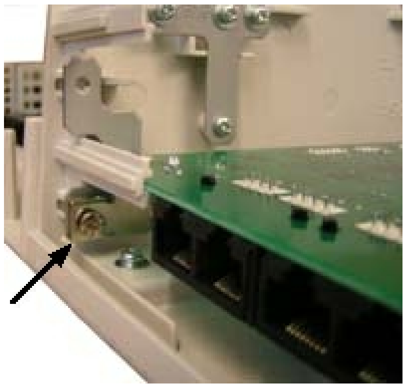
***!* You must power off the system before the 2OPBOX is connected onto the XN120 unit.**

1	You will need a minimum of 690mm x 680mm (W x H) wall space for each XN120 cabinet with the 2OPBOX connected. The XN120 unit with 2OPBOX connected is 490mm x 279mm (W x H) ***!*** You will need more space if you are installing the optional expansion units. Refer to the installation instructions supplied with the unit.	The 2OPBOX does not require fixing to the wall, it is fixed directly to the XN120 unit. 
2	Remove the lid of the 2OPBOX	

3	Power off the XN120.	
4	Using a pair of small cutters or pliers carefully remove the 3 breakouts on the right side of the XN120base unit. You will need to remove the lid of the XN120 first.	 Remove the 3 breakouts on the right side of the XN120 unit
5	Push the 2OPBOX into the connector on the right side of the XN120cabinet	
6	Secure the 2OPBOX using the screws supplied. There are two screws with self tapping thread and four with a machined thread.	

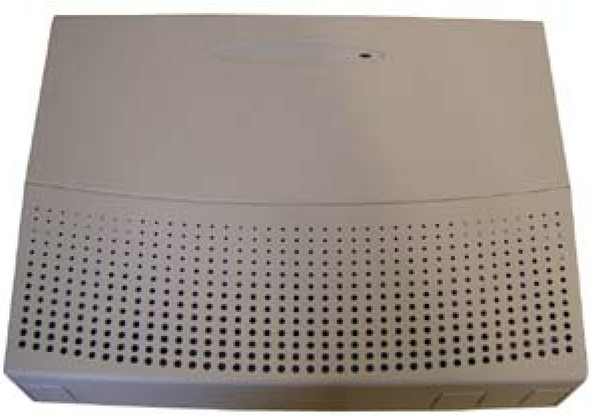
Install the Option Cards into the 2OPBox

1	Install the BRI or VOIP option cards into the 2OPBOX. Refer to the instructions for each card later in this guide.	The card can be installed into either slot of the 2OPBOX. If you are only installing one option card, it is recommended (for logical port configuration) that you fit the card into the slot marked CN1.

		 Fit the option card into the guides and push firmly onto the connector.
2	Secure the card to the ETH plate at the front edge of the 2OPBOX.	

XN120 Expansion Unit

Top



The XN120 expansion unit is connected to the XN120 main unit to increase the capacity of the system.

The XN120 main unit must have the Applications Card (EXIFU-A1) installed before you can connect an expansion unit. The system will allow up to two expansion units to be connected to the XN120 main unit.

The expansion unit is supplied with three exchange lines and eight telephone ports. The quantity of ports can be increased by adding optional 008 or 308 expansion cards.

The expansion unit does not have the external music input/external page output socket; this is only on the XN120 main unit. Additional external audio devices are connected via the 2PGDU cards that can be installed into any unit.

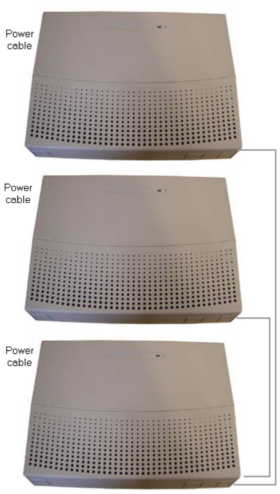
It can also accommodate a 2OPBox for installation of ISDN BRI cards or VOIPU cards.

Identifying the Expansion Unit

The expansion unit looks identical to the main unit when the covers are fitted.
The units can be identified by the label on the left side of the unit.
The main unit is labelled **IP2AT-924M KSU**
The expansion unit is labelled **IP2AT-924ME KSU**

System Connection Diagram

The diagram shows the maximum size of the XN120 system with two expansion units connected to the main unit.
The main unit must have the Applications Card installed (EXIFU-A1) to provide the connections for the expansion units.
The table shows the maximum capacities of each unit within a one, two or three unit system. Refer to the system capacity table at the beginning of this manual for any additional limitations.



		9 analogue lines 24 Telephones 8 BRI and/or VOIP circuits 1 2OPBOX 1 2PGDU card
	9 analogue lines 24 Telephones 8 BRI and/or VOIP circuits 1 2OPBOX 1 2PGDU card	9 analogue lines 24 Telephones 8 BRI and/or VOIP circuits 1 2OPBOX 1 2PGDU card
9 analogue lines 24 Telephones 8 BRI and/or VOIP circuits 1 2OPBOX 1 2PGDU card 1 EXIFU card 1 DSPDB	9 analogue lines 24 Telephones 8 BRI and/or VOIP circuits 1 2OPBOX 1 2PGDU card 1 EXIFU card 1 DSPDB	9 analogue lines 24 Telephones 8 BRI and/or VOIP circuits 1 2OPBOX 1 2PGDU card 1 EXIFU card 1 DSPDB
Main unit only	**Main unit plus one expansion unit**	**Main unit plus two expansion units**

Optional Items	
008 Expansion Card 8 Extension ports Installed onto the main board in the XN120 expansion unit. **308 Expansion card** 3 Exchange lines + 8 Extension ports Installed onto the main board in the XN120 expansion unit. **2PGDU card**	**2OPBOX** For ISDN BRIU and/or VOIPU cards. 2 slots available per unit. Plugs onto the right side of the XN120 expansion unit. **BRIU Card** ISDN Basic Rate cards (2 or 4 circuits available) can be installed in the 2OPBOX. **64 Button DSS console** Provides 64 programmable keys and 14 fixed feature

2 Door phone units with relay contacts 2 Audio ports (External MOH and Paging) **DSPDB** Voice Mail and Voice response unit. **EXIFU** Serial, Ethernet, Expansion unit connection and compact flash socket. There is also an EXIFU with only the serial port.	keys. Gives operator type functionality. **24 Button Add On console** Provides 24 additional function keys for the XN120 display phones.

Installation Procedure

1	**Unpack all items.**	
2	**Mount the XN120 expansion unit on the wall.**	***!*** Within suitable cabling distance from the exchange lines. ***!*** Within suitable distance from a power socket and Earth point. ***!*** Within suitable distance for the RJ45 connecting cables. ***!*** Check the other installation considerations in the Wall Mount section. ***!*** If you have any optional parts to install please check any related installation considerations.
4	**Connect the telephones.**	
5	**Connect the exchange lines.**	
6	**Power off the XN120 Main Unit.**	
7	**Connect the expansion unit to the EXIFU-A1 card installed in the main unit.**	
8	**Connect the power and switch on the XN120 system.**	

Unpack the Expansion Unit.

1 x XN120 expansion unit
1 x Wall mounting template
1 x Power cord 4 x Fixing screws
1 x RJ45 connection cable (1.25 metre RJ45 to RJ45 straight through cable)

<u>Additional Items Required:</u>

- Cross head screwdriver.
- 4 Wall fixing plugs suitable for the type of wall.
- Solid wire for extending telephone cabling:
  Recommended cable type: Twisted pair (CW1308 or similar specification)
  Conductor diameter: 0.4 to 0.6 mm

Maximum cable length: (with 0.5 mm diameter cable)
XN120 system telephone – 300 metres
Normal telephone – 1500 metres

If you need to extend the exchange line cables:

- Solid wire for exchange line cables:
- Recommended cable type: Twisted pair (CW1308 or similar specification)
- Conductor diameter: 0.4 to 0.6 mm

Wall Mount the XN120 Expansion Unit

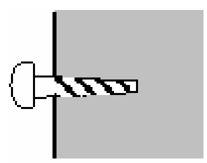
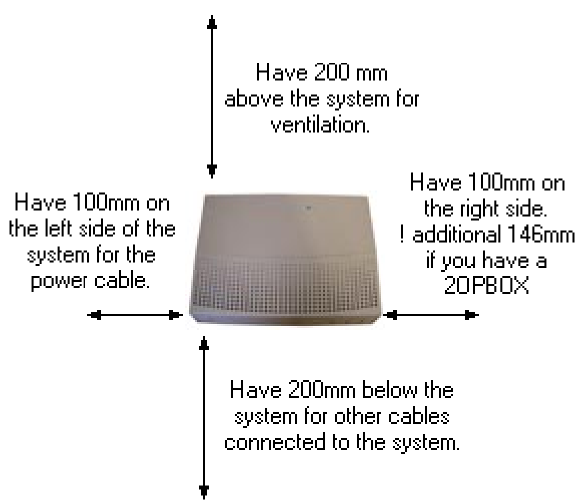
Installation Considerations:

- To avoid electric shock or damage do not plug in or turn on the system power before completing the installation.
- Avoid working with the system during electrical storms.
- Use the power cord supplied with the product.
- Do not bundle power cords together, the cords may overheat.
- Ensure the system has a suitable Earth Ground connection.

Environmental Considerations – Be sure the system is not:

- In direct sunlight or in hot, cold or humid places.
- In dusty areas or in areas where sulphuric gasses are produced.
- In places where shocks or vibrations are frequent or strong.
- In places where water or other fluids may come into contact with the equipment.
- In areas near electric welders or machines that emit high frequency radiation.
- Near computers, microwaves, air conditioners etc.
- Near radio antennas (including shortwave).

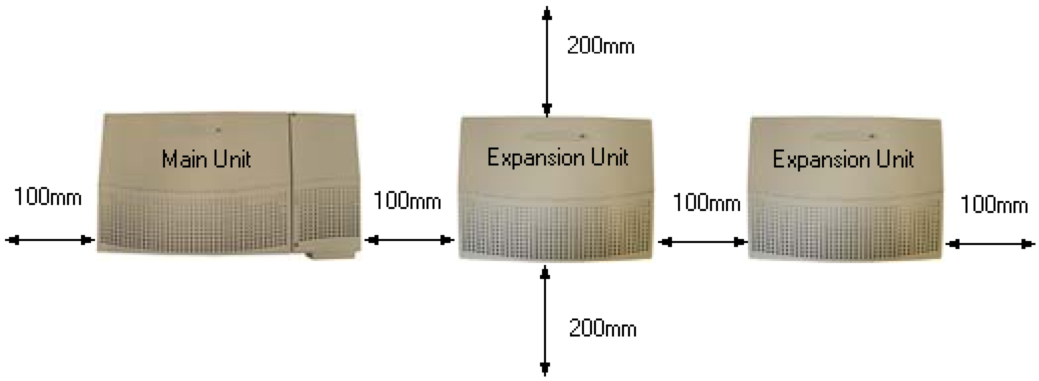
- If you are installing the optional 2OPBOX or an additional expansion unit ensure there is sufficient wall space and ventilation. Refer to the wall mounting diagrams below.

1	The expansion unit is 360mm x 279mm (W x H). ***It can be installed in any position near the XN120 main unit within the length of the RJ45 connecting cable supplied.*** See the examples below. ***!*** You will need more space if you are installing the optional 2OPBOX. Refer to the installation instructions supplied with the unit.	Use the wall mount template supplied to mark the four screw locations. Ensure that you use the correct wall plugs for the type of wall. Leave approximately 3~5mm of the screw protruding from the wall.  

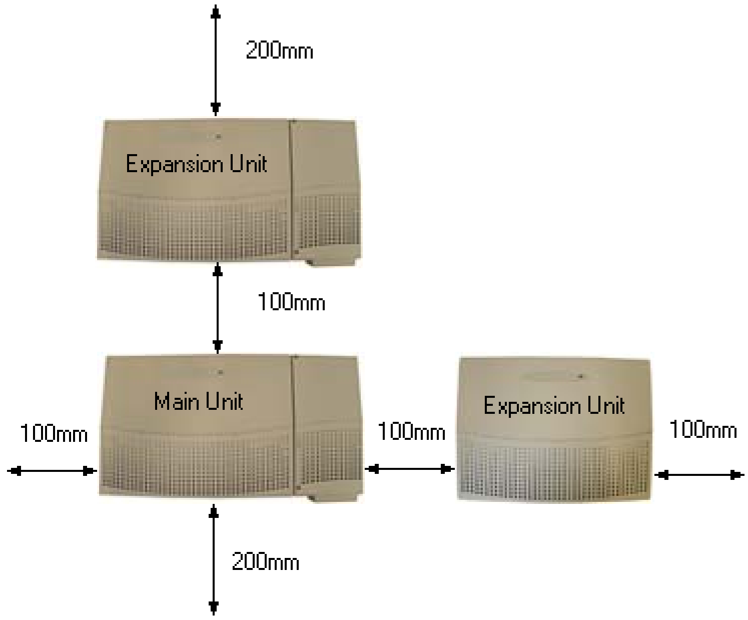
Example installations

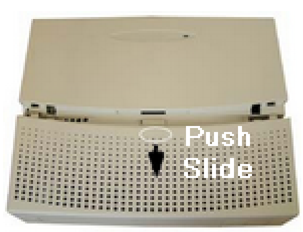
The XN120 expansion unit can be installed in any position near the XN120 Main Unit within the length of the RJ45 connecting cable supplied.
The following diagrams show some examples.
Measurements shown are the minimum recommended for ventilation and cable access.
***!** Ensure the RJ45 connecting cables will reach from the main unit to each expansion unit before you decide on the final positions.*

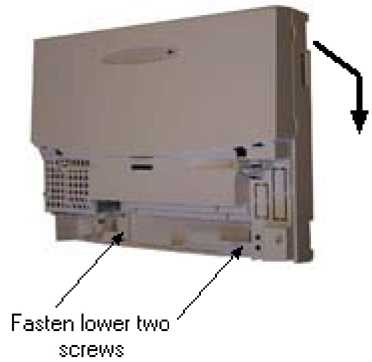
Units installed side by side



Expansion units arranged to the side and above the main unit

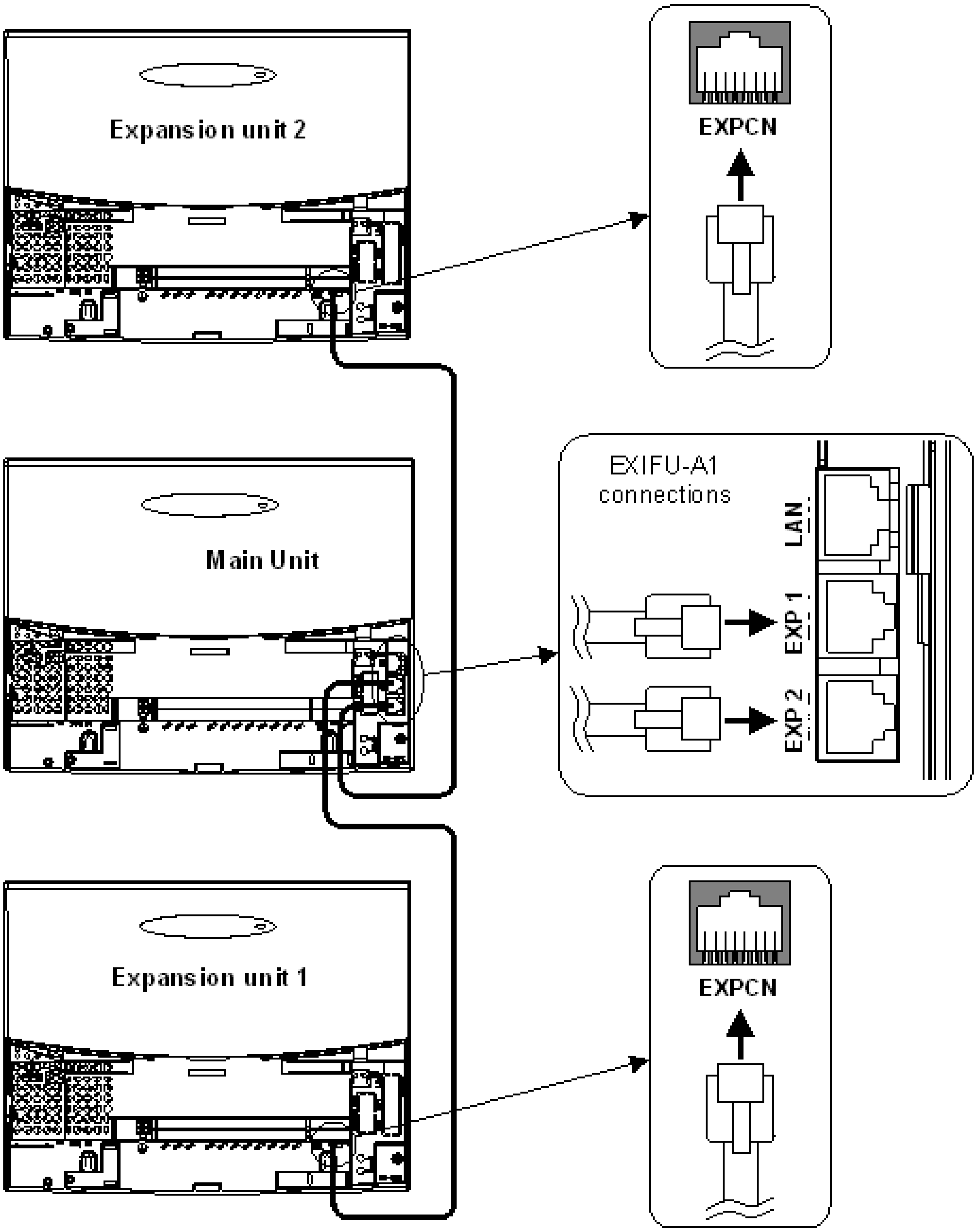


2	Remove the sub cover of the Expansion Unit. Hook the XN120 onto the wall mount screws. Tighten the lower screws to	

	secure the unit to the wall.	
3	Earth the XN120 system. ***Caution:*** If this cable is not installed or requires disconnection, then the telecommunication network connection(s), CO and/or ISDN, must be disconnected first from the XN120.	***Important.*** Each unit must have a permanent Earth Ground connection to a verified Earth point using a minimum of 2.5mm² green/yellow cable. The Earth connection must have no other purpose than connecting to the XN120 unit. The Earth point is located on the left side under the sub cover of each unit. 

Connecting the Expansion Units to the EXIFU card

Each expansion unit must be connected to the EXIFU card within the XN120 main unit.
If you are installing one expansion unit connect to the EXIFU socket labelled EXP1.
If you are installing two expansion units connect to the EXIFU sockets labelled EXP1 and EXP2.
The EXPCN socket on the expansion unit is located to the right of the ST sockets on the main board.



External Battery Box

Top

External Battery Box

- The battery box is an optional unit that connects to the XN120 main unit or expansion unit.
- One battery box is required for each of the XN120 main/expansion units.
- The battery box provides battery backup when the power fails to the XN120 system.
- Each battery box will require two Valve Regulated Lead Acid batteries (Yuasa 2.6Ah 12V)

Battery Specification

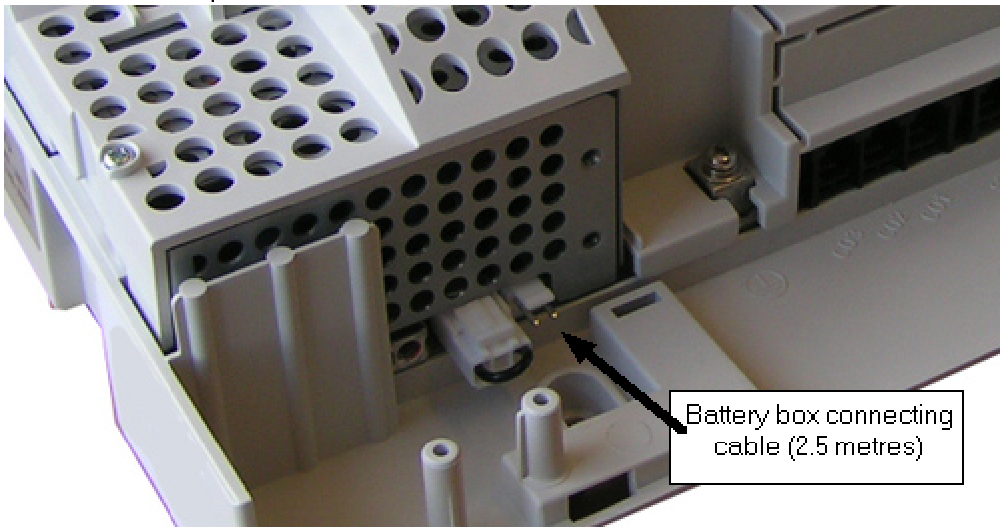
Battery capacity	12V, 2.6Ah or equivalent (Voltage must be 12V)
Battery *Only the manufacturer and type of battery specified must be used.*	Yuasa NP2.6-12 (134x67x64 mm / 1.12kg)
Number of Batteries (per a Box)	2
Backup Duration (Estimated)	1 Hour
Charging time	12 hours (from fully discharged)

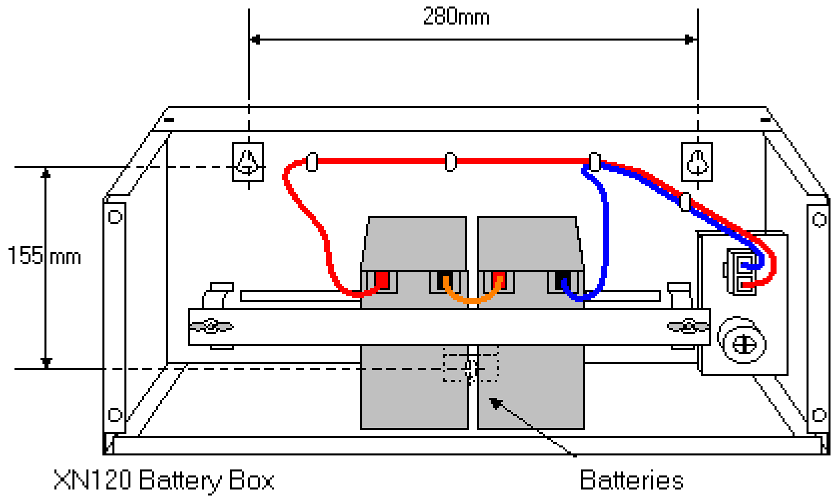
Installation Procedure

1	**Unpack the battery box**	
2	**Floor/shelf mount the battery box**	The battery box should also be secured to a wall.
3	**Install the batteries**	
4	**Power off the XN120**	
5	**Connect the battery box to the XN120 unit**	
6	**Power on the XN120**	
7	**Test the power fail operation**	

Connection Diagram

XN120 Main or Expansion Unit





Unpack the Battery Box

The battery box is a universal part that can be used to provide battery backup for other products therefore, there are parts supplied that are not required when used with the XN120.

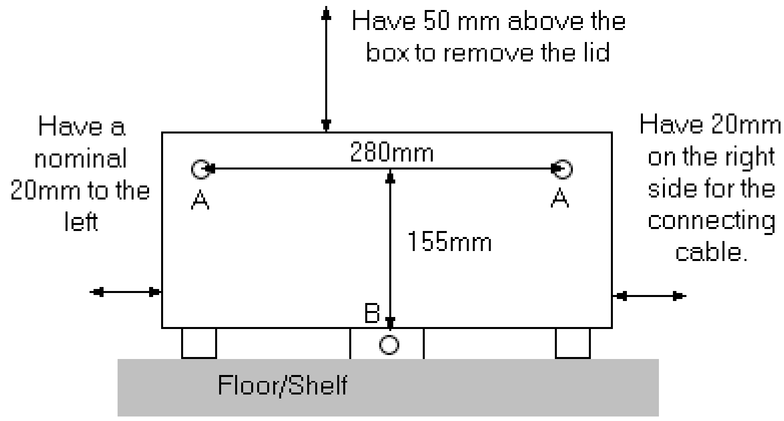
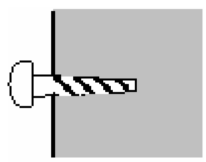
The battery box is supplied with the following items.
Battery box with approximately 2.5 metres of connecting cable attached.
3 screws for securing the box to the wall
Battery retaining bar with 2 fixing brackets.
2 wing nuts for the battery retaining bar.
2 Orange cables (1 required when used for the XN120)
1 red/blue two terminal connecting cable (required when used for the XN120)
1 red/blue four terminal connecting cable (not required when used for the XN120)
1 spare fuse (20mm 4 Amp, taped to the inside of the battery box)

Additional Items Required:
Cross head screwdriver.
Screws and wall plugs if wall mounting the battery box.
2 Batteries (12V 2.6Ah)

Wall/Floor Mount the Battery Box

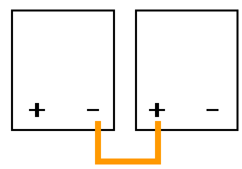
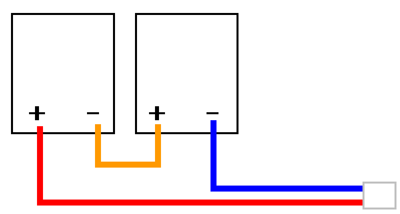
The battery box can be mounted on the floor or a suitable shelf and secured to the wall or mounted directly onto the wall.

***!** If you are wall mounting the battery box you must ensure the screws and wall fixings used are suitable for the weight of the battery box (5.2Kg when the batteries are installed). The screws supplied are suitable for wall mounting the battery box onto a wooden board only. If you are mounting the battery box directly onto a brick/block or plaster board wall you must provide suitable wall fixings and screws.*

1	You will need a minimum of 424mm x 149mm (W x H) wall space for each battery box. The battery box is 384mm x 99mm (W x H). Ensure the battery box is within 2 metres of the XN120 unit; the connecting cable is 2.5 metres long.	
2	Loosen the 4 screws on the front of the battery box and remove the front cover.	
3a	Floor/Shelf mounting	Position the battery box on the floor/shelf. Secure to the wall using two screws through holes marked A in the diagram above. Fit suitable wall plugs if necessary.
	OR	
3b	Wall mounting	Loosen the two screws and the back of the battery box and lower the metal bracket to reveal hole B. Re-tighten the two screws. Mark the three holes (A and B), you may need to use the battery box as the hole template. Fit suitable wall plugs if necessary. Fit two screws into holes A, use suitable length/type of screws. Leave approximately 3~5mm of the screw protruding from the wall.  Hook the battery box onto the two screws. Tighten the two screws and secure the battery box by fitting a third screw into hole B.

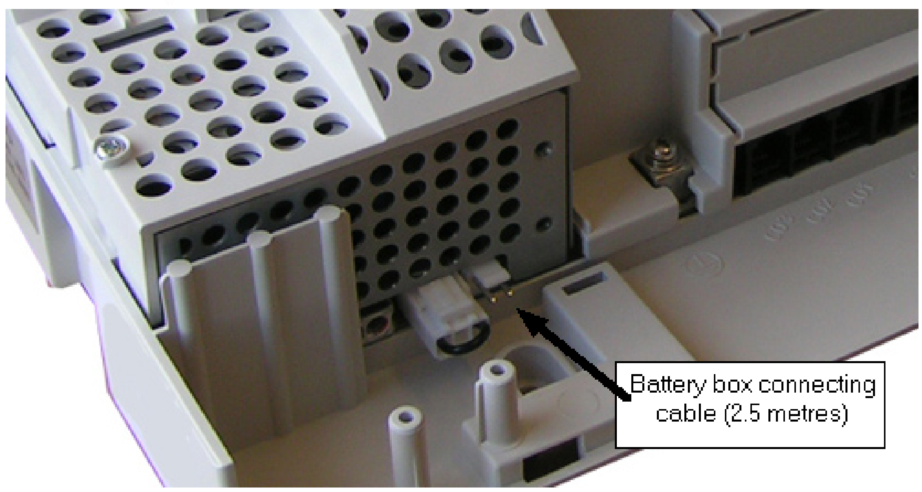
Install the batteries

***!* CAUTION - do not short-circuit the battery terminals.**

1	Fit the two batteries into the battery box.	Refer to the Connection Diagram earlier in this guide. Ensure the battery terminals are towards the front.
2	Secure the batteries using the retaining bracket.	Locate the two fixing brackets supplied with the bracket into the key slots on the inside of the back panel of the battery box. Fit the retaining bracket across the front of the batteries. Secure the retaining bracket with the two wing nuts.
3	Connect the batteries together.	Using one of the orange cables connect the +'ve terminal of one of the batteries to the –'ve terminal of the OTHER battery. CAUTION - DO NOT SHORT-CIRCUIT the battery terminals. 
4	Connect the batteries to the socket within the battery box. The plug on the red/blue cable is polarity sensitive; ensure you connect correctly.	Using the red/blue cable with two terminals connect the remaining +'ve and –'ve battery terminals to the 2-pin socket inside the battery box.  **CAUTION – Ensure you:** **Connect the red cable to the + battery terminal.** **Connect the blue cable to the – battery terminal.** There are plastic retaining clips on the inside of the back panel of the battery box to secure the red/blue cables.
5	Replace the front cover of the battery box. Write the date of installation on the label on the front cover of the battery box.	

Connect the Battery Box to the XN120

! You must power off the XN120 system before the battery box is connected to the XN120 unit.

1	Power off the XN120 system.	You must also power off any expansion units if they are installed.
2	Remove the sub cover of the XN120 Unit.	
3	Connect the cable from the battery box to the socket in the XN120 unit. The plug on the battery box connecting cable is polarity sensitive; ensure you connect correctly. ***!*** If you have any XN120 Expansion units you must connect a separate battery box to each XN120 expansion unit.	 Ensure the connecting cable from the battery box can not become damaged; tie up any loose cable.
3	Re-fit the XN120 sub cover.	

Test the Power Fail Operation

You should test each battery box/XN120 main and expansion unit separately.

1	With the XN120 unit switched ON, powered by mains power and also connected to the battery box.	

2	Turn OFF the mains power to the XN120 unit	The XN120 unit should continue to operate as normal. Leave the power OFF for 30 seconds to ensure the system continues to operate correctly. Check that you can make/receive a call from any of the telephones to confirm the system is working correctly.
3	Turn ON the mains power to the XN120 unit	The XN120 unit should continue to operate as normal.

Ensure the mains power is ON to all XN120 units after you complete the tests.

System Operation when Battery Box is connected

- The battery box will provide battery back up when the mains power fails to the XN120 unit.
- The XN120 charges the batteries whilst it is mains powered.
- Each XN120 main/expansion unit must have a separate battery box connected.

When the mains power fails the XN120 unit will continue to operate as normal until the batteries become discharged. The XN120 unit is considered to be in *battery back up mode* at this time, it is not in power fail mode.
When the batteries are discharged the XN120 unit will stop working, the XN120 is now considered to be in *power fail mode*.

! If you switch off the XN120 unit whilst in battery back up mode it will not power back on until the mains power is supplied. The XN120 will not start up in battery back up mode.

Power Fail Telephones

If the XN120 has power fail telephones connected; they will become active when both the mains power fails and the batteries are discharged i.e. when the XN120 is in power fail mode.

Battery Maintenance and Replacement.

The batteries should be replaced every 3 years to ensure the maximum battery back up duration (1 hour) is available.
The maximum operating life of the batteries is 5 years, the battery back up duration will begin to reduce after 3 years.
The label on the front cover of the battery box should indicate the battery installation and replacement dates.
Follow the instructions in this guide when replacing the batteries.
Follow the instructions supplied by the battery manufacturer for disposal of the old batteries.

Replacing the fuse
Replace with the same type of fuse – 20mm 4Amp fast acting.

Battery Care

This is a guide only; please follow any information that is supplied by the battery manufacturer.

Never leave the batteries in a discharged state.
The battery back up duration will be reduced if the battery is left discharged for long periods of time.
Replace the batteries if you suspect they have been left discharged

Recommended Operating Temperature = 20^{o}C.
Higher temperatures will reduce the battery life.
Lower temperatures will reduce the battery back up duration.

Ventilation.
Although the batteries are of a sealed (valve regulated) type gases may be released from the batteries.
Do not install the battery box in a sealed enclosure.

Never short-circuit the battery terminals.

There is a risk of explosion.
The battery may be damaged and will need to be replaced.

Batteries are heavy – 1.12Kg each.
Although the batteries used with the battery box are relatively small the weight of the battery box is 5.2Kg when the batteries are installed.

Do not tamper with the battery.
The battery contains hazardous liquid and gas.
Never try to open the battery case.
Do not bend the battery terminals.

Disposing of batteries.
Return old batteries to the battery supplier or a licensed battery dealer.
Do not dispose of batteries into a waste bin.
Do not incinerate batteries.

308 and 008 Expansion Cards

Top

008 and 308 Expansion Cards

The 008 and 308 option cards provide additional exchange lines and telephone ports to the XN120 system. The cards are installed onto the base board within the XN120 main or expansion units. Each unit allows up to two of either type of option card to be installed.
The exchange lines and telephone ports have the same features and functionality as those on the XN120 base board, refer earlier in this Guide for details.

The 008 option card provides:

- 8 telephone ports for the connection of XN120 system phones, a 64 button console or normal telephones.
  Each connected via a separate RJ11 plug

The 308 option card provides:

- 3 Analogue exchange lines.
  Each connected via a separate RJ11 plug.
- 8 telephone ports for the connection of XN120 system phones, a 64 button console or normal telephones.
  Each connected via a separate RJ11 plug.
- 1 Power fail port.
  The first exchange line is connected to the first telephone port on the same card.

Installation Procedure

	Unpack the 008 or 308 card	
	Power off the XN120 and install the card(s)	
	Connect the telephones and exchange lines	
	Power on the XN120	

Unpack the Card.

- The cards are supplied with the following to fix the card into the system:

2 x metal threaded mounting bars
2 x plastic mounting bars
2 x screws

Additional Items Required:

- Cross head screwdriver.
- Small cutters or pliers.

- Solid wire for extending telephone cabling:
  Recommended cable type: Twisted pair (CW1308 or similar specification)
  Conductor diameter: 0.4 to 0.6 mm
  Maximum cable length: (with 0.5 mm diameter cable)
    XN120 system telephone – 300 metres
    Normal telephone – 1125 metres

If you need to extend the exchange line cables:

- Solid wire for exchange line cables:
- Recommended cable type: Twisted pair (CW1308 or similar specification)
- Conductor diameter: 0.4 to 0.6 mm

Install the 008 / 308 Card

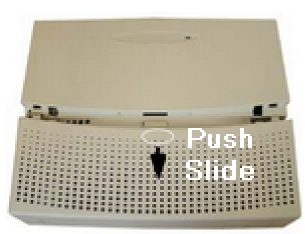
! Observe anti-static precautions when handling the 008 or 308 card.

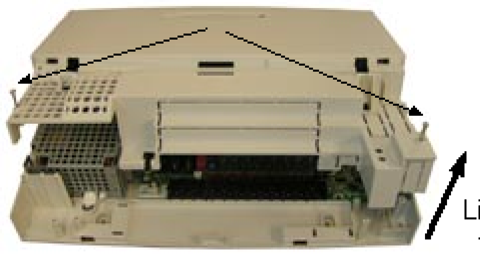
- Wear a suitable anti-static strap connected to an Earth point.

The 008 or 308 card is installed onto the base board within the XN120 main or expansion units.
Up to two cards can be installed into each main/expansion unit.
! When installing the card the installation order will determine the port assignment:
The 008 card will be assigned eight telephone ports.
The 308 card will be assigned three exchange line ports and eight telephone ports.
The port assignment is done automatically when the system is powered on, after the card is installed.
The assignment can be confirmed by Program 10-03-01.

If you have a XN120 expansion cabinet.

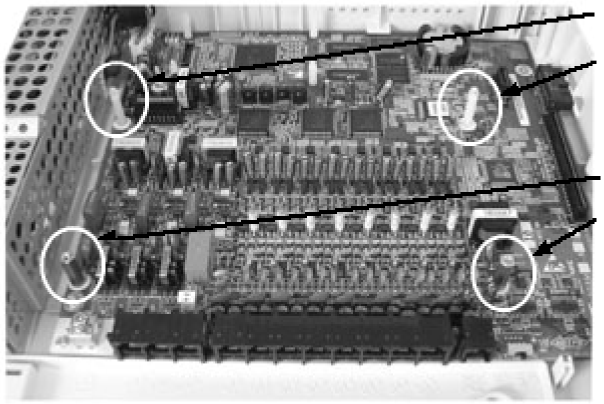
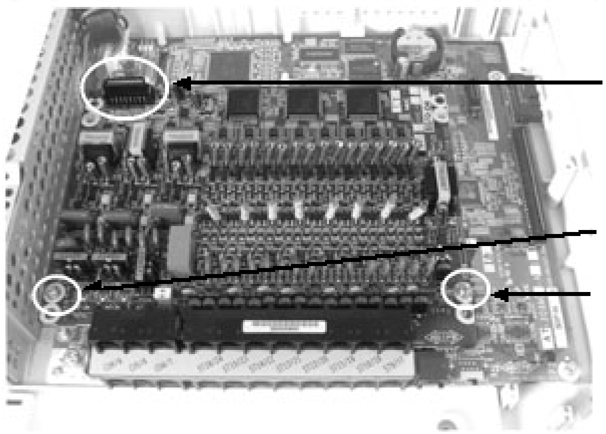
Up to two 008/308 cards can be installed in either the main or expansion cabinets.
If you have spare capacity in either cabinet then it is recommended that you fit the 008/308 card(s) in the main cabinet first.
This will maintain a logical port assignment for the 008/308 cards.

1	Power off the XN120 system	You must also power off each XN120 expansion unit if you have any installed.
2	Remove the sub cover and main cover of the XN120 unit that will have the card installed.	

<table>
<tr><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td></tr>
<tr><td>3</td><td>Fit the first card.
The first 008/308 card is installed onto the connector on the base board within the XN120 unit.

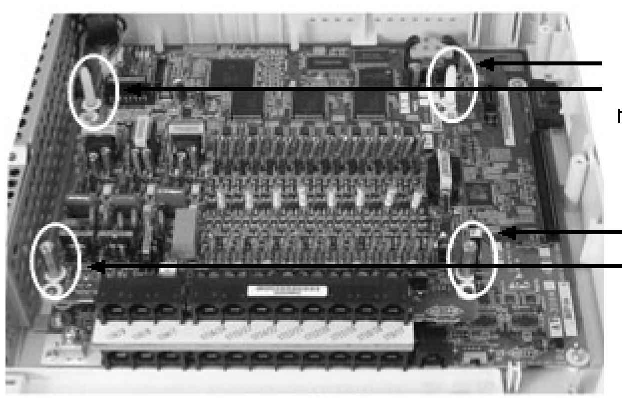
Fit the mounting bars.

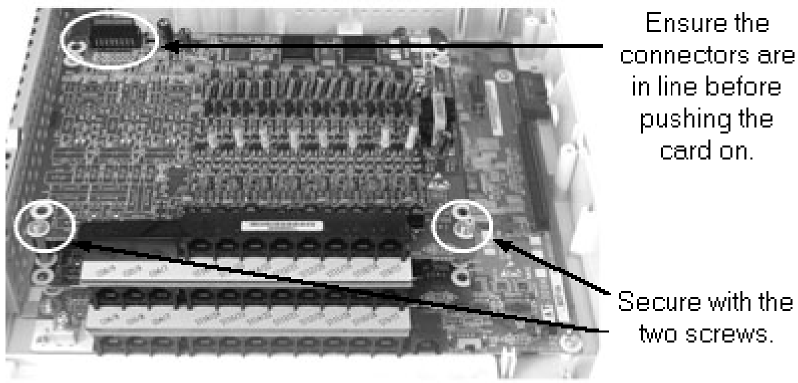
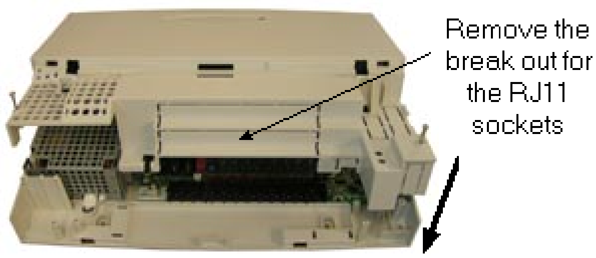
Fit the 008/308 card.

If you are not installing a second card then secure the first card with the two screws.</td><td></td></tr>
<tr><td></td><td></td><td></td></tr>
<tr><td>4</td><td>The second 008/308 card is installed onto the connector on the first 008/308 card within the XN120 unit.

Fit the mounting bars (onto the first 008/308 card).

Fit the second 008/308 card (onto the first 008/308 card).

Secure with the two screws.</td><td></td></tr>
</table>

		
5	Before you refit the main cover break out the panels for the RJ11 sockets.	

2PGDU Card

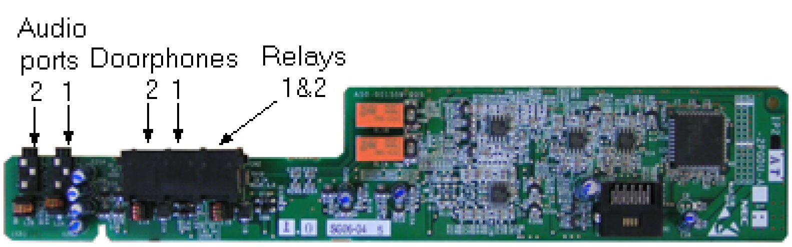
Top

2PGDU Card (Paging/Doorphones/Audio ports)

The 2PGDU card provides:

- 2 connections for door phone units.
  Each connected via a separate RJ11 plug.
- 2 audio ports that can be configured for either external paging, external music input or background music.
  Each connected via a separate 3.5mm jack plug.
- 2 Relay contacts. These can be assigned to any of the door phones or audio ports of the 2PGDU card.
  Both relay contacts are connected via a single RJ11 plug.

You can install one 2PGDU card into each of the IPC 100 main or expansion units giving a total of three cards.



Installation Procedure

1	**Unpack the 2PGDU**	

2	**Power off the XN120 and install the 2PGDU card**	
3	**Power on the XN120**	
4	**Configure the 2PGDU ports**	
5	**Connect the external device(s)**	

Unpack the Items.

Additional Items Required:

- You will need a door phone unit (part code DX4NA) for each door entry point connected via a single pair cable terminated with an RJ11 plug to connect to the PGDU card.
- You will need an external door lock device that is compatible with a normally open contact. The control for the door lock unit must be connected via a single pair cable terminated with an RJ11 plug.
- You will need an external music device connected by a single pair cable terminated with a 3.5mm jack plug for each BGM/External Music Source.

Cross head screwdriver.
Small cutters or pliers.

Optional Items

- Door phone unit.
- Cable to connect the door phone unit:
  Single pair (2 wire) standard telephone cable.
  Maximum cable length: 150 metres with 0.5mm diameter conductor.
- Cable to connect the external audio devices:
  Single pair audio cable. Refer to the instructions supplied with the external device for cable type.
  Maximum cable length may also be determined by the instructions supplied with the external device. If not, keep the cable length as short as possible.

Install the 2PGDU Card

! Observe anti-static precautions when handling the 2PGDU card.

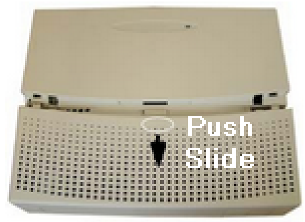
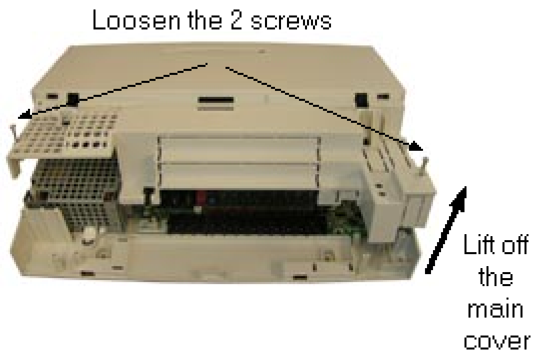
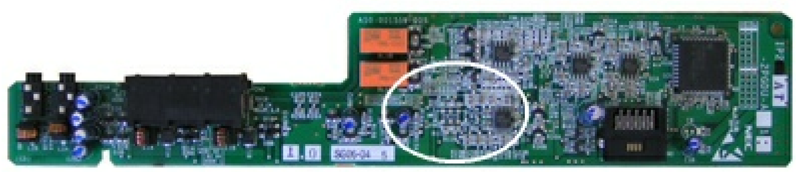
- Wear a suitable anti-static strap connected to an Earth point.

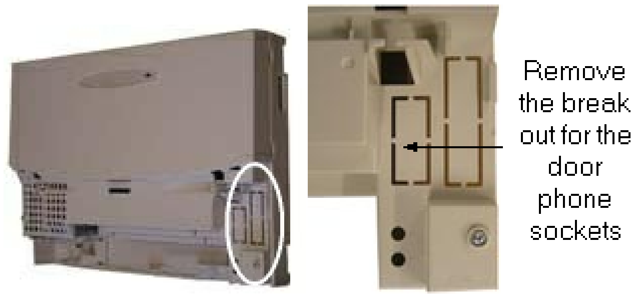
The 2PGDU card is installed onto the base board within the XN120 main or expansion units.
One 2PGDU card can be installed into each main/expansion unit.
! When installing the 2PGDU card the installation order below shows the port assignment:

Order of 2PGDU card installed.	Door Phone numbers	Audio Port numbers
First	1 & 2	1 & 2
Second	3 & 4	3 & 4
Third	5 & 6	5 & 6

Xn120

The port assignment is determined when the system is powered on, after the 2PGDU card is installed.

1	Power off the XN120 system	You must also power off each XN120 expansion unit if you have any installed.
3	Remove the sub cover and main cover of the XN120 unit that will have the 2PGDU card installed.	 
4	Set the door phone switch to the correct mode. The 2PGDU is compatible with two types of door phone unit. The door phone unit used with the XN120 is the 270 type.	Set switch SW1 to the 270 setting.  
5	Fit the 2PGDU card	

6	Before you refit the main cover break out the panels for the door phone sockets.	
7	Re-fit the main and sub covers	

Door Phones and Door Locks

The door phone unit has a push button, built in loudspeaker and microphone. The door phone unit is installed close to the outside of the door where you have visitors to your building.
The visitor can push the button to alert a group of telephones that they are at the door, similar to a conventional door bell. You can answer the door phone call and have a conversation with the visitor. (There is also an option to connect a door lock device to the XN120, you can then release the door lock and allow the visitor to enter, refer to the Door Lock section within this guide).

At default the door phone connections of the 2PGDU are enabled, when the button of the door phone is pressed the call will alert telephone 200. The call will ring for 30 seconds.
The XN120 can have up to 6 door phones connected (2 per XN120 unit).
You can assign a group of phones to each door phone and adjust the ring duration used by all door phones.

Connect the Door Phone Unit

Installation Considerations:

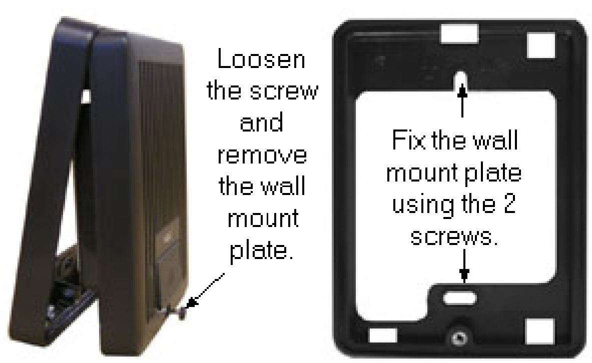
- The maximum cable length to the door unit is 150 metres with 0.5mm diameter conductor.

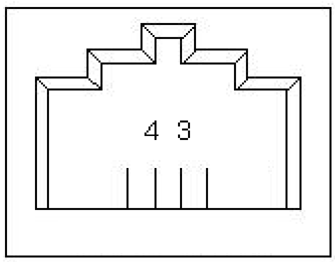
Environmental Considerations:

- Ensure the door unit is not installed where it will be exposed to water or extreme weather conditions.
- The door unit is not weather proof.
- The door unit is not vandal proof.

Door Unit specification:

- Temperature: -20°C to +60°C (-4°F to +140°F)
- Humidity: 20 to 80% (non condensing)

1	Wall mount the door unit. Remove the wall mount plate by loosening the screw at the bottom of the door unit. Break out one of the cable access points on the wall mount plate. Fix the wall mount plate to the	

	wall. There are 2 screws supplied, you will need wall plugs if fixing onto brick etc.	
2	Run a single pair cable from the 2PGDU card to each door unit. Maximum cable length is 150 metres with 0.5mm diameter conductor. The connection is not polarity sensitive.	Door unit. Connect to screw terminals marked - & I  2PGDU Door phone RJ11 socket (DPH1 or DPH2). Connect pins: 3 red 4 green 

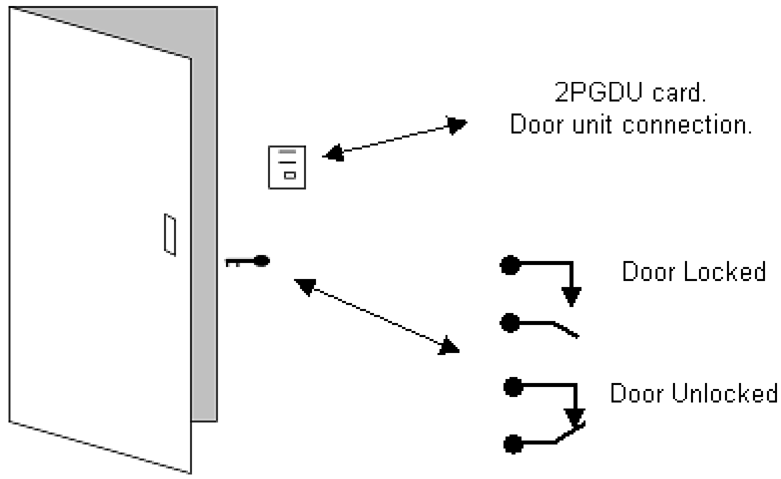
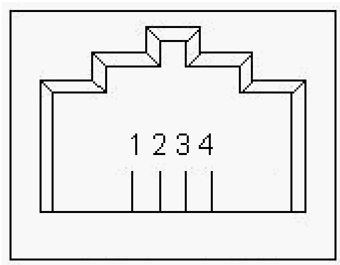
Connect the Door Lock Unit

You can connect an external door lock that can be controlled by the XN120 2PGDU card.

- The 2PGDU card provides two pairs of normally open relay contacts. The contacts are closed when the door lock is released. The 2PGDU card does not provide power at the relay contacts.
- Relay 1 is assigned to door unit 1, relay 2 is assigned to door unit 2.
- The contacts are rated at 24 VDC, 0.5A and 120VAC, 0.25A.

! The external door lock must be compatible with the 2PGDU relay contact operation.
XN120 relay contact open: Door locked.
XN120 relay contact closed: Door unlocked.

The relay assigned to each door phone is set by Program 10-03-01.
At default:
Relay RY1 is assigned to door phone DPH1.
Relay RY2 is assigned to door phone DPH2.
! The relays can only be assigned to one of the ports on the same 2PGDU card.

1	Connect the door lock control to the 2PGDU card relay contacts. ***!*** If you have a door phone unit installed and door lock at the same door then ensure you connect the correct door lock relay RY1/RY2.	 Pin connections of the RJ11 socket on the 2PGDU card. Relay RY1 – pins 2 and 3 Relay RY2 – pins 1 and 4 

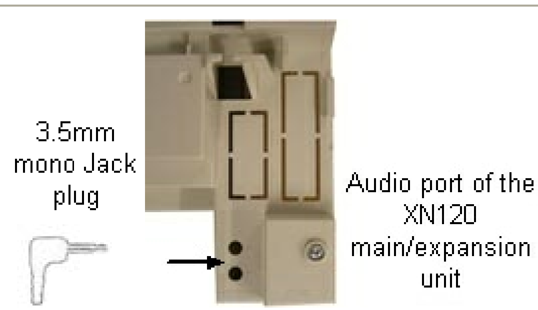
External Paging

You can connect the audio ports of the XN120 to an external Public Address (PA) system. Any telephone can then place an external paging call over the PA system.
Each 2PGDU card has two audio ports, you can install one 2PGDU card into each XN120 main/expansion unit giving a total of six audio ports.

There is also an external paging port on the XN120 main unit, refer to the section at the beginning of this manual for connection details.

2PGDU audio port specification:
Impedance: 600 Ohm @ 1KHz
Nominal output level: 250mV (-10dBm)
Maximum output level: 400mV RMS

<u>Connect the External Paging System</u>

1	Run a single pair audio cable to the external paging system. Terminate the cable with a 3.5mm mono Jack plug to connect to the 2PGDU audio port. Note, there is only 26mm of clearance for the jack plug inside the sub cover.	

External Music on Hold

You can connect the audio ports of the XN120 to an external music device. The external music can be played to calls placed in hold.
Each 2PGDU card has two audio ports, you can install one 2PGDU card into each XN120 main/expansion unit giving a total of six audio ports. When you set a 2PGDU audio port as music input it is referred to as an Audio Communication Interface (ACI).

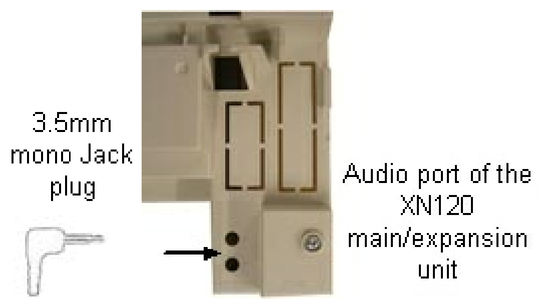
There is also an external music port on the XN120 main unit, this is the System's Music on hold input.

2PGDU audio port specification:
Impedance: 600 Ohm @ 1KHz
Nominal input level: 250mV (-10dBm)
Maximum input level: 1V RMS

Music on Hold Options

- You can have a single external music device that will provide the music on hold for every type of held call on the system.
  The system music source is connected to the MOH input on the XN120 main unit.
  The system music source (XN120 tune or external music input) is defined in Program 10-04-01.
- It is also possible to have more than one external music device connected to the XN120 via a 2PGDU audio port. You could then assign a different music device per trunk or DDI number.
  The music source used for each trunk is defined in Program 14-08-01.
  The music source used for each DDI is defined in Program 22-11-09.
  The 2PGDU audio ports are set as music input in Program 10-03-01.
  You will also need to set the 2PGDU audio ports to Music Input. See the configuration details in this section.

Connect the External Music System

1	Run a single pair audio cable to the external music device. Terminate the cable with a 3.5mm mono Jack plug to connect to the 2PGDU audio port. Note, there is only 26mm of clearance for the jack plug inside the sub cover.	

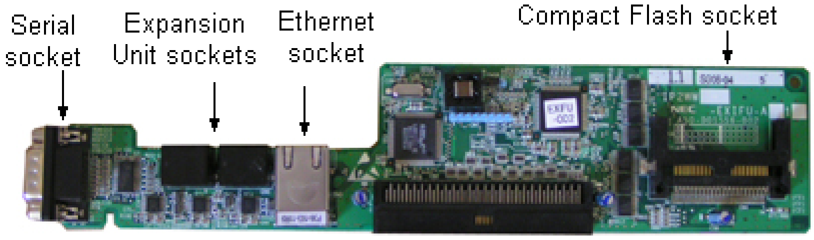
EXIFU Card (Applications Card)

Top

EXIFU Card (Applications Card)

There are two types of EXIFU card available for the XN120 system.
Note that only one EXIFU card can be installed.

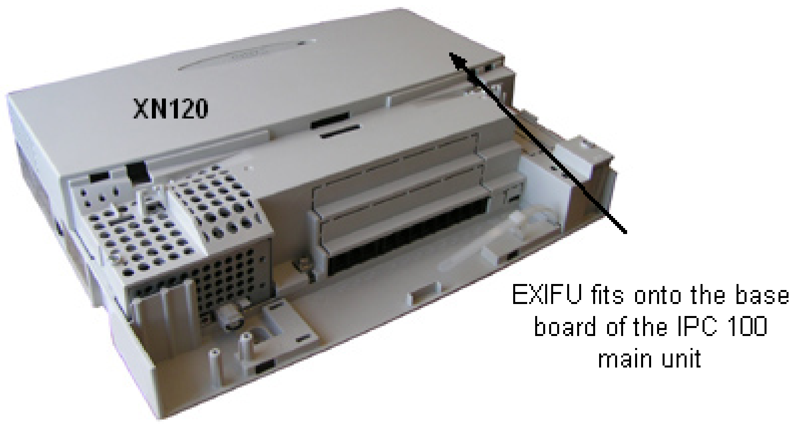
The EXIFU-A1 option card provides:
- Ethernet port
- Expansion Unit Connections
- Serial RS232 port
- Compact Flash socket



<u>The EXIFU-B1 option card provides:</u>

- Serial RS232 port

System Connection Diagram



Installation Procedure

1	**Unpack the EXIFU card**	
2	**Power off the XN120 and install the card**	You will also need to power off any expansion units if they are installed.
3	**Power on the XN120**	
4	**Test the EXIFU**	
5	**Configure the EXIFU**	

Unpack the Card.

There are no cables supplied with the EXIFU.

<u>Additional Items Required:</u>

- Cross head screwdriver.

The EXIFU-A1 card will require:

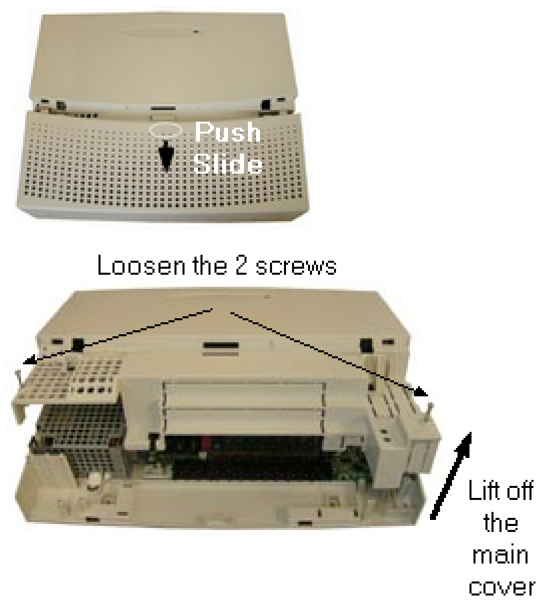
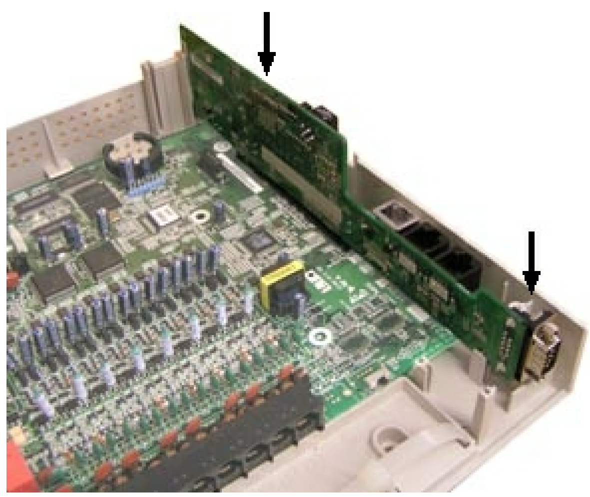
- LAN Cross Cable (or LAN Straight cable connected to a hub) for PC Programming.
- Serial Null Modem Cable (9 pin female to 9 pin female) for SMDR.
- Compact Flash card for saving the system configuration/upgrading the system software.

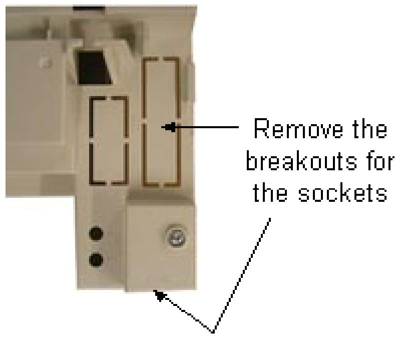
The EXIFU-B1 card will require:
- Serial Null Modem Cable (9 pin female to 9 pin female) for SMDR.

Install the EXIFU

! Observe anti-static precautions when handling the EXIFU card.
- Wear a suitable anti-static strap connected to an Earth point.

One EXIFU card is installed onto the base board within the XN120 main unit.
The EXIFU will be automatically assigned when the system is powered on, after the card is installed.

1	Power off the XN120 system	
2	Remove the sub cover and main cover of the XN120 unit that will have the card installed.	
3	Fit the EXIFU card. The card is installed onto the connector CN2 on the right side of the base board within the XN120 main unit. Ensure the connector is in line before pushing on the card.	
4	Refit the main cover and the sub cover.	Remove the breakouts for the Ethernet/Expansion sockets and the RS-232 socket before refitting the main cover.

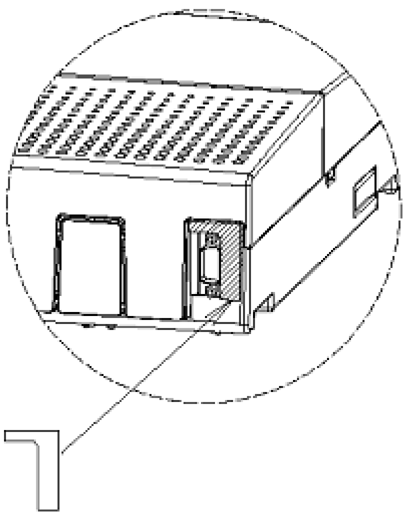
		

Install the Plastic Spacer

The plastic spacer supplied with the XN120 Main Unit must be installed when you have an EXIFU card installed. The spacer is required for Electrostatic Discharge (ESD) protection for the EXIFU card.

<u>Installing the plastic spacer.</u>

1. You do not need to remove any covers from the XN120 Main Unit.
2. Disconnect the cable from the serial port of the EXIFU card.
3. Remove the backing tape from the plastic spacer.
4. Fix the spacer onto the plastic case around the serial port as shown below.
5. Re-connect the cable to the serial port of the EXIFU card.



Test the EXIFU

1	Test the Ethernet port See later in this guide for cable connections of the Ethernet socket.	You will need to connect the Ethernet port of the XN120 to a Network Interface Card (NIC) on your PC. Connect directly via the LAN cross cable. Set your NIC card to the following fixed IP address. IP Address = 172.16.0.11 SubNet Mask = 255.255.0.0 Test the XN120 EXIFU Ethernet port with the PING command. The default IP address of the XN120 is 172.16.0.10 Example. **PING 172.16.0.10** If the Ethernet port is working correctly you will see several replies from

		the XN120, as shown below. **Reply from 172.16.0.10: bytes=32 time=2mS TTL=30** If you do not get a correct reply: Check your NIC card and settings Check the LAN cross cable is plugged in correctly Check the IP address and Sub Net Mask of the XN120, see the Ethernet Port section later in this guide.
2	Test the Serial port See later in this guide for cable connections of the Serial socket.	You will need to connect the serial port of the XN120 to a COM port on your PC. Connect via a null modem serial cable (serial cross cable). Set your COM port to the following: Baud rate = 19200 Data bits = 8 Parity = None Stop bits = 1 Flow control = None You will need a Terminal Application (e.g. Windows HyperTerminal) running to view the output from the XN120. Test the XN120 Serial port by outputting the system report. Using Program 90-13-01 set the Output Type to 1 (serial) Press HOLD to confirm. The display will show OUTPUT COMMAND Enter 1 and press HOLD The system report will be output to the serial port of the EXIFU and will be displayed by the Terminal Application, as shown below. << System Information >> 14/02/2004 04:34 slot location type assign port condition note -- 1 1-1 008/308 1- 8 Running 1- 3 Trunk port 2 1-2 008/308 9- 16 Running 4- 6 Trunk port 3 1-3 -none- -none- Not Install 4 1-4 -none- -none- Not Install 5 1-5 BRIU 17- 20 Running Station port 7- 14 Trunk port 6 1-6 -none- -none- Not Install 7 2-1 -none- -none- Not Install 8 2-2 -none- -none- Not Install 9 2-3 -none- -none- Not Install 10 2-4 -none- -none- Not Install 11 2-5 -none- -none- Not Install 12 2-6 -none- -none- Not Install 13 3-1 -none- -none- Not Install 14 3-2 -none- -none- Not Install

		15 3-3 -none- -none- Not Install 16 3-4 -none- -none- Not Install 17 3-5 -none- -none- Not Install 18 3-6 -none- -none- Not Install
3	Test the Expansion ports	There is no specific method of testing the Expansion ports. They can only be tested by connecting the XN120 expansion unit.
4	Test the Compact Flash socket	There is no specific method of testing the compact flash socket. Refer to the Compact Flash section in this guide for instructions on using the socket.

Ethernet Port

The Ethernet port of the XN120 is used for the following feature.
PC Programming – PC based application used to configure the XN120 system

The Ethernet port will operate when the EXIFU-A1 card is installed, there is no configuration required in order to use PC Programming.
The Ethernet port is assigned the following fixed IP address and subnet mask.
IP Address = 172.16.0.10
Sub Net Mask = 255.255.0.0

This guide shows the configuration required to connect the XN120 to the customers Ethernet LAN. This will enable you to access PC Programming from another PC connected to the customers LAN.

***!* You will need a fixed IP address and the appropriate Sub Net Mask for the XN120, this must be specified by the Network Administrator responsible for the customers LAN.**

Ethernet Port Specification

Standard	IEEE802.3 10Base-T and 100Base-TX compliant	
Access	CSMA/CD	
Interface	Speed	10Mbps/100Mbps Auto Negotiation
	Cable	CAT5 or better Straight/Cross

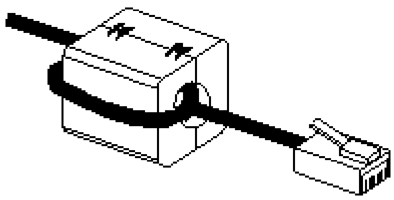
Ethernet Cable

The connections of the Ethernet cable depend on the device that you connect. When connecting to an Ethernet hub you would normally use a straight cable (some hubs will take either a straight or cross cable)
When connecting directly to the NIC card in your PC you must use a cross cable.

RJ45 Ethernet socket on the EXIFU-A1	RJ45 colour code	Ethernet Straight Cable	Ethernet Cross Cable
1	White/Orange	1	3
2	Orange/White	2	6
3	White/Green	3	1
6	Green/White	6	2

Fit the Ferrite Core to the Ethernet Cable

Ensure that you make 1 turn of the Ethernet cable through the ferrite core supplied with the EXIFU-A1 card.
The diagram shows 1 turn of the cable through the ferrite.



Maximum Cable Length

The maximum cable length is 100 metres for straight and cross cables.

Serial Socket - Call Logging (SMDR)

The serial port of the EXIFU-A1 or EXIFU-B1 can be used to output call information for the telephones connected to the system.
You would normally collect the call information with a compatible call logging application loaded onto a separate PC.

You must enable call logging on the XN120 system before the information will be output via the serial port of the EXIFU.

Which trunks will output to the call logger.

You must enable SMDR output for each trunk on the XN120 with program 14-01-06.

Which telephones will output to the call logger.

You must enable SMDR output for each telephone on the XN120 with program 15-01-03.

Enable the call logger output for the system.

You must enable SMDR output for the XN120 with program 35-01-01.

Set the Baud rate for the serial port.

You must set the Baud rate of the XN120 serial port to the same as the COM port of the PC connected to it with program 10-21-02.

Sample SMDR Report

```
                                                             09/01/03 PAGE 001
   CLASS  TIME    LINE    DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
  01 POT   10:44 LINE 001   00:00:30 STA 224    12039265400        8841
  02 POT   10:46 LINE 001   00:00:45 STA 224    18874521              0
  03 POT   10:47 LINE 001   00:00:29 STA 218    12039265441           0
  04 PIN   10:48 LINE 002   00:01:39                                    NO ANSWER
  05 ALB   10:50 02         00:01:40
```

Definitions

Call Record Number	SMDR record number (consecutive)
CLASS	Type of call (see Class Definitions below)
TIME	Time call placed or answered. (For Transferred calls, shows time user picked up Transfer.)
LINE	Trunk number used for call
DURATION	How long call lasted. (For Transferred calls, shows how long user was on call after answering the Transfer.)
STATION	Extension number of call "owner" (i.e., extension that first placed or answered call) (For Transferred calls, there can be more than one owner - depending on how many extensions shared the call.)
DIALLED No./CLI	For outgoing calls, the number dialled or, for incoming calls, the Caller ID information
RD/COST	For outgoing calls, the cost if enabled. For incoming calls, the ring duration.
ACCOUNT	Account Code number entered by extension user
Class Definitions	
POT	Outgoing trunk call
POTA	Outgoing trunk call placed using Toll Restriction Override
PIN	Incoming trunk calls
ALB	All lines in group are busy (group number follows TIME field)
BRD	Call blocked due to Toll Restriction
PTRS	Transferred call
IVIN	Incoming ISDN trunk call
IVOT	Outgoing ISDN trunk call

SMDR Report Format

Character Position	Field Definition

Header Line

1-60 Spaces
61-70 MM/DD/YYYY
71 Space
72-75 PAGE
76 Space
77-79 Report page number (e.g., 001)
CR & LF
Header Line 2
1-3 Spaces
4-8 CLASS
9-10 Spaces
11-14 TIME
15-18 Spaces
19-22 LINE
23-26 Spaces
27-34 DURATION
35-36 Spaces
37-43 STATION
44-46 Spaces
47-53 DIALLED
54 Space
55-61 No./CLI
62-63 Spaces
-70 RD/COST
71 Space
72-78 ACCOUNT
CR & LF Carriage return and line feed
SMDR Record
1-2 Call record number 01-55
3 Space
4-8 Call type (e.g., POT for outgoing)
9 Space
10-14 Time in 24 hour clock (HH:MM)
15 Space
16-25 Line number (e.g., 001)
26 Space
27-34 Call Duration (HH:MM:SS)
35 Space
36-45 Station number (STA, space, nnnn) or name
46 Space
47-61 Number dialled (15 digits maximum)
62-63 Spaces
64-70 Ring duration for incoming. Cost for outgoing
71 Space
72-80 Account number or NO ANSWER

Conditions

1. The SMDR report does not include Intercom calls.
2. The SMDR call buffer stores calls records when the SMDR device is unavailable. When the buffer fills, each new call is not recorded. The alarm display telephone assigned in Program 90-11-01 shows “SMDR Buffer Full,” indicating that the buffer is full. To clear the buffer, the SMDR information must be output. When not using SMDR, make sure Program 90-13-01=0 or Program 90-11-01=0 otherwise the SMDR alarm will display to the extension in Program 90-11-01 or to the operator’s extension.
3. SMDR requires a connection to the EXIFU via a COM port.

SMDR Output Options

You can change the information output via the SMDR. The following options are available.

Omit dialled digits 35-01-04

The SMDR will replace the dialled digits with an X character.
This will prevent the administrator seeing the exact number dialled but will not prevent any billing operation of the call logging

application.

```
   CLASS   TIME    LINE     DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 01 POT    10:44 LINE 001    00:00:30 STA 224    120392654XX         8841
```

Minimum quantity of dialled digits 35-01-05

The SMDR will only output the call record if the minimum quantity of digits is dialled for an outgoing call.

Minimum call duration 35-01-06

The SMDR will only output the call record if the minimum call duration is reached for incoming and outgoing calls.

Minimum ring duration 35-01-07

The SMDR will only output the call record if the minimum ring duration is reached for incoming abandoned calls. An abandoned call is shown as NO ANSWER on the SMDR.

This timer does not affect calls that are answered.

Trunk name or number 35-02-03

The SMDR can either enter the trunk name or trunk number in the LINE field for each call record.

The trunk name is set in program 14-01-01.

```
   CLASS   TIME    LINE     DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 01 POT    10:44 Line 001    00:00:30 STA 224    120392654XX         8841
```

The trunk number is from 001 to 051.

```
   CLASS   TIME    LINE     DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 01 POT    10:44 001         00:00:30 STA 224    120392654XX         8841
```

Extension name or number 35-02-09

The SMDR can either enter the extension name or extension number in the STATION field for each call record.

The extension name is set in program 15-01-01.

```
   CLASS   TIME    LINE     DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 01 POT    10:44 001         00:00:30 P.Jones    120392654XX         8841
```

The extension number is set in program 11-01-02.

```
   CLASS   TIME    LINE     DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 01 POT    10:44 001         00:00:30 224        120392654XX         8841
```

All Lines Busy 35-02-10

The SMDR will output a call record each time a trunk group has all lines busy simultaneously.

The LINE field will show the trunk group number that is busy.

The DURATION field shows how long all lines were busy.

```
   CLASS   TIME    LINE     DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 05 ALB    10:50 002         00:01:40
```

DDI Name output 35-02-12

The call records for incoming ISDN DDI calls can show the name assigned to the DDI in the LINE field.

The DDI names are assigned in program 22-11-03 for each DDI number.

```
   CLASS   TIME    LINE     DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 01 IVIN   10:44 Main No     00:00:30 STA 224    01765768691
```

Date output for each call record 35-01-14

The call records can include the date in the LINE field.

The date is output as dd/mm followed by a space and then three digits to indicate the line number.

Note. When you enable date output the following options are ignored.

Program 35-02-03, Trunk name or number

Program 35-02-12, DDI name output

```
   CLASS   TIME    LINE     DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 01 IVIN   10:44 09/03 001   00:00:30 STA 224    01765768691
```

DDI or CLIP number output 35-02-15

The call records for incoming ISDN DDI calls can show the DDI number instead of the CLIP number in the DIALLED No./CLI field.

```
   CLASS  TIME    LINE    DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 01 IVIN  10:44 001         00:00:30 STA 224   643111
```

DDI number or Line information output 35-02-16

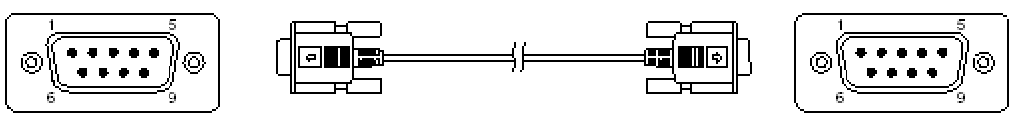
The call records for incoming ISDN DDI calls can show the DDI number instead of the line name or number in the LINE field.

```
   CLASS  TIME    LINE    DURATION  STATION   DIALLED No./CLI  RD/COST ACCOUNT
 01 IVIN  10:44 643111      00:00:30 STA 224    01765768691
```

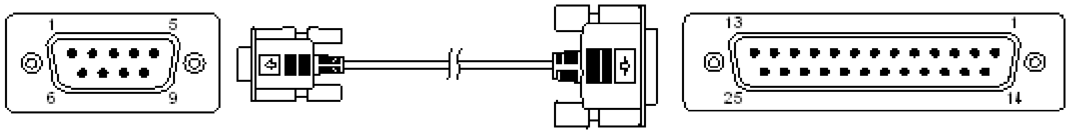
Serial Cable

The serial cable used to connect the EXIFU card to the PC must be a null modem type (cross cable).
The following connections are required.
The EXIFU card has a 9 pin male D-Type connector.
The cable must not exceed 15 metres long.

System		SMDR Device	
Signal Name	Pin No.	Pin No.	Signal Name
RXD	2	2	RXD
TXD	3	3	TXD
DTR	4	4	DTR
SG	5	5	SG
DSR	6	6	DSR
RTS	7	7	RTS
CTS	8	8	CTS



System		SMDR Device	
Signal Name	Pin No.	Pin No.	Signal Name
RXD	2	2	TXD
TXD	3	3	RXD
DTR	4	4	RTS
SG	5	5	CTS
DSR	6	6	DSR
RTS	7	7	SG
CTS	8	20	DTR



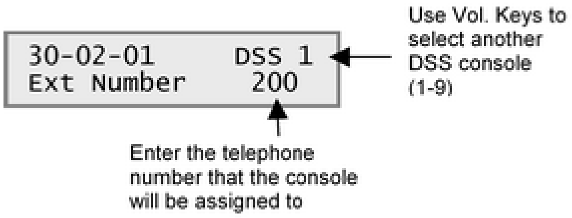
Compact Flash Socket

The compact flash socket of the EXIFU-A1 card can be used for upgrading the XN120 main unit's firmware.

Upgrade the XN120 Main Unit Software

The operating system software of the XN120 system can be upgraded by installing a compact flash card into the socket of the EXIFU card.
The compact flash card has the new revision of XN120 system software copied onto it.

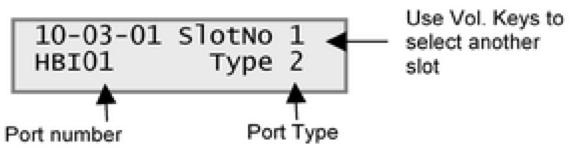
You can confirm the revision of the XN120 system software at any XN120 telephone with a display.
With the phone idle press OPAC and dial 3
The top row of the display will show the software revision
(example below is Version 01.30)



The display will revert to normal after 3 seconds.
Or with program 90-16 via PCPro.

Flash card type

You will need a compact flash card of 16Mbytes or larger.

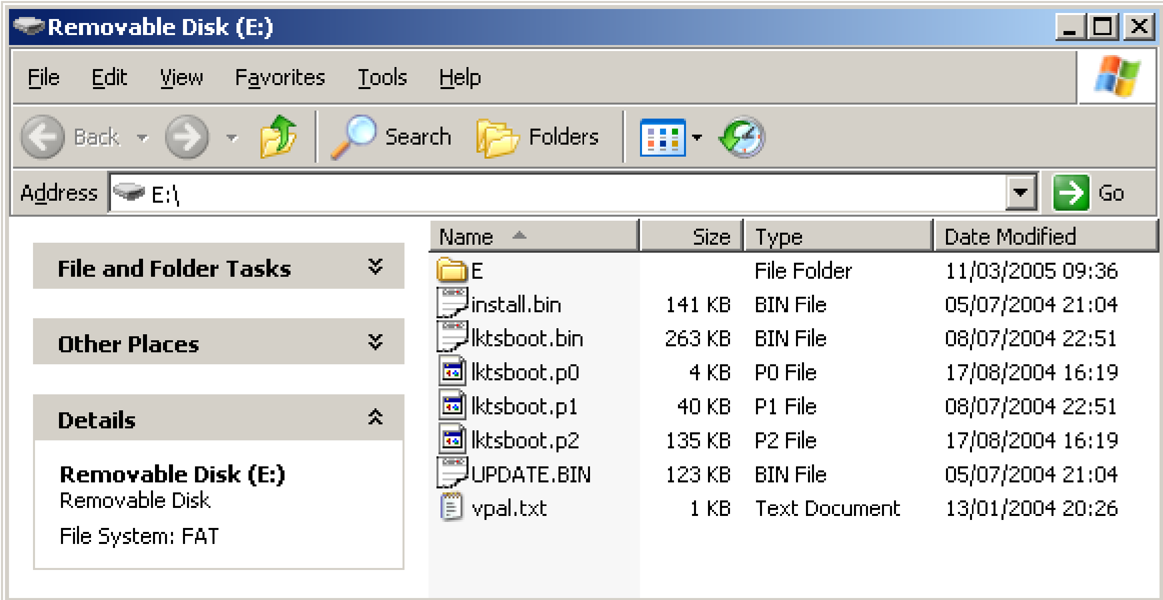


How to build the compact flash card for the XN120 system software.

- The files on the compact flash card must be taken from the XN120 Technical CD.
- Ensure there are no other files on the compact flash card.
- The compact flash must be formatted in FAT. The XN120 will not read the files if the compact flash card is formatted in another type e.g. FAT32.

To build the compact flash card you will need the XN120 system software from the XN120 Technical CD.

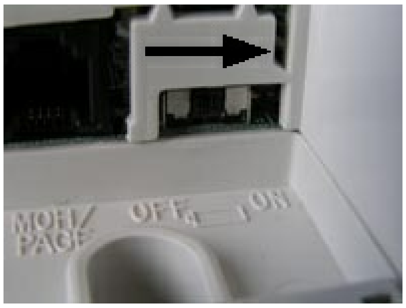
1. Insert the blank compact flash card into your PC.
2. Insert the CD into your PC and from the main menu screen select the XN120 System Software link.
3. You will see each version of XN120 software in separate folders.
4. Double click the folder that contains the version of XN120 software you want to use.
5. Within the folder select ALL files then copy them onto the compact flash card.
6. The structure of the compact flash card will look similar to the screen shot below.
7. The compact flash card is now ready to be inserted into the EXIFU card to perform the software upgrade.



To upgrade the XN120 system software.

It is recommended that you save the system configuration with PCPro before you perform the upgrade.

1	Remove the sub and main covers of the XN120 main unit.	
2	Insert the compact flash card into the socket on the EXIFU	Ensure the lamp LED1on the EXIFU card comes ON, this confirms the compact flash card is inserted correctly.
3	Power off the XN120 system	You must also power off any expansion units if they are installed.
4	Set the NORMAL switch to OFF. With the compact flash card inserted the system will not perform a cold start; you will not erase the customer configuration.	The NORMAL switch is located on the right side of the MOH/PAGE socket on the XN120 main unit. 
5	Power on the XN120 main unit	The new system software will be copied from the compact flash card to the XN120 main unit. Lamp LED4 will flash during the upgrade.

		
6	When LED4 stops flashing power off the XN120 main unit	The upgrade will take 1-2 minutes.
7	Remove the compact flash card	
8	**Set the NORMAL switch to ON** ***!*** If you do not set the NORMAL switch to ON you will erase the customer configuration when you power on the system	The NORMAL switch is located on the right side of the MOH/ PAGE socket on the XN120 main unit. 
9	Power on the XN120 system	
10	Confirm the XN120 system software has upgraded by pressing OPAC 3 at any display phone.	

<u>If the Upgrade fails.</u>
Check that the compact flash card contains all the files from the XN120 Technical CD.
Check the compact flash is formatted in FAT. This is shown in the properties when you insert the card into your PC.
Try another compact flash card.

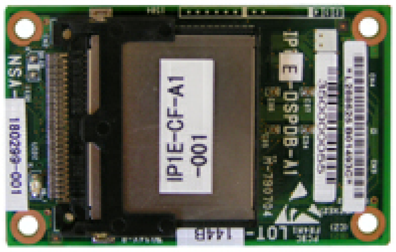
Expansion Socket

Refer to the section for the XN120 Expansion unit within this manual.

DSPDB Card (Voice Mail/VRS)

Top

DSPDB Card (Voice Mail/VRS)



The DSPDB option card provides:

- 8 Channel Voice Mail.
  1 hour storage time (upgradeable to 15 hours).
  300 Mail Boxes.
- 16 Channel Queue Announcement.
  48 user recorded messages.
  Queue announcement for incoming callers waiting
  at a Ring Group or Department Group
- Automated Attendant Operation.
  The 48 user recorded messages are used for the Auto-Attendant prompts.
  Multi-level greetings.
  Single digit translation.
- Pre recorded announcements for various system features.
  Call forward with greeting.
  Park and Page.

One DSPDB card is installed in the XN120 main unit.

Installation Procedure

1	**Unpack the DSPDB card**	
2	**Power off the XN120 and install the card**	You will also need to power off any expansion units if they are installed.
3	**Power on the XN120**	

Unpack the Card.

- The DSPDB card is supplied with:

1 x 64MB Industrial Grade Compact Flash card pre-installed onto the card
4 x Plastic mounting bars

<u>Additional Items Required:</u>

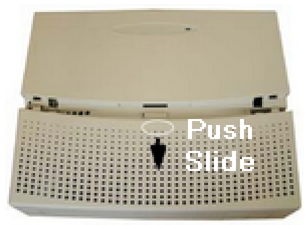
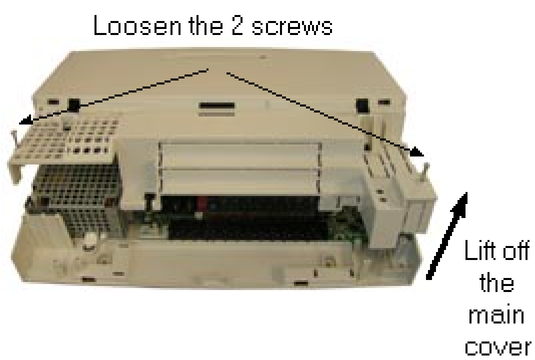
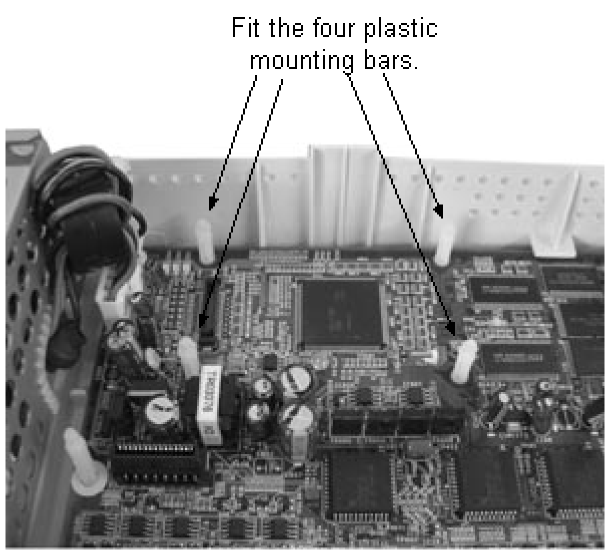
- Cross head screwdriver.

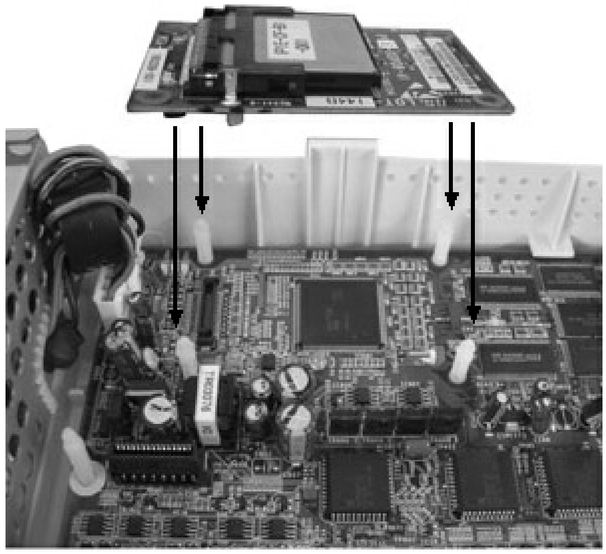
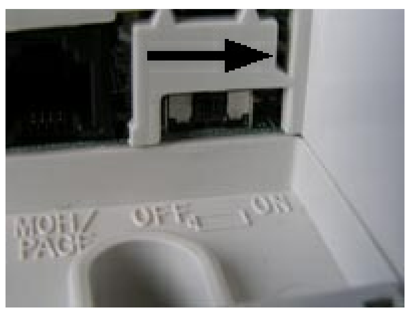
Additional Documents Available:
- Voice Mail and Voice Response System User Guide

Install the DSPDB Card

! Observe anti-static precautions when handling the DSPDB card.
- Wear a suitable anti-static strap connected to an Earth point.

One DSPDB card is installed onto the base board within the XN120 main unit.
The DSPDB will be automatically assigned when the system is powered on, after the card is installed.

1	Power off the XN120 system	You must also power off each XN120 expansion unit if you have any installed.
2	Remove the sub cover and main cover of the XN120 unit that will have the card installed.	 
3	Fit the DSPDB card. The card is installed onto the connector CN6 on the base board within the XN120 main unit. Fit the plastic mounting bars. Install the DSPDB card. Ensure the connector is in line before pushing on the card.	

		
4	Refit the main cover and the sub cover.	If you have any expansion units installed you must power these on first. The DSPDB card will be automatically configured. **! System Start Up – Retain Customer Configuration** This is the normal operation for powering the XN120 on. Before you power on the system check that the NORMAL switch is set to ON. This will ensure that the system memory retains your configuration. The NORMAL switch is located to the right of the MOH/Page connector on the main unit. 
5	Power on the XN120 system	

Test the DSPDB

1	At any XN120 telephone press digit 8 while the phone is idle.	You will hear a pre-recorded announcement of the time and date. If you do not hear the time and date announced first check Program 40-10-01 is set to 1 (enabled).

Upgrading the Compact Flash Card to 15 Hour Capacity

The standard Compact Flash card supplied with the DSPDB card will provide approximately one hour of recording capacity. There is a larger capacity Compact Flash card available that will provide approximately 15 hours recording.

<u>Fit the new CF card – Do not keep the customers messages</u>

If this method is used at an existing customer it will not keep any of the customers recorded messages.

1	Power off the XN120 system	You must also power off each XN120 expansion unit if you have any installed.
2	Remove the Standard Compact Flash card from the DSPDB card	The card pulls out, there are no retaining clips. 
3	Fit the Upgrade Compact Flash card into the DSPDB card	The card pushes in, there are no retaining clips. 
4	Power on the XN120 system	If you have any expansion units installed you must power these on first. The DSPDB card will be automatically configured. ***! System Start Up – Retain Customer Configuration*** This is the normal operation for powering the XN120 on. Before you power on the system check that the NORMAL switch is set to ON. This will ensure that the system memory retains your configuration. The NORMAL switch is located to the right of the MOH/Page connector on the main unit.

Fit the new CF card – Keep the customers messages

This method is recommended if you are upgrading at an existing customer as it will keep the customers recorded messages.

1	Power off the XN120 system	You must also power off each XN120 expansion unit if you have any installed.
2	Remove the Standard Compact Flash card from the DSPDB card	The card pulls out, there are no retaining clips. 

3	Use your PC to make a copy of the Standard Compact Flash card	Fit the Standard Compact Flash card into a compact flash compatible socket of your PC. Copy the folder named VM from the Standard Compact Flash card onto your PC. Remove the Standard Compact Flash card.
4	Use your PC to delete the VM folder on the Upgrade Compact Flash card	Fit the Upgrade Compact Flash card into the compact flash socket of your PC. Delete the VM folder on the Upgrade Compact Flash card.
5	Use your PC to copy the Standard Compact Flash card VM folder to the Upgrade Compact Flash card	Copy the folder named VM from your PC to the Upgrade Compact Flash card. Remove the Upgrade Compact Flash card.
6	Fit the Upgrade Compact Flash card into the DSPDB card	The card pushes in, there are no retaining clips. 
7	Power on the XN120 system	If you have any expansion units installed you must power these on first. The DSPDB card will be automatically configured. ***!* System Start Up – Retain Customer Configuration** This is the normal operation for powering the XN120 on. Before you power on the system check that the NORMAL switch is set to ON. This will ensure that the system memory retains your configuration. The NORMAL switch is located to the right of the MOH/Page connector on the main unit.

BRIU Card (ISDN Basic Rate)

Top

BRIU Card (ISDN Basic Rate)

Basic Rate ISDN is a digital service provided over ordinary copper telephone cables.
Basic Rate ISDN is generally referred to as ISDN BRI.

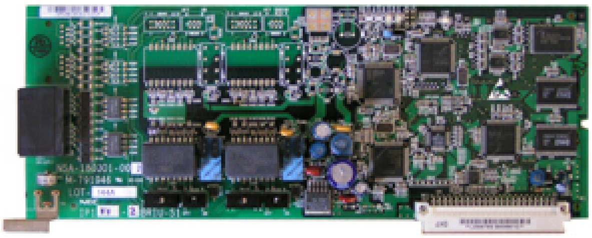
ISDN BRI Lines
ISDN BRI can provide:
Direct Dial In (DDI) facilities. This will enable outside callers to ring directly at the telephones within your system. This can improve efficiency by not having callers transferred through your operator.
Caller ID. The calling number can be displayed at the XN120 system phone or normal phone that supports caller ID. The XN120 can also show the callers name if their number matches an entry in the system speed dials.
You can also specify the caller ID that is sent for your outgoing calls, this can be used to indicate your own DDI number for calls that you make.

ISDN BRI S0

ISDN BRI can provide an S0 interface that will allow the user to connect an ISDN Basic Rate device (e.g. a terminal adapter).
Each circuit of a BRI card can be set individually to S Mode.
The system can route voice or data calls to the devices connected to S0 interface.

XN120 BRI Cards Available

The XN120 has ISDN BRI cards with either 2 or 4 circuits. Each circuit will provide you with two lines, so if you want six lines on the XN120 you will need three ISDN BRI circuits.



The BRI cards are installed in the 2OPBOX unit, refer to the guide supplied with the 2OPBOX for installation instructions.
Each 20PBOX unit can have up to two ISDN cards installed.

System Connection Diagram

ISDN BRI Trunk Mode

Each ISDN BRI circuit is connected to a separate ISDN BRI circuit supplied by your network provider.
When requesting the ISDN BRI circuits from your network provider they may request a number of options:

Point to Point or Point to Multi-Point?

You can connect either type to the XN120.
Check with your network provider that they can support the services you require on the type of line you have selected.
For example some network providers:
Will only supply Direct Dial In (DDI) numbers if you have Point to Point circuits.
Will only supply Multiple Subscriber Numbers (MSN) on one ISDN Point to Multi-Point circuit.

DDI or MSN?

You can use either type with the XN120.
DDI and MSN are very similar in that they allow you to route outside callers directly to telephones on your system.
Check with your network provider that they can support the services you require on the type you have selected.
For example some network providers:
Will provide MSN on one circuit only, so each circuit will have different numbers.
Will provide DDI spread over more than one circuit, so you can have multiple BRI circuits with DDI numbers presented on any line.

How Many DDI or MSN Digits Presented?

You can support up to 8 digits on the XN120.

Calling Line Identity Presentation (CLIP)?

This is supported on the XN120 but you may need to request the service from your network provider.

ISDN BRI S Mode
Each ISDN BRI circuit can be set individually to S Mode (by Program 10-03-01 shown later in this guide).
When connecting the ISDN BRI devices you will have a number of options:

Point to Point or Point to Multi-Point?
You can use either type with the XN120. In P-MP mode you can connect up to 8 S0 devices to one BRI circuit.
DDI or MSN?
The S Mode circuit will send MSN on a Point to Multi-Point circuit and DDI on a Point to Point circuit.
How Many DDI or MSN Digits Presented?
You can send up to 4 digits on the S Mode circuit.
Calling Line Identity Presentation (CLIP)?
This is supported on the XN120. If CLIP is received on a trunk call it will be passed on to the S0 device.

Installation Procedure

1	**Unpack the BRI card(s)**	
2	**Power off the XN120 system**	
3	**Fit the 2OPBOX if not already installed**	
4	**Install the BRI card(s) into the 2OPBOX**	
5	**Connect the BRI circuits to the network**	
6	**Power on the XN120**	
7	**Configure the BRI circuits for trunk mode**	
8	**Configure the BRI circuits for S mode**	

Unpack the Card.

The BRI card is not supplied with any connection cables.

Additional Items Required:
Cross head screwdriver.
RJ45 patch cable for each BRI circuit. (straight through connections).

Optional Items
You must have a spare slot available in a 2OPBOX to install each BRI card.

Power Off the System.

You must power off the XN120 system before you install the BRI card into a slot in the 2OPBOX.

Fit the 2OPBOX.

Refer to the installation instructions supplied with the 2OPBOX.
The BRI card is installed into a free slot in a 2OPBOX, ensure you fit the 2OPBOX before you install the BRI card.

Install the BRI Card

! Observe anti-static precautions when handling the BRIU cards.

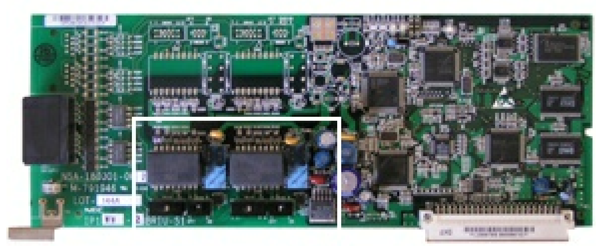
- Wear a suitable anti-static strap connected to an Earth point.

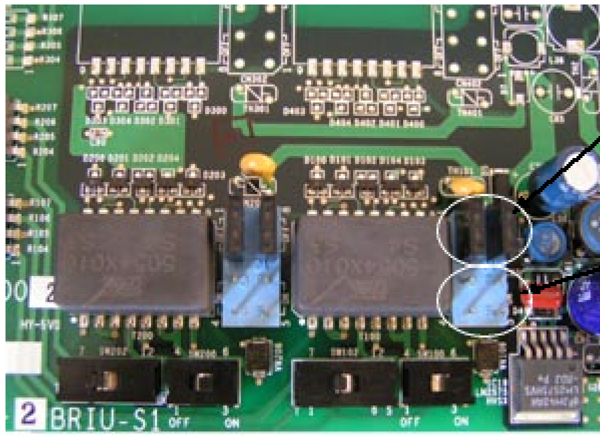
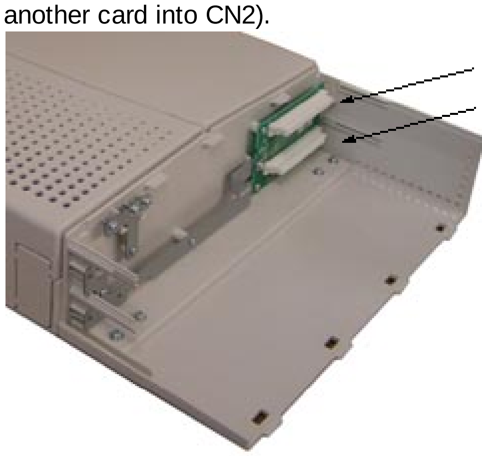
Installation Considerations:

- Each BRI card will use a slot in a 2OPBOX. Each 2OPBOX has two slots.
- The XN120 can have up to three 2OPBOXs, one box can be connected to each XN120 Unit (1 main unit plus 2 expansion units).

Are there sufficient trunk ports available for the BRI card?

- The XN120 can have a maximum of 51 trunk ports, including the trunks used by the XN120 unit(s).
- A two circuit BRI card will use 4 of the XN120's trunk ports.
- A four circuit BRI card will use 8 of the XN120's trunk ports.
- *If there is not sufficient trunk ports available on the system when the BRI card is installed the card will not initialise.*

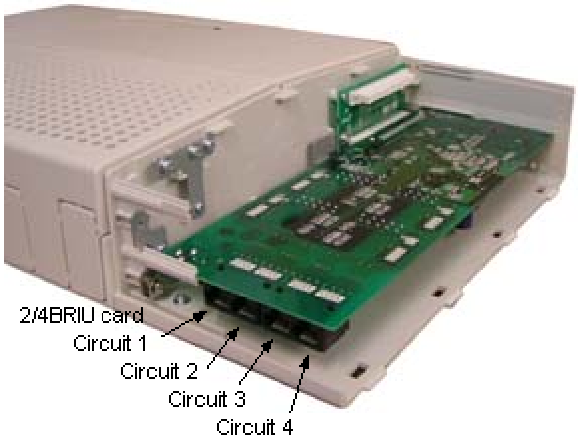
1	Set the switches on each circuit of the BRI card *You do not normally need to change any of these settings.* The factory setting of each circuit is: Termination: ON Power feed: OFF Polarity (Tx/Rx): T mode There are a set of switches for each BRI circuit on the card. Circuit 1 uses: SW100, SW102, CN102 Circuit 2 uses:	**Termination** should be set to ON for either a trunk mode or S mode circuit. You need only set this to OFF if you are connecting the BRI trunk mode circuit onto a P-MP bus and it is not the last device on the bus. **Power feed** should be set to OFF for a trunk mode BRI circuit. Set to ON for an S mode circuit only if the device requires power feed. **Polarity** should be set to T for each BRI trunk mode circuit. Set to S for an S mode circuit. 

	SW200, SW202, CN202 Circuit 3 uses: SW300, SW302, CN302 Circuit 4 uses: SW400, SW402, CN402 *! There is one other link on the BRI card, this must be set to NORMAL.*	
2	Fit the BRI card into a slot in the 2OPBOX ! Don't refit the lid of the 2OPBOX until you have configured the circuits	Ensure the power is switched OFF to the XN120 system. Loosen the two screws and remove the lid of the 2OPBOX. Fit the BRI card into any of the slots. (If you have just one card to install then use Slot CN1 as this will keep the ports assigned to the card in order should you later install another card into CN2). 

Connect the BRI Circuits

1	Connect each BRI circuit to the network If you have BRI trunk circuits that are assigned different dialling numbers by the network provider then make a note of which BRI card/circuit you will connect them to. You will need this when you configure the XN120.	Each BRI circuit is connected via a straight through RJ45 patch cable.

When connecting a Point to Multi-Port S Mode bus you must install a separate RJ45 socket for each device. Connect the blue & green pairs to each socket, number each socket as you will need to terminate the last device on the bus.



The BRI circuit uses the blue and green pairs of the RJ45 patch cable.

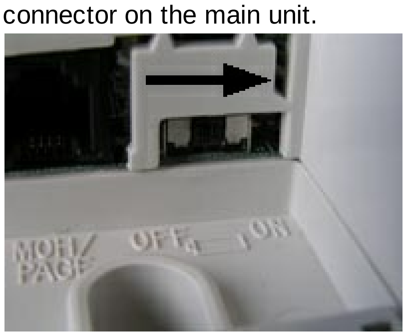
RJ45 Pin number	RJ45 colour	Connection Polarity = T mode	Connection Polarity = S mode
3	White/green	TA	RA
4	Blue/white	RA	TA
5	White/blue	RB	TB
6	Green/White	TB	RB

Maximum cable length of the S-bus using CAT5 cable.
Point to Multi-Point = 300 metres
Point to Point = 500 metres

Power on the XN120

1 Power on the XN120.

! System Start Up – Retain Customer Configuration
This is the normal operation for powering the XN120 on.
Before you power on the system check that the NORMAL switch is set to ON. This will ensure that the system memory retains your configuration.
The NORMAL switch is located to the right of the MOH/Page connector on the main unit.



2	The BRI card(s) will be automatically configured.	Check the BRI card(s) has automatically configured by the LIVE lamp flashing green on each card. If the LIVE lamp is off the card is not initialised correctly. Check the card is fully inserted into the slot (power off the system first). Check the slot has not had a different type of card installed, see Program 10-03-01 below. Check there are sufficient trunk ports available on the system, maximum of 51 trunks.

Configure the Circuits for Trunk Mode

The default configuration for all BRI circuits is Point to Multi-Point, Trunk mode.

Changing the BRI card setting

You can set each circuit to Trunk or S-bus mode on the XN120 BRI cards via the Card Configuration screen within PCPro.

Point to Point or Point to Multi-Point

! You must set the XN120 circuit to the same type as the ISDN BRI Network it is connected to otherwise the circuit will not operate correctly.

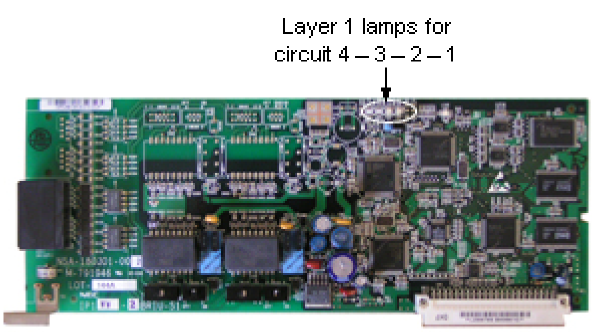
If you are not certain of the setting of the Network:
If the network is supplying DDI's the circuit should be Point to Point.
If the network is supplying MSN the circuit should be Point to Multi-Point.

Test the BRI Lines

Once each BRI circuit is configured and connected to the Network you should test each line on the XN120 before you continue.

Test 1 – Layer 1 Lamp.

Each BRI circuit has a lamp on the BRI card to indicate the condition of the connection to the network.
! This is a basic confirmation that the connection is made, it does not necessarily mean that the circuit is working fully.



Each lamp will indicate the status of the circuit as follows:
Lamp off = No connection
Lamp on = Layer 1 connected correctly

Test 2 – Outgoing Call.

If you have layer 1 you should be able to seize each line of the BRI circuit from a phone connected to the XN120.
You must determine which lines each BRI circuit has been allocated on the XN120, this was done when you configured each circuit in the previous section. Refer to the Configuration Sheet: BRI Card Assignment.
The easiest way to test a BRI line is to assign a line key at a XN120 system phone, there are two lines per BRI circuit.
Press the line key (for each line of the BRI circuit)

The line key flashes green, you hear exchange dial tone and you can continue to dial a test number.
The line is working correctly.
The line key shows red and you do not hear exchange dial tone.
The BRI circuit may be out of service at the Network.
The line can not be seized when the key is pressed.
There is no BRI connection, check the layer 1 lamps.
The line key goes green for a few seconds and then goes off.
The BRI card/network may not both be set to P-P or P-MP. Try changing the type in Program 10-03-03 for the BRI circuit.

Test 3 – Incoming Call.

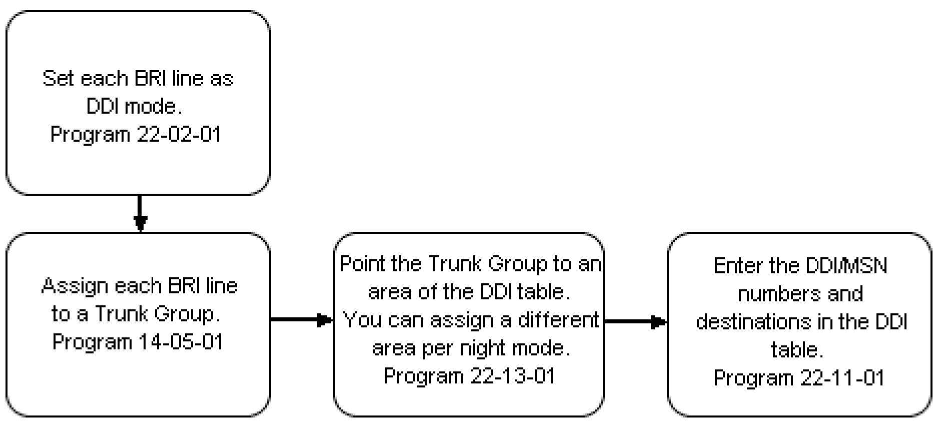
If you have outgoing calls on each line you should be able to receive an incoming call also.
At default the XN120 will accept any type of call on a BRI line and route it to telephone 200.
If incoming calls receive busy indication the BRI card/network may not both be set to P-P or P-MP. Try changing the type in Program 10-03-03 for the BRI circuit.

DDI/MSN Operation.

The XN120 will route both DDI and MSN numbers, it does not distinguish between them.
The only difference is that the network can present the same DDI numbers on more then one BRI circuit. With MSN they will only present them on one circuit, therefore each circuit will have different MSN digits.
With DDI/MSN operation the network will pass the last digits of the dialled number to the XN120. The XN120 then compares these digits to a look up table, if the number matches any entry in the look up table it will follow the routing specified.

In brief,

- Set each BRI line as DDI mode in Program 22-02-01, ensure you do this for all night modes.
- You then place the lines into Trunk Groups in Program 14-05-01, route each Trunk Group to an area within the DDI table in Program 22-13-01.
- Then enter all DDI/MSN numbers in the area specified with their associated destination in Program 22-11-01.



Configure the BRI Lines for Caller ID

Receiving Caller ID

The BRI lines will receive Caller ID at default, the Caller ID will be displayed at the LCD display of the XN120 telephones or normal telephones with caller ID enabled (Caller ID is enabled by Program 15-03-09 for each normal telephone).

For the correct display of the CLIP you must enable the automatic insertion of the prefix digits (0 for national calls and 00 for international calls).
This is enabled by Program 20-19-03, set this option to 1 to enable the insertion of the 0 or 00 prefix digits.

If the Caller ID received on an incoming call matches an entry in the system Telephone Book then the name assigned in the Telephone Book will also be displayed.

Sending Caller ID

The system can send caller ID for outgoing calls.

The Caller ID can be specified both per line and per telephone, if both are set then the telephones will take priority.
Caller ID per line is set in Program 21-12-01.
Caller ID per telephone is set in Program 21-13-01.
Ensure that you enter the full telephone number including local area code (e.g. 01509123456).
The Caller ID you send must also be a valid number for the BRI line that you are making the call on e.g. if the BRI line has a DDI range of 01509 123450 to 01509 123459 then you can only send a Caller ID within this range.

Note that Caller ID may not be passed on by your network supplier, you may need to request this as an additional service.

Configure the Circuits for S Mode

The default configuration for all BRI circuits is Point to Multi-Point, Trunk mode. You will need to change this to S Mode in Program 10-03-01 to S Mode.

Changing the BRI card setting

You can set each circuit to Trunk or S-bus mode on the XN120 BRI cards in the Card Configuration screen within PCPro.

Point to Point or Point to Multi-Point

! You must set the XN120 circuit to the same type as the ISDN device to be connected otherwise the device will not operate correctly.

If you are not certain of the setting of the device:
The most common type for terminal adapters and other ISDN devices is Point to Multi-Point.

Power Feed to the S0 Device

The S0 device may require power feed to be supplied by the XN120 BRI S mode circuit.
Power feed is set by the links on each BRI circuit by the links on the card, refer to Installing BRI Card on page 87 for details of the links.

S Mode Polarity

The polarity of the Transmit and Receive can be reversed by T/S Polarity switch on each BRI circuit on the card, refer to Installing BRI Card on page 87 for details of the polarity switch.
Setting the switch to S will allow you to use a straight through patch cable to connect the S0 device to the BRI circuit.

Termination

You must set the termination to ON for each S Mode circuit. This is the case for both Point to Point and Point to Multi Point type.

Maximum Cable Length

The maximum cable length from the BRIU card when using CAT5 (0.5mm diameter conductor) cable.
Point to Multi-Point = 300 metres
Point to Point = 500 metres

Configure the DDI/MSN digits sent to the S0 device

The DDI/MSN digits are determined by the extension numbers of the two extension ports assigned to the BRI S-Mode circuit. The extension ports can be verified by Program 10-03-02.
In the example above the BRI circuit has been assigned extension ports 11 and 12.
The default extension numbers of these ports are 210 & 211 (set by Program 11-02-01).
Therefore to route a call to the BRI S mode circuit you can dial either extension number 210 or 211, the system will send the number you dialled as the DDI/MSN digits.
You then configure the S0 device to respond to one of these numbers, usually within the MSN setup of the S0 device.

What if I have more than two S0 devices connected to the same S Mode circuit?

It is possible to connect up to eight S0 devices to a single Point to Multi-Point BRI S Mode circuit.
To route calls to each device they will need a separate MSN number, at default there are only two numbers assigned to the BRI S mode circuit.
To increase the amount of DDI/MSN numbers available the XN120 allows you to add an additional digit onto the end of the two extension numbers. This will give 20 DDI/MSN numbers per BRI S Mode circuit.
In the example one additional digit will give DDI/MSN numbers:

Extension number set by Program 11-02-01 is **210**, one additional digit will give **210**0 to **210**9.
Similarly, extension number **211** will give **211**0 to **211**9.

The additional digit is added for each BRI S Mode circuit by Program 10-03-07.

CLIP sent from the XN120 to the S0 device

The system will pass the CLIP received on an incoming trunk call to the S0 device.
The system will pass the telephone number of the calling telephone for internal calls made to an S0 device.

CLIP sent by the S0 device to the XN120

The system will pass the CLIP received from the S0 device to an outgoing ISDN trunk call.
Note that the network provider may not pass on the CLIP information, you may need to request the service.

Voice and Data calls

The system will route both voice and data calls to the S0 devices, it is the responsibility of the S0 device to respond appropriately.
For outside calls made from the S0 devices the system will select an ISDN trunk (even if an analogue trunk is available). If you want the BRI S mode circuits to be able to seize an analogue trunk for voice calls then you must enable Program 20-25-10.

Routing DDI calls to an S0 device

To route a DDI to an S0 device simply enter the DDI/MSN number of the S0 device as the target in Program 22-11-02. If you have assigned additional DDI/MSN digits in Program 10-03-07 you must include these within the target number.
In the example above we have DDI/MSN extension numbers 2100 to 2109 and 2110 to 2119, any of these numbers can be entered as the target for the DDI call.
When you route a trunk call (for example a DDI call) to an S0 device the system will use the target number entered in Program 22-11-03 as the DDI/MSN number for the call to the BRI S Mode circuit (not the DDI number received from the network).

For example:
The system has an ISDN BRI trunk with DDI number 643123 routed to target number 2109.
The system will send 2109 as the DDI/MSN number on the S Mode circuit.
The S0 device that is configured to respond to MSN 2109 will then receive the DDI call.

VOIPU Card (Voice Over IP)

Top

VOIPU Card (Voice Over IP)

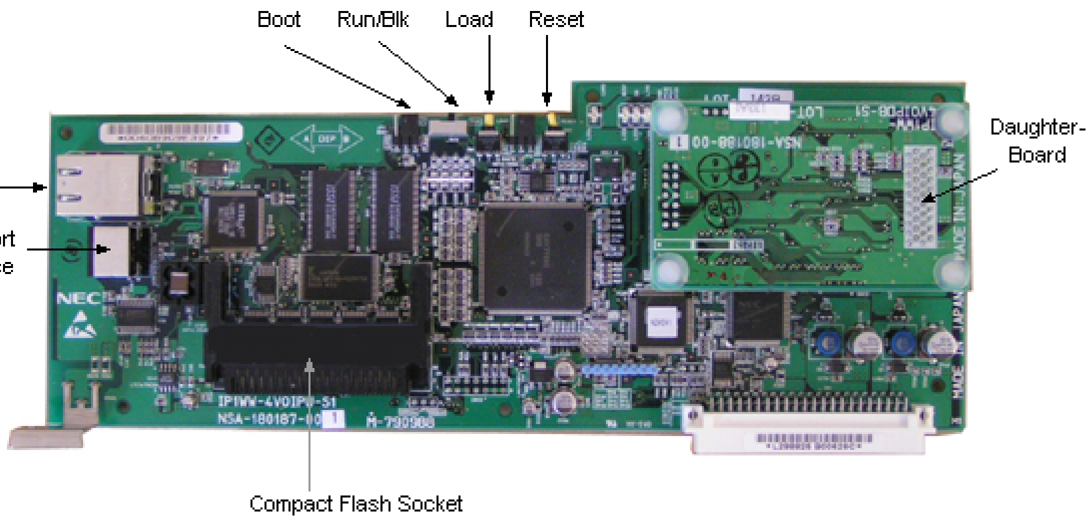
Voice Over IP is a general term used to describe the transmission of voice over IP-based data networks. There are several different VoIP protocols in use worldwide. The most recent protocol, and the protocol used by the XN120, is called Session Initiation Protocol (SIP). SIP is defined by the Internet Engineering Task Force (see http://www.ietf.org).

SIP Trunks can be used to connect the XN120 to other XN120 systems (or some other SIP-compliant telephone systems). This allows calls to be made between the systems over existing data networks rather than using the PSTN.

XN120 VOIPU Cards Available

Each XN120 VOIPU card has 4 Digital Signal Processors (DSP) on board. Each DSP can handle one VoIP call. There is also an optional 4 port daughterboard that can be installed onto the VOIPU to increase the number of DSPs to 8.

The photograph below shows a 4VOIPU card with the 4VOIPDB installed. The card provides 8 DSP resources and therefore supports up to 8 simultaneous VoIP calls.



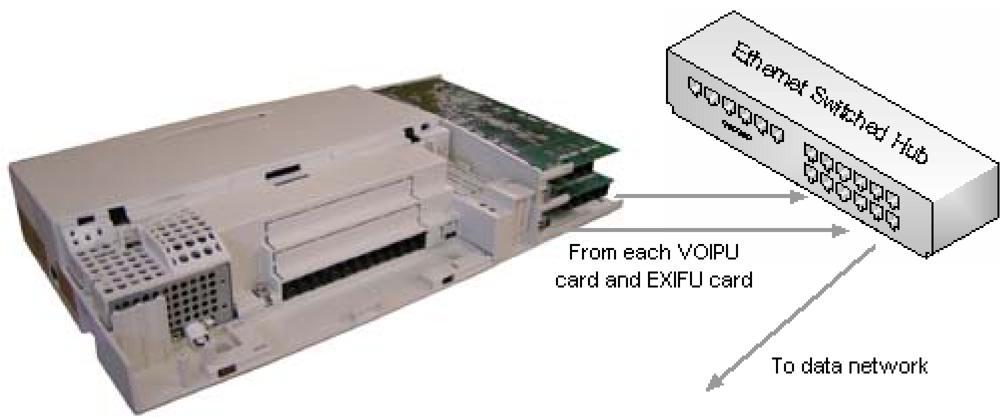
Note:
An EXIFU-A1 is also required for VoIP on the XN120. Please refer to the EXIFU Guide for further information about this card

The VOIPU cards are installed in the 2OPBOX unit, refer to the guide supplied with the 2OPBOX for installation instructions. Each 20PBOX unit can have up to two VOIPU cards installed.

System Connection Diagram

A VOIPU card has one Ethernet port only, even if a VOIPDB is installed. This port is used to transmit/receive the IP packets for all calls active on the card.

Each VOIPU card, and the Ethernet port on the EXIFU-A1 connects to an Ethernet hub (or Switched Hub) provided by the customer.



Installation Procedure

1	**Unpack the VOIPU card(s)**	
2	**Install Daughterboard (if required)**	

3	**Power off the XN120 system**	
4	**Fit the EXIFU-A1 if not already installed**	
5	**Fit the 2OPBOX if not already installed**	
6	**Install the VOIPU card(s) into the 2OPBOX**	
7	**Connect the VOIPU cards and EXIFU-A to the data network**	
8	**Power on the XN120**	
9	**Configure the IP Addresses**	
10	**Test the IP network connectivity**	
11	**Configure SIP Trunks**	
12	**Configure the Incoming Calls**	
13	**Configure the Outgoing Calls**	
14	**Configure the Caller ID**	

Unpack the Card.

The VOIPU card is not supplied with any connection cables.

Additional Items Required:
Cross head screwdriver.
RJ45 patch cable for each VOIPU. (straight through connections).

Optional Items

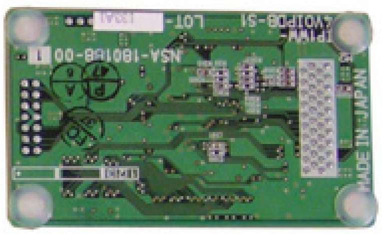
You must have a spare slot available in a 2OPBOX to install each VOIPU card.

Install Daughterboard (if required).

! Observe anti-static precautions when handling the VOIPU cards.

- Wear a suitable anti-static strap connected to an Earth point.

- Place the VOIPU on a flat surface, with the PCB facing upwards.
- Remove the daughterboard from its packaging and locate the black 40-pin connector on the PCB (CN11 on the VOIPU and CN6 on the Daughterboard).
- Gently place the daughterboard onto the VOIPU so that the 4 plastic legs locate into the holes on the VOIPU and the 40-pin connectors meet up.
- Apply firm, even pressure to the four plastic areas on the daughterboard until the legs click into place on the VoIPU.



Fit the EXIFU-A1.
Refer to the installation instructions for the EXIFU-A1.

Fit the 2OPBOX.
Refer to the installation instructions for the 2OPBOX.
The VOIPU card is installed into a free slot in a 2OPBOX, ensure you fit the 2OPBOX before you install the VOIPU card.

Install the VOIPU Card

! Observe anti-static precautions when handling the VOIPU cards.

- Wear a suitable anti-static strap connected to an Earth point.

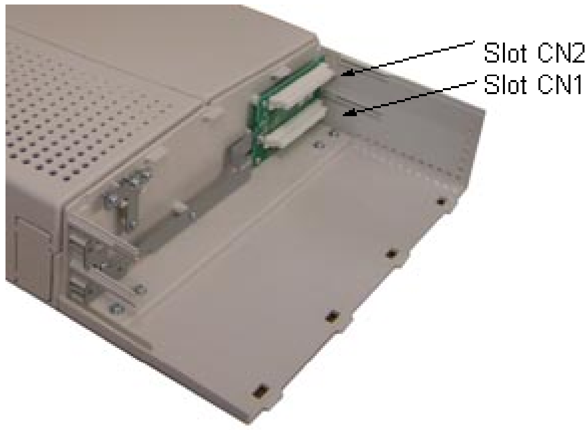
Installation Considerations:

- Each VOIPU card will use a slot in a 2OPBOX. Each 2OPBOX has two slots.
- The XN120 can have up to three 2OPBOXs; one box can be connected to each XN120 Unit (1 main unit plus 2 expansion units).

Are there sufficient trunk ports available for the VOIPU card?

- The XN120 can have a maximum of 51 trunk ports, including the trunks used by the XN120 unit(s).
- A VOIPU will use 4 of the XN120's trunk ports.
- A VOIPU with a VOIPUDB installed will use 8 of the XN120's trunk ports.
- *If there are not sufficient trunks ports available on the system when the VOIPU card is installed the card will not initialise.*

1	Fit the VOIPU card into a slot in the 2OPBOX	nsure the power is switched OFF to the XN120 system. Loosen the two screws and remove the lid of the 2OPBOX. Fit the VOIPU card into any of the slots. *(If you have just one card to install then use Slot CN1 as this will keep the ports assigned to the card in order should you later install another card into CN2).*

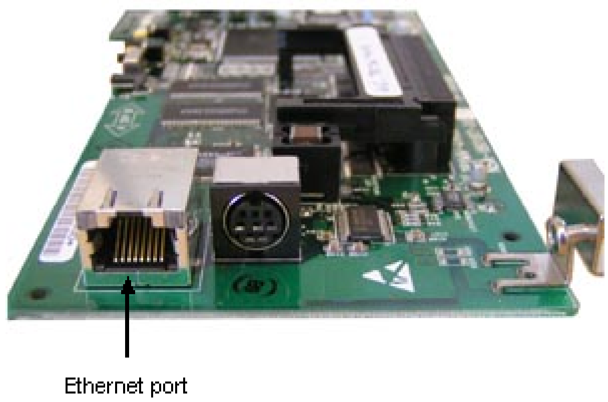


2	Tighten the screw at the bottom edge of the VOIPU to secure the card to the 2OPBOX mounting rail	
3	Refit the 2OPBOX Lid	

Connect the VOIPU Circuits to the Data Network

4 Connect each VOIPU circuit to the data network.

At this point you should also connect the EXIFU-A1 Ethernet port to the data network.



Ethernet Port Specification

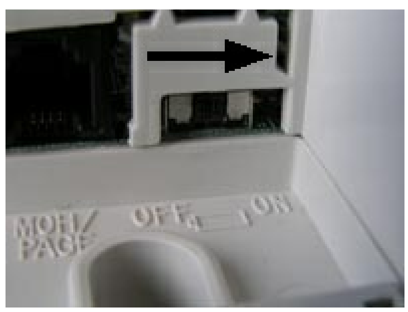
Standard	IEEE802.3 10Base-T and 100Base-TX compliant	
Access	CSMA/CD	
Interface	Speed	10Mbps/100Mbps Auto Negotiation
	Cable	CAT5 or better Straight/ Cross

Ethernet Cable

The connections of the Ethernet cable depend on the device that you connect. When connecting to an Ethernet hub you would normally use a straight cable (some hubs will take either a straight or cross cable)

<table>
<tr><td></td><td></td><td>
When connecting directly to the NIC card in your PC you must use a cross cable.

<table>
<tr><td>RJ45 Ethernet socket on the VOIPU</td><td>RJ45 colour code</td><td>Ethernet Straight Cable</td><td>Ethernet Cross Cable</td></tr>
<tr><td>1</td><td>White/ Orange</td><td>1</td><td>3</td></tr>
<tr><td>2</td><td>Orange/ White</td><td>2</td><td>6</td></tr>
<tr><td>3</td><td>White/ Green</td><td>3</td><td>1</td></tr>
<tr><td>6</td><td>Green/ White</td><td>6</td><td>2</td></tr>
</table>

<u>Maximum Cable Length</u>
The maximum cable length is 100 metres for straight and cross cables.
</td></tr>
<tr><td></td><td></td><td></td></tr>
<tr><td>5</td><td>Power on the XN120.</td><td>
! System Start Up – Retain Customer Configuration
This is the normal operation for powering the XN120 on.
Before you power on the system check that the NORMAL switch is set to ON. This will ensure that the system memory retains your configuration.
The NORMAL switch is located to the right of the MOH/Page connector on the main unit, refer to the Getting Started Guide supplied with the XN120 for further information.


</td></tr>
<tr><td></td><td></td><td></td></tr>
<tr><td>6</td><td>The VOIPU card(s) will be automatically configured.</td><td>
Check the VOIPU card(s) has automatically configured by the LIVE lamp flashing green on each card.

If the LIVE lamp is off the card is not initialised correctly.
Check the card is fully inserted into the slot (power off the system first).
Check the slot has not previously had a different type of card installed Check the Card Configuration screen within PCPro).
Check there are sufficient trunk ports available on the system, maximum of 51 trunks.
</td></tr>
</table>

Configure the IP Addresses.

All Ethernet ports need to have valid IP addresses assigned. IP Address details are usually supplied by the customer's Network Administrator. Note that the XN120 has default IP addresses assigned to each Ethernet port. If these addresses conflict with addresses used on the customers existing network, network disruption will occur.

Default IP details:

EXIFU-A1 IP Address	172.16.0.10	
VOIPU Slot 5	172.16.0.24	(Cabinet 1, CN1)
VOIPU Slot 6	172.16.0.25	(Cabinet 1, CN2)
VOIPU Slot 11	172.16.0.30	(Cabinet 2, CN1)
VOIPU Slot 12	172.16.0.31	(Cabinet 2, CN2)
VOIPU Slot 17	172.16.0.36	(Cabinet 3, CN1)
VOIPU Slot 18	172.16.0.37	(Cabinet 3, CN2)
Subnet Mask	255.255.0.0	
Default Gateway	0.0.0.0	

This guide shows the configuration required to connect the XN120 to the customers Ethernet LAN.

! **You will need a fixed IP address and the appropriate Subnet Mask for the XN120, this must be specified by the Network Administrator responsible for the customers LAN.**

Program 10-12-01

Enter the IP Address of the XN120 system.

Program 84-05-01

Enter the IP Address for each VOIPU card.

Test IP Connectivity

Once the VOIP card and EXIFU-A1 Ethernet ports have been configured and connected to the Data Network you should test connectivity.

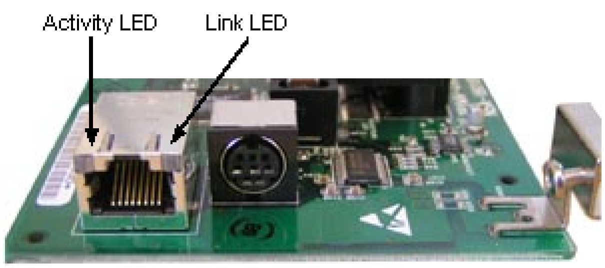
Test 1 – Layer 1 Lamp.

Each Ethernet port has two lamps next to the RJ45 connector to indicate the status of the port.

! This is a basic confirmation that the connection is made, it does not necessarily mean that the circuit is working fully.

The orange link LED indicates that there is a Layer 1 link with the Ethernet hub/switch. This should be on when a connection is present.

The green activity LED indicates Ethernet activity. Every frame that is transmitted or received by the Ethernet port will cause the activity LED to light temporarily. When calls are in progress, or on a busy Ethernet network you will notice that this LED is lit almost constantly.



Test 2 – PING.
If there is a layer 1 connection and the IP details are correct, it should be possible to “ping” the VOIPU card and the EXIFU-A1 Ethernet port. PING (Packet InterNet Groper) is a standard utility that is run from a PC that sends a small packet of data to a specified IP address. When the “ping” is received, the receiving device returns the data to the sender. The original sender can then determine if the device is available on the network and can indicate the network delay between the two endpoints.

To ping from a Microsoft Windows PC (method may vary slightly dependant on Operating system):

- Ensure the PC has a valid IP address and is connected to the data network
- Click on the Start button
- Click on Run…
- Type cmd in the Open dialogue box and click OK. A command prompt window will open.
- Type ping 172.16.0.10 (replace 172.16.0.10 with the actual IP address that you want to test) and then press Enter.

The output will resemble one of the following outputs:

a) Successful – The PC receives a reply back from the XN120 and the reply is received in approx. 1milli-second

```
Microsoft Windows XP [Version 5.1.2600]
(C) Copyright 1985-2001 Microsoft Corp.

C:\>ping 172.16.0.10

Pinging 172.16.0.10 with 32 bytes of data:

Reply from 172.16.0.10: bytes=32 time=1ms TTL=30
Reply from 172.16.0.10: bytes=32 time<1ms TTL=30
Reply from 172.16.0.10: bytes=32 time<1ms TTL=30
Reply from 172.16.0.10: bytes=32 time<1ms TTL=30

Ping statistics for 172.16.0.10:
   Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
Approximate round trip times in milli-seconds:
   Minimum = 0ms, Maximum = 1ms, Average = 0ms
```

b) Unsuccessful. No reply is received to the ping. This could indicate a cabling problem, or an IP addressing problem.

```
Microsoft Windows XP [Version 5.1.2600]
(C) Copyright 1985-2001 Microsoft Corp.

C:\>ping 172.16.0.10

Pinging 172.16.0.10 with 32 bytes of data:

Request timed out.
Request timed out.
Request timed out.
Request timed out.

Ping statistics for 172.16.0.10:
   Packets: Sent = 4, Received = 0, Lost = 4 (100% loss),
```

Note:
If the Data Network is separated into different subnets, it is advisable to ping from PCs on all relevant subnets. This ensures that the IP routing is functioning correctly.

Configure SIP Trunks

Changing the VOIPU trunks to SIP mode
It is necessary to change each VOIPU trunk to SIP mode by Program 10-03-01.

Note. When you enter Program 10-03-01 you will see the setting for Port 01 of Slot 01 which is always a Hybrid extension port. So before you make any changes you must use the VOLUME ▲ ▼ keys to scroll to the correct slot number that your VOIPU card is plugged into, see below.

The 2OPBOX connected to the Main XN120 unit is assigned slots 5 and 6.
The 2OPBOX connected to the First XN120 expansion unit is assigned slots 11 and 12.
The 2OPBOX connected to the Second XN120 expansion unit is assigned slots 17 and 18.

Setting Global SIP Parameters
The XN120 has several system-wide settings that have to be programmed to enable SIP to work. These items are located in Program 10-28.

Setting CODEC Options
The XN120 uses a CODEC (Coder/Decoder) to convert speech into IP packets. Industry standard CODECs are used to allow interoperability between un-like devices. The choice of CODEC to use will depend on various factors, including the type of data network, the amount of bandwidth available and the capabilities of connected equipment.

The XN120 CODEC (called the “Audio Capability Priority”) is changed in PRG84-13-28 for SIP Trunk calls.

The example configuration gives an example of the configuration required on a typical SIP Network installation.

Program 10-03-01
You will need to change this setting to match the network setting.
Program 10-28-01
Domain Name. This setting is the name of the SIP domain.

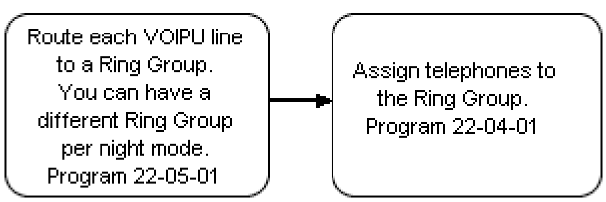
Configure Incoming Calls

The XN120 supports two methods of handling incoming SIP calls; normal (non-DDI) or DDI.

Non-DDI Operation.

If the SIP network does not supply any DDI digits then the BRI lines can be routed in exactly the same way that you would route an analogue trunk.

In brief,
- Route each VOIPU line to an Incoming Ring Group (IRG) in program 22-05-01. A line can route to a different IRG in each night mode.
- You then place telephones into IRG’s in program 22-04-01, a phone can be a member of more than one IRG.



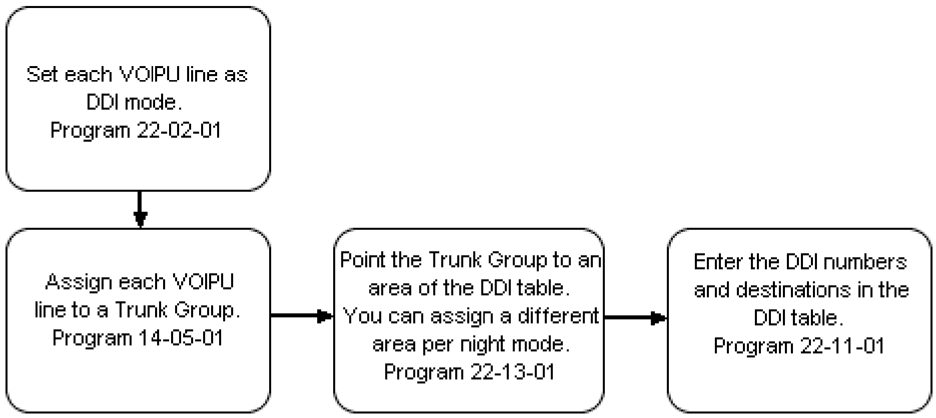
DDI Operation.

The XN120 can also route DDI numbers. This allows calls to be presented to individual extensions (desk-to-desk dialling). This is the most common method of handling incoming SIP calls.

With DDI operation the network will pass the last digits of the dialled number to the XN120. The XN120 then compares these digits to a look up table, if the number matches any entry in the look up table it will follow the routing specified.

In brief,

- Set each VOIPU line as DDI mode in Program 22-02-01, ensure you do this for all night modes.
- You then place the lines into Trunk Groups in Program 14-05-01, route each Trunk Group to an area within the DDI table in Program 22-13-01.
- Then enter all DDI numbers in the area specified with their associated destination in Program 22-11-01.



Configure Outgoing Calls

There are several methods of dialling SIP destinations using the XN120. All of these methods (detailed below) use a standard dialling table to determine the IP address of the remote destination.

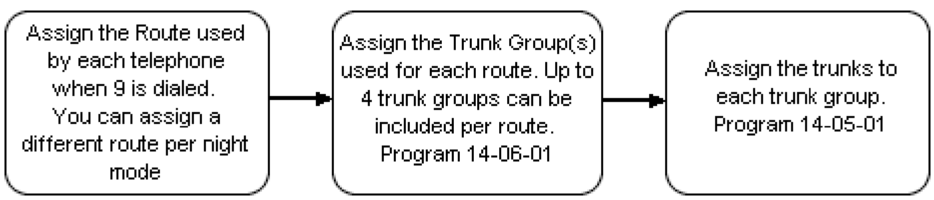
It should be noted that SIP calls can only be made using en-bloc dialling. This means that there will be a delay after dialling the digits before the call is set up. This delay is called the inter-digit timer and is configured in PRG21-01-03. This timer can be reduced but it is a system-wide setting so this will impact on other trunks in the system. An alternative is to dial a # after the required digits. # indicates the end of the dialled digits and this causes the call to be set up immediately.

Manually seize a VOIPU trunk (line key)

The exchange lines of each VOIPU circuit can be presented under a function key in the same way as the analogue/BRI lines. The lines provided by each VOIPU card can be confirmed by Program 10-03-01, explained previously.

Automatically seize a VOIPU trunk (dial 9)

To place an outgoing call the user can dial 9 to seize any free line.
At default dialling 9 will seize any free trunk in Trunk Group number 1. If you have changed the Trunk Group number of the VOIPU lines (in Program 14-05-01) you will also need to include the new trunk group number within the route used when 9 is dialled. You can assign up to 4 trunk group numbers per route by Program 14-06-01.



This method is usually used when all calls are to be made via the VOIPU circuits.

Flexible Routing (f-route)

Many XN120 installations utilise SIP trunks for site-to-site calls and the PSTN for calls to external numbers. Typically 9 is used as the first digit for calls to PSTN numbers and internal calls have a different numbering scheme. To allow calls to be routed the relevant trunk automatically, F-route can be used.

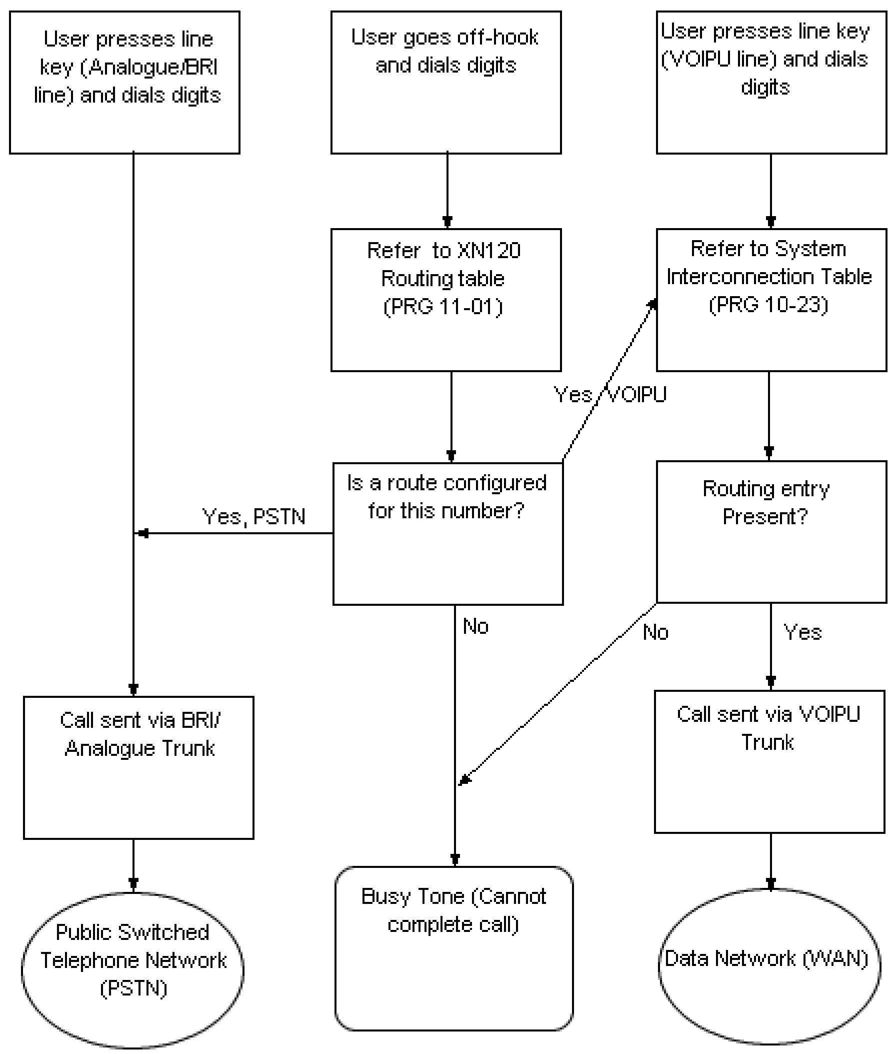
F-Route configuration is outside the scope of this document. Please refer to the "XN120 Programming Reference Manual" for further information.

System Interconnection Table

When calls are made via Analogue or BRI trunks, the network provider routes the call to the required destination. The XN120 just has to send the digits to the provider and their network is configured with the routing information to set up the call. This is essential as there are millions of possible telephone numbers that you could dial.

When SIP trunks are used, typically there are no more than a hundred remote destinations. As the destinations are usually all located within a private data network there needs to be a method of translating dialled numbers to IP addresses – these cannot be automatically determined as in the previous example.

The XN120 has a dialling table that is referred to whenever a call is attempted via a VOIPU trunk. The dialling table allows up to 1000 telephone number/IP address entries. The flowchart below shows how the routing decisions are made in a typical XN120 installation.



Configure Caller ID

Receiving Caller ID

The VOIPU lines will receive Caller ID at default, the Caller ID will be displayed at the LCD display of the XN120 telephones or normal telephones with caller ID enabled (Caller ID is enabled by Program 15-03-09 for each normal telephone).

For the correct display of the CLIP you must enable the automatic insertion of the prefix digits (0 for national calls and 00 for

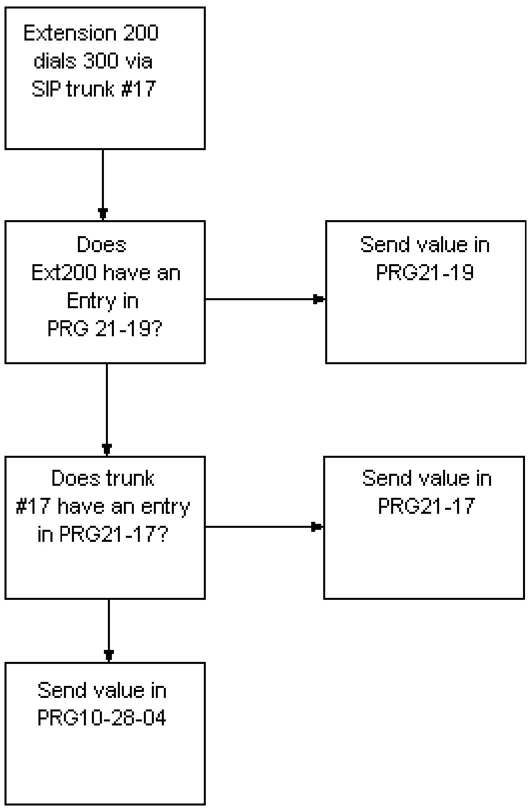
international calls).
This is enabled by Program 20-19-03, set this option to 1 to enable the insertion of the 0 or 00 prefix digits.

If the Caller ID received on an incoming call matches an entry in the system Telephone Book then the name assigned in the Telephone Book will also be displayed.

Sending Caller ID

The system can send caller ID for outgoing calls.
The Caller ID can be specified per VOIPU line (PRG21-17), per telephone (PRG21-19) and per system (PRG10-28-04).

The flowchart below shows which Caller ID setting will be used for a SIP call.

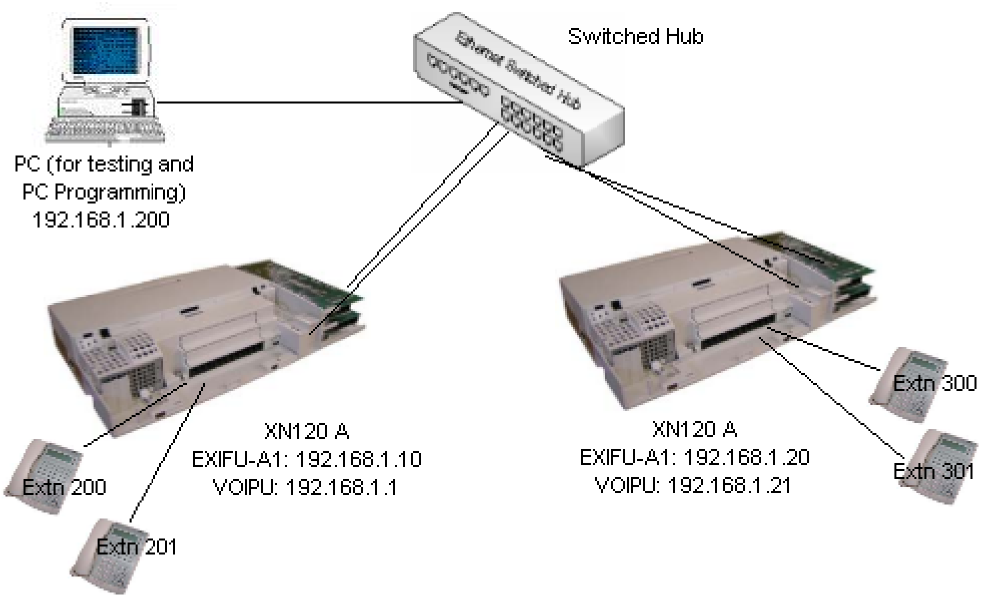


Example Configuration – SIP Trunks

This example shows two XN120 systems, connected via an Ethernet network. The extension numbers are different on both systems and the users on each system should be able to dial each other by using their extension numbers alone. SIP Trunks will be used, along with F-route to simplify dialling.

The configurations are based on default systems (i.e. Cold Start) and assume that a VOIPU (no daughterboard) is installed in CN1 (Slot 5). A system restart is required after applying these changes.

Xn120

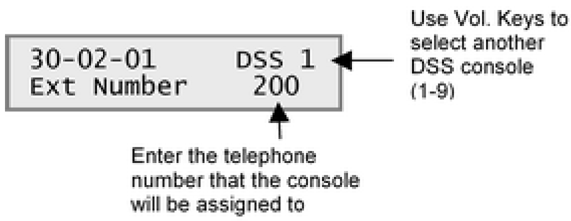


Program	XN120 A	XN120 B	Comments
10-03-02 (Slot 5)	All VOIPU Ports to Type 1:SIP		Assigns ports as SIP
10-12-01	192.168.001.010	192.168.001.020	System IP Address
10-12-02	255.255.255.0	255.255.255.0	Subnet mask
10-23-01 (Sys 1)	1	1	Enables this table entry
10-23-02 (Sys 1)	192.168.001.020	192.168.001.010	IP Address of remote system
10-23-04 (Sys 1)	3	2	Digits to associate with remote system
10-28-01	XN120.co.uk		SIP Domain Name
10-28-02	systema	systemb	SIP Host Name
10-28-04	200	300	UserID for system
11-01-01 (Dial 3)	3 Digits, Type 6	<default>	Uses F-Route to handle calls to numbers beginning with 2 or 3
11-01-02 (Dial 2)	<default>	3 Digits, Type 6	
11-02-01	<default>	300, 301, 302, etc	Change extension numbers on Sys. B
14-05-01 (Trk4-7)	2		Put VOIPU trunks into own trunk group
21-19-01	200, 201, etc	300, 301, etc	Match CLI to extension number
22-02-01 (Trk4-7)	3: DID (for all time modes)		Set trunks to DDI mode
22-09-01 (TRG 2)	2		Set number of received DDI Digits
22-11-02 (Table1)	<default>	300	Define DDI translations for incoming calls
22-11-02 (Table2)	<default>	301	
44-02-01 (Table1)	3	2	Dialled digits
44-02-02 (Table1)	2: F-Route table		Routes to F-route tables (PRG44-05)
44-02-03 (Table1)	1		Specifies F-Route table 1
44-05-01 (F-Rte1)	2		Uses Trunk Group 2 (VOIPU Trunks)
84-05-01 (Slot 5)	192.168.001.011	192.168.001.021	IP Address for VOIPU card
84-13-28	2: G729		Set voice CODEC to G.729

Power Fail Options

Top

Power Fail Options

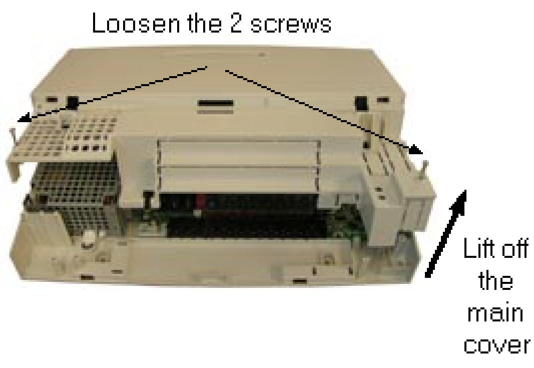


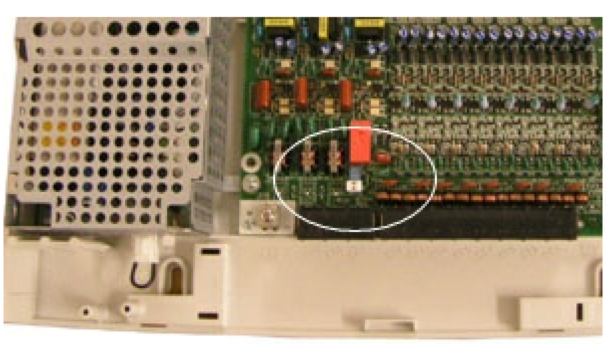
There are various options available for power failure operation.

1. Plug a normal telephone directly into the exchange line socket.
If your exchange lines are provided with a normal telephone socket you can unplug the XN120 connection and plug a normal telephone into the socket to make/receive calls.
You can plug a phone in each exchange line socket if you wish.

! Do not attempt to plug an XN120 system phone into an exchange line socket – you will damage the phone.

2. Use the power fail setting on the XN120.
This will require a normal telephone connected to telephone socket ST8 of the XN120 and an exchange line connected to CO1.
You must set the power fail link (CN500) on the XN120 main board: see the diagram below.
Remove the sub cover of the XN120 unit.
Loosen the two screws and remove the Main cover of the XN120.





Power fail ON

Power fail OFF

CN500=PF CN500=KT

When Power Fail is set to ON (marked PF on the card):
When the power fails (or the XN120 is switched off) Exchange line CO1 is automatically connected to the normal telephone plugged into socket ST8.
When the XN120 is powered on the telephone in socket ST8 will operate as a normal XN120 telephone.
You can not use an XN120 system phone or 64 Button Console in socket ST8 with the power fail link set to PF.

When Power Fail is set to OFF (marked KT on the card):
When the power fails (or the XN120 is switched off) Exchange line CO1 is not automatically connected to socket ST8.
When the XN120 is powered on, the telephone in socket ST8 will operate as either a normal XN120 telephone or XN120 system phone, depending on which type is plugged in.
You can use an XN120 system phone in socket ST8 with the power fail link set to KT.

Power Fail Options when the 64 Button Console is also Installed

- The 64 Button Console must be connected to socket ST8 of the XN120.
- You must ensure the power fail link (CN500) is set to KT for the console to work correctly.
- You cannot use socket ST8 for power fail operation when the 64 Button Console is connected to ST8.

If you have a 64 Button Console installed

- You can use the power fail operation of the optional 308 expansion card, if you have one installed.
  There must be an exchange line connected to the first CO port of the 308 expansion card.
- You can connect the normal telephone directly into the exchange line socket – See instruction above.

Battery Back Up Options

There are two options available to provide back up during power failure.

Battery Box
Each XN120 Main and expansion unit can have the optional battery box connected. This will provide approximately one hour of battery back up. You will need a separate battery box for each XN120 unit.

Uninterruptible Power Supply (UPS)
Each XN120 Main and Expansion unit can be plugged into a UPS. The UPS is will provide AC power to the XN120 system.
You can plug all XN120 units into a single UPS.
Each XN120 unit can take a maximum of 240VA.

Connect the Power & System Start Up Top

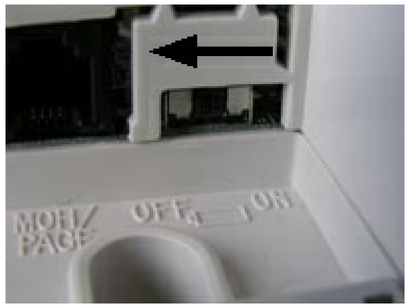
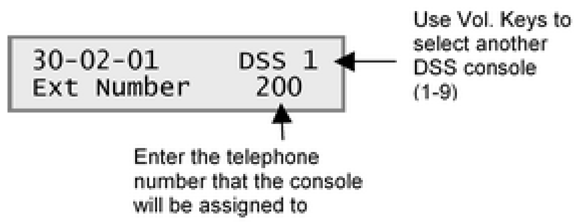
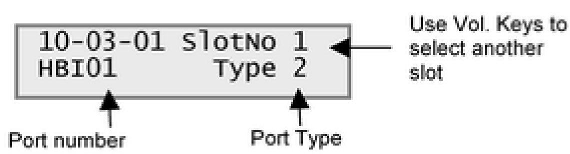
Connect the Power & System Start Up

The power cable is plugged into the left side of each main and expansion unit via an IEC-C13 socket using the cable supplied with the XN120.

Before connecting the power:

- Ensure the power switch on the left side of the unit is OFF.
- Ensure the power is switched off at the source.

System Start Up – First Time
! The first time you start up the XN120 it is important to clear the system memory. This will ensure that the system is set to the default configuration.

1	Set the NORMAL switch to the OFF position. This will ensure that the system is set to the default configuration.	The NORMAL switch is located under the sub cover of the XN120 main unit; on the right side of the ST1-ST8 connection sockets. 
2	Switch ON the power at the source. Switch ON the power switch on the left side of the XN120 unit.	The lamp within the cover of XN120 will come on
3	Wait for 2 minutes while the system starts up.	The XN120 system telephones will 'click' while the system starts up. If you have an XN120 system telephone connected that has an LCD display it will show:  while the system is starting up.
4	When the start up is complete the XN120 phones will show time and date. You can set the time/date when you configure the system, see later in this guide.	If you have an XN120 system telephone connected that has an LCD display it will show the time/date, for example:  when the system has started up successfully.
5	***! Important*** Set the NORMAL switch to the ON position.	The NORMAL switch is located under the sub cover on the right side of the station connection sockets. 

Switching the XN120 OFF
! Be sure that no calls are in progress otherwise they will be cut off.
Turn the power switch off on the left side of the XN120 unit.

System Start Up – Retain Customer Configuration
This is the normal operation for powering the XN120 on.
Before you power on the system check that the NORMAL switch is set to ON. This will ensure that the system memory retains your configuration.

System Start Up – Default Configuration
CAUTION! This will erase any customer configuration in the battery backed memory.
Before you power on the system set the NORMAL switch to OFF. This will ensure that the system is set to the default configuration.
Set the NORMAL switch to ON after the system has started up.

Configure the XN120

Top

Configure the XN120

The getting started guide will cover the most frequently used configuration options. For advanced configuration please refer to the XN120 Programming Manual.

Before you configure your system it is important that you:

- Ensure the power will not be turned off to the XN120, otherwise you will lose any changes you have made that were not previously saved to battery backed memory.
- Ensure that the NORMAL switch is set to ON before you commence, otherwise you could lose your entire configuration. Refer to System Start Up for information.
- Have a diagram of your exchange lines and telephones.
- Plan your requirements before you start.

While you configure your system it is important that you:

- Exit configuration mode periodically, this will save your changes into battery backed memory. They will not be lost if the power is removed.
- Fill out the configuration sheets as you go so that you have a record of your configuration.
- Make small changes, exit configuration mode and test the changes. Do not make all your changes at once as this can make testing very difficult.
- Record your changes as you can only 'undo' them by re-entering the previous values.
- Do not unplug the phone. If it is unplugged by mistake then plug it back in, wait for the display to show time and date and then press HOLD to return to the configuration mode. Your changes will not be lost.

The XN120 consists of exchange lines and telephones connected to the 008/308 cards you have installed.
Within the XN120 configuration the exchange lines are referred to as trunks and the telephones as extensions.
When the XN120 is started up as shown in this guide all the equipment will operate, it is not necessary to make any changes to the system configuration.
With the default settings:

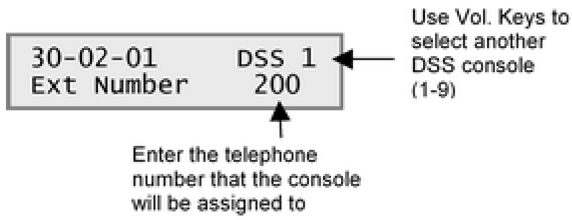
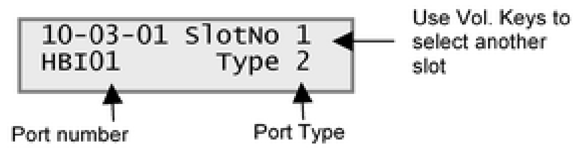
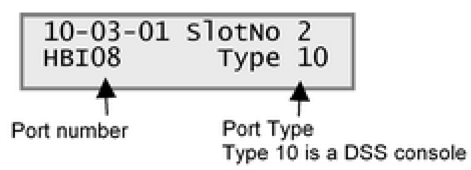
- Each telephone will function and is assigned an extension number.
- Calls received on the exchange lines will ring at telephone number 200.
- Each telephone can make exchange line calls by dialling 9.
- Each exchange line is presented at a Function Key with busy lamp indication.

How to change the XN120's Configuration
The configuration is stored in battery backed memory within the XN120. You can change the configuration via any XN120 system phone that has an LCD display.
When you have made your changes the XN120 will automatically save the configuration into memory.

Check the User Guide for other options
There are some options that are set via normal service codes, for example:
System Phone Book – with service code 853.
Telephone Names – with service code 800.

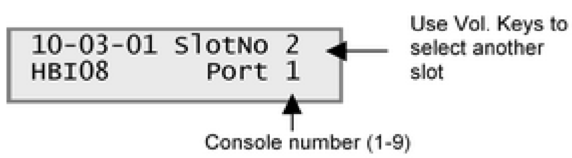
Entering Configuration Mode

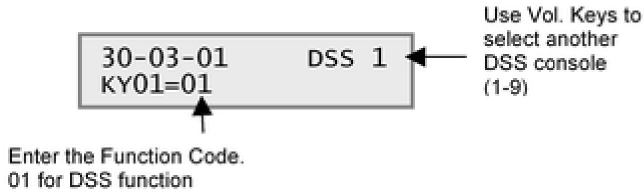
- You will need an XN120 system phone with an LCD display.
- The phone should be idle (no call in progress).

1	Press SPK (do not lift the handset)	 You will see -
2	Dial Service Code # * # *	 You will see -
3	Dial password 12345678	 You will see -
4	Press HOLD	 You will see -

Selecting the Program Number

- Each configuration setting within the XN120 is identified by a Program Number (e.g. 22-05-01).

1	Ensure the LCD display shows:	
2	If it is not displayed press the DC key several times to step back.	
3	Now enter the Program Number e.g. 22 05 01 with the numeric keys of the XN120 phone.	

Using the System Phone Keys to Make Changes

- During configuration mode use the keys at the system phone to select the program item and change its value etc.
- The LCD display will show the current program item, the editing point is shown by a flashing curser.

SPK	Save changes and exit configuration mode.
Numeric keys 1 to 9 * and #	Alphanumeric entry keys. Entered at the curser location.
HOLD	Confirm the entry and step on to the next.
DC	Step back one level. Current entry is not confirmed.
DND/CONF	Delete one character to the left of the curser.
CLEAR	Delete all characters to the right of the curser.
LND	Move the curser one character to the left.
OPAC	Move the curser one character to the right.
Vol. up	Confirm the entry and step to the next item shown at the top right of the display.
Vol. down	Confirm the entry and step to the previous item shown at the top right of the display.
FLASH	Move the curser to the next entry point.

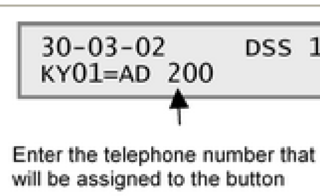
Making Changes

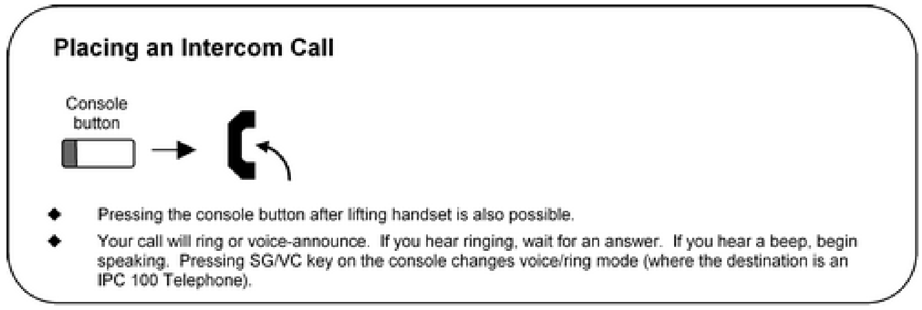
- With the Program Number entered and the curser positioned at the first entry you can change the value by entering the new one with the numeric keys of the XN120 phone.
- When you have entered the new value press HOLD to confirm it and move to the next entry.
- You can also press HOLD to step on to the next if you did not make any changes to the entry.

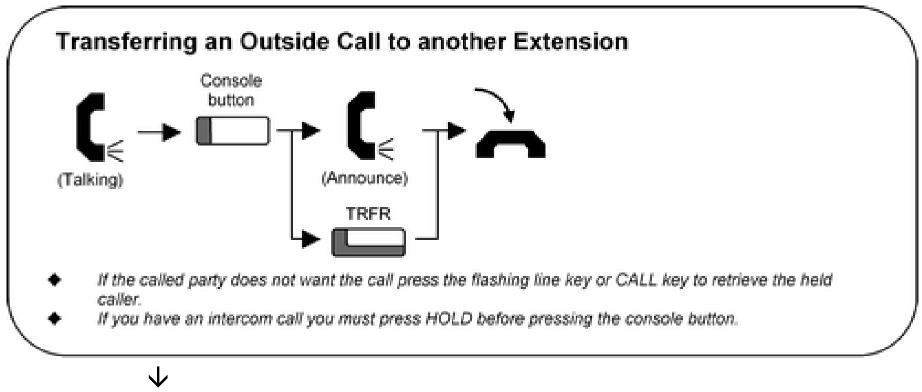
Exiting Configuration Mode

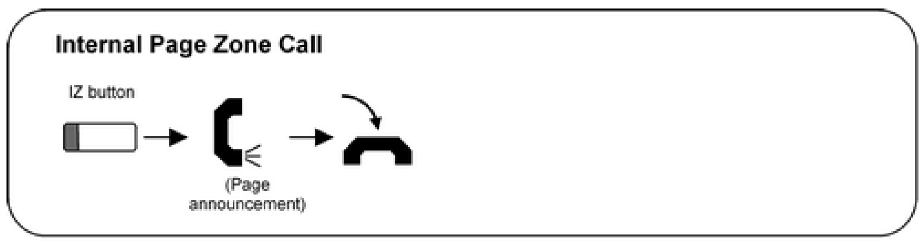
- When you exit configuration mode your changes will be saved into the battery backed memory.

! Until you exit any changes you have made are stored in temporary memory and will be lost if the power is switched off to the XN120.

1	Press HOLD to confirm your current entry.	

<table>
<tr><td>2</td><td>Press DC several times.</td><td>
</td></tr>
<tr><td></td><td></td><td></td></tr>
<tr><td>3</td><td>Press SPK.

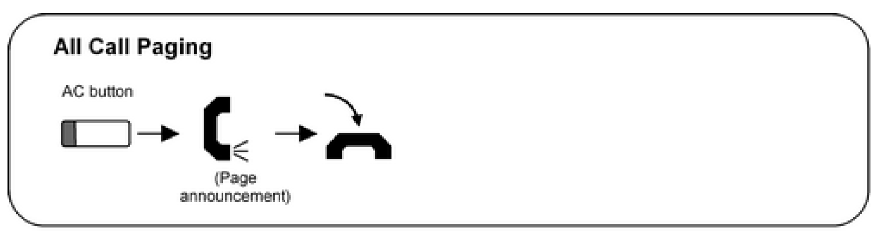
When the save is complete the phone returns to normal operation.</td><td>


</td></tr>
<tr><td></td><td></td><td></td></tr>
</table>

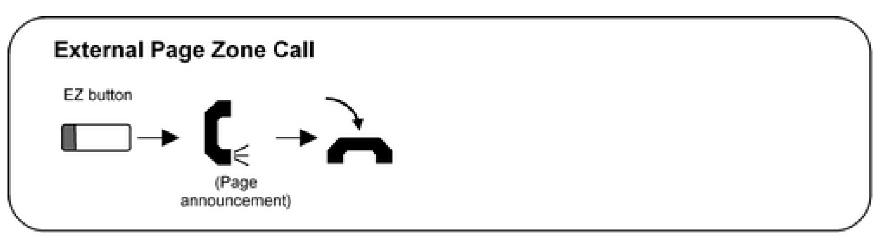
Time & Date Setting

You may want to change this setting if:

- The time and date on the XN120 phones is not correct.

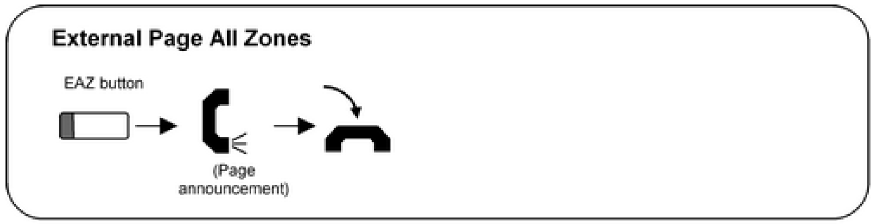
<table>
<tr><td></td><td>Program 10-01-01
Time and date setting for the system.</td><td>

Enter the last two digits of the year (e.g. 04) you can overwrite the current entry.
Press HOLD to confirm the entry and step to the next option.

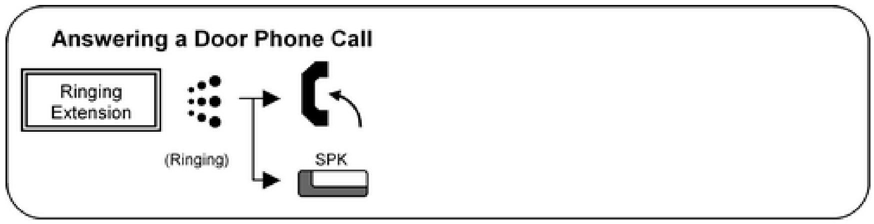


Enter the two digits of the month (01-12).
</td></tr>
</table>

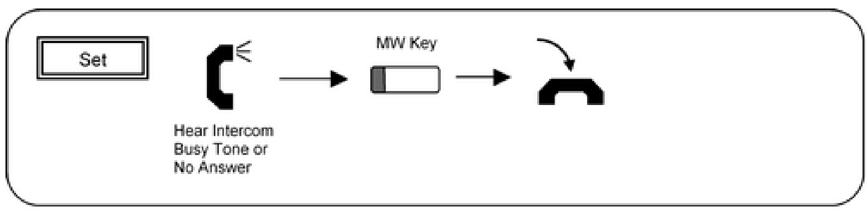
- Press HOLD to confirm the entry and step to the next option.

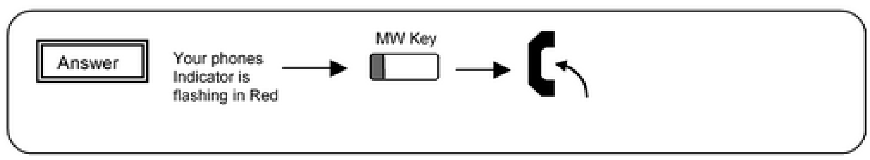


- Enter the two digits of the date (01-31).
- Press HOLD to confirm the entry and step to the next option.



- Enter one digit for the day of the week (1-7).
  1 = Sunday, 2 = Monday, 3 = Tuesday etc.
- Press HOLD to confirm the entry and step to the next option.





- Enter the two digits for the hour (24h format).
- Press HOLD to confirm the entry and step to the next option.

```
10-01-06
Minute          48
```

- Enter the two digits for the minutes (00-59).
- Press HOLD to confirm the entry and step to the next option.

```
10-01-07
Second          00
```

- Enter the two digits for the seconds (00-59).
- Press HOLD to confirm the entry and step to the next program number.
- Press DC several times when you are done to return to the Program Mode.
- (Press SPK to save changes and exit if you are finished).

Telephone Ringing Assignment.

You may want to change this setting if:

- You want one, or more, exchange lines to ring at one or more telephones.
- You want a dedicated exchange line to ring at a specific telephone.
- You want exchange lines to ring at different locations throughout the day or at the weekend.

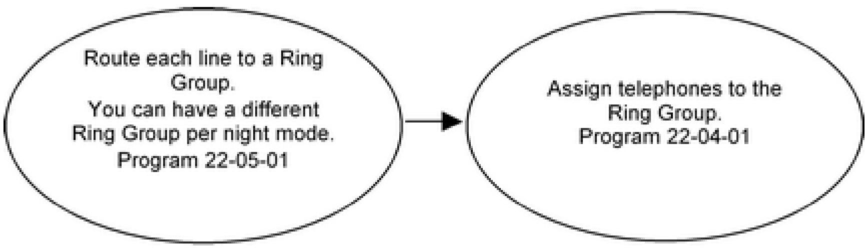
System Operation:

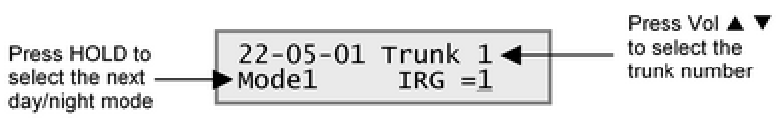
The ringing assignment is achieved by pointing the exchange line to Incoming Ring Groups. The ring group then contains the telephones that will ring.

- Route each exchange line to an Incoming Ring Group (IRG) in Program 22-05-01. A line can route to a different IRG in each night mode.
- You then place telephones into IRG's in program 22-04-01, a phone can be a member of more than one IRG. Up to 32 telephones can be entered per IRG.

The default operation is for each exchange line to ring at telephone 200.

The day/night mode is selected by keys at an XN120 telephone, simply press the key that corresponds to the mode you require.



1	***Program 22-05-01*** The Incoming Ring Group (IRG) number is assigned to the exchange lines. Different IRG's can be assigned to each exchange line to give different ring assignment throughout the day.	Default Setting: Each line (CO1, CO2 & CO3) is assigned to Incoming Ring Group (IRG) number 1. Default Setting: There are up to 8 settings available. IRG number 1 is used for all 8 settings. Therefore, each exchange line will ring at IRG 1 (and therefore telephone number 200, regardless of the time of day.  ➢ Enter the new IRG number (1-25), you can overwrite the current entry. ➢ To remove an entry press CLEAR. ➢ Press HOLD to confirm the entry and step to the next day/night mode. ➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).
2	***Program 22-04-01*** The telephones are placed into ring groups (IRG's).	Default Setting: Telephone number 200 is a member of IRG number 1. 

		➢ Enter the new telephone number, you can overwrite the current entry. ➢ To remove an entry press CLEAR. ➢ Press HOLD to confirm the entry and step to the next member. ➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).
3	***Program 20-07-01*** Turn on the day/night mode option.	Default Setting: Day/night modes can not be changed by any telephones.  ➢ Enter 1 to enable the option, you can overwrite the current entry. ➢ Press HOLD to confirm the entry. ➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).
4	***Assign a key to each mode*** Choose a telephone that will be able to change the day/night mode. ***!*** Exit Configuration Mode. The keys are set at the telephone itself, not within XN120 configuration mode.	Default Setting: There are no keys set to day/night modes on any of the telephones. 1. At the telephone that you want to be able to change the day/ night mode press SPK. 2. Dial service code 851 3. Press the Programmable Function Key you want to set (its current setting is shown if you have an LCD display). Use one of the block of 10 keys for your day/night mode keys. 4. Dial 09 followed by the mode number 1 to 8. 1 = Day 5 = Day 2 2 = Night 1 6 = Night 2 3 = Mid-Night 1 7 = Mid-Night 2 4 = Rest 1 8 = Rest 2 5. Repeat steps 3. and 4. for further keys/modes.

See also:
Outgoing Exchange Line Access to give a telephone a dedicated line for outgoing calls.

Configuration Sheet: Telephone Ringing Assignment
With defaults shown.

Place the telephones into Ring Groups.

IRG Number	**List of telephones that will ring** Prg 22-04-01
IRG 1	Default=200
IRG 2	
IRG 3	
IRG 4	
IRG 5	
IRG 6	

IRG 7	
IRG 8	
IRG 9	
IRG 10	

Up to 25 IRG’s are available, only 10 are listed as this is normally sufficient.
Up to 32 telephones can be entered per IRG. Try to keep the number of ringing telephones to a minimum.

Assign the Ring Groups to the Exchange Lines.

Day/Night Mode Number	**Trunk 1** Prg 22-05-01 default=IRG1	**Trunk 2** Prg 22-05-01 default=IRG1	**Trunk 3** Prg 22-05-01 default=IRG1
1			
2			
3			
4			
5			
6			
7			
8			

Turn on the ability to change the mode.

Mode Change	**Setting** Default=0 (off)
20-07-01	

Assign Keys at the telephone(s) that will change the mode. A phone with an LCD is preferable.

- This is not done within the XN120 Configuration mode; Keys are changed by dialling a Service Code at the telephone itself. This is explained further in the User Guide.
- You will need a separate key for each mode you are using.
- The key for the current mode will light red, to change the mode just press the appropriate key.

Telephone	**Mode 1**	**Mode 2**	**Mode 3**	**Mode 4**	**Mode 5**	**Mode 6**	**Mode 7**	**Mode 8**
	Key	Key	Key	Key	Key	Key	Key	Key
	Key	Key	Key	Key	Key	Key	Key	Key

Note:

- Plan your requirements as the ring assignment is the most important operation of your telephone system. It’s your customers that will be ringing you!
- Try to keep the number of ring modes to a minimum, 3 per day is sufficient (normal day working, lunch times and evenings for example). You may want an additional mode to cover the weekend.
- Do not have too many phones in a ring group, remember that calls can be answered by pressing the Function Keys at the XN120 phones, see also Call Pickup in the Feature Guide.

Example Configuration.

- You have 3 exchange lines (trunks) connected.
- Trunks 1 & 2 (CO1) needs to ring at extensions 200 and 202 during the day time working.
- At lunch times they should ring at extension 205.
- In the evenings and at weekends they should go to an answer phone, the answer phone is connected to ST8 so is extension number 207.
- Trunk 3 (CO3) is a dedicated line and should go to telephone 206 at all times.

· Telephone 200 will have Function keys to change the mode for day, lunch etc.

Step 1
Place the telephones into an IRG for each of the modes (day time, lunch time, evenings and weekends).

IRG Number	List of telephones that will ring Prg 22-04-01
IRG 1	200, 202 (day time)
IRG 2	205 (lunch time)
IRG 3	207 (evenings and weekends)
IRG 4	206 (at all times)

Step 2
Assign the IRG number to each trunk for the modes you will use.

Mode	Trunk 1 Prg 22-05-01	Trunk 2 Prg 22-05-01	Trunk 3 Prg 22-05-01
1 Day	IRG 1	IRG 1	IRG 4
2 Lunch	IRG 2	IRG 2	IRG 4
3 Evening & Weekend	IRG 3	IRG 3	IRG 4

Step 3
Turn on the ability to change the ring mode for day time, lunch time, evenings etc.

Mode Change	Setting
20-07-01	1 (on)

Step 4
Assign modes 1 (Day), 2 (Lunch) and 3 (Evenings/Weekends) to keys at telephone 200.

1. At telephone 200 press SPK.
2. Dial service code 851
3. Press Key 7 (Its current setting is shown if you have an LCD display).
4. Dial 09 followed by 1 for the mode number.
5. Repeat steps 3. and 4. for Key 8 = mode 2 and Key 9 = mode 3.

Telephone	Mode 1	Mode 2	Mode 3
200	Key 7	Key 8	Key 9

Telephone Ringing Style

You may want to change this setting if:

- You want to change the ringing style of outside and internal calls to telephones.

System Operation:
The ringing patterns are set for the system by Program 20-15-01
The default setting is:
Outside calls have a single ring pattern of 2 seconds on / 1 second off.
Internal calls have a double ring pattern of 0.4 seconds on / 0.2 seconds off / 0.4 seconds on / 2 seconds off.
The ringing pattern for normal phones is fixed to the single ring pattern; the fixed ringing can be turned off individually by Program 15-03-12 for each normal phone.

1	***Program 20-15-01*** 20-15-01 sets the ring pattern for outside calls. (Most users would expect a	20-15-01 TRK Normal INC 8

	double ring for outside calls so set this option to 8)	➢ Enter the ring pattern for outside calls, you can overwrite the current entry. (3 is a single ring, 8 is a double ring). ➢ Press HOLD to confirm the entry and step to the next option. ➢ Leave this item set to 8, Press HOLD to step to the next item.
	20-15-03 sets the ring pattern for internal calls. (Most users would expect a single ring for internal calls so set this option to 3)	20-15-03 Internal INC 3 ➢ Enter the ring pattern for internal calls. (3 is a single ring, 8 is a double ring). ➢ Press HOLD to confirm the entry and step to the next option. ➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).

1	***Program 15-03-12*** Turn off fixed ringing for each normal phone. You set this option for each phone. Turning on fixed ringing gives a single ring pattern for all calls. Turning off will use the ring pattern set in Program 20-15-03.	15-03-12 205 Fixed Cadence 0 Press Vol ▲ ▼ to select the telephone number Select the telephone number of the normal phone you want to set using the VOL. ▲ ▼ keys. Set to 0 to turn off fixed ringing for the telephone. (Set to 1 to turn on fixed ringing). Press VOL. ▲ ▼ keys to select the same option for the next telephone. Press HOLD to confirm the entry(s). Press DC several times when you are done to return to the Program Mode. (Press SPK to save changes and exit if you are finished).

Internal Call Ringing Mode

You may want to change this setting if:

- You want to change the way internal calls to telephones are presented. Internal calls can either ring the phone or voice announce where the caller can speak directly to the loudspeaker of the phone they are calling

System Operation:

The mode is set for the system by Program 20-02-12

The default setting is Voice Announce mode.

1	***Program 20-02-12*** Select the mode for internal calls. ***!*** Most users prefer ring mode so set this option to 1.	20-02-12 ICM Call Type 1 ➢ Enter the mode for internal calls, you can overwrite the current entry. (1 is Ring mode, 0 is Voice Announce mode). ➢ Press HOLD to confirm the entry. ➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).

Configuration Sheet: Ringing Style

Program	Description	Setting 3=single ring, 8=double ring
20-15- 01	Ring pattern outside calls	Default=3 single ring
20-15-03	Ring pattern internal calls	Default=8 double ring

Telephone	20-02-12 Voice call or Ring call mode for internal calls to system phones. 0=Voice call 1=Ring call	15-03-12 Fixed ringing for normal phones. 0=Not fixed 1=Fixed
200	Default=0	Default=1
201	Default=0	Default=1
202	Default=0	Default=1
203	Default=0	Default=1
204	Default=0	Default=1
205	Default=0	Default=1
206	Default=0	Default=1
207	Default=0	Default=1

Outgoing Exchange Line Access

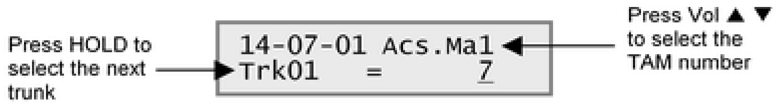
You may want to change this setting if:

- You want a dedicated exchange line for one of the telephones (e.g. FAX and PDQ machines).

System Operation:

Each telephone is assigned a Trunk Access Map (TAM) number.
The TAM number is then given the access properties for each of the exchange lines.
The default operation is that all telephones have access to any exchange line.

1	***Program 15-06-01*** Give the telephone a TAM number. You can specify a different TAM number for each day/night mode. See Changing the Telephone Ringing Assignment for the modes you may be using.	Default Setting: All telephones have TAM number 1.  ➢ Enter the TAM number (1-51) for each mode (1-8), you can overwrite the current entry. ➢ Press HOLD to confirm the entry and step to the next mode. ➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).

2	***Program 14-07-01*** Give each exchange line the access properties for the TAM number.	Default Setting: Each exchange line (CO1, CO2 & CO3) has full access (Property type 7) for TAM number 1. Therefore every telephone can access any of the trunks.  ➢ Enter the access properties (0-7) for each trunk, you can overwrite the current entry. 0 – No access 1 – Outgoing only 2 – Incoming only 3 – Retrieve held call only 4 – Outgoing and retrieve held call 5 – Incoming and retrieve held call 6 – Incoming and outgoing 7 – Incoming, outgoing and retrieve held call ➢ Press HOLD to confirm the entry and step to the next mode. ➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).

See Also:
Changing the Telephone Ringing Assignment.

Configuration Sheet: Outgoing Exchange Line Access
With defaults shown.

Give each telephone a TAM number.

Prg 15-06-01	**TAM Number for each day/night mode** Default=TAM1 for all modes							
Telephone	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**
200								
201								
202								
203								
204								
205								
206								
207								

There are 51 TAM numbers available.

Give each exchange line the access properties for the TAM number.

Prg 14-07-01				**Values available:** 0 – No access 1 – Outgoing only 2 – Incoming only 3 – Retrieve held call only
TAM Number	**Trunk 1**	**Trunk 2**	**Trunk 3**	

1	7	7	7	4 – Outgoing and retrieve held call 5 – Incoming and retrieve held call 6 – Incoming and outgoing 7 – Incoming, outgoing and retrieve held call
2	0	0	0	
3	0	0	0	
4	0	0	0	
5	0	0	0	

Although there are 51 TAM numbers available only 5 are listed as this is normally sufficient.

Caller ID

You will need to enable this setting if:

- You have Caller ID service supplied on your outside lines.
- You have normal telephones that are Caller ID compatible.

The XN120 can detect the Caller ID and display it on the LCD display of the XN120 system phones.
It can also be available at a normal phone that is Caller ID compatible.

System Operation:
You will need to turn on the Caller ID detection for each trunk that it will be received on.
You will also need to turn on Caller ID for each of the normal telephones that are Caller ID compatible.

Program 14-02-10

Turn on Caller ID for each trunk.

Default Setting:
14-02-10 is set to 0 (Caller ID is turned off) for each trunk.

```
14-02-10 Trunk 1
Caller ID      0
```

Press Vol ▲ ▼ to select the trunk number

➢ For each trunk enter 1 to turn on Caller ID, you can overwrite the current entry.
➢ Press HOLD to confirm the entry and step to the next option.
➢ Press DC several times when you are done to return to the Program Mode.
➢ (Press SPK to save changes and exit if you are finished).

Program 15-03-09

Turn on Caller ID for each normal telephone that is Caller ID compatible.

You do not need to change this option for the XN120 system phones.

Default Setting:
15-03-09 is set to 0 (Caller ID is turned off) for each telephone.

```
15-03-09 Tel 200
Ext No Display 0
```

Press Vol ▲ ▼ to select the telephone number

➢ For each telephone enter 1 to turn on Caller ID, you can overwrite the current entry.
➢ Press HOLD to confirm the entry and step to the next option.
➢ Press DC several times when you are done to return to the Program Mode.
➢ (Press SPK to save changes and exit if you are finished).

See Also:
There are no other related settings.

Configuration Sheet: Caller ID

With defaults shown.

Turn on the Caller ID for each trunk.

Prg 14-02-10	
Trunk number	**Setting** Default= 0 Off
Trunk 1	
Trunk 2	
Trunk 3	

Turn on the Caller ID for each normal telephone.

Prg 15-03-09	
Telephone	**Setting** Default=0 Off
200	
201	
202	
203	
204	
205	
206	
207	

Recall for Normal Telephones

You may want to change this setting if:

- You have normal telephones connected and the RECALL key does not work correctly. This is highlighted when you press the RECALL key but the call is not placed on hold.
- The RECALL is also referred to as Timed Break Recall (TBR).

System Operation:

The XN120 must be configured with the correct RECALL timing that matches the normal telephones that you have connected.

	Program 82-04-04 Set the system to detect a RECALL duration of 70 to 125mS (for normal UK time break recall operation). You will need to change three options within this program. 82-04-04=13 82-04-07=14 82-04-08=25	Default Setting: The XN120 will accept a RECALL duration of 105mS to 1000mS 82-04-04 Max.Break TM 13 ➢ Change the setting to 13 (this is equivalent to 65mS). ➢ Press HOLD 3 times to confirm the entry and step to the next option. 82-04-07 Max.Break TM 14 ➢ Change the setting to 14 (this is equivalent to 70mS). ➢ Press HOLD to confirm the entry and step to the next option. 82-04-08 Max.Break TM 25 ➢ Change the setting to 25 (this is equivalent to 125mS). ➢ Press HOLD to confirm the entry and step to the next option.

		➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).

Configuration Sheet: RECALL Timing

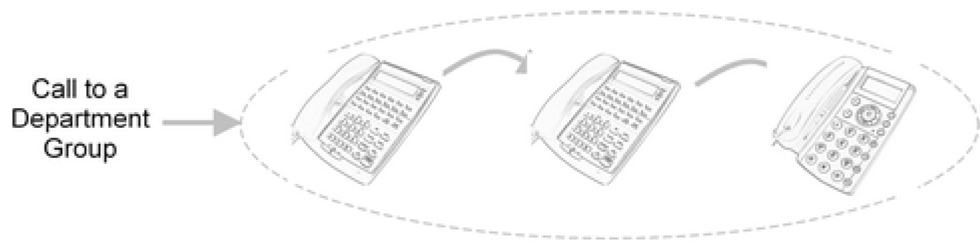
With defaults shown.

	82-04-04 default=20 (100mS)	**82-04-07** default=22 (105mS)	**82-04-08** default=200 (1000mS)
Setting			
Equivalent duration (mS)			

Department Groups

You may want to change this setting if:

- You have people that work within a group and you want to be able to call anyone within the group. The call will ring at anyone that is available within the group. If they do not answer, the call will step to the next member.



System Operation:

The telephones are placed into Department groups. There are 32 groups available.
The group is given a number (pilot number) that you dial to reach the group.
You can choose the following options for each group.

- How the calls will ring around the group – either in a set order of priority or randomly at any telephone.
- Try each telephone once or keep hunting – your call can ring at each available telephone in the group and if not answered stay at the last member or keep trying each member.
- How long each member rings before the call will step on to the next one available.

	Program 16-02-01 Place the telephones into a department group. A telephone can be a member of one group. Priority is in order 1-99 (high-low)	Default Setting: All telephones are in department group 1 16-02-01 TEL 200 ◄── Press Vol ▲ ▼ to select the telephone number Extension Grp 1 ➢ For each telephone enter the group number (1-32) that it is a member of, you can overwrite the current entry. ➢ Press HOLD to confirm the entry and step to the priority option. 16-02-02 TEL 200 ◄── Press Vol ▲ ▼ to select the telephone number Priority 1 ➢ For each telephone enter the priority number (1-99), you can overwrite the current entry. ➢ Press HOLD to confirm the entry and step to the next telephone. ➢ Press DC several times when you are done to return to the Program Mode.

		➢ (Press SPK to save changes and exit if you are finished).

	Program 16-01-02 Select how calls ring around the department group.	Default Setting: Calls ring in priority order within the department group.  For each group select the ring mode, you can overwrite the current entry. 0 = Priority order 1 = Random order Press HOLD to confirm the entry and step to the next option. Press DC several times when you are done to return to the Program Mode. (Press SPK to save changes and exit if you are finished).
	Program 16-01-04 Select how many times the calls try each member of the department group.	Default Setting: Calls try each telephone once.  For each group select the hunting mode, you can overwrite the current entry. 0 = Calls try each telephone once 1 = Calls continue trying the telephones Press HOLD to confirm the entry and step to the next option. Press DC several times when you are done to return to the Program Mode. (Press SPK to save changes and exit if you are finished).
	Program 16-01-09 Select how long calls ring at each member of the department group. You can use this option to turn off the step on operation by setting the time to 0 seconds.	Default Setting: Calls ring each member for 15 seconds.  For each group select the ring duration (0-64800 seconds), you can overwrite the current entry. 0 seconds will stop the call stepping on. Press HOLD to confirm the entry and step to the next option. Press DC several times when you are done to return to the Program Mode. (Press SPK to save changes and exit if you are finished).

	Program 11-07-01 Give the department group a pilot number.	Default Setting: There are no pilot numbers assigned.

	Try to use a number that is easy to remember. For example use: Pilot number 501 for group 1 Pilot number 502 for group 2 etc.	 ➢ For each group enter the pilot number (3 digits required) you will dial to reach the group. ➢ Press HOLD to confirm the entry and step to the next group. If you duplicate a number you will see Duplicate Data, the entry will be removed and you can enter a new number. ➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).

<u>See Also:</u>
There are no related settings.

<u>Configuration Sheet: Department Groups</u>
With defaults shown.

Telephone number	**Department Group number** 16-02-01 default=1	**Priority number** 16-02-02
200		1
201		2
202		3
203		4
204		5
205		6
206		7
207		8

Department Group number 1-32	**Pilot Number** 11-07-01	**Ring in priority/ random** 16-01-02 default=0 Priority	**Try once or continually** 16-01-04 default=0 Once	**Ring time before step on** 16-01-09 default=15 seconds
1				
2				
3				
4				

There are 32 groups available, only 4 are listed as this is usually sufficient.

<u>Create an Internal Paging Group</u>

You may want to change this setting if:

- You want to make a paging call. The paging will be broadcast out of the loudspeakers of the XN120 system phones. Paging is useful if you have staff that leave their desk and you need to contact them.

System Operation:
You add the XN120 system phones into a paging group. There are 32 paging groups available.

	Program 31-02-01
	Place the telephones into paging groups.
	
	You can only broadcast the paging call out of the loudspeakers of XN120 system phones, not normal telephones.
	
	The internal all call page option is not mandatory, you can leave this set to 0.
		Default Setting:
		None of the telephones are in a paging group.
		
		`31-02-01  Tel200` ← Press Vol ▲ ▼ to select the telephone number
		`INT PG GP No   0`
		
		➢ For each telephone enter the paging group number (0-32, enter 0 to remove the phone from a group), you can overwrite the current entry.
		➢ Press HOLD to confirm the entry and step to the next option.
		
		`31-02-02  Tel200` ← Press Vol ▲ ▼ to select the telephone number
		`INT ALL Call PG0`
		
		➢ Enter 1 to place the telephone in the Internal All Call Page option
		➢ Press DC several times when you are done to return to the Program Mode.
		➢ (Press SPK to save changes and exit if you are finished).

See Also:
There are no related settings.

Configuration Sheet:
With defaults shown.

Telephone	**Page Group Number** 31-02-01 default=0 None	**All call page** 31-02-02 default=0 No
200		
201		
202		
203		
204		
205		
206		
207		

External Music Device

You may want to change this setting if:

- You want to play the external music source to callers that are placed on HOLD.

System Operation:
You will need to connect the external music device to the MOH/PAGE socket of the XN120 and select the external device for the music on hold.
The external music device is not supplied with the XN120 system. Refer to Connecting the External Music on Hold Device in

this guide for details of connecting the external music device.

	Program 10-04-01
	Select the device used for the music on hold tune.
		Default Setting:
		The XN120's own tune is played to callers placed on hold.
		`10-04-01`
		`Hold Music Set 0`
		➢ Enter 1 to select the external device (enter 0 to select the XN120 tune), you can overwrite the current entry.
		➢ Press HOLD to confirm the entry and step to the next option.
		➢ Press DC several times when you are done to return to the Program Mode.
		➢ (Press SPK to save changes and exit if you are finished).

Note. You can also use this option to turn off the XN120 tune. If you set the option to 1 (external) but there is no external music device connected to the external music on hold input (MOH input) the held callers will hear silence.

Appendix A - 64 Button Consoles

Top

Appendix A – 64 Button Consoles

Refer to Section 4 – Connecting the 64 Button Console for wiring instructions for the 64 Button Console.
Refer to Section 13 – Power Fail Options for limitations related to the power fail operation when the 64 Button Console is installed.

Which ST sockets can the 64 Button Console be plugged into?
The 64 Button Console MUST be plugged into the eighth ST socket of either the XN120 main unit OR an optional 008/308 expansion card. Refer to Section 4 for wiring instructions.
N.B. the 24 button add-on console is connected to the TEL 2 variant System Phone via an RJ11 style connector and items 4-6 below are not necessary. Ensure that the correct add-on console is available prior to attempting to connect it. Do not attempt to modify the connector to fit an old style console to a new style system phone or vice versa.

- The part number can determine a new or older type, as the new items were given a different part number as indicated in the tables below.

TEL Part Numbers - Pre April 2006 NO RJ11 Connectors for 24DLS / 60DSS		TEL2 Part Numbers – Post April 2006 WITH RJ11 Connectors for 24DLS / 60DSS	
006310-5	IP2AT-6TD TEL (WH)	006341-5	IP2AT-6TD TEL2 (WH)
006316-5	IP2AT-6TD TEL (BK)	006342-5	IP2AT-6TD TEL2 (BK)
006311-5	IP2AT-6TXD TEL (WH)	006343-5	IP2AT-6TXD TEL2 (WH)
006317-5	IP2AT-6TXD TEL (BK)	006344-5	IP2AT-6TXD TEL2 (BK)
006312-5	IP2AT-12TD TEL (WH)	006345-5	IP2AT-12TD TEL2 (WH)
006318-5	IP2AT-12TD TEL (BK)	006346-5	IP2AT-12TD TEL2 (BK)
006313-5	IP2AT-12TXD TEL (WH)	006347-5	IP2AT-12TXD TEL2 (WH)
006319-5	IP2AT-12TXD TEL (BK)	006348-5	IP2AT-12TXD TEL2 (BK)
006314-5	IP2AT-64D DSS CONSOLE (WH)	006349-5	IP2AT-64D DSS CONSOLE2 (WH)

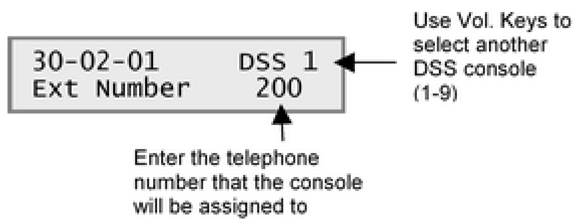
006320-5	IP2AT-64D DSS CONSOLE (BK)	006350-5	IP2AT-64D DSS CONSOLE2 (BK)
006315-5	IP2AT-24DL DLS CONSOLE (WH)	006351-5	IP2AT-24DL DLS CONSOLE2 (WH)
006321-5	IP2AT-24DL DLS CONSOLE (BK)	006352-5	IP2AT-24DL DLS CONSOLE2 (BK)

How many 64 Button Console can be installed?
You can plug one console into socket ST8 of the XN120 Main Unit
You can plug one console into socket ST16/24 of the 008 and 308 expansion cards.
This means you can have a maximum of three 64 Button Consoles connected to the XN120 Main Unit.

What if I have an XN120 Expansion Unit installed?
Each expansion unit can also have a maximum of three 64 Button Consoles connected, following the same rules as the main unit.

Assign the 64 Button Console to a Telephone

Each 64 Button Console must be associated with one of the XN120 system phones; one with a display is recommended as an operator normally uses the console, the display will make their job easier.

<table>
<tr>
<td></td>
<td>Program 30-02-01
Assign each console to a telephone.</td>
<td>Default Setting:
The consoles are not assigned.


Enter the telephone number that each console will be assigned to, you can overwrite the current entry.
Press HOLD to confirm the entry and step to the next console.
Press DC several times when you are done to return to the Program Mode.
(Press SPK to save changes and exit if you are finished).

</td>
</tr>
</table>

In program 30-02-01 you will see that there are 9 DSS consoles available. The console number is determined by the order in which they are installed onto the XN120 NOT the ST socket number they are plugged into.
You can check the console numbering with Program 10-03-01.

64 Button Console Number (1-9)

You can confirm the console number with Program 10-03-01.
A console can be plugged into the eighth socket of each XN120 main unit, expansion unit, 008 card or 308 card.

<table>
<tr><th>XN120 Unit</th><th>Slot number within Program 10-03-01</th></tr>
<tr><td rowspan="2">Main Unit</td><td>Slot 1 Base board</td></tr>
<tr><td>Slot 2</td></tr>
</table>

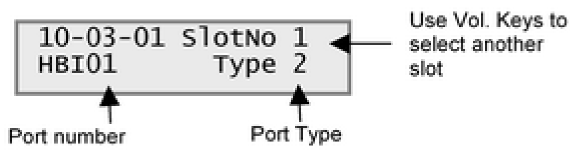
	Slot 3
Expansion Unit 1	Slot 7 Base board
	Slot 8
	Slot 9
Expansion Unit 2	Slot13 Base board
	Slot14
	Slot15

Program 10-03-01 allows you to view each card installed in the system by selecting the appropriate slot number, this table shows the slot numbers of the cards that the consoles can be plugged into.

<table>
<tr>
<td></td>
<td>Program 10-03-01
Confirm the console number 1-9.

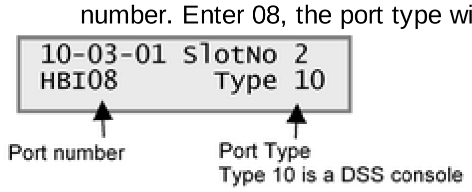
Make a note of the console numbers on the configuration sheet at the end of this section.</td>
<td>

<u>Default Setting:</u>
The consoles are automatically assigned a number when they are installed.

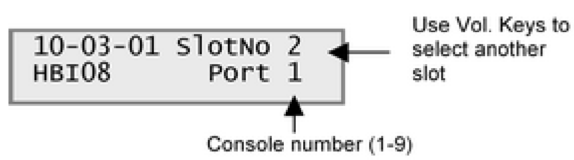


- Use the Volume keys to select the correct slot number of the card that the DSS console is plugged into.

- Press FLASH twice to move the flashing curser to the port number. Enter 08, the port type will be shown.



- If the port is set as a DSS console then the type will be set automatically to 10.
- Press HOLD to display to the console number.



- Press DC several times when you are done to return to the Program Mode.
- (Press SPK to save changes and exit if you are finished).

</td>
</tr>
</table>

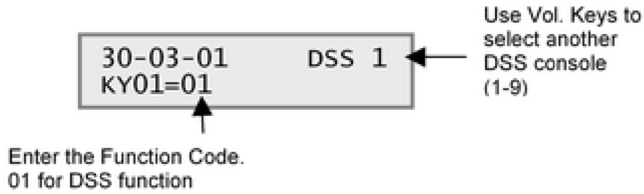
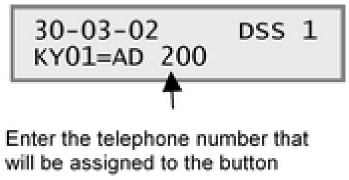
Customising the buttons

When a console is installed the 64 buttons are automatically assigned as Direct Station Select (DSS) keys for extensions 200 to 263.

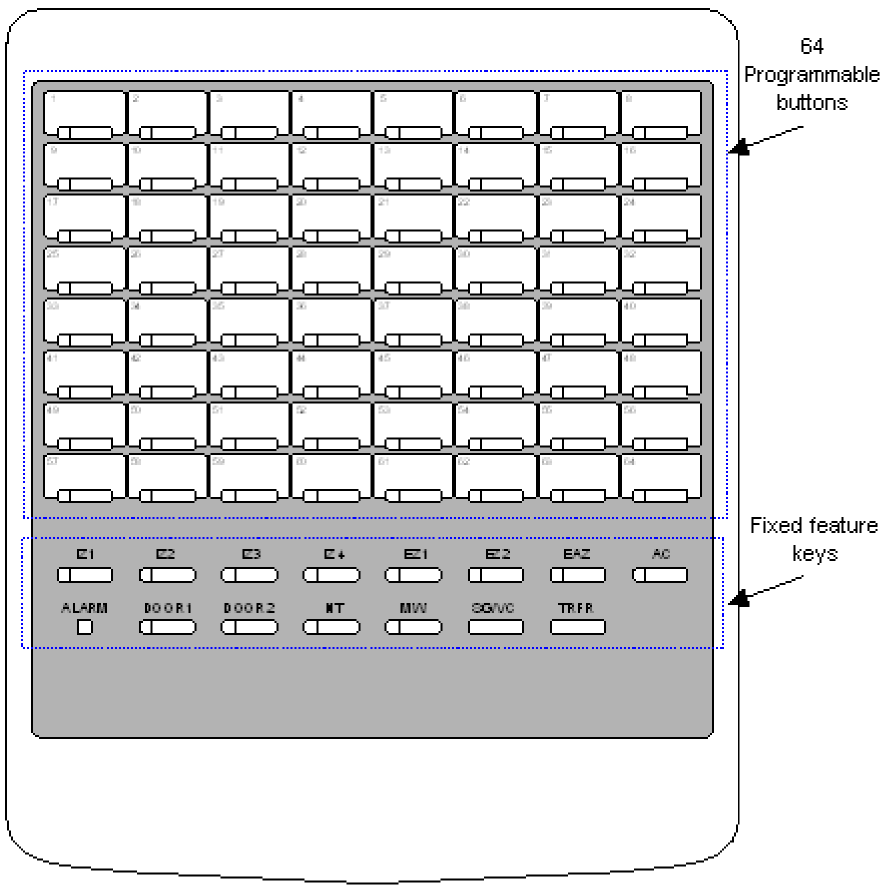
You can customize the buttons of each console with program 30-03-01.

The buttons can have the same functions as the programmable keys on the XN120 system phones. This section will show the setting of the DSS function. For other functions refer to the XN120 Programming Reference Manual.

	Program 30-03-01 Customise the buttons.	<u>Default Setting:</u> The 64 buttons are set as DSS function for extensions 200 to 263.

	Each button has two entries: Function code and Additional data. The 64 buttons are shown as KY01 to KY64. This guide shows function code 01 for DSS function. The additional data can be up to 36 digits long and can be any number including outside numbers. For outside numbers enter the trunk access code followed by the number to dial out e.g. 901509643100	 ➢ Enter the Function Code, 01 for DSS function; you can overwrite the current entry. Enter 000 to set the key to ‘Not Defined’. ➢ Press HOLD to confirm the entry and step to the Additional Data.  ➢ Enter the additional data for the key. This will determine the telephone number that will be assigned to the button. ➢ Press DC several times when you are done to return to the Program Mode. ➢ (Press SPK to save changes and exit if you are finished).

Console User Guide



64 Programmable Buttons

If the button has a telephone number assigned it will light to show the status of the telephone.

Lamp off The telephone is idle

Lamp on The telephone is off hook (busy)

Lamp flashing The telephone has call forward or DND set

If the button has an outside number assigned the lamp will remain off.

Fixed Feature Keys

IZ1 to IZ4	Internal Zone page call for zones 1 to 4. The key lights when the zone is in use.
EZ1 & EZ2	External Zone page call for zones 1 and 2. The key lights when the zone is in use.
EAZ	External All Zone page call. The key lights when in use.
AC	All Call internal page zones. The key lights when in use.
DOOR1 & DOOR2	Doorphone call for door unit 1 and 2. The key lights when the door unit is in use
NT	Flip between Day1 and Night1. The key lights when Night1 is set.
MW	Set Message Waiting when calling another telephone. Also used to reply to message waiting.
SG/VC	Signal/Voice call selection when calling another telephone.
TRFR	Transfer the held call.

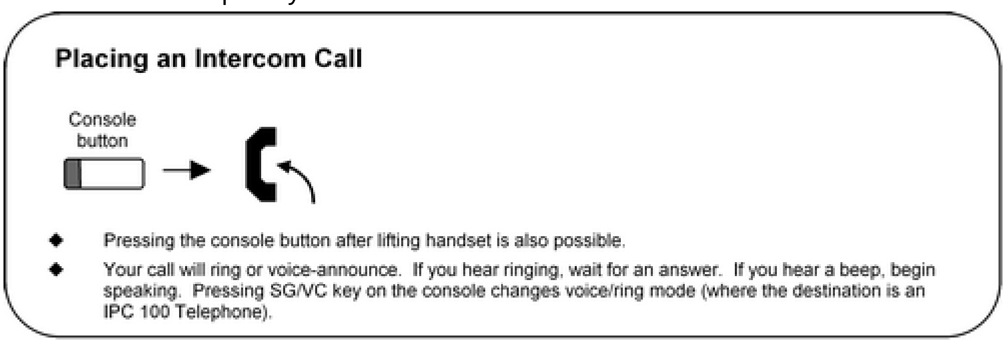
Alarm Lamp

Lights when the low battery or SMDR buffer full alarm are present on the XN120.
Requires program 90-11-01 setting for the telephone that the console is associated with.

Place a call to a telephone

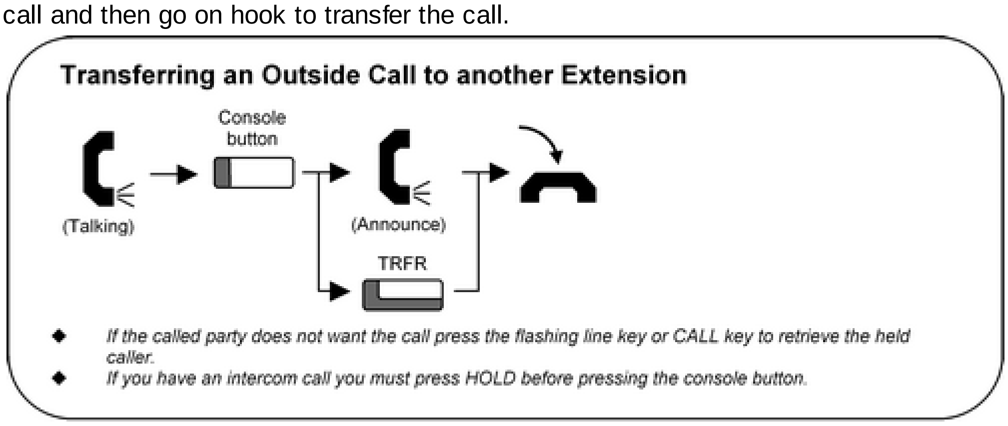
Press the button on the console for the telephone you want to call.
The phone assigned to the console will go into hands free.

You can lift the handset for privacy.



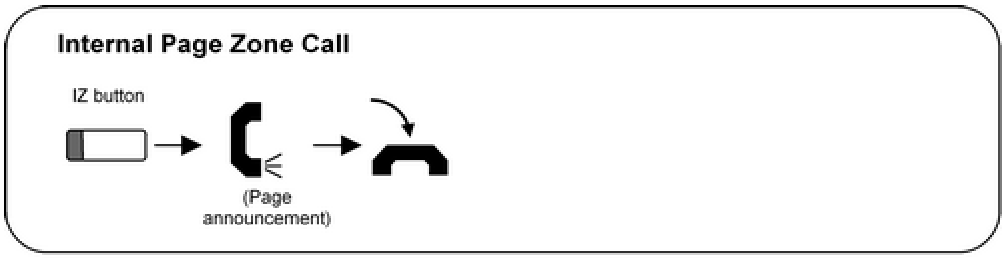
Transferring a call

Press the console button while you are talking to the outside caller; the call will be placed on hold.
Either press TRFR to transfer the call through or wait for the called party to answer, inform them that you wish to transfer a call and then go on hook to transfer the call.



Internal paging zone call

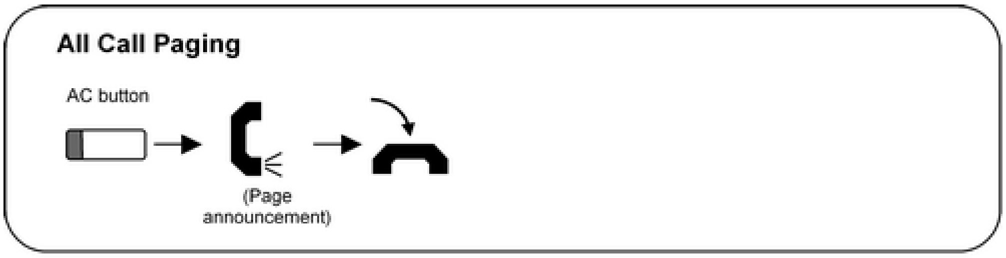
Press the fixed feature button on the console for the internal zone (IZ1 to IZ4) you want to place the page call to.
You will hear a confirmation beep tone and the IZ button will go red.



The XN120 must be configured before the internal paging zones can be used, refer to Create an Internal Paging Group in this guide.

Internal paging - all zones

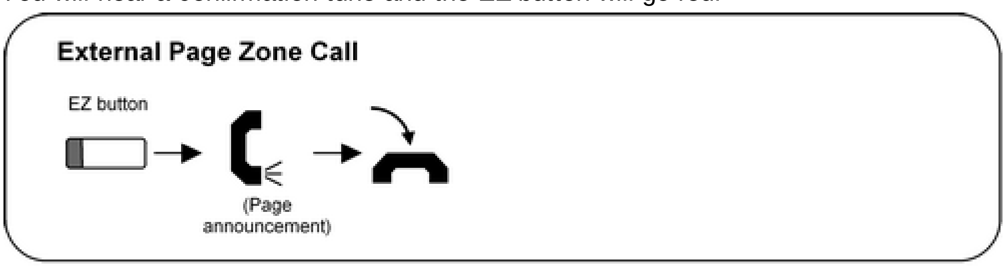
Press the All Call (AC) fixed feature button on the console.
You will hear a confirmation beep tone and the IZ1 to IZ4 buttons will go red.



The XN120 must be configured before the internal paging - all zones can be used, refer to Create an Internal Paging Group in this guide.

External paging zone call

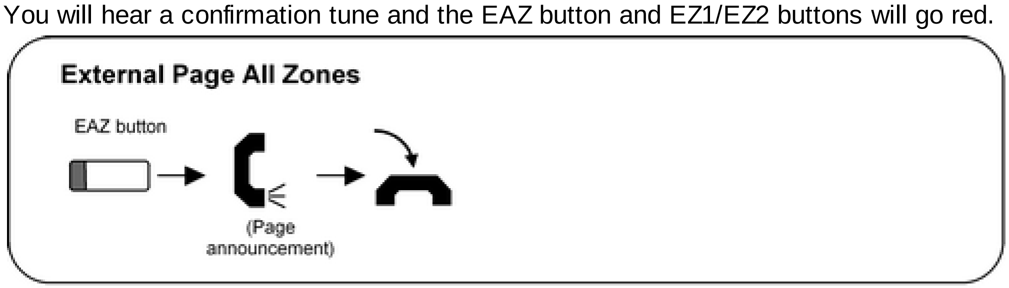
Press the fixed feature button on the console for the external zone (EZ1 or EZ2) you want to place the page call to.
You will hear a confirmation tune and the EZ button will go red.



The XN120 must have an external paging device connected before external paging can be used, refer to Connecting the XN120 to an External Paging System in this guide.

External paging - all zones

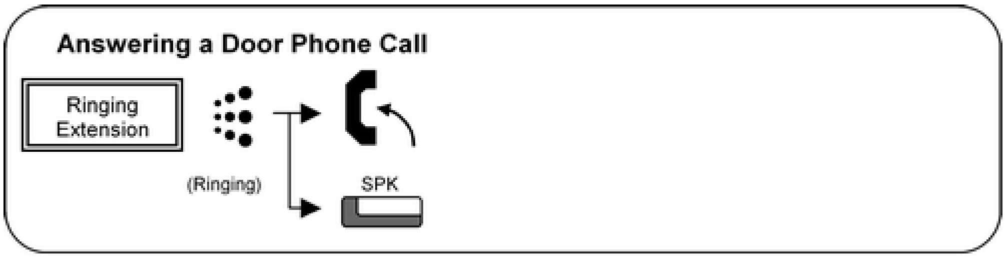
Press the External All Zones (EAZ) fixed feature button on the console
You will hear a confirmation tune and the EAZ button and EZ1/EZ2 buttons will go red.



The XN120 must have an external paging device connected before external paging can be used, refer to Connecting the XN120 to an External Paging System in this guide.

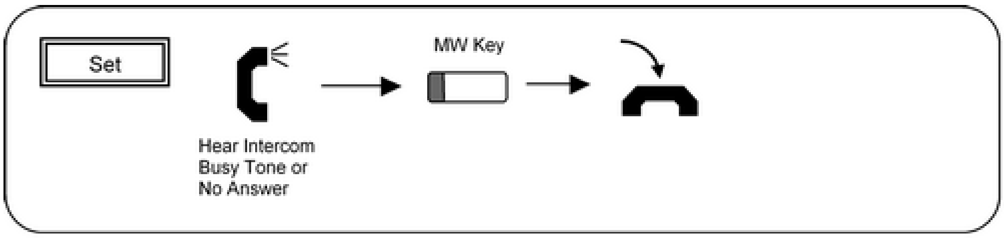
Door phones

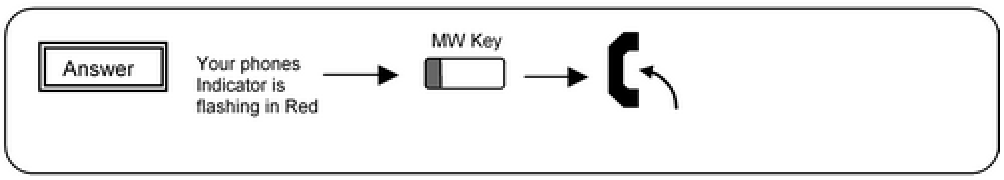
When the call button of the door phone unit is pressed the phones assigned will ring, refer to the separate 2PGDU guide for details. Lift the handset to answer to door phone all at the telephone associated with the console.
When the door phone is in use the DOOR button on the console will go red.



Night Modes
The night mode (NT) fixed feature key on the console will flip the XN120 between Day1 and Night1 mode each time the key is pressed.
If you want to have access to other night modes (Rest1, Mid-Night1 etc) you will need to set one of the programmable function keys on the telephone, refer to Function Keys in the Administrators User Guide within this guide.

Message Waiting
The message waiting (MW) fixed feature key on the console can be used to either set a message waiting or answer a message waiting.





Signal/Voice Call
The signal/voice call (SG/VC) fixed feature key on the console can be used to change the voice/ring mode when calling another XN120 system phone.

Transfer
The transfer (TRFR) fixed feature key on the console can be used to transfer the held call.

Appendix B - 24 Button Consoles

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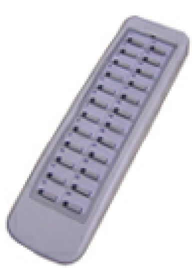
Appendix B – 24 Button Consoles



The 24 button add on console can be connected to any XN120 display system phone.
The console provides 24 additional programmable function keys. The keys can be programmed and used in the same way as the programmable function keys on the XN120 system phone.
Refer to Function Keys in the Administrators User Guide within this guide.

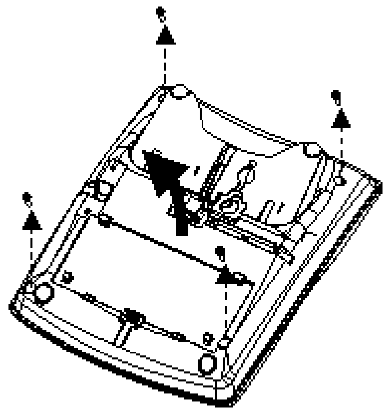
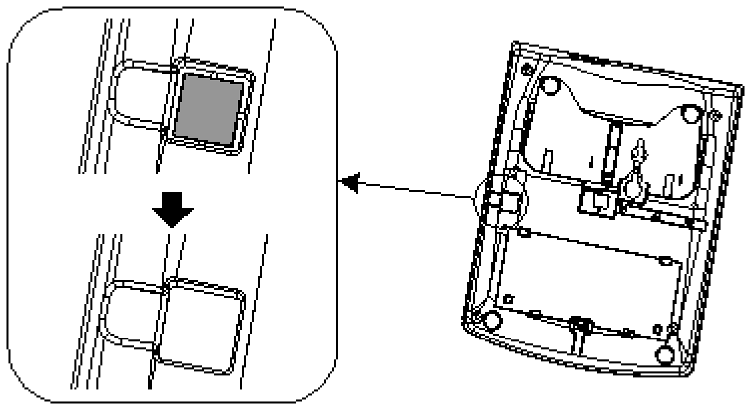
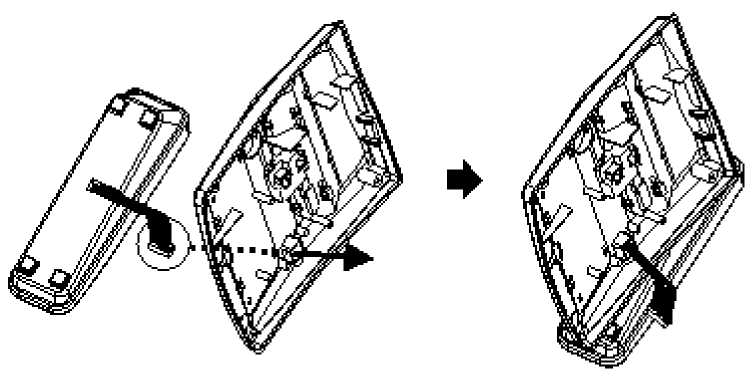
How many 24 Button Console can be installed?
Each XN120 display phone can have a 24 button console connected.

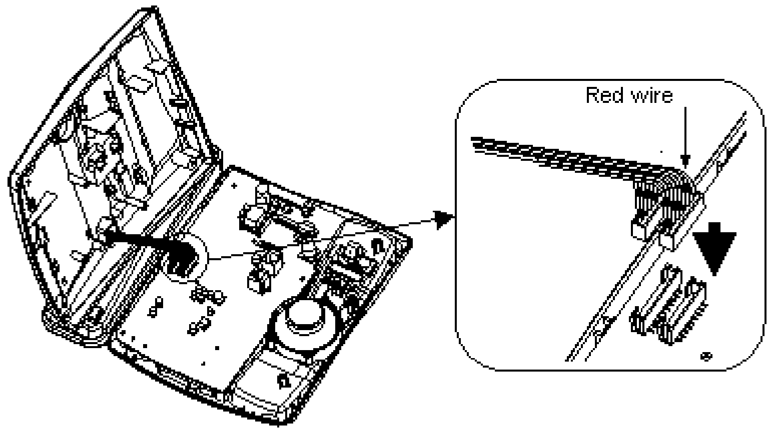
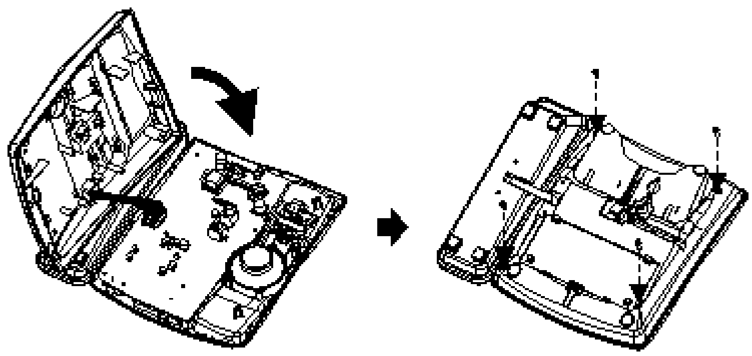
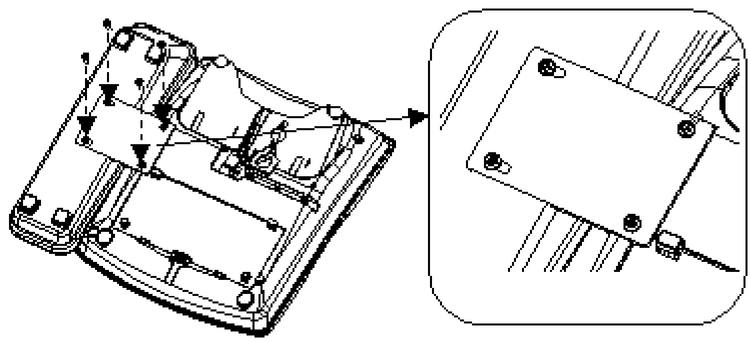
Installing the 24 Button Add On Console
The 24 Button Add On Console is supplied with 4 screws and metal fixing plate to secure it to the XN120 system phone.
The XN120 System phone can not be wall mounted with the 24 button add on console attached.
The XN120 System phone should have the desk stand set to the lower position.

N.B. the 24 button add-on console is connected to the TEL 2 variant System Phone via an RJ11 style connector and items 4-6 below are not necessary. Ensure that the correct add-on console is available prior to attempting to connect it. Do not attempt to modify the connector to fit an old style console to a new style system phone or vice versa.

- The part number can determine a new or older type, as the new items were given a different part number as indicated in the tables below.

TEL Part Numbers - Pre April 2006 NO RJ11 Connectors for 24DLS / 60DSS		TEL2 Part Numbers – Post April 2006 WITH RJ11 Connectors for 24DLS / 60DSS	
006310-5	IP2AT-6TD TEL (WH)	006341-5	IP2AT-6TD TEL2 (WH)
006316-5	IP2AT-6TD TEL (BK)	006342-5	IP2AT-6TD TEL2 (BK)
006311-5	IP2AT-6TXD TEL (WH)	006343-5	IP2AT-6TXD TEL2 (WH)
006317-5	IP2AT-6TXD TEL (BK)	006344-5	IP2AT-6TXD TEL2 (BK)
006312-5	IP2AT-12TD TEL (WH)	006345-5	IP2AT-12TD TEL2 (WH)
006318-5	IP2AT-12TD TEL (BK)	006346-5	IP2AT-12TD TEL2 (BK)
006313-5	IP2AT-12TXD TEL (WH)	006347-5	IP2AT-12TXD TEL2 (WH)
006319-5	IP2AT-12TXD TEL (BK)	006348-5	IP2AT-12TXD TEL2 (BK)
006314-5	IP2AT-64D DSS CONSOLE (WH)	006349-5	IP2AT-64D DSS CONSOLE2 (WH)
006320-5	IP2AT-64D DSS CONSOLE (BK)	006350-5	IP2AT-64D DSS CONSOLE2 (BK)
006315-5	IP2AT-24DL DLS CONSOLE (WH)	006351-5	IP2AT-24DL DLS CONSOLE2 (WH)
006321-5	IP2AT-24DL DLS CONSOLE (BK)	006352-5	IP2AT-24DL DLS CONSOLE2 (BK)

1	Disconnect the line cord and handset cord from the XN120 System Phone	
2	Place the phone onto a flat surface to prevent damage to the front of the phone. Unscrew the 4 screws securing the base of the system phone and remove the base.	
3	Using a blunt tool remove the breakout covering the connector for the add-on console.	
4	Fit the add on console cables through the hole in the base	

5	Plug the add on console cables into the sockets DLCN1 and DLCN2 on the PCB of the upper part of the XN120 system phone Ensure you plug each cable into the correct sized socket. Each cable has a red wire at one end; this must be located towards pin 1 of the socket.	
6	Re-fit the base to the XN120 system phone and secure with the 4 screws.	
7	Using the metal plate and 4 screws supplied with add on console secure the console to the base of the phone. Be careful not the trap any of the wires under the plate.	
8	Reconnect the line cord and handset cord to the XN120 System Phone.	The 24 button console will be automatically recognised. The user can program the function keys using Service code 851 - refer to Function Keys within the Administrators User Guide in this guide.

Appendix C - Wall Mount the XN120 System Phone

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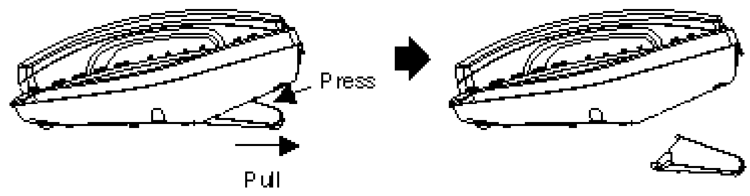
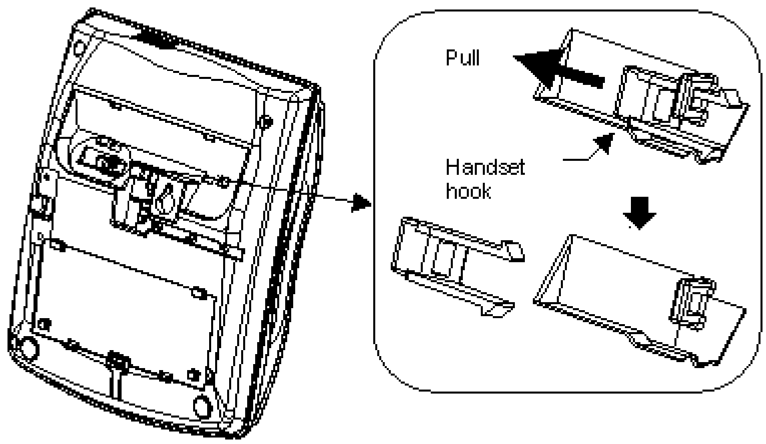
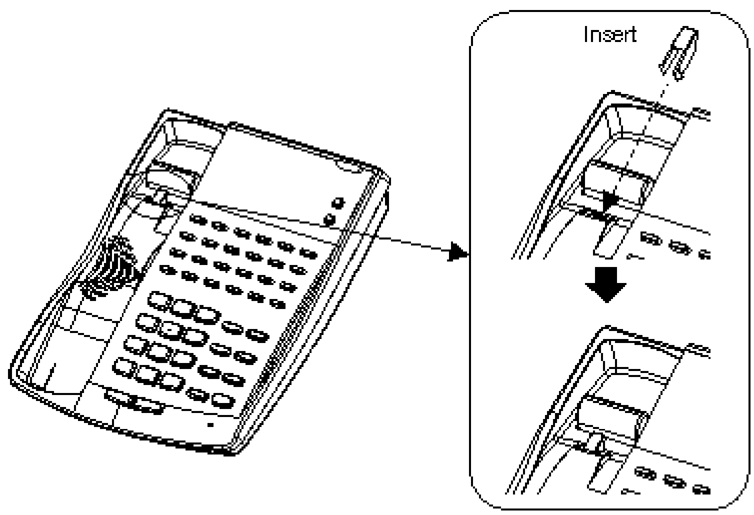
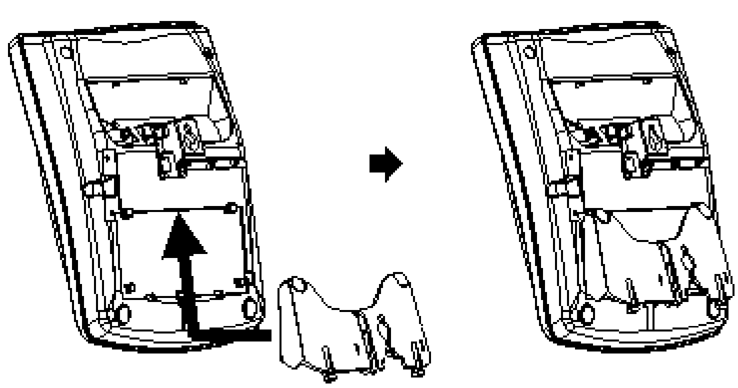
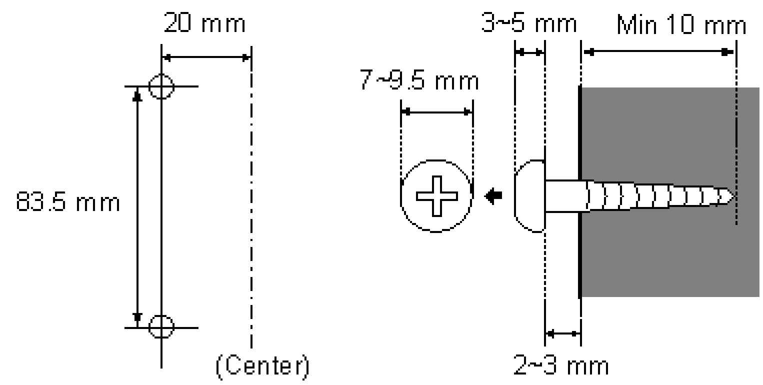
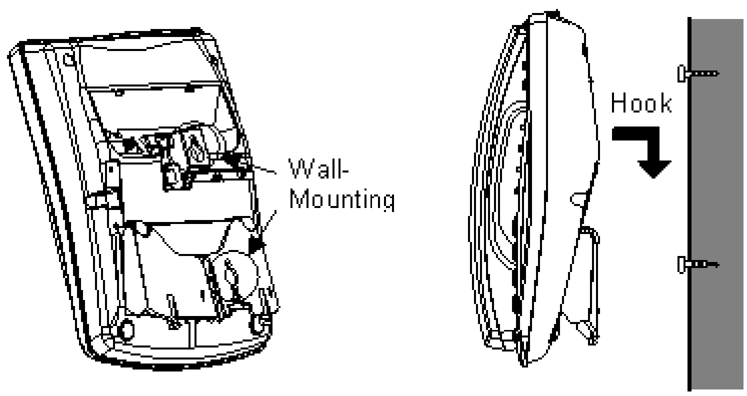
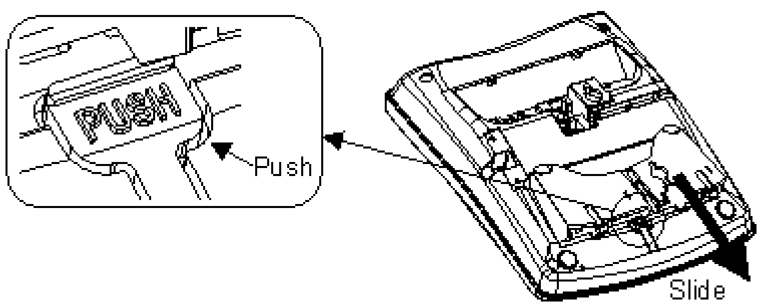
Appendix C – Wall Mount the XN120 System Phone

The XN120 System Phone can be wall mounted using the built in wall mount/desk stand bracket on the base of the phone.
There is a handset hook supplied; attached to the base of the phone.
Two screws (and wall fixing plugs where necessary) will be required to fix the bracket to the wall.

The XN120 system phone cannot be wall mounted when either the 64-button or 24-button console is attached.

1	Disconnect the line cord from the wall socket.	
2	Remove the wall mount/desk stand bracket from the case of the phone	
3	Remove the handset hook from the base of the phone	
4	Insert the handset hook into the slot below the hook switch on the front of the phone. The hook will hold the handset in place when the phone is wall mounted.	
5	Fix the bracket to the base of the phone. Ensure the bracket clips into all 4 of the fixing points on the base of the phone.	

6	Fix 2 screws to the wall leaving approximately 3mm protruding.	
7	Fit the phone onto the 2 screws. Ensure the line cord is not trapped between the phone and the wall.	
8	Reconnect the line cord to the wall socket.	
	If you want to remove the bracket. Push in the retaining clip and slide off the bracket.	

Appendix D - Hard of Hearing Options

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Appendix D – Hard of Hearing Options

Telephone Volume Adjustment

You can increase the speech levels of individual telephones within IPC 100 Configuration Mode Program 10-03.
The levels of IPC 100 System phones and normal phones can be adjusted. You can also adjust the levels to/from the telephone independently.
Refer to the IPC 100 Display and System phone User Guide for further information.

Hotel - System Setup

Room Telephone

Each telephone in a hotel room must be assigned in Hotel Room Extension Setup, this will allow the receptionist to check the room in/out, set room status and hotel features.
You may also want to check with the hotel manager which features (service codes) the hotel room telephones have access to, most features are available at default, you may need to place the hotel room telephones in their own Class of Service number in Class of Service per night mode and disable some features in Class of Service Options.
It is common for the extension number of the telephone to be the same as the hotel room number, use System Numbering to setup the system numbering plan and assign the telephone numbers to the hotel room telephones in Extension Basic Setup.
If the telephones in the room support Caller ID you can enable this per extension in SLT Basic Setup.

Reception Telephone

The hotel reception telephone will be used to check rooms in/out, set room status and assign hotel features, it will also be the extension called when a room dials the Operator/Reception service code.
The reception telephone may also have a DSS console assigned to show the status of all hotel room telephones.
The reception telephone should be assigned the following features in Hotel Class of Service Setup (the Class of Service number is assigned to the telephone in Class of Service per night mode; you may want to put only the reception telephone in its class of service number):
Check In Operation
Check Out Operation
Room Status Output
Changing DND for other extensions
Wake-Up call setup for other extensions
Changing room status for other extensions
Changing restriction class for other extensions
Room to Room Call Restriction

The hotel reception phone may also want the DND/Call Forward override option enabling in Class of Service.

The system also has a room monitor option which must be setup between two single line telephones, therefore you will need to install a seperate SLT at reception for this.
The telephones at reception and in the rooms will need 'SLT Room Monitor' set in Hotel Class of Service Setup.

Operator/Reception Service Code
The system can have a service code assigned to call to operator/reception telephone.
The service code must be assigned as 'Operator' in System Numbering, the extension number to be called is then assigned in Operator Extension.

Hotel DSS Console
The reception telephone can have a DSS console assigned that will show the status of the hotel room telephones (for example checked in, vacant, room clean).
The DSS console is set to hotel mode and assigned to the receptionist's telephone in Hotel DSS Console.
The DSS console keys are assigned in DSS Key Assignment.

Toll Restriction

When a room is checked in the telephone can be given a different Toll Restriction class, this would usually have no restrictions.
When the room is checked out it would typically have a fully restricted.
The toll restriction class when checked in is assigned in Hotel Room Extension Setup.
The toll restriction class when checked out is assigned in Toll Restriction per night mode.

Changing Toll Restriction class while room is checked in
The Toll Restriction class number can be changed for a room telephone when checked in.
The system should be setup with different Toll Restriction levels (for example No restriction, National calls barred, all calls barred) to make use of this feature.
The reception telephone will need the 'Changing Restriction Class for other extensions' option enabled in Hotel Class of Service

Setup.

Class of Service

All extensions (including hotel room telephones) are assigned a Class of Service number in Class of Service per Night Mode.
The class of service number assigned to the hotel room is not effected when the room is checked in or out.
As well as the standard system features in Class of Service Options there are also some hotel related features in Hotel Class of Service setup.

Single Digit Dialling

Hotel room telephones can have single digit dialling to simplify the telephone operation.
The single digit can be translated into any Service Code, extension number or Department Group pilot number.
Hotel single digit dialling is setup in Hotel Single Digit Routing.

DSPDB Card

The optional DSPDB card can provide voice announcements for Wake Up Calls at the room telephone.
The system can play an announcement when the guest sets and answers their Wake Up call.

Print Out

The system can print out the following reports via the EXIFU card(Serial port):
Reports Output by using service code 742 (Refer to Hotel - Receptionist Guide for instructions)
Room Status
Room Call Restriction
Do Not Disturb
Message Waiting
Wake Up Calls
Reports Output automatically if set in Hotel Report Printing
Wake Up Calls not answered
Check Out sheet

Wake Up Call not answered output
The system will print out the following message when a room telephone does not answer their wake up call.
The print out occurs when the wake up call ringing ends; 30 seconds at default set by 'Alarm Clock Duration' in System Timers.

Wake Up Call No Answered --- TEL No.236
Set Time07:00 No Ans Status Time Out

END

Check Out Sheet
The system will print out the following message when a room is checked out

Check Out Information --- TEL No.236
Check In 10/04/06 17:33 Check Out 12/04/06 08:45
Call Count 0

END

Note that the Call Count information does not operate, the value will always be 0. Use the SMDR output for call information.

Call Logging - SMDR

In order to charge the guest for their telephone calls you must connect a Call Logging device to the system, refer to the SMDR section of the system manuals for details.
The system does not support call costing for analogue trunks, this must be done by the external call logging device.
The system does support Advice of Charge for ISDN lines, this is setup in Trunk Call Charging.

The system is supplied with a 30 day licence for Hotel Operation.
To remove the 30 day licence you must enter a licence key by following these instructions.

Make a note of the System's MAC Address.

With the system powered on and operating.
1. At any idle system phone that has a display press the OPAC key and then enter 3 on the keypad.
2. The display will show the MAC address on the bottom row, for example 0060-B9CA-1234 shown below:

Main Ver: 05.xx
0060-B9CA-1234

3. The display will revert to normal after 10 seconds.

Obtain the licence Key.

1. Send the MAC address to your system supplier who can then supply you with the licence Key.
Note that this may not be returned immediately, if this is the case then instruct the customer to use Method 2 to enter the licence Key using a service code (*Ensure you set a Service Code in Hotel License Information to allow Method 2 to be used*).

Enter the licence Key into the System.

There are two methods to enter the licence key into the system:

Method 1 - Enter the licence Key with PRG 20-32-01 from a display system phone.
You should be familiar with the system's programming to use this method.
PCPro can not be used to enter the licence Key.

1. Access PRG 20-32-01

20-32-01
ID _

2. Enter the 8 character licence Key, for example:

20-32-01
ID 1ED3CB76

Use the keypad to enter the characters.
Use the LND to move the curser one space left.
Use the OPAC key to move the curser one space right.

3. Press HOLD.
If the licence Key is not accepted the display will show:

20-32-01
INVALID DATA

Method 2 - Enter the licence Key with a Service Code.
Use this method if the licence Key is not available at the time of system installation.
This method can be used by the customer.

Note - The Service Code must be assigned in Hotel License Information (PRG 11-10-37), there is no entry at default.
For example, set the Service Code to #700.

1. At any display system phone press SPK and dial the licence Service Code e.g. #700.
The display will show:

LICENCE INPUT
-

2. Use the keypad to enter the 8 character licence Key.
For example:

LICENCE INPUT
-1ED3CB76

Press # to move the curser one space to the right when entering characters under the same keypad key e.g. ED.
If you make an incorrect entry Press SPK to exit.

3. Press HOLD to confirm the licence Key.
If the licence Key is accepted the display will show:

LICENCE INPUT
HOTEL SERVICE ON

If the licence Key is not accepted you will hear a reject tone and the display will go back to operation 1.

4. Press SPK to end.

To confirm the Hotel licence period remaining

Use PRG 20-33-01 to show the days remaining for the Hotel licence.
If a valid licence Key has been entered the display will show ON, for example

20-33-01	
Hotel	ON

PCPro can also be used to view the Hotel license in Hotel License Information, if Active(ON) is displayed then a valid license code has been entered into the system and the 30 day limit has been removed.

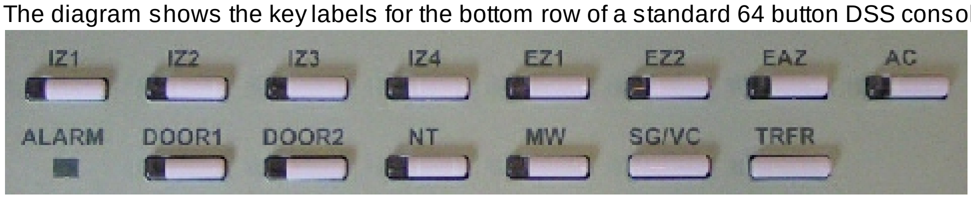
Hotel - Receptionist Guide

Receptionist Quick Guide

Feature	**Operation**
Check In Check Out	**738** + room number **739** + room number
Room Clean Status	**741**+ room number + Status (1-4) 1 = Room Clean 2 = Maid Required 3 = Maid in Room 4 = Inspection Required
Toll Restriction Class (when checked in)	**737** + room number + Class (01-15)
Room to Room Call Restriction	Set = **735** + room number Cancel = **736** + room number
Wake Up Call	Set = **733** + room number + hhmm (24hour clock) Cancel = **734** + room number
Room Status Printout	**742** + Option (0-5) 0 = All Printouts 1 = Room Status List 2 = Call Restriction List 3 = Do Not Disturb and Room Clean List 4 = Message Waiting List 5 = Wake Up Call List
Do Not Disturb	Set = **729** + room number Cancel = **730** + room number
Room Monitor (SLT phone only)	770 + 2 + room number

Hotel DSS Console

The reception telephone can have a 64 button DSS console assigned that will show the status of the hotel room telephones (for example checked in, vacant, room clean).
It will also show the status of other extensions (idle, busy, DND, Call forward).
The bottom row of feature keys have the functions shown below.
The diagram shows the key labels for the bottom row of a standard 64 button DSS console, you may want to rename the keys.



NT The Night Mode key will set the system into night mode (lamp is on when system is in night mode).
If you require selection of other night modes you must set the function keys on the keyphone, see Function Key Programming below.
IZ1 Will show the Message Waiting status and will change the mode of the 64 busy lamp keys to show the message waiting status of the hotel rooms.
IZ2 Will show the The Wake Up Call status and will change the mode of the 64 busy lamp keys to show the wake up call status of the hotel rooms.
IZ3 Will show the Check In/Out status and will change the mode of the 64 busy lamp keys to show the check in/out status of the hotel rooms.

The Hotel reception telephone can use the Programmable Function keys on their keyphone or have the 24 button DLS console installed but these will not provide DSS lamp indication for hotel room telephones.

DSS Console Lamp Indications
The DSS lamps will show the status of the room telephone:
When IZ1, IZ2 and IZ3 lamps are off the 64 keys will show the busy lamp information (idle/off hook etc) for all extensions on the system, including non-hotel room telephones.
Note, when the DSS console is set to Hotel mode buttons IZ1 to IZ3 are used for hotel STATUS and IZ4 is not used. If you want keys for Internal Paging zones then you must use the Programmable Function keys on the keyphone.

Message Waiting (IZ1 key lit on DSS console)
ON = A Message Waiting
OFF = No messages

Wake Up Call Status (IZ2 key lit on DSS console)
ON = A Wake Up Call set
OFF = No Wake Up Call set
FLASH = Wake Up Call missed

Check in/Out Status (IZ3 key lit on DSS console)
ON = Checked In and Clean
OFF = Checked Out (Clean and Available)
FAST FLASH = Maid Required
MEDIUM FLASH = Maid in Room
LONG FLASH = Inspect

Function Key Programming To change the function of a General Function programmable key:
1. Press idle CALL key.

2. Dial 851.
3. Press the key you want to program.
4. Enter the 2-digit key function, any additional information needed for the key and press HOLD.
Available functions are 00-99 and line keys 001-200 (for a DSS key enter 01+room telephone number+HOLD, for Night Mode enter 09 +mode number).
To undefine a key, enter 00.

To change the function of an Appearance Function programmable key:
1. Press idle CALL key.
2. Dial 852.
3. Press the key you want to program.
4. Enter the 3-digit key function and any additional information needed for the key.
Available functions are *00-*99 and line keys 001-200.
To undefine a key, enter 000.
When a key is programmed using service code 852, that key cannot be programmed with a function using the 851 code until the key is undefined (852+000).

Check In and Check Out

When hotel room telephones are checked in their toll restriction class can be changed automatically by the system to allow the guest to make calls.

To Check In a hotel room telephone:
1. Lift the handset.
2. Dial 738.
3. Dial the extension number of the room you want to check in.
You hear confirmation tone.
4. Hang up.
In the STATUS mode, the DSS Console key for the room is on.

To Check Out a hotel room telephone:
You can set a room as checked out only if you have previously checked it in.
1. Lift the handset.
2. Dial 739.
3. Dial the extension number of the room you want to check out.
You hear confirmation tone.
4. Hang up.
In the STATUS mode, the DSS Console key for the room is off.

When the room is checked out the system will automatically cancel any Message Waiting, Do Not Disturb, Room to Room Call Restriction, Toll Restriction and Wake Up calls that may be set at the room telephone.

Room Clean Status

You can use the DSS console lamps to indicate the status of the hotel room, commonly Clean (occupied or not), Maid Required, Maid in Room, Inspection Required.

1. Lift the handset.
2. Dial 741.
3. Dial the extension number of the room you want to set.
4. Dial the room status code:
1 = Room Clean
2 = Maid Required
3 = Maid in Room
4 = Inspection Required
5. You hear confirmation tone.
6. Hang up.

In the STATUS mode, the DSS Console shows the room's status.
ON = Checked In and Clean
OFF = Checked Out and Clean
FAST FLASH = Maid Required
MEDIUM FLASH = Maid in Room
LONG FLASH = Inspect

The room status can also be set from the room's telephone, refer to the Hotel Staff - Room Telephone Guide.

Toll Restriction

When a room is checked in the telephone can be given a different Toll Restriction class, this would usually have no restrictions.
When the room is checked out it would typically have a fully restricted.
This operation is automatic if setup in the system configuration.

Changing Toll Restriction class while room is checked in
The Toll Restriction class number can be changed for a room telephone when checked in.
The Toll Restriction classes must be pre-defined in the system configuration, check with your system maintainer for the Toll Restriction class numbers you can use.

To change a room telephone's Toll Restriction (When Checked In) level:
1. Lift the handset.
2. Dial 737.
3. Dial the extension number of the room which you want to change the Toll Restriction (When Checked In) level.
You hear a single beep.
4. Enter the new Toll Restriction (When Checked In) level (01-15).
You hear confirmation tone.

Room to Room Call Restriction

You can prevent a guest from placing calls to other hotel rooms.
This will not prevent the guest placing outside calls or calls to non-hotel room telephones.

To enable Room-to-Room Call Restriction for a guest's phone:
1. Lift the handset.
2. Dial 735.
3. Dial the guest's phone number.
You hear confirmation tone.
The guest can not dial any other Hotel room extension.

To disable Room-to-Room Call Restriction for a guest's phone.
1. Lift the handset.
2. Dial 736.
3. Dial the guest's phone number.
You hear confirmation tone.

Wake Up Call

A Wake Up call is like an alarm clock for the guest.
The guest can also set and cancel their own Wake Up calls.

To set a Wake Up for a room:
1. Lift the handset.
2. Dial 733.
3. Dial the number of the room phone that should receive the wake up.
4. Dial the time for your wake up.
Use a 24-hour clock. For example, 1:00 PM = 13:00. You hear confirmation tone.
5. Hang up.

To cancel a Wake Up for a room:
1. Lift the handset.
2. Dial 734.
3. Dial the number of the room phone whose wake up you want to cancel.
You hear confirmation tone.

The system can also be setup to alert the Reception telephone when a Wake Up call is not answered. The display will show the room number of the missed Wake Up call.

To check the Wake Up calls set
If you have a DSS console:
1. Press the Wake Up Call Status key (IZ2 key)
ON = A Wake Up Call set
OFF = No Wake Up Call set
FLASH = Wake Up Call missed

To clear the missed Wake Up call indication place a call to the room from the Reception telephone, when the call is answered the lamp will go out.

If you have a printer connected:
You can print out a list of Wake Up calls.
1. Lift the handset.
2. Dial 742.
3. Dial 5 (the Wake Up Call List option).

The output will show the room numbers and the time set for the Wake Up call.
A missed Wake Up call will be shown by a * before the room number, as shown below for room 236.

Wake Up Call List -------- 11/04/2006 12:55
*236 -12:48

The printer can also output missed Wake Up calls automatically if setup in the system configuration.

Room Status Printouts

The system can have a printer connected to the EXIFU card (serial port) that can be used to printout the following reports.
Room Status List
Call Restriction List
Do Not Disturb and Room Clean List
Message Waiting List
Wake Up Call List

To have your printer output the Room Status Printout:
Your printer should be location conveniently next your phone.
1. Lift the handset.
2. Dial 742.
3. Dial the Room Status Printout option:
0 = All Printouts
1 = Room Status List (Check-in and House Cleaning Status)
2 = Call Restriction List
3 = Do Not Disturb and Room Clean List
4 = Message Waiting List
5 = Wake Up Call List
4. Hang up.

Room Status Printout example
Room Status List -------- 11/04/2006 14:06
Room Clean(Occupied) --- Check In
236 , 238
Room Clean(Vacant) --- Check Out
237 , 239 , 240
Maid Required
241
Maid in Room

Inspection Required
242 , 243 , 244 , 245

END

Call Restriction List example
Shows rooms that have Room to Room Call Restriction set and the current Toll Restriction class number of each hotel room.
Calling Class List -------- 11/04/2006 14:21
Room to Room Barring

236
Outside Call Class
236 -02, 237 -02, 238 -01, 239 -02,
240 -02, 241 -02, 242 -02, 243 -02,
244 -02, 245 -02

END

Do Not Disturb and Room Clean example
DND and Clean Up Check -------- 11/04/2006 14:23
Do No Disturb
242
Clean Up Check
241

END

Message Waiting List example
Message Service List -------- 11/04/2006 14:39
236

END

Wake Up Call List example
Wake Up Call List -------- 11/04/2006 14:40
236 -14:50, 239 -07:30, 243 -06:45

END

Do Not Disturb

You can set Do Not Disturb for a room telephone to prevent calls being made to the room telephone.
The room telephone can also set/cancel Do Not Disturb, refer to Hotel Guest - Room Telephone Guide.

To enable DND for a room telephone:
1. Lift handset.
2. Dial 729.
3. Dial the number of the extension for which you want to enable DND.
You hear confirmation tone.
4. Hang up.

If you need to contact the guest you may be able to override the Do Not Disturb:
1. Lift handset.
2. Call the room telephone.
3. Dial 809 (if override is enabled the telephone will ring).

To cancel DND for a room telephone:
1. Lift handset.
2. Dial 730.
3. Dial the number of the extension for which you want to disable DND.
You hear confirmation tone.
4. Hang up.

Message Waiting

You can leave a Message Waiting indication (flashing lamp) for a guest.
When the guest replies to the Message Waiting the system will automatically setup a call to the Reception telephone.

To leave a Message Waiting:
1. Call the room telephone. There is no answer.
2. Dial 841.
You hear confirmation tone. The Message Waiting lamp flashes on the room's telephone.
3. Hang up.
If you want to cancel the message you just left, lift the handset and dial 871 and then the room number.

To Leave a Message Waiting Without First Calling the Extension:
1. Lift the handset.
2. Dial 726.
3. Dial the number of the room telephone at which you want to leave the message waiting.
You hear confirmation tone.

Room Monitor

The room telephone can be used to monitor the audio within the hotel room.
The monitored room telephone and the monitoring telephone at reception must be single line telephones, not keyphones.
The room telephone must first be setup to be monitored, refer to the Hotel Guest - Room Telephone Guide.

1. Lift the handset.
2. Dial 770 + 2.
3. Dial to room number.
You hear confirmation tone and can hear the audio near to the room telephone.
If the room telephone is not setup to be monitored you will hear busy tone.
4. You can monitor other room telephones, if setup to be monitored, by going on hook and repeating steps 1 to 3.

Hotel Guest - Room Telephone Guide

Room Telephone Quick Guide

Feature	**Operation** Lift the handset and dial the code to set any feature, when done replace the handset.
Wake Up Call	To set: Dial **731** + time (hhmm) using 24 hour clock
	To cancel: Dial **732**
Do Not Disturb	To set: Dial **727**
	To cancel: Dial **728**
Reply to Message Waiting	Dial **841**
Room Monitor	Dial **770** + 1 + telephone number that will be used to monitor
Call to Reception	Dial 0
Outside Line	Dial 9 and wait for dial tone then dial the number you require

Wake Up Call

A Wake Up call is like an alarm clock.
You can also ask Reception to set a wake up call for you.

To set a Wake Up call:
1. Lift the handset.
2. Dial 731.
3. Dial the time for your wake up (hhmm).
Use a 24-hour clock. For example, 7:30 AM = 0730.
You hear confirmation tone (you may also hear the time repeated back to you).
4. Hang up.

To cancel a Wake Up call:
1. Lift the handset.
2. Dial 732.
You hear confirmation tone.

To answer a Wake Up call:
Your room telephone will ring at the time set for the Wake Up call.
1. Lift the handset.
You hear simulated music (you may hear a pre-recorded announcement instead).

Do Not Disturb

You can stop calls to your telephone by setting Do Not Disturb.
Reception may be able to override your Do Not Disturb if they need to contact you urgently.

To set Do Not Disturb:
1. Lift handset.
2. Dial 727.
You hear confirmation tone.
3. Hang up.

To cancel Do Not Disturb:
1. Lift handset.
You hear interrupted dial tone when you lift the handset.

2. Dial 728.
You hear confirmation tone.
3. Hang up.

Message Waiting

Reception can leave a Message Waiting indication for you, this will be shown by a flashing lamp on your telephone.
When you reply to the Message Waiting a call will be placed to Reception.

To answer a Message Waiting left at your phone:
1. Lift the handset.
Listen for dial tone.
2. Dial 841.
You will automatically call the extension that left you a message.

Room Monitor

The room telephone can be used to monitor the audio within the hotel room.
The room telephone must first be setup to be monitored.

1. Lift the handset.
2. Dial 770 + 1.
3. Dial to telephone number that will be used to monitor the room (ask Reception for the number).
You hear confirmation tone.
4. Leave the handset off hook near to the sound you would like monitored.

Note - while your telephone is being monitored your calls will be overheard.
To cancel the room monitor the telephone that is monitoring your room must be placed on hook, ask Reception to do this.

Hotel Staff - Room Telephone Guide

Room Telephone Quick Guide (Hotel Staff)

Feature	Operation
Room Clean Status	**740** + Status (1-4) 1 = Room Clean 2 = Maid Required 3 = Maid in Room 4 = Inspection Required
Common Cancel Code	**720**

Room Clean Status

You can change the status of the hotel room: Clean (occupied or not), Maid Required, Maid in Room, Inspection Required.
This option may not be available due to system configuration.

1. Lift the handset.
2. Dial 740.
3. Dial the room status code:
1 = Room Clean
2 = Maid Required
3 = Maid in Room
4 = Inspection Required
4. You hear confirmation tone.
5. Hang up.

The room status can also be set from the Reception telephone, refer to the Hotel - Receptionist Guide.

Common Cancel Code

This code will cancel the following features if set at the room telephone.
Wake Up Call, including missed.
Message Waiting
Do Not Disturb
1. Lift the handset.
2. Dial 720.
3. You hear confirmation tone.

General Settings

When configuring the XN120 for Voice over IP (VoIP), there are certain general settings that are required.
These settings are required regardless of the modes of VoIP that will be used.

Before connecting the XN120 to a data network, it is necessary to obtain the relevant IP Addressing information. This information will be supplied by the IT Manager or Network Administrator at the customer site.

IP Addressing

The EXIFU-A1 and VOIPU card(s) require IP addresses. The following screens are used to configure the IP Network information. They can be found in the *VoIP - General Settings - IP Addressing section*

IP Network Setup

This screen allows you to change the IP configuration of the EXIFU-A1. In most cases it is necessary to obtain these details from the Network Maintainer or System Administrator.

IP Address - This sets the IP address to be used by the system. The address must be valid for the network that the system will be connected to.

Subnet Mask - The Subnet mask should be selected to determine the Network address for the system.

Default Gateway - The default gateway is the local router that will allow access to external networks. In some cases (i.e. a LAN only environment) this parameter is not required.

Time Zone - This sets the time zone. The options range from 0 to 24 which is the equivalent of -12 through to +12 hours (12 is equal to GMT)

NIC Setting - Ethernet port speed and duplex. Auto-negotiate can be used in majority of cases.

NAT Router - This item switches on the NAT feature (see next item)

Default Gateway (WAN) - If a NAT Router is used, it is possible to identify the public IP address that will be used for external connections. This item determines the public IP address to use. This feature only applies to SIP Trunks.

ICMP Redirect - Some routers use ICMP Redirect messages to re-route packets via a different router to the one specified by the host. In some cases this is an undesirable feature and this item allows the system to ignore (Disable) these messages.

VOIPU IP Address Setup.

This screen allows you to change the IP address for each VOIPU installed in the system. The settings are on a per-slot basis.

IP Address - This sets the IP address to be used by the VOIPU card. The address must be valid for the network that the system will be connected to.

NIC Setting - Ethernet port speed and duplex. Automatic Detection can be used in majority of cases.

VOIPU Configuration

When an IP Phone or IP Trunk calls a legacy device (Keyphone, SLT, trunk) the speech has to be converted from IP to TDM technologies.
The VoIPU card provides this function. Each VoIPU card has a number of DSP resources on board; each one can convert a speech channel from IP to TDM and vice versa.

It is necessary to configure the VOIPU to meet the requirements of the Voice over IP application. The configuration options can be found in VoIP - General Settings - VOIPU Configuration

DSP Resource Selection
Each VOIPU card contains a number of DSP (Digital Signal Processors). Each DSP can perform one TDM to IP conversion.

By default, each DSP can be used for any type of VoIP call (Extensions or Trunks). It is possible to configure specific DSPs to be available only for extensions or only for trunks.

For example:

You have a 4VOIPU in Slot 5 and are using IP Extensions and IP Trunks. You always want to have at least two IP trunks available - the rest of the resources will be available for any VoIP call. In this scenario you would set

Slot5

DSP 01 = IP Trunks
DSP 02 = IP Trunks
DSP 03 = IP Extensions / Trunks
DSP 04 = IP Extensions / Trunks

VOIPU Configuration Setup
Each VOIPU card contains a number of ports. Each port can be set to H.323 or SIP depending on the type of VoIP application required.
By default, each port is set to H.323. If you are using SIP applications you will need to change the required ports to SIP. It is possible to have a mixture of SIP/H.323 ports on the same card. The appropriate port will automatically be used by the VoIP application.

VOIPU Setup
This screen configures some of the VOIPU specific features.

RTP Port - UDP Port number used for Realtime Transport Protocol (speech packets). It is not usually necessary to change this port number from the default.

RTCP Port - UDP Port number used for Realtime Transport Control Protocol (RTP signalling). It is not usually necessary to change this port number from the default.

H.245 Port - TCP Port number used for H.245 signalling of call control and setup messages. It is not usually necessary to change this port number from the default.

DTMF Behaviour - This setting determines how DTMF signals are handled by the VOIPU. It is possible for the XN120 to detect DTMF tones and convert them to data, then regenerate the DTMF tones at the receiving side.

This option determines whether the DTMF relay occurs within the RTP stream (in band, using RFC2833), as a separate signal (out of band, using H.245) or is disabled.

If DTMF relay is disabled then DTMF will be treated as "voice" and sent within the RTP stream. This can cause the DTMF to become distorted and incorrectly interpreted at the receiving end, particularly when transmitted over unreliable data networks.

Ready/Answer Port - This UDP Port is used by IP Extensions. It is not usually necessary to change this port number from the default.

VOIPU Firmware information
This page displays the current firmware version for all VOIPU cards in the system. This information is "read only".

VOIPU Upgrade Procedure
A new version of VOIPU firmware may occasionally be released. The procedure for upgrading the firmware is very simple. Firstly, check the existing firmware version in VoIP - General Settings - VOIPU Configuration - VOIPU Firmware Information.

If this is different from the version that has been released:

- Copy the firmware file (voipu.bin) to the root directory of a Compact Flash card.
- Power off the system
- Move JP6 from the top position (pin 1&2) to the bottom position (pin 2&3). This jumper is located between the LOAD and RESET switches.
- Insert the Compact Flash card into the slot on the VOIPU
- Power on the system
- Check that the firmware version has changed in *VOIPU Firmware Information*
- Replace JP6 back to its original position

- Upgrade complete

QoS Settings

When transmitting Voice over an IP network it is important to consider Quality of Service (QoS).
This is the perceived quality of speech after being transmitted over the network.
It is recommended that you consult the "Voice over IP Reference Guide" as this discusses the issues that should be considered when implementing a VoIP network.

QoS is implemented by the network hardware, not the XN120.
The XN120 can "mark" its data with appropriate tags and the network equipment has to be configured to prioritise that data over other (non VoIP) data.
The Network Administrator should supply the XN120 installer with the relevant QoS settings.

QoS can be implemented at Layer 2 (within the Ethernet Frame) or Layer 3 (within the IP packet).

Layer 2 QoS and VLAN
QoS is implemented at Layer 2 by using VLAN (IEEE 802.1pq) tags.

VLAN Mode - By default the system does not use VLAN tags so these have to be enabled.
Once enabled, all frames transmitted by the EXIFU-A1 and VOIPU cards use the VLAN tags.

VLAN ID - This is the VLAN that the system belongs to. Valid values are 0 to 4094 tags

Priority - The priority should be configured between 0 (no prioritisation) to 7 (highest priority)

Layer 3 QoS
This is the most common form of QoS. It utilises the Type of Service (ToS) field within the IP packets, and can be configured based on two different QoS standards: IP Precedence and Diffserv.
These configuration items can be found within the *VoIP - QoS Settings - Layer 3 QoS* section.

Although IP Precedence and Diffserv are both supported on the XN120, it is becoming more common to use Diffserv only. The two methods of QoS are interoperable (IP precedence values can be mapped to Diffserv values).

The ToS value can be set for each type of VoIP packet.

	SIP Extensions	H.323 Extensions	SIP Trunks	H.323 Trunks
Voice Control (H.245) Sets up voice parameters for a voice call	√	√	√	√
H.323 Used for H.323 signalling information, including GK registration		√		√
RTP/RTCP Speech packets	√	√	√	√
SIP Used for SIP signalling, including registration	√		√	

ToS Mode - This setting determines if Diffserv or IP Precedence is to be used. If using Diffserv, all items on the page can be ignored, except for the last item.

IP Precedence Priority - The priority should be set between 0 (not prioritised) and 7 (highest priority). If Diffserv has been selected above, this setting will be ignored. (RFC 791)

IP Precedence Delay - Normal Delay or Low Delay (RFC 791)

IP Precedence Throughput - Normal Throughput or High Throughput (RFC 791)

IP Precedence Reliability - Normal Reliability or High Reliability (RFC 791)

IP Precedence Cost - Normal Cost or Low Cost (RFC 791)

Priority (Diffserv) - This is the Diffserv Code Point (DSCP) value as defined in RFC2474. This can be a value between 0 and 63. It is recommended that you refer to the table in Appendix C for further information.

Extensions

This section describes the procedure for connecting SIP and H.323 extensions to the PBX.

SIP and H.323 are industry standard protocols and therefore there are many manufacturers hardware and software based phones. As these phones are not developed by us, and are not designed specifically for use on the PBX, they do not support majority of the features that you would find on a Keytelephone.

Various types of SIP/H.323 Phones are available, including:

- Software application for PCs (usually used on laptop computers)
- Software applications for PDA (Personal Digital Assistant)
- Messaging software with integrated SIP/H.323 capability (e.g. Windows Messenger)
- Hardware based telephone
- Analogue adapter - allows connection of analogue POT telephone to SIP/H.323 network

The SIP/H.323 Extension will register itself to the system. The registration creates a map between the IP phone and an extension port on the PBX. This means that any programming related to that extension port (for example, Class of Service) will apply to the IP Extension.

To allow registration of an IP Extension to a particular extension port it is necessary to assign an extension number to the port using VoIP – Extensions – [SIP Extensions / H.323 Extensions] – Extension Setup. This will be the extension number assigned to the IP Phone.

The IP Phone should be configured with a valid IP Address and should be connected to the same data network as the PBX. The procedure for configuring the IP Extension varies depending on the Manufacturer – this guide does not cover the configuration of third party equipment.

SIP Extensions

The configuration options for connection of SIP Extensions can be found in *VoIP - Extensions - SIP Extensions*.
The PBX is compliant with the RFC3261 SIP standard.
The features available to SIP extensions are detailed in the SIP Cetificate of Compatibility for the relevant SIP terminal.

SIP Extension Setup

This page can be used to change extension numbers and names for all extensions on the system (including SIP Extensions).

Extension Number - Set the Extension Number for this port. This is the "username" or "SIP ID" that should be configured on the SIP phone.
The actual terminology used by the third-party device for this item varies.

Extension Name - Set the Name to be associated with this extension number

Terminal Type - This will show the type of terminal that has been registered to the Extension port. This is a read only setting.

MAC Address - When an IP extension has registered to the system, its MAC address will appear in this area

IP Address - When an IP extension has registered to the system, its IP address will appear in this area

Authentication Password - A password can be assigned to each extension. This password must match the password configured on the IP Phone

Calling Party Display Info - It is possible to use the Calling Party number supplied by the IP Phone. The IP Phone may have the Calling Party Number in various SIP Fields.
This item determines which field to use.
IP Duplication Allowed Group - There are 10 IP groups available. These groups allow multiple registrations with the same IP address.

For example, if a device has two ports and one IP address (for example a Cisco ATA186), this device will typically send two registration attempts - one for each port.
When the system receives the second registration and determines that it is from the same IP address it will overwrite the first registration.

By adding the two extension numbers into the same "IP Duplication Group" this will not occur and multiple registrations from the same IP address will be allowed.

<u>SIP System Information Setup</u>
This page defines the basic SIP identity of the system.

Domain Name - This is the domain that the system belongs to. For example: CustomerXZY or mysipnetwork.com.
If there is no connection to external SIP providers this item can be set to any valid value.

Host Name - The host name that will be combined with the Domain Name above. For example "SystemA". If there is no connection to external SIP providers this item can be set to any valid value.

Transport Protocol - This must be set to UDP

User ID - This is the SIP UserID. If there is no connection to external SIP providers this item can be set to any valid value.

Domain Assignment - The SIP messages can use the system IP address or the domain name for addressing

<u>SIP Codec Setup</u>
VoIP can use various CODECs. A CODEC is a standard for converting an analogue signal to digital. This conversion process is handled by the DSP (Digital Signal Processors) on VOIPU cards.

Each CODEC has different voice quality and compression properties. The correct choice of CODEC will be based on the amount of bandwidth available, the amount of calls required and the voice quality required.

CODECS:
G.729. Low bandwidth requirement, and is used on most Wide Area Network links
G.711. High bandwidth requirement - usually used on Local Area Networks.
G.723. Low bandwidth requirement, but not commonly used.

This page configures the voice parameters to be used with SIP Extensions.

Peer to Peer Mode - Switches SIP Extension Peer to Peer mode on or off.
This allows RTP (speech) to be sent directly between SIP extensions, which reduces the number of DSP resources required.

Audio Capability Priority - This setting determines the preferred CODEC to be used for SIP Extensions.

G.711 Audio Frame Number - The amount of 10ms periods to encapsulate into one packet. For example 3 = 30ms. This setting has an effect on bandwidth and voice quality so careful consideration should be made before changing this item.

G.711 Voice Activity Detection Mode - VAD detects when there is silence during a call and when this occurs the RTP stream is temporarily stopped and replaced with "comfort noise". This reduces the bandwidth consumption but can sometimes be noticeable at the point of switching between speech and comfort noise. This item enables/disables VAD for G.711

G.711 Type - This should be set to A-Law companding

G.711 Jitter Buffer (min) - If a dynamic Jitter buffer is used; this is the minimum length of the buffer

G.711 Jitter Buffer (typical) - This is the normal Jitter buffer size

G.711 Jitter Buffer (max) - If a dynamic Jitter buffer is used; this is the maximum length of the buffer

The above settings are also applied to G.729 and G.723 CODECS (with the exception of the companding method).

Jitter Buffer Mode - The Jitter Buffer can use a static setting (based on the Typical setting) or can adjust based on network conditions. The automatic adjustment can be made Immediately or During Silence.

VAD Threshold - This is the threshold for silence detection. The threshold attempts to differentiate between background noise and actual speech. Range is 0 (-20db) to 30 (+10db)

Idle Noise Level - It is possible to set the level for comfort noise generated whilst VAD is in operational. Range 5000 (-5000dbm) to 7000 (-7000dbm).

TX Gain - Sets the transmit gain from 0 (-14dbm) to 28 (+14dbm)

RX Gain - Sets the receive gain from 0 (-14dbm) to 28 (+14dbm)

DTMF Payload Number - When DTMF is sent via RTP it is possible for SIP Extensions to use various payload types. This value must match the type used by the SIP Extension

<u>SIP Basic Information Setup</u>

Registrar/Proxy Port - This is the port number for SIP Extensions. When configuring a third-party SIP Extension it is usually necessary to change the port number from 5060 to 5070.

Session Timer Value - This determines the timer that is used for INVITE and UPDATE requests. (See draft-ietf-sip-session-timer-xx for further info)

Minimum Session Timer Value - This determines the minimum value that can be used for Session Timers (see draft-ietf-sip-session-timer-xx for further info)

Called Party Info - When calling from the system to the SIP Extension it is possible to send the Called Party number based on the URI header or the TO Header.

Expire Value of Invite - If there is no response to an INVITE request within this time (in seconds) the request will be cancelled.

<u>SIP Registrar/Proxy Setup</u>

Registration Expiry Timer - This determines how long a SIP Extensions registration is valid for. After this timer, the SIP Extension must re-register

Authentication Mode - If this item is set to Disabled, it is not necessary for SIP Extensions to use the password information specified in VoIP - Extensions - SIP Extensions - Extension Setup. This means that there is no authentication of extensions.

Registrar/Proxy Domain - It is possible to specify the systems SIP Registrar domain name. This setting should be configured on the SIP Extension. If this is not configured it is still possible for the SIP Extensions to register using the system IP address

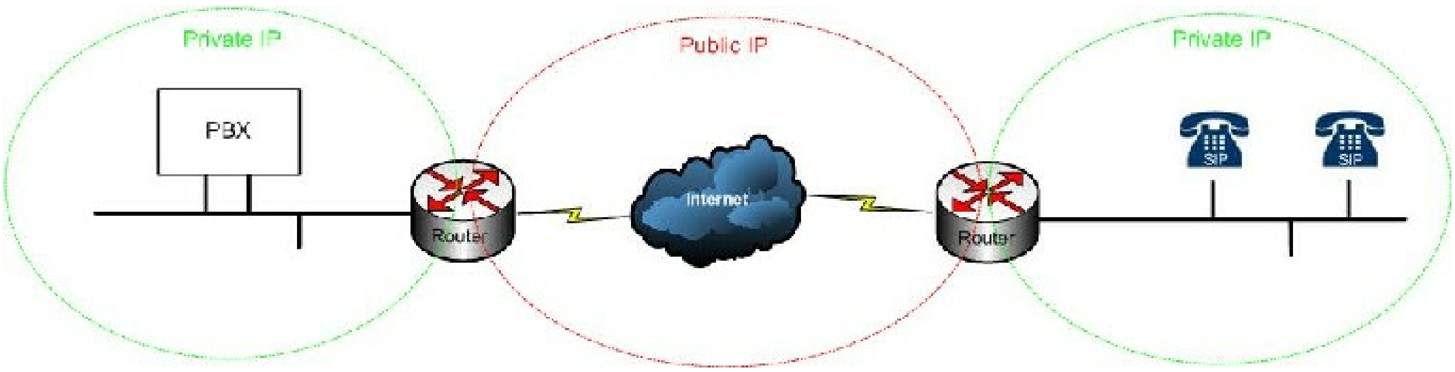
Registrar/Proxy Host Name - Systems Registrar Host Name - See item above

<u>Configuration of third-party SIP phones</u>
The PBX uses UDP/5070 for SIP Extension registrations. Most SIP Extensions use UDP/5060 by default so this needs to be changed on the SIP Extensions.
The SIP Extension should be configured to login as the Extension Number assigned in *VoIP - Extensions - SIP Extensions - Extension Setup.*

<u>SIP Extensions and Network Address Translation (NAT)</u>

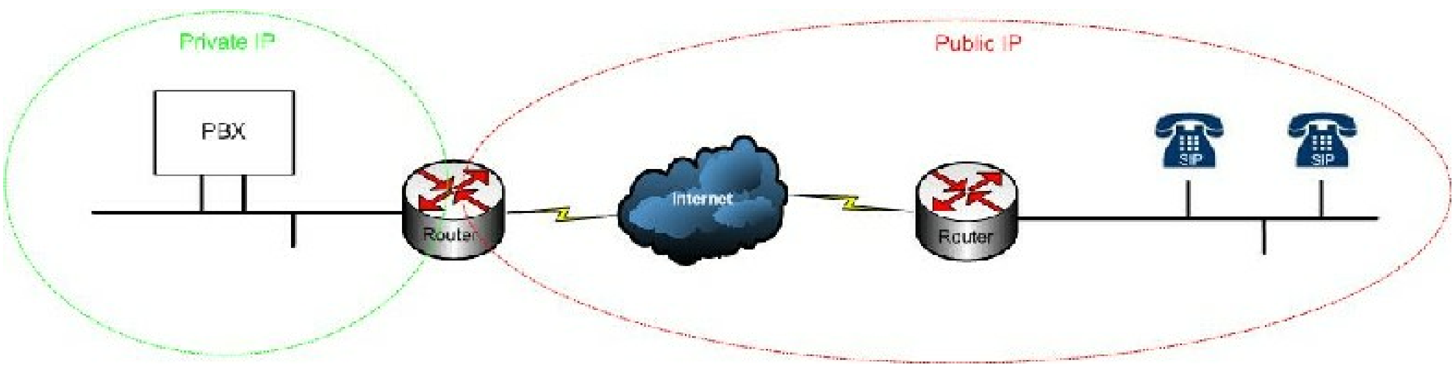
The following issues need to be considered when connecting SIP extensions to the telephone system over a NAT-enabled network:

Scenario 1
Private IP addresses on the telephone system and SIP extensions. Public IP Addresses on the Internet connection. NAT enabled on both routers.



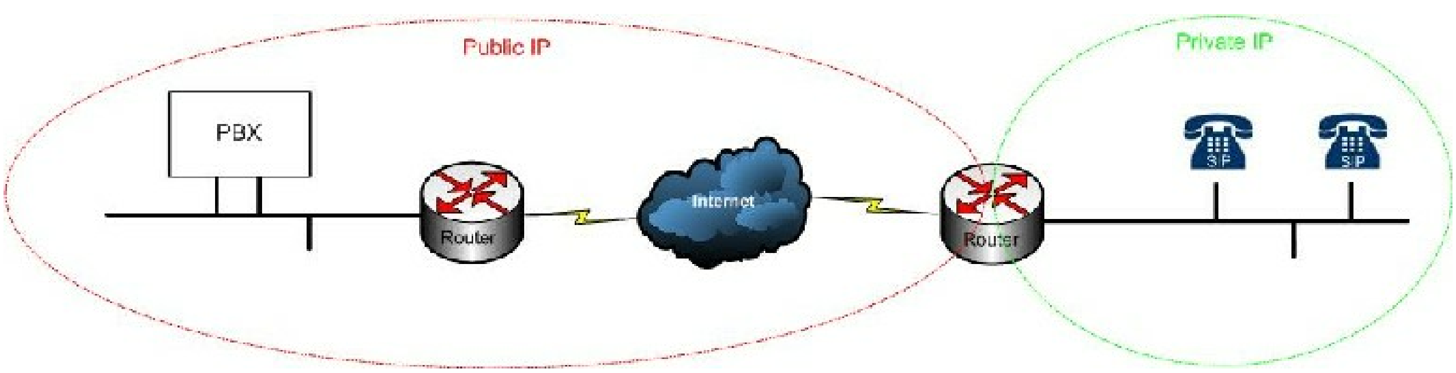
- The telephone system cannot perform NAT traversal
- A VPN tunnel must be implemented between the two Routers. This allows the SIP extensions and telephone system to use private IP addressing.

Scenario 2
Private IP addresses used on the telephone system. Public IP addresses on the Internet connection and SIP extensions. NAT enabled on the router at the telephone system side.



- The telephone system cannot perform NAT traversal
- A VPN tunnel must be implemented between the two Routers. This allows the SIP extensions and telephone system to use private IP addressing.

Scenario 3
Public IP addresses used on the telephone system and Internet connection. Private IP addresses used on the SIP extensions. NAT enabled on the extension side.



- It must be possible to assign unique local port numbers for SIP and RTP/RTCP to each SIP extension.
- The SIP phone must support NAT traversal. This means that the extension should be able to specify its public IP address as the RTP destination, rather than its private IP address.
- The router at the extension side must support port forwarding
- Note: This method is not possible on all SIP extensions - some devices do not support the above requirements.

Example Configuration

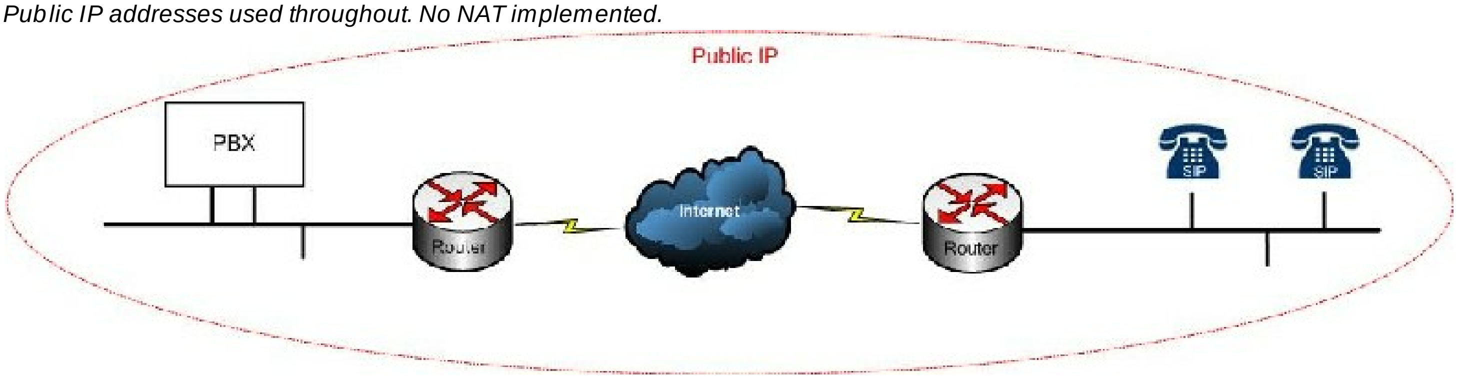
	SIP Extension	SIP Extension 2

SIP Local Port	5060	5064
SIP RTP/RTCP Port	5062-5063	5066 - 5067
IP Address	172.16.0.100	172.16.0.101
NAT Address	80.1.2.3	80.1.2.3

Port forwarding (on router):

Destination	**Forward to**	**Description**
80.1.2.3:5060	172.16.0.100:5060	SIP Extension 1 - SIP signalling
80.1.2.3:5062	172.16.0.100:5062	SIP Extension 1 - RTP
80.1.2.3:5063	172.16.0.100:5063	SIP Extension 1 - RTCP
80.1.2.3:5064	172.16.0.101:5064	SIP Extension 2 - SIP Signalling
80.1.2.3:5066	172.16.0.101:5066	SIP Extension 2 - RTP
80.1.2.3:5067	172.16.0.101:5067	SIP Extension 2 - RTCP

Scenario 4
Public IP addresses used throughout. No NAT implemented.



- No VPN or port forwarding required
- Ensure that all necessary UDP ports are enabled on firewalls at both ends

H.323 Extensions
It is possible to connect third party H.323 (Version 3) phones to the PBX

The configuration options for connection of H.323 Extensions can be found in *VoIP - Extensions - H.323 Extensions.*

H.323 Extension Setup
This page can be used to change extension numbers and names for all extensions on the system (including H.323 Extensions).

Extension Number - Set the Extension Number for this port. This is the "alias", "username" or "UID" address that should be configured on the H.323 phone. The actual terminology used by the third-party device for this item varies.

Extension Name - Set the Name to be associated with this extension number

Terminal Type - This will show the type of terminal that has been registered to the Extension port. This is a read only setting

MAC Address - When an IP extension has registered to the system, its MAC address will appear in this area

IP Address - When an IP extension has registered to the system, its IP address will appear in this area

Authentication Password - A password can be assigned to each extension. This password must match the password configured on the IP Phone

Calling Party Display Info - This is not applicable for H.323

IP Duplication Allowed Group - This is not applicable for H.323

H.323 Codec Setup
This page configures the voice parameters to be used with H.323 Extensions. The Codec Setup section for SIP Extensions (above) describes the available CODECS.

Audio Capability Priority - This setting determines the preferred CODEC to be used for SIP Extensions.

G.711 Audio Frame Number - The amount of 10ms periods to encapsulate into one packet. For example 3 = 30ms. This setting has an effect on bandwidth and voice quality so careful consideration should be made before changing this item.

G.711 Voice Activity Detection Mode - VAD detects when there is silence during a call and when this occurs the RTP stream is temporarily stopped and replaced with "comfort noise". This reduces the bandwidth consumption but can sometimes be noticeable at the point of switching between speech and comfort noise. This item enables/disables VAD for G.711

G.711 Type - This should be set to A-Law companding

G.711 Jitter Buffer (min) - If a dynamic Jitter buffer is used; this is the minimum length of the buffer.

G.711 Jitter Buffer (typical) - This is the normal Jitter buffer size

G.711 Jitter Buffer (max) - If a dynamic Jitter buffer is used; this is the maximum length of the buffer

The above settings are also applied to G.729 and G.723 CODECS (with the exception of the companding method).

Jitter Buffer Mode - The Jitter Buffer can use a static setting (based on the Typical setting) or can adjust based on network conditions. The automatic adjustment can be made Immediately or During Silence.

VAD Threshold - This is the threshold for silence detection. The threshold attempts to differentiate between background noise and actual speech. Range is 0 (-20db) to 30 (+10db)

Idle Noise Level - It is possible to set the level for comfort noise generated whilst VAD is in operational. Range 5000 (-5000dbm) to 7000 (-7000dbm).

TX Gain - Sets the transmit gain from 0 (-14dbm) to 28 (+14dbm)

RX Gain - Sets the receive gain from 0 (-14dbm) to 28 (+14dbm)

DTMF Relay Mode - When DTMF is sent via H.323 it is possible to use "relaying" so that DTMF is sent as data packets rather than as part of the RTP stream.
There relaying methods available are: RFC2833 (this is an Internet standard - the receiving host must also support this) and VOIPU (this uses the setting within *VoIP - General Settings - VOIPU Configuration - VOIPU Setup* in Section 1)

Configuration of third-party H.323 phones
The PBX uses port TCP/1719 for H.323 RAS. This means that most H.323 IP Phones will be able to connect to the PBX using their default settings.
The H.323 Extension should be configured to register as the Extension Number assigned in *VoIP - Extensions - H.323 Extensions - Extension Setup.*

It is important to note that third-party H.323 phones have many configuration items, and some of these are given slightly different names by different manufacturers.
The list below shows some common items and the suggested setting, to allow compatibility with the PBX.

Fast Start	Enabled
H.245 Tunnelling	Disabled
H.245 in Setup	Disabled
DTMF - Q.931 Keypad	Disabled
DTMF - H.245 Signal	Disabled
DTMF - H.245 Alphanumeric	Enabled
DTMF - In Band	Enabled (if no other supported option is available)
DTMF - RFC2833	Enabled

To delete a telephone registration:
From telephone programming, enter program 90-23-01, then enter the extension number of the IP Keyphone. Press 1 and then HOLD to delete the registration.

Networking

The following networking modes are supported on the XN120:

- SIP Networking
- H.323 Networking

Each of these connection modes are described in detail below.

Numbering Plan
When planning a networking implementation, it is important to determine the numbering plan for the sites. A properly planned VoIP network will allow flexibility and will simplify additions or changes to the network.

There are two approaches to the numbering scheme;

Open Numbering

In this case the sites are identified by dialling codes that are appended to the dialled digits. This allows the extension numbers to be the same at multiple sites.
For example:

	System A:	System B:
Extension Number	200-299	200-299
Site Code	001	002

Ext200 at System A can dial Ext200 at System B by dialling the site code, followed by the extension digits - 002200

This numbering scheme is ideal when there are a large number of sites in the VoIP network, or when there are lots of extension numbers in use at each site.

Closed Numbering

In this case, the first digit(s) of the extension number are used for routing calls. This alleviates the need for dialling a site code and makes a more unified numbering plan.
This is only feasible if there are a relatively small number of sites, or the sites have very few extensions.

Dialling via IP Trunks
It is possible to seize an IP trunk and dial the remote destination number.

Note: XN120 uses en-bloc sending for IP Trunk calls. This means that the call will not proceed until the External Call Interdigit Timer has expired, or # is entered.
It is possible to reduce the timer, but is should be noted that this will also affect ISDN calls.

Example 1:
Seize a VOIPU trunk and dial 200
External Call Interdigit Timer in Trunk Setup - Timers (default: 10 secs) starts.
After timer expires, call proceeds

Example 2:
Seize a VOIPU trunk and dial 200#
Call proceeds immediately

It may be advantageous to configure f-route to route calls to remote destinations via the IP trunks (this can simplify dialling for the users), see "call routing including DDI/DDO" (below).
The External Call Interdigit Timer still applies when dialling via f-route.

SIP Networking

SIP (Session Initiation Protocol) is a protocol used for Voice over IP. It is defined by the IETF (Internet Engineering Task Force) in RFC3261. XN120 can use SIP to connect to another XN120 system, an XN120 or a third party product.

The configuration options for SIP Networking can be found in *VoIP - Networking - SIP Networking*.

System Information Setup
This page defines the basic SIP identity of the system.

Domain Name - This is the domain that the system belongs to. For example: CustomerXZY or mysipnetwork.com. If there is no connection to external SIP providers this item can be set to any valid value.

Host Name - The host name that will be combined with the Domain Name above. For example SystemA. If there is no connection to external SIP providers this item can be set to any valid value.

Transport Protocol - This must be set to UDP

User ID - This is the SIP UserID. If there is no connection to external SIP providers this item can be set to any valid value.

Domain Assignment - The SIP messages can use the system IP address or the domain name for addressing

SIP Server Setup
Configures items related to SIP server registration

Default Proxy (Transmit) - If a SIP Proxy is used, and all SIP calls should be forwarded to the proxy, this item should be switched on

Default Proxy (Receive) - If a SIP Proxy is used, and all SIP calls are received from the proxy, this item should be switched on

Default Proxy IP Address - Configure the IP address for the SIP Proxy (if used)

Default Proxy Port No - Configure the UDP port used by the SIP Proxy

Registrar Mode - If a SIP Proxy is used you should leave this as None. If you are registering to a SIP Server you should usually set this to Automatic.

Registrar IP Address - Enter the IP address. If this is not known, and a domain name has been supplied (e.g. sipserver.company.com) then you should leave this as 0.0.0.0 and use the DNS settings below.

Registrar Port - Enter the UDP port number used by the SIP Registrar. This will usually be 5060

DNS Server Mode - If the SIP Server is to be accessed by domain name rather than IP address, a DNS server is required and this item has to be set to On.

DNS Server IP Address - Enter the IP Address of the DNS server. This should be supplied by the network Administrator.

DNS Server Port Number - The port number used by DNS. This is almost always set to 53.

Registrar Domain Name - Enter the fully qualified domain name (FQDN) for the SIP registration server (e.g. sipserver.company.com)

Domain Name - Enter the Domain name of the SIP registration server (e.g. company.com)

Host Name - Enter the Host name of the SIP registration server (e.g. sipserver)

SIP Carrier Choice - This item alters the format of outbound CLIP.
For Networking Mode, this item should be set to 2
For Carrier Mode, please refer to the SIP Certificate of Compatibility

Registration Expiry Time - This is the amount of time in seconds that should expire before the system attempts to re-register with the SIP Server.

<u>Authentication Information</u>
This screen provides the login authentication information to the SIP server. Usually this information would be supplied by the SIP Server maintainer.

Username - Enter the username
Password - Enter the Password

<u>Trunk Registration Information</u>
It is possible to register up to 32 different User IDs to the SIP server. In most cases only one ID is used and this item is configured in Authentication Information (above). If further IDs are required they are configured in this screen.

Registration - Enables or Disables this User ID

User ID - The user name or number which has been supplied by the SIP server administrator.

Authentication User ID - The authentication user name or number which has been supplied by the SIP server administrator. This is usually the same as the User ID.

Authentication Password - The password supplied by the SIP server administrator.

<u>Codec Setup</u>
This page configures the voice parameters to be used with SIP Networking. The Codec Setup section for SIP Extensions (above) describes the available CODECS.

Audio Capability Priority - This setting determines the preferred CODEC to be used for SIP Networking.

G.711 Audio Frame Number - The amount of 10ms periods to encapsulate into one packet. For example 3 = 30ms. This setting has an effect on bandwidth and voice quality so careful consideration should be made before changing this item.

G.711 Voice Activity Detection Mode - VAD detects when there is silence during a call and when this occurs the RTP stream is temporarily stopped and replaced with "comfort noise".
This reduces the bandwidth consumption but can sometimes be noticeable at the point of switching between speech and comfort noise. This item enables/disables VAD for G.711

G.711 Type - This should be set to A-Law companding

G.711 Jitter Buffer (min) - If a dynamic Jitter buffer is used; this is the minimum length of the buffer

G.711 Jitter Buffer (typical) - This is the normal Jitter buffer size

G.711 Jitter Buffer (max) - If a dynamic Jitter buffer is used; this is the maximum length of the buffer

- The above settings are also applied to G.729 and G.723 CODECS (with the exception of the companding method).

Jitter Buffer Mode - The Jitter Buffer can use a static setting (based on the Typical setting) or can adjust based on network conditions. The automatic adjustment can be made Immediately or During Silence.

VAD Threshold - This is the threshold for silence detection. The threshold attempts to differentiate between background noise and actual speech. Range is 0 (-20db) to 30 (+10db)

Idle Noise Level - It is possible to set the level for comfort noise generated whilst VAD is in operational. Range 5000 (-5000dbm) to 7000 (-7000dbm).

TX Gain - Sets the transmit gain from 0 (-14dbm) to 28 (+14dbm)

RX Gain - Sets the receive gain from 0 (-14dbm) to 28 (+14dbm)

DTMF Payload Number - When DTMF is sent via RTP it is possible for SIP Networking to use various payload types.

DTMF Relay Mode - When DTMF is sent via SIP Networking it is possible to use "relaying" so that DTMF is sent as data packets rather than as part of the RTP stream.

The relaying method used is RFC2833 (this is an Internet standard - the receiving host must also support this)

Remote Destinations
This screen allows configuration of up to 1000 remote systems. This table is shared with H.323 Networking.
If a SIP server registration has been configured then this table will not be utilised.

System Interconnection - Enables or Disables this entry

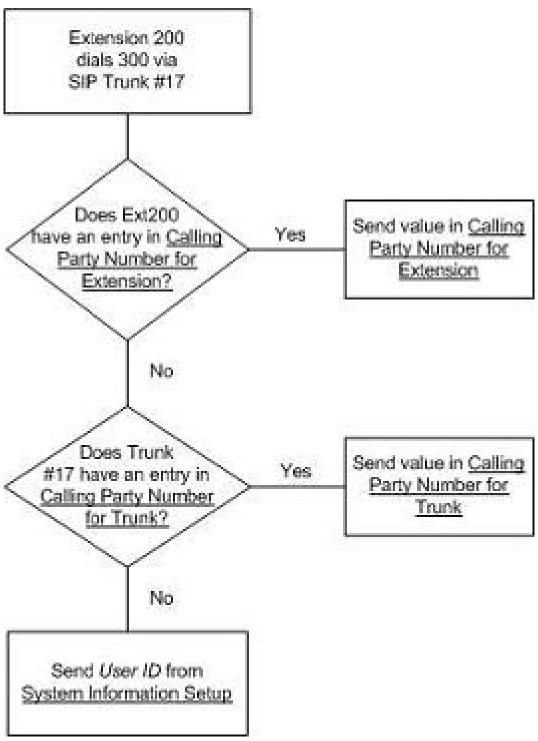
IP Address - The IP address of the remote system

Dial Number - The number that will be dialled to reach the remote system. This does not have to be the full dialled number. (E.g. if this item is set to 2, and the dialled number is 200 then this would be a "match")

Calling Party Number for Trunk
CPN can be configured on a per-trunk or a per-extension basis. This screen sets the CPN for trunks. The table below shows the priority for sending CPN numbers.

Calling Party Number for Extension
CPN can be configured on a per-trunk or a per-extension basis. This screen sets the CPN for extensions.
The table below shows the priority for sending CPN numbers.

Priority for Calling Party Number



H.323 Networking

H.323 is an ITU (International Telecommunication Union) standard for "Packet Based Multimedia Communication Systems". XN120 can use H.323 to connect to another XN120 system, an XN120 or a third party product for Voice over IP. The XN120 uses H.323 Version 3.

The configuration options for H.323 Networking can be found in VoIP - Networking - H.323 Networking

Gatekeeper Setup
H.323 requires a gatekeeper to be configured to manage registration of H.323 endpoints. XN120 has a built-in H.323 gatekeeper and also supports connection to an external gatekeeper. In most cases, the XN120 internal Gatekeeper will be sufficient. External gatekeepers are usually only used in very large H.323 networks.

Gatekeeper Mode - The system can register to a Gatekeeper using a fixed address (Manual) or can use multicasting to obtain a

Gatekeeper address (Automatic).

Gatekeeper IP Address - Enter the IP address of the Gatekeeper. When using H.323 networking within a private network this is normally set to the IP address of the EXIFU-A1. This means that the system registers to its own Gatekeeper.

Preferred Gatekeeper - If the Gatekeeper Mode is set to automatic (Multicast mode) there may be several Gatekeepers discovered. The Preferred Gatekeeper's alias can be entered here and this will be registered to if it is available.

Alias Address - When registering to a Gatekeeper, this alias will be used. This can be thought of as the user ID for the system. The value has to be a number (e.164)

<u>Gateway Prefix</u>
Some Gatekeepers can be configured to allow a "Gateway Prefix". The system can send this prefix along with the registration if required.

Gateway Prefix Registration - Enables and Disabled the prefix option

Gateway Prefix - Enter the prefix (e.164)

<u>Basic Information Setup</u>
This page configures the voice parameters to be used with H.323 Networking. The Codec Setup section for SIP Extensions (above) describes the available CODECS.

Audio Capability Priority - This setting determines the preferred CODEC to be used for H.323 Networking.

G.711 Audio Frame Number - The amount of 10ms periods to encapsulate into one packet. For example 3 = 30ms. This setting has an effect on bandwidth and voice quality so careful consideration should be made before changing this item.

G.711 Voice Activity Detection Mode - VAD detects when there is silence during a call and when this occurs the RTP stream is temporarily stopped and replaced with "comfort noise". This reduces the bandwidth consumption but can sometimes be noticeable at the point of switching between speech and comfort noise. This item enables/disables VAD for G.711

G.711 Type - This should be set to A-Law companding

G.711 Jitter Buffer (min) - If a dynamic Jitter buffer is used; this is the minimum length of the buffer

G.711 Jitter Buffer (typical) - This is the normal Jitter buffer size

G.711 Jitter Buffer (max) - If a dynamic Jitter buffer is used; this is the maximum length of the buffer

- The above settings are also applied to G.729 and G.723 CODECS (with the exception of the companding method).

Jitter Buffer Mode - The Jitter Buffer can use a static setting (based on the Typical setting) or can adjust based on network conditions. The automatic adjustment can be made Immediately or During Silence.

VAD Threshold - This is the threshold for silence detection. The threshold attempts to differentiate between background noise and actual speech. Range is 0 (-20db) to 30 (+10db)

Idle Noise Level - It is possible to set the level for comfort noise generated whilst VAD is in operational. Range 5000 (-5000dbm) to 7000 (-7000dbm).

TX Gain - Sets the transmit gain from 0 (-14dbm) to 28 (+14dbm)

RX Gain - Sets the receive gain from 0 (-14dbm) to 28 (+14dbm)

Bandwidth Limitation Mode - The system can restrict the number of simultaneous calls based on the bandwidth utilisation. This does not take into account the Layer 2 protocol or any signalling requirements (it is just based on the RTP data). This item Enables and Disables this feature.

Band Max - The maximum amount of bandwidth available (in Kbps)

Fax Relay Function - Faxes can be transmitted using a relaying protocol (T.38), rather than as part of the RTP stream. This item switches relaying on or off.

DTMF Relay Mode - When DTMF is sent via H.323 Networking it is possible to use "relaying" so that DTMF is sent as data packets rather than as part of the RTP stream.
There relaying methods available are: RFC2833 or H.245 (these are Internet standards - the receiving host must also support this) and VOIPU (this uses the setting within *VoIP - General Settings - VOIPU Configuration - VOIPU Setup in Section 1*)

Fast Start Mode - Some H.323 equipment supports Fast Start mode (sometimes call Fast Connect). This allows the media stream to be set up prior to the call being connected. The XN120 uses Fast Start by default - if the connected equipment does not support this you should disable it using this item.

Remote Destinations
This screen allows configuration of up to 1000 remote systems. This table is shared with SIP Networking. If an external Gatekeeper has been configured then this table will not be utilised.

System Interconnection - Enables or Disables this entry

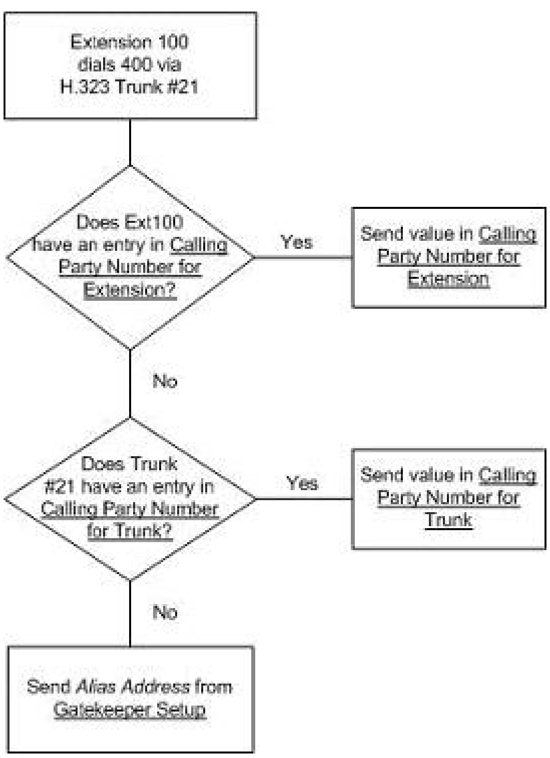
IP Address - The IP address of the remote system

Dial Number - The number that will be dialled to reach the remote system. This does not have to be the full dialled number. (E.g. if this item is set to 2, and the dialled number is 200 then this would be a "match")

Calling Party Number for Trunk
CPN can be configured on a per-trunk or a per-extension basis. This screen sets the CPN for trunks. The table below shows the priority for sending CPN numbers.

Calling Party Number for Extension
CPN can be configured on a per-trunk or a per-extension basis. This screen sets the CPN for extensions. The table below shows the priority for sending CPN numbers.

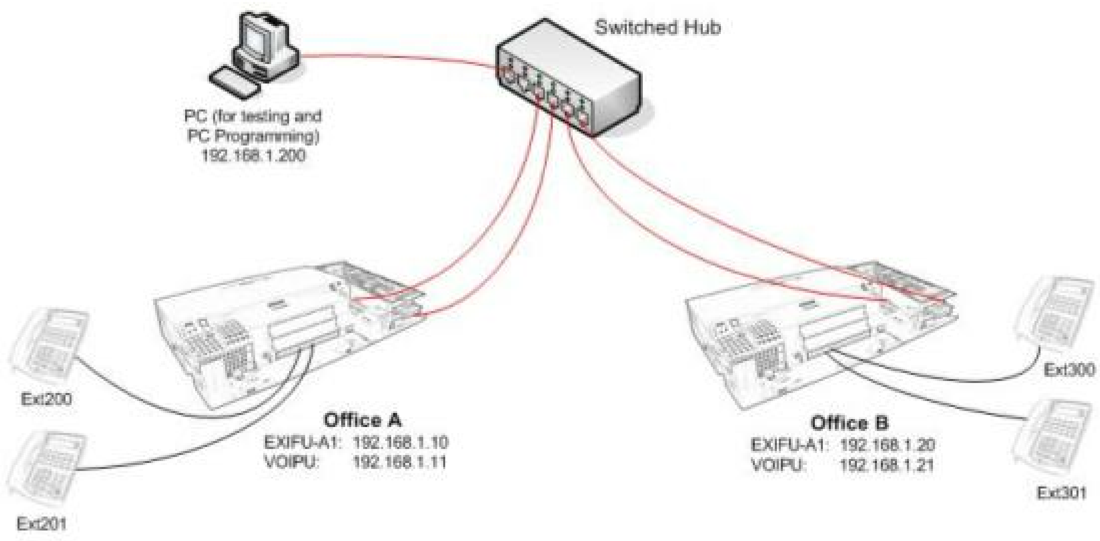
Priority for Calling Party Number



Example Configuration

The diagram below shows four sites networked via IP trunks. Closed Numbering is used so that it is possible to dial extension numbers directly.

The configuration for Office A and B is shown below. This would be sufficient configuration to call between Ext200 and Ext300 via SIP and H.323.



General Settings	**Office A**	**Office B**
IP/EXIFU Network Setup		
Configures the IP address and Subnet mask	for the system (EXIFU)	
IP Address	192.168.1.10	192.168.1.20
Subnet Mask	255.255.255.0	255.255.255.0
VOIPU IP Address Setup		
Configures the IP Address for the VOIPU	card	
IP Address	192.168.1.11	192.168.1.21
Remote Destinations		
Configures the IP dialling/ routing table		
System Interconnection (System 1)	Yes	Yes
IP Address (System 1)	192.168.1.20	192.168.1.10
Dial Number (Syatem 1)	3	2
Extension Setup		
Configures the local extension numbers		
Extension Number	200, 201, etc	300, 301, etc
H.323 Settings		
VOIPU Configuration Setup		
Sets the VOIPU to H.323 mode		
Trunk Type	H.323	
Gatekeeper Setup		

Configures the XN120 to register to its	own internal Gatekeeper	
Gatekeeper Mode	manual	manual
Gatekeeper IP Address	192.168.1.10	192.168.1.20
Alias Address	2	3
SIP Settings		
VOIPU Configuration Setup		
Sets the VOIPU to SIP Mode		
Trunk Type	SIP	SIP
System Information Setup		
Configures the systems SIP identity	information	
Domain Name	sipdomain	sipdomain
Host Name	sysa	sysb
User ID	2	3

Call Routing including DDI/DDO

To configure basic incoming and outgoing routing it is necessary to only configure the following:

- Add all VOIPU trunks into a separate Trunk Group from the normal (PSTN) trunks
- Configure F-Route to send calls for the remote system out via the VOIPU trunk group
- Set the VOIPU Trunks to DDI mode and configure the DDI table to match incoming digits with extension numbers (eg. 200-250)

Additionally, it is possible to configure the VoIP trunks as tie lines.
By doing this the received digits route according to the settings in system numbering and f-route.
This means that if the received digits match the digits set as extensions they will route automatically to the connected extension, similarly it is possible to enable trunk to trunk breakout providing the trunk access is configured using f-route on both sites..

Example 1
System A has extensions 100-110 (IP address 192.168.1.10) and System B has extension 200-250 (IP address 192.168.1.20).
Both sites have SIP trunks set as tie lines.

System A has a numbering plan that routes 2 and 9 to f-route and digit 1 as extension access.
System B has a numbering plan that routes 1 and 9 to f-route and digit 2 as extension access.

System A has f-route configured to route digits 2 and 9 to the IP address of system B.
System B has f-route configured to route digit 1 to the IP address of system A and digit 9 to route to its own external trunks.

This would mean that system A could dial the extensions of system B directly and also break out onto system B's trunks.

The example above can be expanded even further by increasing the entries within the remote destinations on the originating side or f-route on the remote site.
This could enable a selective form of trunk breakout routing.

Example 2
The systems are configured as above but with the addition of system C which has extension 300-310 (IP address 192168.1.30).

System B is local to the 01509 area code and system C is local to the 0161 area code.
System A configures their numbering plan to route digits 2, 3 and 9 to f-route.
The remote destinations of System A is configured to route 901509 to IP address 192.168.1.20 and route 90161 to IP address 192.168.1.30.

Additional routing, via f-route, could also be enabled on the remote site but would require more indepth configuration.

Appendix A: Port List

Appendix A: Port List

If the XN120 is located behind a firewall, it may be necessary to open the following TCP/UDP Ports

Application	Type	Port
CTI Server	TCP	8181
DHCP Server	TCP	67
DIM	TCP	5963
H.323 Extension GRQ	TCP	1718
H.323 Extension RAS	TCP	1719
H.323 Extension Signalling	TCP	1730
H.323 H.245	TCP	5600
H.323 Trunk RAS	TCP	20001
H.323 Trunk Signalling IP Keyphone DRS (Registration)	TCP UDP	1720 3458
IP Phone H.245	TCP	10100
IP Phone Ready/Answer	UDP	4000
IP Phone Signalling	UDP	3456
PCPro (PC Programming Application)	TCP	8000
RTCP	UDP	10021
RTP	UDP	10020
SIP Extension	UDP	5070
SIP Trunk	UDP	5060
SMDR	TCP	4001

WebPro (HTML Based programming)	TCP	80

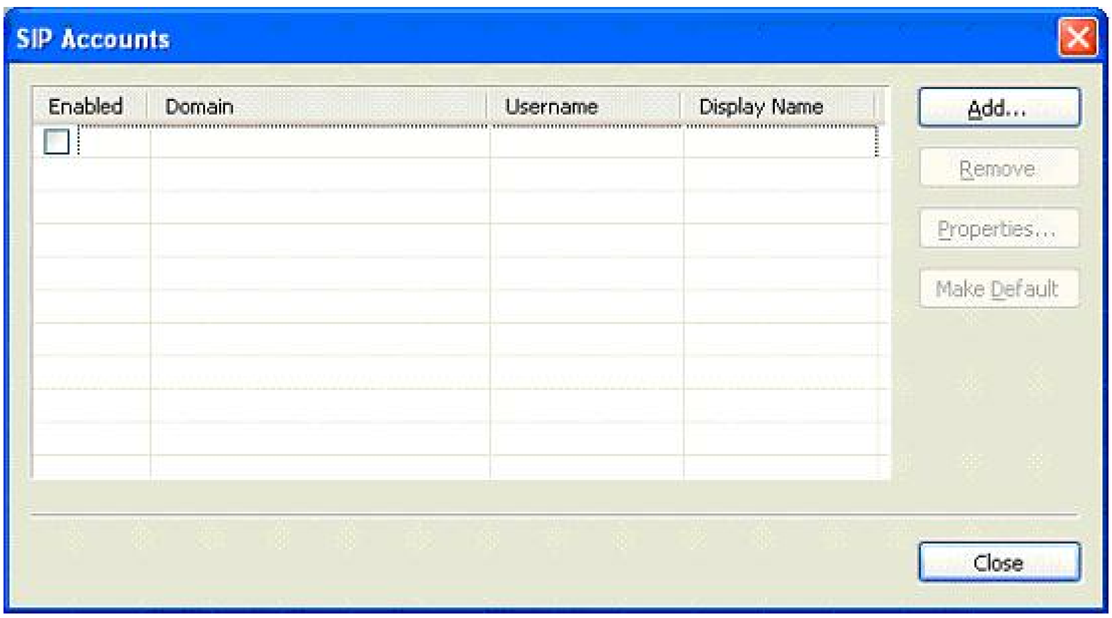
Appendix B: Example IP Phone Configurations

Appendix B : Example IP Phone Configurations

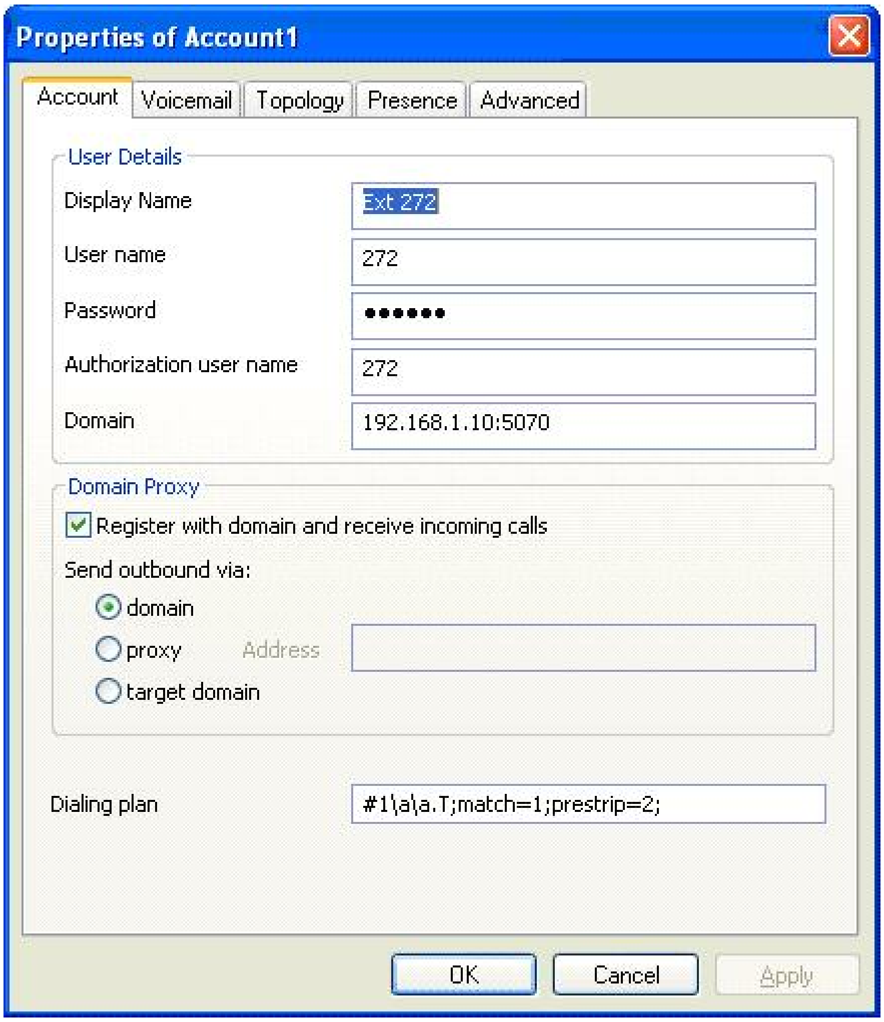
SIP Softphone Configuration (Xten Networks, X-Lite)

X-Lite is a software based SIP phone, available from Xten Networks, Inc. The software is freely available form their website. X-Lite is a reduced feature version of their commercial software, X-Pro.
If G.729 CODEC is required it is necessary to use the X-Pro software.

1) Install the application using the Installation Wizard
2) The SIP Accounts dialog will open. Click the Add button



3) Configure the Account Properties as per the example below:



Display Name: The name that will appear on the Softphone display
User name: The extension number assigned in *VoIP - Extensions - SIP Extensions - Extension Setup*
Password: The extension number assigned in *VoIP - Extensions - SIP Extensions - Extension Setup*
Authorization User name: Same as Username
Domain: The telephone system IP address and port number.
4) Click on OK and Close to exit the configuration dialog
5) The Softphone should register to the telephone system

Please refer to http://www.counterpath.com/ for further information on this product

SIP Telephone Configuration (Cisco 7940)

The Cisco 7940 Series IP Phone is available with various different firmware versions installed.
For this example, the SIP firmware is required.

The Cisco 7940 can be configured using various methods, such as TFTP, web browser or manually using the buttons on the telephone. For this example, the settings are changed using the telephone. Only the basic configuration items are described below. Please refer to the Cisco documentation for further information.	

Step 1 - Enter programming mode
1. Press the settings button
2. Use the scroll buttons or press "9" to go to the "Unlock Config" option
3. Enter the password using the alpha-numeric keys and then press the Accept soft key

Step 2 - Configure Network Information
This step sets up the IP address information for the IP Phone

1. Use the scroll buttons or press "25" to go to the "DHCP Enabled" option
2. Press the No soft key and then the Save soft key
3. Use the scroll buttons or press "5" to go to the "IP Address" option
4. Press the Edit soft key, enter the IP address (192.168.1.51) and then the Accept soft key
5. Use the scroll buttons or press "6" to go to the "Subnet Mask" option
6. Press the Edit soft key, enter the Subnet Mask (255.255.255.0) and then the Accept soft key
7. Use the scroll buttons or press "7" to go to the "TFTP Server" option
8. Press the Edit soft key, enter the default a valid IP address (eg. 192.168.1.1) and then the Accept soft key.
   Note that this TFTP server will not be used for this configuration, but it is required to have a value in this field.

Step 3 - Configure SIP Information
1. Press the Back soft key to return to the main menu
2. Use the scroll buttons or press "4" to go to the "SIP Configuration" option
3. Use the scroll buttons or press "1" to go to the "Line 1 Settings" option
4. Use the scroll buttons or press "1" to go to the "Name" option
5. Press the Edit soft key and enter the extension number (273) then the Accept soft key.
   The extension number is assigned in *VoIP - Extensions - SIP Extensions - Extension setup.*
6. Use the scroll buttons or press "3" to go to the "Authentication Name" option
7. Press the Edit soft key and enter the extension number (273) then the Accept soft key.
   The extension number is assigned in *VoIP - Extensions - SIP Extensions - Extension setup*.
8. Use the scroll buttons or press "4" to go to the "Authentication Password" option
9. Press the Edit soft key and enter the extension password (273) then the Accept soft key.
   The authentication password assigned in *VoIP - Extensions - SIP Extensions - Extension setup*
10. Use the scroll buttons or press "6" to go to the "Proxy Address" option
11. Press the Edit soft key and enter the address of the NTCPU (192.168.1.10) then the Accept soft key
12. Use the scroll buttons or press "7" to go to the "Proxy Port" option
13. Press the Edit soft key and enter the SIP port number (5070) then the Accept soft key. 5070 is the port used for SIP Extensions.
    This item is not usually changed but can be found in *VoIP - Extensions - SIP Extensions - Basic Information Setup*
14. Press the Save, the Back soft keys

The phone should now return to its idle state and the extension number should appear on the display. To call an XN120 extension, lift the handset (or press the Speaker button) and dial the number.

Please refer to http://www.cisco.com/ for further information on this product

H.323 Analogue Terminal Adapter Configuration (Cisco ATA186)

The ATA186 is an analogue gateway that allows connection of an analogue device (eg. POT/Fax) to the IPC 500 via VoIP. The Cisco ATA186 can be supplied with SIP or H.323 firmware.

The basic steps for configuration are to:

1. Allocate an IP address to the ATA via DHCP or assign via the Voice commands (refer to manual)
2. Connect to the IP address from a PC, using a web browser.
3. Make changes to the configuration as per the example below:

Cisco ATA 186 (H323) Configuration

UIPassword:	•	UseTftp:	0
TftpURL:	0	CfgInterval:	3600
EncryptKey:	•	EncryptKeyEx:	00000000000000000000
Dhcp:	0	StaticIP:	192.168.1.99
StaticRoute:	0	StaticNetMask:	255.255.255.0
UID0:	273	PWD0:	•
UID1:	0	PWD1:	•
GkOrProxy:	192.168.1.1	Gateway:	0.0.0.0
UseLoginID:	0	LoginID0:	0
LoginID1:	0	AltGk:	0
AltGkTimeOut:	0	GkTimeToLive:	0
GkId:	.	NATIP:	0.0.0.0
MediaPort:	10020	LBRCodec:	3
AudioMode:	0x00150015	RxCodec:	2
TxCodec:	2	NumTxFrames:	2
CallFeatures:	0xffffffff	PaidFeatures:	0xffffffff
CallerIdMethod:	0x00019e60	FeatureTimer:	0x00000000
FeatureTimer2:	0x0000001e	Polarity:	0x00000000
ConnectMode:	0x00002405	AutMethod:	0x00000000
TimeZone:	1	NTPIP:	0.0.0.0
AltNTPIP:	0.0.0.0	DNS1IP:	0.0.0.0
DNS2IP:	0.0.0.0	TOS:	0x0000a8b8
SigTimer:	0x21418564	OpFlags:	0x00000042
VLANSetting:	0x0000002b	FXSInputLevel:	-1
FXSOutputLevel:	-4	NPrintf	0.0.0.0.0
TraceFlags:	0x00000000	SyslogIP:	0.0.0.0.514
SyslogCtrl:	0x00000000	RingOnOffTime:	2,4,25
IPDialPlan:	1	DialPlan:	*St4-|#St4-|911|1>#t8.r9t2
DialPlanEx:	0	DialTone:	2,31538,30831,1380,174(
BusyTone:	2,30467,28959,1191,151:	ReorderTone:	2,30467,28959,1191,151:
RingBackTone:	2,30831,30467,1943,211	CallWaitTone	1,30831,0,5493,0,0,2400,
AlertTone:	1,30467,0,5970,0,0,480,4	CallCmd:	Af;AH;BS;NA;CS;NA;Df;I

The extension number assigned in *VoIP – Extensions – H.323 Extensions – Extension setup*

The authentication password assigned in *VoIP – Extensions – H.323 Extensions – Extension setup*

The EXIFU IP address as specified in *VoIP – General Settings – IP Addressing – IP / EXIFU Networking setting*

Appendix C : Type of Service Field Values

Appendix C : Type of Service Field Values

Description (*1)	Layer 3 QoS (*2)	ToS Field Bit Pattern (*3)
Best Effort / Class Selector 0	0	00000000
Class Selector 1 (CS1)	8	00100000
Assured Forwarding 11 (AF11)	10	00101000
Assured Forwarding 12 (AF12)	12	00110000
Assured Forwarding 13 (AF13)	14	00111000
Class Selector 2 (CS2)	18	01001000
Assured Forwarding 21 (AF21)	18	01001000
Assured Forwarding 22 (AF22)	20	01010000
Assured Forwarding 23 (AF23)	22	01011000
Class Selector 3	24	01100000
Assured Forwarding 31 (AF31)	26	01101000
Assured Forwarding 32 (AF32)	28	01110000
Assured Forwarding 33 (AF33)	30	01111000
Class Selector 4 (CS4)	32	10000000
Assured Forwarding 41 (AF41)	34	10001000
Assured Forwarding 42 (AF42)	36	10010000
Class Selector 5 (CS5)	40	10100000
Expedited Forwarding (EF)	46	10111000

Class Selector 6 (CS6)	48	11000000
Class Selector 7 (CS7)	56	11100000

Notes:

1. The description field lists commonly used "code-points". Other values can be implemented but it is unusual for a Network Maintainer to use anything other than the code-points listed.
2. Layer 3 QoS. These are the values that would be entered into the Priority (Diffserv) field on the VoIP - QoS Settings - Layer 3 QoS page
3. This column is provided for information only. It shows the actual bit pattern for the ToS byte. Note that the last to bits are not used for QoS.

Revision History

For On Line manuals.
Release dates of the manuals may not be aligned with the release date of the associated system software.

Update Trunk to Trunk Forwarding in Feature Manual to include Step Transfer operation.

PRG 14-01-26
Automatic trunk to trunk transfer mode (normal or step transfer)
Add to 1529 Trunk Basic Data Setup

Update 1522 Automatic trunk to trunk transfer target to include the operation when step transfer is enabled.

PRG 24-02-11
No answer timer for step transfer
Add to 1369 Hold and Transfer
Add to 2013 Trunk to Trunk Options
Add to 2098 Timers

PRG 24-02-12
No answer timer for automatic trunk to trunk transfer
Add to 1369 Hold and Transfer
Add to 2013 Trunk to Trunk Options
Add to 2098 Timers

Flexible Routing by Caller ID / Caller ID refuse setup to include in the Feature Manual.
Note that PRIVATE call options will not be included.

PRG 14-01-27
Caller ID Refuse option
Add to 1529 Trunk Basic Data Setup

PRG 40-10-06
VRS message for Pivate Call Refuse
This item will not be included in Easy Edit

PRG 22-18
Private Call Setup for Caller ID routing / refuse
This item will not be included in Easy Edit

PRG 40-10-07
VRS message for Caller ID refuse
Add to 1023 VRS Options

PRG 22-16
labelled as Private but is used for normal CLIP rejection. This is renamed in Easy Edit to Caller ID Refuse Target Area'.
This option sets the Abbreviated Dial Group to be used for Caller ID Reject.
3230 New help page required called 'Reject call by Caller ID'

PRG 13-04
Abbreviated Dial name and number
Add 13-04-03 Transfer Mode
Add 13-04-04 Destination Number
Add 13-04-05 Incoming Ring Pattern to 1552 help page

PRG 11-10-33
PRG 11-10-34
Caller ID Edit Service code and Caller ID Refuse sevice code
Already included in xml, no help required.

PRG11-10-32 Private Call Refuse
Not included in Easy Edit

PRG 15-07
Function key code 86, Private Call Refuse function key
Will not be included in Easy Edit help

PRG 15-07
Function key code 87, Caller ID Refuse
Already included in xml, no help required.

DDI Time Mode Operation
Add to Features manual

PRG22-02
Add type 8 DDI Time Mode option to Help 1530/2530

2530 is currently not included in the French Easy Edit, this has been included (FR1280).

PRG 22-17
DDI Time Mode Table
3232-3235 New help page required, this will be added to the DDI section of Easy Edit

PRG 15-07 undefine
DDI Time Mode Change
No help required for this page

PRG 20-07-26
COS option for DDI Time Mode Change
No help required for this page

PRG 11-10-35
Service code for DDI Time Mode change
no help required for this page

Hotel
Help pages for Hotel Easy Edit need adding to Xn120

3050 Licence Information to write for XN120

2092 to 2096 are the same as the Aspire

2097 edited for CTA change to EXIFU

Include the Hotel License Input information for the XN120 in the feature section

ACD
Remove any association with ACD features within the XN120 Features Manual:
Programmable Function Keys
Ring Groups
Headset
Swap Extension
DISA - Remote Feature setup
Music on hold
Toll Restiction
Preface in Hardware manual

ISDN Options
PRG 10-03-17, 18 and 19
option 17 (ISDN ringback tone)
option 18 Type of Number. Outgoing SETUP message contains the ToN in the Calling Party number element.
option 19 Numbering Plan Identification. Outgoing SETUP message contains the Num Plan in the Calling Party number element.

Add to 1361 BRI setup
Add to 2361 BRI setup
Add to 2000 PRI Setup

Area Number for NGT Information Setup (selects the tones for the IP keyphone)
PRG 84-03-06
No change to help needed.
Needs adding to xml 1077

Caller ID for Forwarded calls to an SLT
PRG 15-03-14
Add to help for 2472 / 1472

Disconnect without dialling after hooking hold
PRG 15-03-15
This option will not be added to Easy Edit.

Trunk call dial sending Forced Time
PRG 20-03-07
This option will not be added to Easy Edit.

SMDR output of caller ID name
PRG 35-02-17
PRG35-02-18
This option will not be included in Easy Edit as it requires the North American format which we do not include in Easy Edit

Ring tone selection for Tel2 keyphone
PRG15-02-02
PRG 15-02-03
Add to Help for 1006

Virtual extension ring patterns for Tel2 keyphones
PRG 15-08-01
Add to Help for 1507

Ring tone Selection for Trunk
PRG 22-03
Add to Help for 1531

Loop/Ground start
PRG14-02-14 (XN120 only)
This option is not available in the EU systems, it will not be added to Easy Edit

Trunk to Trunk limitation
PRG 14-02-13
This is not used so will not be included in Easy Edit

Busy tone detection during call
PRG 14-02-18
This is not used in EU so will not be included in Easy Edit

MOH from the DSPDBU card
Add to Music on Hold in Feature manual

PRG 10-04-01
new option added to select the DSPDB card
PRG 10-04-02
now allows the entry of 0-48 to allow the DSPDB message to be selected.
Add to Easy Edit 2941 for Aspire/
Add to Easy Edit 1441 for XN120

Operation of the Service Tone option in 10-04-01 on XN120 for beep tones for MOH included. (Esc1155)
Add to 1441 Music On Hold Setup

Toll Restriction in Credit
Add to Features manual (Tech0549)

Correct CLI for SLT ports (fixed ring pattern) esc0843
Already included in Ring pattern (2506/1506) and SLT section of the Feature manual, but will be added to SLT Basic Setup 2472/1472 also.

Add power points for EXIFU and AspireMail DMS to Load Factor in Aspire Hardware manual (Tech1072)

SMDR enhancements
PRG 35-02-19
Display from first or last digit
Add a new option to 1027 Display first or last digit.

PRG 35-02-16
Trunk name, Received DDI number or both
add the option for 'both' to 1027 Trunk Name/Received Number option.

Remote Conference (conference bridge)
Add to the Feature manual
Add as a related feature to Conference feature

PRG 20-13-46 COS option for remote conference
3237 help page

PRG 14-01-02/03/04/05 Trunk Gain
No changes required, already included in help

PRG 20-01-03 DSP resource
This option is currently not included in Easy Edit, the default is set to conference which suits this feature.

PRG 20-34 Remote Conference Group Setup
3236 new help page called Remote Conference.

PRG 11-19 Pilot Codes
3236 new help page called Remote conference Pilot Code.

PRG 47-01-01 PVMU mode for DSPDB
This is an option within Intramail

EU Character Set
Add a single page to Easy Edit that.
Add a reference to the Extension Basic Setup page (1326) to the character set page.

Call Deflection
Add to the Feature Manual 'Call Deflection'

PRG 14-15 ISDN call forward method
3238 Add a new page called 'ISDN Call Forward Method' in the Cards/ISDN menu.

PRG 13-04-06 Call Deflection on/off
Add to 1552 Abbreviated Dial Number and Name.
This is currently called CR/PR Feature which is not correct, this should be renamed to 'ISDN Call Forward Method'

QSIG
Add new section to the Features Manual.

Timer Class of Service
PRG 20-29 Timer class for extensions
3241 new hwlp page called Timer Class for Extensions.

PRG 20-30 Timer class for trunks
3240 new help page called Timer Class for Trunks.

PRG 20-31 Timers for Class of Service
3242 new help page called Timer Class of Service.

ACD Overflow to DSPDB Voice Mail box
Add to ACD section of Feature manual

PRG 41-08-02 Overflow destination
Now allows the entry of 69 which specifies overflow to the DSPDB VM.
Add to 2067 ACD Overflow and Announcements

PRG 41-08-07 new option to define the DSPDB mail box number 0-300.
Add to 2067 ACD Overflow and Announcements

ACD Queue Announcement from IntraMail
PRG 41-08-03 Overfow Options
Two new options added
03 = CVM, this can be either a local VM or Centralised VM, which we will only indicate is the IntraMail.
04 = Flexible, this uses PRG 41-08-08 & 09 to specify a different source for announcements 1 and 2.
Add to 2067 ACD Overflow and Announcements

PRG 41-08-03 Overflow options
02 = VMI In Skin, this is currently shown as not available, this in now available
Add to 2067 ACD Overflow and Announcements

PRG 41-19 ACD VM box assignment for queue announcement
This is currently included in xml 2071 but the help indicates that it is not available.
The xml page 2071 renamed to VM Box Setup for Queue Announcements

Flexible Queue message source
PRG 41-08-03 Overfow Options
New option added
04 = Flexible, this uses PRG 41-08-08 & 09 to specify a different source for announcements 1 and 2.
Add to xml 2067 ACD Overflow and Announcements

PRG 41-08-08 new option to specify the 1st queue announcement (ACI, VRS only) VM and CVM are not to be included for the French and German Aspire
Add to xml 2067 ACD Overflow and Announcements

PRG 41-08-09 new option to specify the 2nd queue announcement (ACI, VRS only), VM and CVM are not to be included for the French and German Aspire
Add to xml 2067 ACD Overflow and Announcements

VRS Dialling Options for Extension Call from Auto Attendant (DUD/DISA)
Add to feature manual ‘DUD/DISA Extension Call’

PRG 11-11-59 Service code for busy
PRG 11-11-60 Service code for no answer
Add to 1410 3 Digit Codes
No help required

PRG 15-07 Code 94 Call Attendant key
Add to Programmable Function Keys/template
No help required

PRG 15-01-08
PRG 15-01-09 Options for busy and no answer for each extension, assign the DSPDB message to be used for attendant message.
3243 New help page called ‘Extension Call - Extension Message’
Rename the options to:

Extension Call Busy Message
Extension Call No Answer Message

PRG 40-10-08
PRG 40-10-09 Options for busy and no answer for the system, assign the DSPDB message to be used for attendant message.
3244 New help page called 'Extension Call - System Message
Rename the options to:
Extension Call Busy Message
Extension Call No Answer Message

PRG25-06-01 three new options for Extension Call
13031 Auto Attendant Single Digit Options edited to include the new options.
The ATT Msg columns need to allow the entry of the new options codes, the 'help' in the top row changed to include the new options.
New option codes are 101, 104 and 105

SMDR FWD off premise shows extn that set the fwd in SMDR
SMDR 35-02-20
Add to 1027 SMDR Output Options

DSPDB VM Single digit options
Add to feature section
Not required for Philips

PRG 40-01-09 VRS 1 Digit Access Setup
Add to 1538 Voice Mail Basic Setup

PRG 40-01-07 Escape from mail box
Add to 1538 Voice Mail Basic Setup
This should be included for all systems

PRG 40-12 New PRG for 1 digit translations
3239 new help page called 'DSPDB VM 1 Digit Translation'

Edit help for 1544 to indicate that 50# is not fixed if VRS 1 Digit Access Setup is enabled.

Update PRG cross reference for any new PRG numbers added

Virtual Extension Ehanced Setup
PRG 15-18
3247 new help page called 'Virtual Extension Ehanced Setup' in the Virtual Extension menu.
Rename the option of 15-18-01 from 'Land on the key' to 'Remain Busy'.
There is also a comment added to 1492 'Virtual Extension Options' related to this PRG.

User Pro
3245 new help page called Extension Password
3246 new help page called UA Mode Commands

Edit Caller ID & Redial
Add to Features manual

No pages added to easy edit

Flexible Transfer – Virtual Loopback
Add to features manual

PRG10-42
Virtual loopback setup
Added to Virtual Loopback Setup

Call Restriction Between Department groups

PRG16-04-01
Call restriction between department group
Added to 1208 Department group – departmental call restriction

SIP Station Enhancement Peer To Peer
Add to IP Manual

PRG 10-26-03
SIP Peer to Peer mode
Added to 1063 SIP Code Setup

SIP Message Waiting Indication
Add to IP Manual

No change to help needed.

Call Park Searching
Add to features manual

PRG 20-11-27
Call Park Automatically search
Added to 1502 Class of service options.

Hot Keypad
Added to features manual

PRG 20-08-20
Hot Keypad enable.
Added to 1502 Class of service options.

Mobile Extension
Added to features manual

PRG 15-22-01
Mobile Extension target setup
Added to 4010 Mobile Extension setup

PRG 15-22-02
Connect mode
Added to 4010 Mobile Extension Setup

PRG 15-22-03
Trunk Access Code
Added to 4010 Mobile Extension Setup
This option will not be included in Easy Edit as it requires the North American format which we do not include in Easy Edit

OPAC Key Enhancement
Added to features manual

PRG 20-39
Added to 1381 OPAC Key Operation
Added to 1382 OPAC Key Operation

PRG 20-40
Added to 1381 OPAC Key Operation
Added to 1382 OPAC Key Operation

PRG 20-41
Added to 1381 OPAC Key Operation
Added to 1382 OPAC Key Operation

Doorphone External Call Forward
Added to Features Manual

Added to Door Box section

Incoming Ring By VRS Channel
Added to Features Manual

PRG 13-04-07
Added to 1552 Abbreviated Dial Name and Number

PRG 15-02-55
Added to 1006 Keyphone Options