### PLEXTEL IP-PBX

## USER MANUAL

Version 2.1 (Release 03/03/2011)

#### CONTENT

Chapter1: Extensions Management	
1.1 Group Manager	4
1.2 Add Extensions	5
1.3 Add IAX Extensions	12
1.4 Add Analog Extensions	15
1.5 Add Follow Me Extensions	16
Chapter2: External Line Configuration	
2.1 Voice Interface Hardware	19
2.2 PSTN Trunk Setting -	22
2.3 SIP Trunk Setting	30
2.4 Gateway	32
2.4.1 GSM Gateway Integration	32
2.4.2 Setting Gateway to work with Soundwin	38
2.5 Site to Site Setup	41
Chapter3: System Features	
3.1 Conference	43
3.2 Feature Key / Call Parking	47
3.3 Paging	53
3.4 Virtual Number	56
Chapter4: Incoming Line Configuration	
4.1 Schedule	59
4.2 IVR (Interactive Voice Response)	61
4.3 Incoming Call Setting	73
Oberster 5: Outland Management	
Chapter5: System Management	70
5.1 Manage System Administrator	76
5.2 LAN Network Setup	79
5.3 WAN Network Setting	79
5.4 HA Setup	80
5.5 FIREWALL/NAT Setup	80
5.6 DNS Setup	81
5.7 Dynamic DNS	82
5.8 DHCP Server	82
5.9 Backup & Restore	83
5.10 License Management	83

Chapter:

2

5.11 Call Details Record	84
5.12 Auto Provision	89
5.13 SSH Terminal	90
5.14 Asterisk CLI	92
5.15 Screen pop-up Account	94
Chapter6: System Monitoring	
6.1 Show Status	95
6.2 Show Graph	96
6.3 Service Status	97
6.4 List DID Number	97
6.5 Phone's Connection Status	98
6.6 Phone Status Panel	99
6.7 Active Call Status	99
Chapter7: System's Sound Files	
7.1 Create Voice	100
7.2 Call Record Files	105
7.3 Music On Hold	106
7.4 Backup Voice Record	107
7.5 External Storage	110
Chapter8: Call Center Technology	
8.1 Queue and Agent Management	113
8.2 Customer Satisfaction Report	118
Chapter9: FAX System	
9.1 Add virtual fax	120
Chapter10: CallerID Routing & Call Back	
10.1 CallerID Routing	128
10.2 Call Back	130
10.3 External Database Connector	135
Chapter11: Help	136

#### **Chapter 1: Extensions Management**

**1.1Group Manager:** See "Group" Menu on the left hand size menu. This page is used to classifiy group of extension or users in the system. It will be useful to define priority and billing term for any company or organization. By default, the group that has been created by the system is called "Default", and please remind that this group can not be deleted.

splay <b>1</b> - :	3 From 3				
GID	Company	Department	Position	Music on hold	Edit
0	Default	Default	Default	Default	Ż
1	poise	eng	test	eng	Ø.

If we want to add a new group, we select "Add New Group", the following screen will be shown:

	Manager Add New Group ]			
GID	Company	Department	Position	Music on hold
3 🔻 🛛				Default 🔻 Add Group
Cancel	1	I	I	

"Music on Hold" is the name of music group, which it will be played while calling every extension in this group, and different group can be assigned with different music . When we input all values, we can press "Add Group" button, then the system will show like the following screen:

#### **Group Manager**

[ View ] [ Add New Group ]

GID	Company	Department	Position	Music on hold	Edit
0	Default	Default	Default	Default	I I
1	poise	eng	test	eng	Ż
2	poise	sale	sale	Default	1 De

**1.2 Add Extensions :** See "Extensions  $\rightarrow$  Add Extensions" Menu, this menu is used to add extensions into the system. It will be classified to be two types:

1.Add Sip Extension: Adding one by one like 1001, 5002, 7065, .....

**2.Add Multiple Sip:** Adding 2 and more extensions in range like 1001-1020, 4010-4050, .....

When we describe 2 types of adding, we can observe that all parameters look same except first parameter, "Extension".

Add SIP Extension Phone Setting:		8
Group Name	Default:Default:Default 👻	
Phone Number		
Caller ID		
Password		

Phone Number : This blank is used to assign phone number like 1001,3001, 6441,.....

**Caller ID**: This value is used to assign caller name or number which this value will be shown when we use this phone call to another phone in Plextel IP-PBX system. We can assign same as phone number like "1001", "5005", or assign with name like "John", "Jane", or assign both like " John 1001"

**Auto Provisioning**: This menu is used to setup ip-phone by pass these parameters to specific ip-phone without direct setup on that ip-phone.

ENABLE		
Phone Type	NONE -	
Phone MAC Address		
Allow Firmware Upgrade		
Custom Command		*

**ENABLE**: Click for Enable this function.

**Phone Type**: Select model of ip-phone from list menu, SNOM 300, SNOM 320, SNOM 360, SNOM370, SNOM820, SNOM821, SNOM870, Yealink SIP-T12P, Yealink SIP-T18P, Yealink SIP-T20P, Yealink SIP-T26P, and Yealink SIP-T28P

Phone MAC Address: Input MAC address of this ip-phone.

**Allow Firmware Upgrade**: If we would like to allow firmware upgrade automatically by the system, we have to click here.

**Custom Command**: This blank is used to add some further command. (only advanced user or developer)All parameters left same as "Add Multiple Sip" like this example

#### **Extension Manager**

[View Extensions] [Add SIP Extension] [Add Multiple SIP] [Add IAX Extension] [Add Analog Extension] [View All, Add Follow-Me Extension]

Extensions Range	through			
Group Name	Default:Default:Default -			
Password				
Enable BLF	no 🔻			
Codec	G.711u     G.711a     GSM     G.729     G.723.1     ILBO     Speex     Ipc10     adpcm     G.726			
Video Codec	Гн.261 Гн.263 Гн.263р Гн.264			
dtmf mode	rfc2833 🔻			
Extension Monitor	yes -			
Concurrent Call Support	12			
Enable REINVITE	default 👻			
NAT Support	default 👻			
Support T.38 FAX	default 👻			
SIP Additional Setting				
Dial Option:				
Dial Option:	<ul> <li>✓ Allow calling user to Transfer (T)</li> <li>✓ Allow called user to Transfer (t)</li> <li>✓ Allow calling user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (w)</li> <li>✓ Generate a ringing tone (r)</li></ul>			
	<ul> <li>✓ Allow called user to Transfer (t)</li> <li>✓ Allow calling user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (w)</li> </ul>			
Dial Option: Call Features: Ring Timeout	<ul> <li>✓ Allow called user to Transfer (t)</li> <li>✓ Allow calling user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (w)</li> </ul>			
Call Features:	<ul> <li>✓ Allow called user to Transfer (t)</li> <li>✓ Allow calling user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (w)</li> <li>✓ Generate a ringing tone (r) </li> </ul>			
<b>Call Features:</b> Ring Timeout Pickup Call from	<ul> <li>✓ Allow called user to Transfer (t)</li> <li>✓ Allow calling user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (W)</li> <li>✓ Generate a ringing tone (r) </li> <li>✓ Provide Music on Hold (m)</li> </ul>			
<b>Call Features:</b> Ring Timeout Pickup Call from	<ul> <li>✓ Allow called user to Transfer (t)</li> <li>✓ Allow calling user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (W)</li> <li>✓ Generate a ringing tone (r) </li> <li>✓ Provide Music on Hold (m)</li> </ul>			
<b>Call Features:</b> Ring Timeout Pickup Call from Record Incoming Calls Record Outgoing Calls	<ul> <li>✓ Allow called user to Transfer (t)</li> <li>✓ Allow calling user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (w)</li> <li>✓ Generate a ringing tone (r) </li> <li>✓ Provide Music on Hold (m)</li> </ul> 30  30  Default:Default:Default poise:eng:test poise:sale:sale ✓ Yes  ✓ No			
Call Features: Ring Timeout Pickup Call from Record Incoming Calls Record Outgoing Calls Support Intercom	<ul> <li>✓ Allow called user to Transfer (t)</li> <li>✓ Allow calling user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (w)</li> <li>✓ Generate a ringing tone (r) </li> <li>✓ Provide Music on Hold (m)</li> </ul> 30 ▼ Default:Default:Default     poise:eng:test     poise:sale:sale ✓ Yes  ✓ No ✓ Yes  ✓ Yes  ✓ Yes  ✓ Yes  ✓ Yes  ✓ No ✓ Yes  ✓ Yes  ✓ Yes  ✓ No ✓ Yes  ✓ Yes </td			
Call Features: Ring Timeout	<ul> <li>✓ Allow called user to Transfer (t)</li> <li>✓ Allow calling user \"One Touch Record\" (W)</li> <li>✓ Allow called user \"One Touch Record\" (w)</li> <li>✓ Generate a ringing tone (r) </li> <li>✓ Provide Music on Hold (m)</li> </ul> 30  30  Default:Default:Default poise:eng:test poise:sale:sale ✓ Yes  ✓ No			

Mailbox:					
User Email Address :					
Enable Message Center User Login : Voice Mailbox :	Yes 👻				
Enable Voicemail Box :	Enabled	-			
Send Voice Message To Email :	C Yes 🖬	lo			
Voice Mailbox Description :					
/oicemail Password (fix) :					
Voice Mailbox Size (messages) :					
Fax Mailbox			_		
Enable Fax Mailbox :		No	•		
Send Fax Message Notification To En	nail :	Yes	•		
Attached Fax File To Notification Ema	ail :	Yes	•		
Send Voicemail Notification For Incom	ing Fax :	Yes	•		
Fax Mailbox Size :		20		+	
Contact Information					
Picture :				Browse	
Business Phone :					
Home Phone* :					
Business Fax :					
Mobile :	-		-		
			_		
Address :					
Note :					

**Extension range**: This value is used to add extension number in range, sequencially, like 1001-1009, 2001-2099.

**Group Name**: This value is used to assign group name, which assigned group is shown here has been assigned from "Group Manager Menu". We can use a mouse click on drop down list then all assigned group will be shown like this example

Group Name	Default:Default:Default	•
Password	Default:Default:Default	
Enable BLF	poise:eng:test poise:sale:sale	

By automatically, if we don't select any group, the system will assign to be default group like "Default: Default: Default". If we would like to assign other group, we can click mouse on that specific group.

**Password**: This value is used to assign ip-phone password, which this password is used to verify when ip-phone make registration to the system.

**Enable BLF**: Click for enable "Busy Lamp Field", which this function will be available as shown in list menu like SNOM 300, SNOM 320, SNOM 360, SNOM370, SNOM820, SNOM821, SNOM870, Yealink SIP-T12P, Yealink SIP-T18P, Yealink SIP-T20P, Yealink SIP-T22P, Yealink SIP-T26P, and Yealink SIP-T28P. This function is used to monitoring status of another phone, if monitored phone show busy status, we can see red light at the button which we assign to monitor.

**Codec:** This menu allow us to select codec type, if we don't select anyone, the system will assign to be G711u and G711a, default codec used in sip protocol.

**Video Codec:** This menu allow us to select video codec, but anyway we should select this by reference from video phone type.

**DTMF Mode:** This value is the frequency used on this ip-phone panel, by default, in Thailand we use RFC2833.

**Extension Monitor:** If we select "yes", we can monitor ip-phone 's registry status at menu Status -> Phone monitor like this example.

No	Number	Name (Callerid)	IPaddress	Natsupport	VideoSupport	Status	Туре
1	192.168.0.124		192.168.0.214	eno	eno	Unmonitored	SIP
2	4000		-none-	eno	eno	Unmonitored	SIP
3	sipsite_wwwww		-none-	eno	eno	UNKNOWN	SIP
4	trunk%%sip%%asdf		0.0.83.97	yes	eno	Unmonitored	SIP
5	3004	3004	-none-	eno	eno	UNKNOWN	SIP
6	3003	3003	-none-	eno	eno	UNKNOWN	SIP
7	3002	3002	-none-	eno	🔵 no 👝	UNKNOWN	SIP
8	3001	3001	192.168.0.190	eno	eno	OK (48 ms)	SIP
9	3000	3000	-none-	eno	🔴 no 🤚	UNKNOWN	SIP

The green bar show 3001 is registered.

**Concurrent Call Support**: This value is number of lines that we allow to call this phone in same time.

**Enable REINVITE**: If we select "yes", ip-phone can communicate each other without server, which it can reduce server load, but however we can not send signal like transferring anymore.

Nat Support: This value will be set as "yes" when this phone register from different network.

**Support T.38 Fax**: This value will be set as "yes" when we set this extension to work with T.38 fax system.

**Sip Additional Setting**: This value is reserved for advanced user or developer who want to add more qualification to this extension.

**Dial Option**: This value is used to set option related with dialing like line transfer, ringing option, and if we don't select this, the system will set as proper default value.

**Ring Timeout** : Time in second unit that we assign this phone ring when there is incoming line, when timeout the system will perform assigned fuction like voicemail or follow me, depends on our setting.

**Pickup Call From**: List menu here is all assigned group name. we can select group if we want to allow call pickup all phones in that group from this extension. Anyway this menu allow us to select more than one group.

Pickup Call from	Default:Default:Default Poise:IT:Programmer Poise:IT:Engineer

**Record Incoming Call**: If we select "yes", the system will record all conversation when there is incoming line to this phone. To see recorded file, go to menu Sound  $\rightarrow$  Call Record File

**Record Outgoing Call**: If we select "yes", the system will record all conversation when there is outgoing line from this phone. To see recorded file, go to menu Sound  $\rightarrow$  Call Record File

**Default Language**: Select the default language, which it will be displayed in voicemail, conference and any provided system.

**Allow Roaming Station Feature**: Select "yes" to allow registration from other phone. (To use this feature please see menu "Key Features".

**Support Intercom:** Intercom is the two way communication between 2 ip-phone by press only one programmed button, so only listed ip-phone here can use this function.

Mailbox: If select "Enable", this phone will have own voice mailbox.

Yes 🔻	
Enabled 💌	
C Yes 🖲 No	
	Enabled 💌

**User Email Address**: Input email address for this user to link with Plextel 's message center system.

**Enable Message Center User:** Click here to allow this extension (user) has it own personal message center.

Enable Voicemail Box: Click here to enable personal voicemail box.

**Send Voice Message to Email**: If select "yes", mean assign system to forward voicemail message to specific mailbox.

L Chapter:

Voice Mailbox Description: Input the description of this voicemailbox.

Voicemail Password(fix): Input password to access this mailbox.

Voice Mailbox Size (Messages): Input number of mailbox.

Contact Information: Input information for this user.

Picture :	Browse	
Business Phone :		
Home Phone* :		
Business Fax :		
Mobile :		
Address :		*
		-
Note:		*
		-

12

Chapter

#### **1.3 Add IAX Extensions**

IAX abbreviate from Inter-Asterisk eXchange, which has been developed for used in asterisk system only, and can not communicate with other techlogy except devices which have been designed to work with IAX. The main purpose of this protocol is elimination of NAT problem founded in SIP protocol.

Normally, when we implement IP-PBX in organization, security issue is the most important issue to concentrate, so every device have to pass firewall. When we use ip-phone with sip protocol we always see "Register Fail" message, that means sip ip-phone can not work properly with some firewall, even the admin said every ports have been opened. To avoid this probem, asterisk developer has designed IAX protocol, which run on only one port, 4569, and can save 50% of bandwidth, if compare with sip.

To add IAX extension, go to menu Extensions  $\rightarrow$  Add IAX Extension , the following screen will be shown.

#### **Extension Manager**

[View Extensions X] [Add SIP Extension X] [Add Multiple SIP X] [Add IAX Extension X] [Add Analog Extension X] [View All X, Add Follow-Me Extension X]

Add IAX Extension Phone Setting :				
Group Name	Default:Default:	Default 🔻		
Phone Number				
Caller ID				
Password	·			
	no 🔻			
Phone IP-Address	Selvere 1			
Codec	dynamic 👻			
Codec	G.711u G.7			
Video Codec	□ H.261 □ H.26			
Phone Monitor	yes 👻			
IAX Additional Setting	100			
Dial Option :	0			*
	Allow calling us	er to Transfe	r (T) X	
	Allow called use	er to Transfer	(t) X	
	Allow calling use	er \"One Tou	ch Record\" (W) X	
	Allow called use	Store and the store of the store	Contraction of the second second	
C-II Factoria	Generate a ring	ging tone (r)	🕻 🙆 Provide Musi	c on Hold (m) X
Call Features : Ring Timeout	20			
Pickup Call from	30 -			
	Default:Default: poise:eng:test poise:sale:sale	Detault		
Record Incoming Calls	C Yes • No			
Record Outgoing Calls	C Yes  No			
Default Language	Thai 🔻			
Allow Roaming Station	( Internet and Int			
Feature	No 👻			
Mailbox :				
User Email Address :				
Enable Message Center Us	er Login Yes 🔻			
Voice Mailbox :				
Enable Voicemail Box :	Enabled			
Send Voice Message To Em	ail : C Yes 🕫 M	No		
Voice Mailbox Description	:			1
Voicemail Password (fix)				
Voice Mailbox Size (messag	es) :			
Fax Mailbox	v es - 22 - 21			
Enable Fax Mailbox :		No 🔻		
Send Fax Message Notifica	ation To Email :	Yes 👻		
Attached Fax File To Notifi		Yes 👻		
Send Voicemail Notification		Yes 👻		
Fax Mailbox Size :				
Tax Mailbox Size ;		20	- 1993 ( )	

Chapter:

1

		1 /
Contact Information X Picture X :	Browse	
Business Phone X :		•
Home Phone X*:		
Business Fax X :		
Mobile X :		
Address X :	*	
Note X :	÷	
APPLY Cancel		

The previous screen present every parameters look like "Add Sip Extension", actually everything same, different only its protocol. When we complete adding, press Add Ext, then the following screen will be shown:

#### **Extension Manager**

isplay 20	/ <b>1 - 21</b> F	rom	21 •		E	kten	sion	Imp	ort				Brow	se	Data Import	Extensi	on Expor	t
	umber	Туре	Group	Pickup Call From	BLF	I/R	0/R	INT	Lang	AP	Phone Type	DIDF	ollowme	Roaming	User Email	Vmail	FAX mailbox	,Edit
	000 : on nousand	sip	Default:Default:Default		×	×	~	x	Thai	x	none		-	x	nicrora@hotmail.com		-	Ż
10	001 : two	sip	Default:Default:Default		1	1	1	x	Thai	1	snom360		-	x		1	1	Ż
	002 : 002	sip	Default:Default:Default		x	1	1	x	Thai	x	none		-	x		×	~	Ż
	003: 003	sip	Default:Default:Default		x	<	•	x	Thai	x	none		-	x		×.	×	Ż
	004: 004	sip	Default:Default:Default		x	×	×	x	Thai	x	none		-	x		×.	-	Ż
	005: 005	sip	Default:Default:Default	Default:Default:Default	x	×	Ľ	x	Thai	x	none		-	x		×	×.	Ż
	006 : 006	sip	Default:Default:Default		x	×	Ľ	x	Thai	x	none		-	x		×.	-	2
	007: 007	sip	Default:Default:Default	Default:Default:Default	x	×	×	x	Thai	x	none		-	x		×	×.	2
	)08 : )08	sip	Default:Default:Default		x	×	×	x	Thai	x	none		-	x		×	-	2
	009: 009	sip	Default:Default:Default		x	×	×	x	Thai	V	yealinkt28		-	x		×.	×.	2
	010: 010	sip	Default:Default:Default		x	×	×	×	Thai	V	yealinkt26		-	x		~	-	2
	119: 119	sip	Default:Default:Default		x	×	1	x	Thai	V	yealinkt22		~	x		×	-	2
	144 : 144	sip	Default:Default:Default		x	×	1	x	Thai	1	yealinkt20		-	x		Ń	-	Ż
П	000 : Mr. Two noudsand	iax	Default:Default:Default	Default:Default:Default	x	x	x	x	Thai	x	none		-	x		V	-	Ż
	000 : 000	sip	-	Default:Default:Default poise:eng:test	x	×	×	•	Thai	1	yealinkt18		-	x		×.	1	2
	001: 001	sip		Default:Default:Default poise:eng:test	x	V	×	1	Thai	x	none		-	x		×.	1	2
	002: 002	sip	-	Default:Default:Default poise:eng:test	x	V	×	1	Thai	x	none		-	x		×.	~	2
	003: 003	sip		Default:Default:Default poise:eng:test	x	<	×	1	Thai	x	none		-	x		×	×.	ļ
	004:	sip		Default:Default:Default	1	1	1	1	Thai	x	none		-	x		V	1	ļ
- 60		iax )	Default:Default:Default		1	x	x	~	Thai	~			-	x		1	-	Q

This example we can see that Ext 6000 present as iax protol.

Chapter:

#### 1.4 Add Analog Extension

Analog Extension is a type of extension that using analog phone connect with analog card (FXS port), while analog card can be pci card or external device like channelbank.

To add analog extension, everything same as sip extension adding so we can go to menu **Extension**  $\rightarrow$  **Add Analog Extension**, the following screen will be shown:

Extension Manage	er				
[View Extensions ] [ Add SIP Ex Add DAHDI Extension Phone Setting :	tension ] [ Add M	ultiple SIP ] [ Ad	d IAX Extension ] [ Add Analog	Extension ] [View )	All ,Add Follow-Me Extension
Group Name	Default::Defa	ult::Default 👻			
Phone Number					
DAHDI Port					
Caller ID					
Transmit Volume Gain	0.0				
Receive Volume Gain	0.0				
DAHDI Additional Setting			*		
			-		
Dial Option :					
		g user <mark>to Transfe</mark> r			
		user to Transfer			
		g user \"One Touc			
		user \"One Touch	<ul> <li>Provide Music on Hold (m) X</li> </ul>		
Call Features :	· Generate a	ringing tone (r) x	Provide Music on Hold (III) X		
Ring Timeout	30 👻				
Pickup Call from	Default::Defa	ult::Default			
	poise::eng::t				
	poise::sale::s	ale			
	Gamma and				
Record Incoming Calls	No 👻				
Record Outgoing Calls	No 👻				
enable callwaiting	Yes 👻				
enable callwaiting-callerid	No 👻				
enable threewaycalling	No 👻				
Default Language	English -				
Allow Roaming Station Feature	No 👻				
Mailbox :					
User Email Address					
Enable Message Center user login		Yes 🔻			
Voice Mailbox :					
Enable Voicemail box :		Disabled -		56. -	
Send Voice Message to Email :		No 🔻			
Voice Mailbox Description					
Voicemail Password (fix)		-			
Voice Mailbox Size(messages) Fax Mailbox :		1			
-		blo -			
Enable Fax Mailbox :	-	No 🔻			
Send Fax Message Notification to		No 🔫			
Attached Fax file to Notification E	mail :	No 🔫			
Send Voicemail Notification for inco	oming Fax :	No 👻			
Fax Mailbox Size :		20 👻			

re :	Browse	
ess Phone :		
Phone *:		
ess Fax :		
	*	
ss :		
	*	
	*	
i l		

#### 1.5 Add Follow-Me Extension

Follow-Me here means assign forwarding when some one call this number and no response in specific time, we can assign automatically forward in this menu.

To set Follow-Me, go to menu **Extensions**  $\rightarrow$  **Add Follow-Me Extension**, the following screen will be shown:

Show All   Close All	Add Follow-Me Extension:				
Status	Extensions Number	None	•		
Report			•		
Group Manager	Enable	yes 🔻			
Extensions	Music On-Hold	Default 🔻			
Fax	First Level Number Dial Timeout	30			
Call Control	Number	1:	2:	3:	
Call Features	Second Level Number	1:1	2:	2:1	
Sounds	Dial Timeout	30			
Incoming Call	Number	1:	2:	3:	
Outgoing Call	Third Level Number			,	
Schedules	Dial Timeout	30			
IVR	Number	1:	2:	3:	
Site to Site Setup					
Manual Config	APPLY Cancel				
Voice Interface					Plextel Enterprise Version 2
Log					
2 Advanced					

There are three levels of forwarding, if First Level can not answer can not answer on time (Dial Timeout), line will be forword to Second and Third level automatically.

All parameters can be explaine as these following:

**Extension Number**: Input specific extension, which we have to select once from drop down list.

L Chapter:

Enable: If select "Yes", enable this function, and "No" for disable.

Music On-Hold: Select Music On-Hold from list.

**Dial Timeout**: Unit of time in second, which the system will countdown, and perform next level when time up.

**Number**: Three of destination number which they will ring all when assigned, or we can assign one or two number.

An example of Follow-Me:

Extensions Number	SIP/1001	•	
Enable	yes 👻		
Music On-Hold	Default 👻		
First Level Number Dial Timeout	15		
Number	1: 1002	2: 1003	3:
Second Level Number Dial Timeout	15		
Number	1: 2001	2: 2002	3: 2003
<b>Third Level Number</b> Dial Timeout Number	15	-	-
Number	1:90866121905	2:	3:
APPLY Cancel			

This example is shown followme of 1001, if 1001 does not answer in 15 s, call will be forwarded to 1002 and 1003 (First level), if 1002 and 1003 do not answer in 15 s, call will be forwarded to 2001,2002,2003 (second level), then third level. In this example, at third level, destination number is mobile phone number with prefix "9", because we have to apply acceptable rule of the existing system.

When we finish, press "Save" and the following screen will be shown:

#### **Extension Manager**

18[ View Extensions ] [ Add SIP Extension ] [ Add Multiple SIP ] [ Add IAX Extension ] [ Add Analog Extension ] [ View All, dd Follow-Me Extension] Follow-Me Extension:

🗌 Numbe	er	Enable	List Number	Edit
1001		yes	Level1> 1002,1003, Level2> 2001,2002,2003 Level3> 90866121905,,	Ż
Page 1:	Delete Select	ted		

If we would like to see what extension have been set follow-me, we can see at menu Extensions →View All

8 Chapter:

#### **Chapter 2: External Line Configuration**

# 19

#### 2.1Voice Interface Hardware

This menu is the most important part of the system and it should be setup at beginning because this part is the interface part between IP-PBX and PSTN.

PSTN channels can be classified into 2 types, as following:

1. Analog Channel : The telephony line inform of copper line with RJ11

connector. There are 2 types of modules :

1.1 FXO Interface: Connect with PSTN line.



1.2 FXS Interface: Connect with normal analog telephone.



In term of PCI analog telephony card, there are many types which have been classified by number of ports (FXS or FXO) used.



**2. Digital Channel:** Digital telephone system which in Thailand the popular one is called ISDN PRI or E1, and there are 30 channels, concurrent call, per 1 line.

Due to IP-PBX system, to implement E1 system, we need E1 card, which one card can be reached 4 E1 line, depends on requirement.



Next, to configure card, go to menu Voice interface  $\rightarrow$  Voice Interface Hardware, then we press on "Detect Hardware", the system will show detected card which is put in pci slot on this moment.

Plextel System: 🥵	Voice Interface Hardwa		ttings ]	
Show All   Close All Status Report Group Manager Extensions Fax Call Control Call Features Sounds	Group Number : [1] Device Type : Interface Type : analog Port : 1 2 3 4 5 mg2 mg2 mg2 mg2 mg2 selectAll Echo Cancellation Technique mg2 Save Value	Active : yes Location :	Status : OK	
Incoming Call Outgoing Call Schedules IVR		APPLY Can	cel	Plextel Enterprise
Site to Site Setup Manual Config				
Voice Interface Voice Interface Hardware PSTN Trunk Setting SIP Trunk Setting Mobigate Gateway				

This previous picture show analog card with 4 FXO and 1 FXS module, then we press "Apply" and the system will show yellow bar at the top screen with wording "Click here to Restart IP-PBX", click it again.

20

# 21

Group Number : [1] Device Type : Astribank: Unit 0 Subunit 0: E1 Interface Type : digital-E1 Interface Range : 1 to 31	Active : yes St Location : usb-0000:00:1d.7-8	atus : RED
Clock/Sync Source : use incoming SYNC as primary sync source	•	
Cable Length: 0 db (CSU) / 0-133 feet (DSX-		
Frame Type: CCS 🔻		
Line Decoding: HDB3  CRC: Yes CRC: CRC: Yes CRC:		
Sav	ve Value	
	APPLY	ancel

This previous show digital channel type and device is asteriskbank, external channel bank with USB interface. Next, select "use incoming SYNC as primary source", click APPLY, the system will show yellow bar at the top screen with wording "Click here to Restart IP-PBX", click it again.

\*\* If your using card is Sangoma, you have to set it first at menu Advaced Setting and select Sangoma as following example.

Voice Interface Hard					
[ Current Interface ] [ Detect Hardward	[ Current Interface ] [ Detect Hardward [ Advanced Settings ] )				
Advanced Setting					
Interface Type	C Digium Compatible				
Dial-Tone	th 👻				
APPLY Cancel					

#### 2.2 PSTN trunk Setting

Next, go to menu Voice Interface  $\rightarrow$  PSTN Trunk Setting, and choose Add 222 , the following screen will be shown Analog Trunk, the following screen will be shown

Dahdi Interface Sett Interface Type	analog
1) Interface Setup	Description test
	Channel Number 1 🔹 - 4 💌
	Channel Group None -
	Trunk Name trunk:dahdi:test
	Record Incoming Calls 🔿 Yes 🤄 No
	Record Outgoing Calls C Yes  No
	Default Language th 💌
	Usage Type Wizard Voice (FXO)
	Channel TYPE FXO -
	Echo Cancel yes 👻
	Echo Cancel on Pure TDM yes 🔻
	Echo Training no 🔻
	Relax DTMF yes -
	TX Gain 0.0
	RX Gain 0.0
	Busydetect yes -
	Busycount 3
	Country-Tone Yes - Country Code TH
	Pulse Line no
	Caller ID Num
	Caller ID Name
	Immediate(FXS only) C Yes <ul> <li>No</li> </ul>
	Additional Setting
	Available Options: hanguponpolarityswitch=yes/no answeronpolarityswitch=yes/no busypattern=500,500
	usecallerid=no(as answer call immediately)
APPLY Canc	el

Some parameter has been set as default, no need to change it, we will focus on these following:

Channel Number: This value we have to assign related with "Define Raw Interface Card), if

It has 1-4 channel as FXO, we have to input 1-4.



Trunk Name: Input name of this trunk.

**Record Incoming Call**: If we would line to record all incoming line, select "yes", and we can see recorded file at menu Soun  $\rightarrow$  Call Recording File.

**Record Outgoing Call**: : If we would line to record all outgoing line, select "yes", and we can see recorded file at menu Soun  $\rightarrow$  Call Recording File.

#### Channel Type: Choose FXO or FXS

When complete all values, press "Add", the following screen will be shown

PSTN Trunk Setting [ View Trunk ] [ Add Digital Trunk ] [ Add Analog Trunk ] [ Advanced Setting ]					
Trunk Name	Description	Interface Type	Channel	Edit	
trunk:dahdi:test	test	analog	1-4	Ż	

#### **Digital Trunk**

Go to menu Voice Interface  $\rightarrow$  PSTN Trunk Setting  $\rightarrow$ Add Digital Trunk

## 24

[ View Trunk ] [ Add Digital Trun	Add Analog Trunk ] [ Advanced Setting ]
Dahdi Interface Setting	
Interface Type	digital
1) Interface Setup	Channel Number 1 🔻 - 1 🔻
	Channel Group None 👻
	Trunk Name
	Description
	PRI switch type euroisdn 👻
	PRI reset interval (ms)
	PRI Dialplan unknown 🔻
	PRI Local Dialplan unknown 🔻
	International Prefix 00
	National Prefix 0
	Local Prefix
	Private Prefix
	Unknown Prefix
	DID Yes 👻
	Callingpres Yes 👻
	Relax DTMF Yes 🔻
	Clock C Master   Slave
	Echo Cancel Yes 🔻
	Echo Cancel on Pure TDM Yes 🔻
	TX Gain 0.0
	RX Gain 0.0
	Record Incoming Calls C Yes 📀 No
	Record Outgoing Calls 🖸 Yes 💿 No
	Default Language th 🔻
	Caller ID Num
	Caller ID Name
	Additional Setting
	A
APPLY Cancel	

We have to assign these following:

**PSTN Trunk Setting** 

- Channel Number: Assign this related with detected hardware, like sample, start from 1-15, 16 for D channel (signaling), and 17-31.
- Trunk Name:Input name of this trunk.
- Clock: If we generate clock, select "Master", otherwise select "Slave".

To perform this menu completely we have to add 1-15 as beginning, then click "APPLY", and select "Add Digital Trunk" again to add channel 17-31, like this example.

5
J

PSTN Trunk Setting [View Trunk] [ Add Digital Trunk ] [ A	Add Analog Trunk ] [ Advanced Setting ]
Dahdi Interface Setting	
Interface Type	digital
1) Interface Setup	Channel Number 17 - 31 - Channel Group None - Trunk Name trunk:dahdi:Trunk_E1 -
APPLY Cancel	

When adding complete, the following screen will be shown.

#### **PSTN Trunk Setting**

[ View Trunk ] [ Add Digit Page 1 🗸	al Trunk ] [ Add Analog Trunk	] [ Advanced Setting ]		
Trunk Name	Description	Interface Type	Channel	Edit
trunk:dahdi:Trunk_E	1	digital	1-15 17-31	
Delete Selected				

This table present configured value assigned from service provider in Thailand.

	PRI Dialplan	PRI Local Dialplan	International Prefix	National Prefix	Relax DTMF
TOT	unknown	unknown	00	0	Yes
True	unknown	unknown	00	0	Yes

Next , Go to menu Outgoing Call  $\rightarrow$  Add New Outgoing Call.

OutGoing Call			
[ View Outgoing ] [ Add New Out Going C	all ]		
Outgoing Route Information			
Route Name			
Route Description			
Route Password			
Time Based Call Routing			
Default Route			
Add Frefix: Digit to Strip:	di:test  Default Outgoing Number: Dialing Option: T	Dialing	9
First Schedule Enable Schedule © Yes © No Time time			
Add     Trunk:     ZAP/trunk:dahdi:test     ▼       Prefix:     Digit to Strip:	Default Outgoing Number: Dialing Option: T	Dialing	
Second Schedule Enable Schedule C Yes © No Time time ▼			
Add Trunk: ZAP/trunk:dahdi:test 💌		Dipling	
		Dialing	
Prefix: Digit to Strip:	Dialing Option: T		
Call Patterns			
Call Prefix			-
Destination Pattern			
Destination Pattern2(optional)			
Destination Pattern3(optional)			
Dial Timeout	40		
Concurrent Call Limit for this trunk	100		
Strict Time Routing	No 🔻		
Support DID With This Route	No 👻		
APPLY			

#### **Outgoing Route Information:**

**Route Name**: Input name for this routing here.

**Route Password**: If we assign password here, everytime when dial out, the system will ask this password for authentication.

**Route Description**: Input description to make the other understand.



#### Time Base Call routing:

This feature means call routing that related with scheduling. This feature will be applied if we use gsm gateway like a trunk, while we can program call routing priority to match with service provider 's promotion, like if we have call free promotion for 8 am to 5 pm, we can program that this trunk will be first priority to save cost of the company.

In the beginning, we have to assign "Default Route", in this terms mean assign available trunk that will be applied when other trunks can not used.

Add: Select trunk then click this button.

**Trunk**: In this drop down menu show all defined trunk in the system, we have to select sequencially.

**Default Outgoing Number**: This value is used only in E1 system, to specific outgoing callerid as default, if any extension has no own out going caller id, the system will apply this value.

**Dialing Prefix**: Call prefix which this number will not be actually sent out.

**Digit to strip**: Number of digit to strip that related with prefix

Example of create "Default Route" after press "Add".

#### Default Route

Enable	No.	Trunk / Group	Default Outgoing Number	Dialing Prefix	<b>Digit to Strip</b>	dialing_option	
	1	trunk%%dahdi%%test				т	88
Add	Add 🔽 Enable Trunk: ZAP/trunk:dahdi:test 🔻 Default Outgoing Number: Dialing Prefix:						
Digit to	Digit to Strip: Dialing Option: T						

#### First Schedule, Second Schedule:

All values here same as default route, these following have been added

**Enable Schedule**: Select "yes" to apply this schedule.

**Time**: Click at drop down list, created schedule will be shown, if there is no created schedule, we have to create at menu "Schedule" (Topic 4.1 in this manual).

If there is some created schedule, the screen will appear like this example/.

#### First Schedule Enable Schedule • Yes • No Time time time business\_time Add wwwww

Other parts we input same as Default route, and the next example is the sample screen when press "Add".

#### First Schedule

Enable Schedule  Yes  No Time business_time							
Enable	No.	Trunk / Group	Default Outgoing Number	Dialing Prefix	Digit to Strip	dialing option	1
<b>V</b>	1	trunk%%dahdi%%test				т	8 🛛 👕
Add Strip:	Trunk	: ZAP/trunk:dahdi:test - Dialing Option: T	Default Outgoing Number:		Dialing Prefi	x:	Digit to

**Call Pattern**: The following picture present call pattern setting which some value shown here is the default value.

Call Patterns	
Call Prefix	
Destination Pattern	
Destination Pattern2(optional)	
Destination Pattern3(optional)	
Dial Timeout	40
Concurrent Call Limit for this trunk	100
Strict Time Routing	No 🔻
Support DID With This Route	No 🔫
APPLY	

**Call Prefix**: Define prefix of this outgoing rule. Example like press 9 to call PSTN, press 8 to call international.

Destination Pattern: Define destination pattern.

Destination Pattern2: Second destination pattern.

Destination Pattern3: Third destination pattern.

Dial Timeout: Unit of time in second, if timeout, system will be terminated.

**Concurrent Call Limit for this trunk**: Define allow maximum concurrent call for this trunk. **Strict Time Routing**: If select "yes", means there are setting First schedule and Second Schedule.

Support DID with this route: Select "yes" if this trunk need DID to call out. (E1 only)

#### Example of create Outgoing Call

				/	
mple of create Outgoing	Call				
OutGoing Call					
[ View Outgoing ] [ Add New Out Go	ing Call ]				
Outgoing Route Information		-			
Route Name	pstn		_		
Route Description		_			
Route Password					
Time Based Call Routing					
Default Route					
Enable No. Trunk / Group	Default Outgoing Number	Dialing Prefix	Digit to Strip dialin		
✓ 1 trunk%%dahdi%%test			Т	S S 🖻	
Add 🔽 Enable Trunk: ZAP/trun	k:dahdi:test 🔻 Default Outgoir	a Number:	Dialia	g Prefix:	
Digit to Strip: Dialing Option:		ig Nulliber. j	Dialiti	g Pielix. j	
	, ,				
First Schedule					
Enable Schedule C Yes C No					
Time time					
Enable No. Trunk / Group	Default Outgoing Numbe	r Dialing Prefix	Digit to Strip	dialing_option	
Add Trunk: ZAP/trunk:dahdi:test	<ul> <li>Default Outgoing Number:</li> </ul>		Dialing Prefix:	Digit to	
Strip: Dialing Option: T					
Second Schedule					
Enable Schedule C Yes 📀 No					
Time ffddfd 🗸					
Enable No. Trunk / Group	Default Outgoing Numbe	r Dialing Prefix	Digit to Strip	dialing_option	
ZAD/kmuslu daladistaat					
Add Trunk: ZAP/trunk:dahdi:test	<ul> <li>Default Outgoing Number: [</li> </ul>		Dialing Prefix:	Digit to	
Strip: Dialing Option: T					
Call Patterns					
Call Prefix					
Destination Pattern	1x				
Destination Pattern2(optional)					
Destination Pattern3(optional)					
Dial Timeout	40				on
Concurrent Call Limit for this trunk	100				ati
Strict Time Routing					Inc
-	no 🔻				fig
Support DID With This Route	no 🔻				Configuration

6 Chapter: Chapter 2: External Line Configuration

2.3 SIP Trunk Setting Go to menu Voice Interface -> Sip Trunk Setting, this menu is used for configure sip trunk like voip service provider, while we have to apply for any account from any service provider, and bring given information to apply here, like Username, Password, IP-Address (or domain name).

#### SIP Trunk Setting

[ View Trunk ] [ Add SIP Trunk ]
SIP Trunk Setting
Trunk Name
Description
Type SIP Account -
Sip Server Address
Tel Number/Username
Password
Allow Incomming Yes -
Concurrent Call Support 1
Optional: Number of digit to strip
Optional: Dialing Prefix
Record Incoming Calls 😳 Yes 🖷 No
Record Outgoing Calls 🔿 Yes 🖲 No
Default Language th 👻
Advanced Setting no 🔻
APPLY Cancel

All parameter can be explained like these following

Trunk Name: Define trunk name here.

Description: Input descripton of this trunk

Type: 2 type of provider.

**1.**Sip Account: Given Username, Password**2.**Sip Trunk: Username and password is no need.

Sip Server Address: Input IP-Address or domain name.

Tel Number/Username: Input given Tel Number/Username from service provider.

Password: Input given password from service provider

**Concurrent Call Support**: Number of concurrent call for this trunk, normally 1 concurrent / 1 account.

**Optional Number of digit to strip**: Number of digit to strip related with each service provider.

**Optional Dialing Prefix**: Prefix number related with each service provider.

**Record Incoming Call**: If select "yes", all incoming line will be recorded, and we can check recorded file at menu Sound  $\rightarrow$  Call Record Files.

**Record Outgoing Call**: If select "yes", all outgoing line will be recorded, and we can check recorded file at menu Sound  $\rightarrow$  Call Record Files

Default Language: Input default language for this trunk

When we complete all values, click Save, then we have to create Outgoing Call and also set call control same as we have done in PSTN setting.

#### 2.4 Gateway

**2.4.1 GSM Gateway** Integration This part is GSM gateway setting, we will separate into 2 parts:

- PLEXTEL-EE part
- GSM-SIP-Gateway part

(PLEXTEL-EE part) PART 1: Gateway Setting

Go to menu Voice Interface  $\rightarrow$  Gateway

#### Gateway

[ View Gateway ] [ Add Gateway ]

Display 1 - 1 fro	m 1 20	•				
🗌 Name	Number	IP-Addresss	Extension Monitor	Record IN	Record OUT	Edi
test_sys	4000	Dynamic	No	N	N	Ż
Page 1 :	Delete Selected	Cancel				

Chapter: Chapter 2: External Line Configuration

In this menu, we will create extension for gateway to register with Plextel.

Gateway [ View Gateway ] [ Add Gateway	]
Name	
Number / Account	
Default Language	English -
Password	
IP-Address	dynamic 👻
Codec	G.711u      G.711a      GSM      G.729      G.723.1     iLBC      Speex      lpc10      adpcm      G.726
dtmf mode	rfc2833 🔻
Extension Monitor	no 🔻
Concurrent Call Support	1
Enable REINVITE	default 👻
NAT Support	default 👻
CallerID Number	
CallerID Name	
Record Incoming Calls	C Yes 💿 No
Record Outgoing Calls	○ Yes ☉ No
APPLY Cancel	

- 1. Name: Input trunk name related with Plextel.
- 2. Number/Account: Input a Number or Account with can not same as defined extension in Plextel.
- 3. Default Language: Input default language for this trunk.
- 4. Password: Define password which gateway have to use this password to register.
- 5. IP-Address: Input IP-Address of gateway, normally we set to be "dynamic".
- 6. Codec: Define codec type.
- 7. dtmf mode: Define dtmf mode used for this gateway.
- 8. Concurrent Call support: Input maximum call support, which depends on number of ports of this gateway. Example: we use gateway with 4 FXO, so this value will be 4.
- 9 Enable Reinvite: Define packet type to be Reinvite.
- 10 NAT Support: If this device is working behide NAT, we have to set NAT=yes.
- 11 CallerID: Set caller id to be default value.
- 12 Record Incoming Calls: Allow recording incoming call.
- 13 Record Outgoing Calls: Allow recording outgoing call.

#### PART 2: INCOMING CALL

This part is setting incoming call for gateway, everything same as setting PSTN, but for gateway setting we have to input DID as the following example.

#### Incoming Call

[ View Incoming Call ] [ Add New Incoming Call ]

Displa	y 1	- 5	from	5
--------	-----	-----	------	---

Trunk		Destination			tion Edit
<pre>gateway:sip:test_sys</pre>	0 time	ivr	111	202	Ż

This example show setting gateway account as no.201 and trunk name is "test".

Incoming Call	
[ View Incoming Call ] [ Add	New Incoming Call ]
Add Incoming Call	
Trunk	gateway:sip:test_sys 🔻 🔲 PABX-link
Description	
Support DID	yes 🔻
Incoming DID	202
Replace CallerID	
Extensions Ring Timeout(se	c) 40 -
Add time based hand	ller
	Actions
When * time	▼ Destination* IVR ▼
Value 111 🔻	
Remove	
APPLY Cancel	

We have to set Support DID = yes and Incoming DID should not same with account of gateway like 202, while 202 will be set in gateway to send incoming call from gateway to PLEXTEL.

PART 3: CALLOUT WITH GATEWAY Go to menu OutGoing Call

#### OutGoing Call

[ View Outgoing ] [ Add New Out Going Call ]

#### Display 1 - 1 from 1

Route Name	Descriptions	Call Prefix	Call Pattern	Schedule	Enable	Edit
gateway			2xxx	Default	Default	2
				First / time	no	
				Second / time	no	

Page 1:

Delete Selected

We have to add gateway as trunk, then we call out from this gateway.

Tin	ne Based Call Routing					
De	efault Route					
No.	. Trunk / Group	Default Outgoing Number	Dialing Prefix	Digit to Strip	Delete	
1	gateway%%sip%%test				Del	
Ad	d Trunk: gateway:sip:test	🗾 Default Outgoing Number: 🗌		Dialing Prefix:		Digit to Strip:

#### (GSM-SIP-Gateway part) PART 1: SIP Setting

×	Service Do	Service Domain Settings					
Route	Mobile 1	Mobile 1 💌					
Mobile	Realm 1 (Default)						
Network	Active:	ON OFF					
SIP Settings	Display Name:	201					
Service Domain	User Name:	201					
Codec Settings	Register Name:	201					
Codec ID Setting	Register Password:	••••					
DTMF Setting RPort Setting	Domain Server:	192.168.0.140					
SIP Responses	Proxy Server:	192.168.0.140					
Other Settings		192.168.0.140					
NAT Transform	Outbound Proxy:						
Update	Status:	Registered					

Go to menu "SIP Setting  $\rightarrow$ Service Domain", and see previous example which it is setting SIP Account for SIP Gateway. Anyway all values here have to match with we have set in PART1.

Each value can be explained like these following:

35

**Display Name/ User Name/ Register Name**: Input Account no. which we have set in PLEXTEL.

Password: Input password which we have set in PLEXTEL.

**Domain Server/ Proxy Server/ Outbound Proxy** : Input PLEXTEL 's IP-Address.

When finish all, we have to SAVE and REBOOT everytime.

#### PART 2: CALLING GSM FROM PLEXTEL

# 36

#### LAN To Mobile Table

Mobile 1, 2 💌

Page: 1 💌

ltem	URL	Call Num	Select
0	192.168.0.140	#	
1			
2			
3			
4			
5			
6			
7			
8			
9			
Delete 3	Selected Delete All	Reset	

Go to menu "Route -> LAN To Mobile Table"

Input thes following values: Position = 0 URL = "IP-Address of PLEXTEL IP-PBX" Callnum = #
# 37

### PART 3: RECEIVING CALL FROM GSM Mobile To LAN Table

Mobile 1, 2 💌

Page: 1 💌			
ltem	CID	URL	Select
0	*	202	
1		$\searrow$	
2			
З			
4			
5			
6			
7			
8			
9			
Delete Se	lected Delete All	reset	
Add New			
Position:		(0~49)	
CID:		Ex:0911111111, 0911*, *	
URL:		Ex:192.168.0.1, *:2St	
Add reset	]		
Go to	menu Route -> Mobile	e to LAN	

Input these following values:

Position = 0

CID = \*

URL= "Input number from PLEXTEL PART 2, like following screen"

Add Incoming Call		
Trunk	gateway:sip:test_sys 🝷 🗌 PABX-link	
Description	ļ.	
Support DID	yes 👻	
Incoming DID	202	
Replace CallerID		
Extensions Ring Timeout(se	ec) 40 -	, C

#### 2.4.2 Setting Gateway to work with Soundwin

Create Gateway like previous part (GSM Gateway).

Gateway	
[ View Gateway ] [ Add Gateway ]	
Name	3000
Number / Account	3000
Default Language	English 💌
Password	••••
IP-Address	dynamic 💌
Codec	☑ G.711u ☑ G.711a □ GSM □ G.729 □ G.723.1 □ iLBC
	Speex pc 10 adpcm G.726
dtmf mode	rfc2833 💌
Extension Monitor	no 💌
Concurrent Call Support	1
Enable REINVITE	default 💌
NAT Support	default 💌
CallerID Number	3000
CallerID Name	3000
Record Incoming Calls	© Yes ◉ No
Record Outgoing Calls	◎ <sub>Yes</sub> ● <sub>No</sub>
APPLY Cancel	

Then we got like following example.

Gateway						
[ View Gateway ]   Page 1 💌	[ Add Gateway ]					
Name	Number	IP-Addresss	Extension Monitor	Record IN	Record OUT	Edit
3000	3000	Dynamic	No	Ν	N	Ż
Delete Selecte	d					

Then we log on Soundwin, and go to menu Advanced  $\rightarrow$  VoIP Basic, and make registration with PLEXTEL 's assigned number like these following:

VoIP Protocal Setting: Select protocol type as SIP.

No. : Imput port number.

Number: Input extension or gateway no.

Account: Input defined account number.

Password: Input defined password number.

SIP Proxy Setting: Input IP of SIP server like following example:

				tocol Setting SIP    Sele or / Password Setting(MAX 25 di		
N	lo.	Number		Account		Password
	1	3000		3000	•	
	2		]			
	Use P	ublic Account (PORT 1)		© Enable	e 💿 Dis	able
				SIP Hunting Table :		
No	<b>)</b> .	Hunting Member				
1		🗹 Port 1 🔲 Port 2				
2		Port 1  Port 2				
				SIP <u>Proxy Setting :</u>		
		Domain/Realm		192.168.0.130		
	SIP Proxy Server 192.168.0.130/5060			ione Service		
		Register Interval (seconds)		90		
		SIP Authentication		• E	nable 🤇	Disable
		Outbound Proxy Server		192.168.0.	130/5060	

NAT Pass Method	NAT Pass Setting: STUN  Symmetric RTP
STUN Server IP Address	64.69.76.21
STUN Server port	3478
NAT IP Address	0.0.0

	Local Setting	
Local SIP Port	5060	
	Apply	

Click Apply, select Register and Apply again, then save configuration and reboot. If register success, the screen will show like this example:

VoIP Protocol Setting SIP  Select							
		l	Port Number / Password Setting	(MAX 25 digit) :			
No.	Number	Reg	Account	Password	_	Register Status	Reason
1	3000		3000			Success	ОК
2							

Chapter: Chapter 2: External Line Configuration

Setting Hotline for mapping with DID in incoming call of PLEXTEL IP-PBX.

Hotline Delay	● Disable        ● Enable
Hotline Delay Time(Max. 20 sec)	3 Sec
Port 1 number	1111
Port 2 number	None

In this example, defined number is 1111 which have to map with menu "Incoming Call" in PLEXTEL, then we input DID to map with Hotline no. in Soundwin.

Incoming Call

[ View Incoming Call ] [ Add Nev	v Incoming Call ]			
Add Incoming Call				
Trunk	gateway:sip:3000	PABX-link	c	
Description	1			
Support DID			yes 💌	
Incoming DID			1111	1
Replace CallerID				
Extensions Ring Timeout(s	ec)		40 💌	
Enable CallerID-Based Rou	ting Service		No 👻	

Add time based handler
Actions
When * All_Time   Destination* Group
Value Default:Default 💌
Remove

This example, we discuss on Trunk that receive incoming from "gateway:sip:3000", have DID as 1111, and perform assigned action.

In term of Outgoing, we create like normal outgoing, while we add trunk to be gateway trunk.

**2.5 Site-to-Site Setup** This part we will discuss how to create trunking between 2 sites by using PLEXTEL IP-PBX, while we can select protocol between IAX and SIP



- 1. IAX Site to Site
- 2. SIP Site to Site

This screen show site-to-site setup screen.

Site to Site Setup [View Site] [Site to Site Setup] [Site-to-Site (SIP)	]
Add IAX2 Trunk	
Trunk Name	
Remote IP Address	Dynamic 👻
Remote Password	
Local Password	
Codec Used	<ul> <li>✓ G.711u</li> <li>✓ G.711a</li> <li>✓ GSM</li> <li>✓ G.729</li> <li>✓ G.723.1</li> <li>✓ iLBC</li> <li>✓ Speex</li> <li>✓ lpc10</li> <li>✓ adpcm</li> <li>✓ G.726</li> </ul>
Destination Pattern	Dialing Prefix: Pattern: Number of digit(s) to strip:
Destination Pattern2 (Optional)	Dialing Prefix: Pattern: Number of digit(s) to strip:
Destination Pattern3 (Optional)	Dialing Prefix: Pattern: Number of digit(s) to strip:
Destination Pattern4 (Optional)	Dialing Prefix: Pattern: Number of digit(s) to strip:
Destination Pattern5 (Optional)	Dialing Prefix: Pattern:

- Trunk Name: Input trunk name.
- Remote IP Address: Destination IP-Address, normally we use "Static", and define its IP.
- Remote Password: Input password for remoted site.
- Local Password: Input password for registration from another site.
- Codec Used: Select codec type.
- Destination Pattern: Define call out destination pattern.
- Dialing Prefix: Input prefix no.
- Pattern: Input pattern.
- Number of digit(s) to strip: Input no. of digit to strip.

When we have done both sites, the following screen will be shown.



We can check registration status at menu "Phone Connection Status".

S	(P						
No	Number	Name (Callerid)	IPaddress	Natsupport	VideoSupport	Status	Туре
1	4000		-none-	eno	eno	Unmonitored	SIP
2	trunk%%sip%%asdf		0.0.83.97	yes	eno	Unmonitored	SIP
3	3004	3004	-none-	eno	eno	UNKNOWN	SIP
4	3003	3003	-none-	eno	eno	UNKNOWN	SIP
5	3002	3002	-none-	eno	eno	UNKNOWN	SIP
6	3001	3001	-none-	eno	eno	UNKNOWN	SIP
7	3000	3000	-none-	eno	eno	UNKNOWN	SIP
8	1144	1144	-none-	eno	eno	UNKNOWN	SIP
9	1119	1119	-none-	eno	eno	UNKNOWN	SIP
10	1010	1010	-none-	eno	eno	UNKNOWN	SIP
11	1009	1009	192.168.0.190	eno	eno	OK (58 ms)	SIP
12	1008	1008	-none-	eno	eno	UNKNOWN	SIP
13	1007	1007	-none-	eno	eno	UNKNOWN	SIP
14	1006	1006	-none-	eno	eno	UNKNOWN	SIP
15	1005	1005	-none-	eno	eno	UNKNOWN	SIP
16	1004	1004	192.168.0.184	eno	eno	OK (113 ms)	SIP
17	1003	1003	-none-	eno	eno	UNKNOWN	SIP
18	1002	1002	-none-	eno	eno	UNKNOWN	SIP
19	1001	two	-none-	eno	eno	UNKNOWN	SIP
20	1000	on Thousand	-none-	es	eno		SIP

#### **Phone's Connection Status**

#### IAX

No	Number	Name (Callerid)	IPaddress	Natsupport	VideoSupport	Status	Туре
1	server_test_sit		192.168.0.100			(T)OK(1ms)	IAX2
2	6000	6000	(Unspecified)			UNKINOWIN	IAX2
3	2000	Mr. TTwo Thoudsand	(Unspecified)			UNKNOWN	IAX2
4	FaxDSP2/FaxDSP2		127.0.0.1			ОК	IAX2
5	FaxDSP1/FaxDSP1		127.0.0.1			ОК	IAX2

## **Chapter 3 System Features**

#### 3.1.Conference

Go to ment Call Features  $\rightarrow$  Conference, the following screen will be shown

## **Conferences Manager**

 [ View Conferences ] [ Add New Conference Room ]

 Conference Room
 Room Password
 Admin Password
 Record
 Leader
 Edit

 600
 No
 No
 No

 Page1 :
 Delete Selected
 Cancel

Select "Add new Conference", the following screen will be shown



# **Conferences Manager**

Room Number		
Conference Password		
Conference Admin Password		
Custom Voice Greeting for this room	None 👻	
Enable User Based Conference	No 👻	
Record This Conference Room	No 🔫	
Disable All Announcement (MUTE ALL)	No 🔫	
Disable Announcement For Single User In Room	No 🔫	
Announce User Count when Join Conference	Yes 👻	
Announce User Join/Leave	No	•
Enable Voice Menu mode	Yes 👻	
Enable logoff from conference using # key	Yes 👻	
Enable MusicOnHold for Single User login	Yes 🔻 Defa	ılt 🖣
Enable Monitor Mode (# prefix)	No 🔫	
Enable Leader Mode	No 👻	
Conference Leader Dialing Number		
Close the conference when Leader exit	No 🔫	
	No 🔫	

#### APPLY Cancel

All parameter can be explaine as following:

- Room Number: Input room number
- Conference Password: Input password for this room.
- Conference Admin Password: Imput password for admin.
- Custom Voice Greeting for this room : Select greeting voice for this room.
- Enable User Based Conference: Define user and password for participant.
- Record this conference room: If select "yes", all conversation will be recored, and we can check recorded file at menu Sounds -> Call Record Files (conf).
- Disable all announcement (MUTE ALL): If select "yes", all announcement will be closed.
- Disable Announcement for Single User in room: If select "yes", the system will close announcement when first participant enter this room.
- Announce User Count when Join Conference: If select "yes", system will announce number of participants for next entry person.
- Announce User Join/Leave: If select "WihtReview", system will ask joinging participant 's name and announce to the other.

- Enable Voice Menu mode: If select "yes", participant can press "\*" to listen and select menu.
- Enable logoff from conference using # key: If select "yes", "#" button will be applied to use for leaving out from this room.
- Enable MusicOnHold for Single User login: Enable Music on hold, when first participant is waiting the other.
- Enable Monitor Mode (# prefix): If select "yes", any user can listen conference room without talking by press # and follow by conference room number.
- Enable Leader Mode: Click to enable leader mode.
- Conference Leader Dialing Number: Define number for leader to access this room.
- Close the conference when Leader exit: If select "yes", this room will be terminated when leader leave out.
- Run Custom Menu (1 digits) : This part for additional programming by developer only.

When we complete all value, click "Save", click yellow bar at top menu to reload", then we will define permission of conference room by go to menu Call Control.

This following screen present accept permission of conference room for only "Default" group.

Group Name	<pre>] DefaultDefaultDefault</pre>	1 poiseengtest1234	l pstn	gateway	1 SATSCORE	virtual-eng_	virtual-33	virtual-abcd	agent_blf_status	conference	parkcalls	[ testttttttttttt	test_ivr	Paging-55555	dddd	T WWWWW(SIP)
All Group						•			4							
Sefault:Default:Default	M		<b>V</b>			<b>V</b>	V	1	1		<b>J</b>	~	<b>V</b>	V		1
poise:eng:test1234				Г						Μ	K					
ρρρρ										Γ	2				$\overline{\mathbf{N}}$	
wwwww(SIP)																
hello_world						$\overline{}$				Γ						
testttt		1 - A			-				-							
testttt test_fax		, ,		Γ	Γ	Г	Г	Г		Г	Г		Г			
		, 1														

# Call Control

# 46

Next, we can test conference by using ip-phone and press 600 (created room), the system will ask password, if password has been defined, then the system will announce number of participants now.

If we would like to make conference with any person in outside system, we can call their mobile phone and transfer line to created room number, example like #1 600, while 600 is created room.

To monitor conference room, go to menu Report  $\rightarrow$  Conference Status, the following screen will be shown.

Confere Conference F	Room Status 600 -	(1)	<b>∠</b> <sup>3</sup>	
Information Non Kick	for conference room: 600 Users / CallerID	Mute All LOCK Channel	Mute / Unmute	4 Status
2 Kick 2 Kick	1001/two <b>6</b> 1005/1005	SIP/1001-00000045 SIP/1005-00000046	Mute Mute	(unmonitored) (unmonitored)

This menu can be explained as following:

- 1. Select room that we want to monitor.
- 2. Mute All, when we want to mute all participant, this button status will change to red, and
- it will be green when we click to unmute.
- 3. LOCK, press when we need to block other users to join this room.
- 4. Mute, press if we want to mute this person.
- 5. Kick, press when we need to take off this person from this roo,.
- 6. Caller ID or name of each participant.

O P Chapter: Chapter 3 System Features

# 47

### 3.2. Feature Key / Call Parking

Go to menu Call Features  $\rightarrow$  Feature Key & Call Parking, the following screen will be shown:

Features Code / S	Sys	te	m S	etup	
Agent Login (permanent)	*	•	9		
Agent Callback Automatic Login/Logoff	*	•	45		
Agent CallBack Login	*	•	40		
Agent CallBack Logoff	*	•	41		
Pause Agent Prefix	*	•	42		
UnPause Agent Prefix	*	•	43		
Transfer to Agent Prefix	44				
Whisper	*	•	97		
Private Whisper	*	•	98		
Channel Spy	*	•	99		Password: 1234
Call Parking					
Parking Number	700	)			
Parking Position	70:	1		- 720	
Max Parking Time	120	)			
Transfer digit timeout	3				
Features Key Mapping					
Features digit timeout(ms)	300	)0			
Call Pickup	*	•	8		
Extensions Pickup	*	•	*		
Blind Transfer	#	•	1		
Attend Transfer	#	•	2		
Disconnect	*	•	0		
One Touch Record	*	•	3		
Voicemail	100	)			
Phone Lock	99				
Roaming Station Register / Dial- Out Prefix	*	•	**		
Fax Prefix		-	33		
Features Key Mapping					
CUSTOM1	*	•			Enable 🗔
APPLY Cancel Restore Default					

**Call Parking:** In this term, call parking means holding line from concurrent call for a while. In Plextel IP-PBX we have to transfer this line to no.700, and the system will automatically announce our number of parking status like 701, 702 ...., if we want to pick up that line again, we will dial to announced number like 701, 702, related with define parking position.

**Transfer Digit Time Out:** Unit of time in second that allow users to press each digit for using this feature.

**Feature Digit Timeout:** Unit of time in millisecond that allow user to press for access any feature.

**Call Pickup:** In this term we can pick up incoming call by press \*8, to pick up any extension existing in same group.

Normally, we have to set permission for each extension at extension menu.

Plextel System:			Manager [ Add SIP Extension ] [ A	dd Multiple SIP ] [ Add IA	X E	xten	sion	] [ A	dd Ar	nalog	g Extension	I] [View All,Add	Follow-Me	Extension]			
how All   Close All	Display 1 - 30	From	30		_							Brow		Data Innant	Extensi		
Status	20		•		E	xten	sion	Imp	ortj			Brow	se	Data Import	Extensi	on Expo	rt
Report		• 🛛	3														
Group Manager Extensions	Number	Тур	eGroup	Pickup Call From	BLF	I/R	O/R	INT	Lang	JAP	Phone Type	DIDFollowme	Roaming	User Email	Vmail	FAX mailbo	x Edi
Fax	Thousand		Default:Default:Defaul	t	×	1	V	x	Thai	x	none	-	x	nicrora@hotmail.con	1	-	2
Call Control	☐ 1001 : tw	o sip	Default:Default:Defaul	t	1	1	1	x	Thai	1	snom360	1	х		1	1	4
Call Features	□ 1002 : 1002	sip	Default:Default:Defaul	t	x	1	1	x	Thai	x	none	-	x		1	1	Z
Sounds	□ 1003 : 1003	sip	Default:Default:Defaul	t	x	1	1	x	Thai	x	none	-	x		1	1	Į
Incoming Call Outgoing Call	1004 : 1004	sip	Default:Default:Defaul		1	V	V	x	Thai	x	none	-	x		1	-	ý
Schedules	1005 : 1005	sip	Default:Default:Defaul	tDefault:Default:Default	x	1	V	x	Thai	x	none	-	x		1	1	ý
IVR	1006 : 1006	sip	Default:Default:Defaul	t	x	1	1	x	Thai	x	none	-	x		1	1	ļ
Site to Site Setup	□ 1007 : 1007	sip	Default:Default:Defaul	tDefault:Default:Default	x	1	1	x	Thai	x	none	-	x		1	1	Q.
Manual Config	1008 :	sip	Default:Default:Defaul	t	x	1	1	x	Thai	x	none	-	x		1	-	Į,
Voice Interface	1008 1009 : 1009	sip	Default:Default:Defaul	t	x	1	-				yealinkt28	-	x		<b>√</b>	1	Ø,
Advanced	1009 1010 : 1010	sip	Default:Default:Defaul	t	x	1	1				yealinkt26	•	x		1	-	Ż

When clink "Edit", we can check permission of this extension like this following example.

#### Call Features:

Ring Timeout	30 🔻
Pickup Call from	Default:Default:Default poise:eng:test1234
Record Incoming Calls	⊙ Yes C No
Record Outgoing Calls	
Support Intercom	
Allow BARGE Call	
Default Language	Thai 🔻
Allow Roaming Station Feature	No 🔻

This previous picture we can observe that there are many group name form pick up group list, if we would like to select group, we can click mouse on that group.



This menu show how to set pickup call since we create extension.

### **Extension Manager**

[ View Extensions ] [ Add SIP Extension ] [ Add Multiple SIP ] [ Add IAX Extension ] [ Add Analog Extension ] [View All, Add Follow-Me Extension]

Edit SIP Extension Phone Setting:	
Group Name	Default:Default:Default 👻
Phone Number	1001
Caller ID	two
Password	• • • •
Enable BLF	yes 🔻 manual BLF num
Phone IP-Address	dynamic 🔻
Codec	<ul> <li>✓ G.711u</li> <li>✓ G.711a</li> <li>✓ G.723.1</li> <li>✓ ILBC</li> <li>✓ Speex</li> <li>✓ Ipc10</li> <li>✓ adpcm</li> <li>✓ G.726</li> </ul>
Video Codec	□ H.261 □ H.263 □ H.263p □ H.264
dtmfmode	rfc2833
Extension Monitor	yes 👻
Concurrent Call Support	12
Enable REINVITE	default 👻
NAT Support	default 💌
Support T.38 FAX	default 👻
SIP Additional Setting	×
Dial Option:	
	<ul> <li>Allow calling user to Transfer (T)</li> <li>Allow called user to Transfer (t)</li> <li>Allow calling user \"One Touch Record\" (W)</li> <li>Allow called user \"One Touch Record\" (w)</li> </ul>
Call Features:	C Generate a ringing tone (r) • Provide Music on Hold (m)
Ring Timeout	30 💌
Pickup Call from	Default:Default poise:eng:test1234
Record Incoming Calls	⊙ Yes C No
Record Outgoing Calls	⊙ Yes C No
Support Intercom	
Allow BARGE Call	
Default Language	Thai 🗸
Allow Roaming Station Feature	No 🔻

This picture present this user can pick up only extension in same group.

- **Extensions Pickup:** Pickup Call by press specific destination number by pressing \* \* follow by extension. Ex: \*\* 1002, \*\*1003.

**51** 

**Blind Transfer** is a transfer when the person receiving a call transfers the caller to another person without telling the that person anything about the caller or why they are calling. This is usually very frustrating to the caller as they have to introduce themselves and explain why the are calling again. If the no person receive that transfer call, that call will be sent back to the person receiving a call. We can press \*1 to send blind transfer code before sending the call to destination number.

Attend Transfer is a transfer when the person receiving a call transfers the caller to another person and telling the that person anything about the caller or why they are calling before transfer to them. We can press \*2 to send blind transfer code before sending the call to destination number. If the persons whom you transfer to, receives a call, you can hang up. But if they don't, you can press \* to pick that call back to have a conversation.

**Disconnect** when you want to cut your line off while having the conversation, you can press \*0 during the conversation time.

**One Touch Record** You can record voice of talker while you're talking. You can press \*3 to automatic recording and you can bring that file to play later at menu *Sounds -> Call record File* 

**Voicemail** you can set the number to go to listen your voicemail by press on your phone. The number is 100, when you call 100, it asks your password before you listen your voicemail. Make sure that you've created mailbox already. You can check it at Extension menu.

🗆 Nur	mber	Туре	Group	Pickup Call From	BLF	I/R	O/R	INT	Lang	AP	Phone Type	DIDFollowm	eRoaming	User Email	Vmail	FAX mailbo	x Edit
	00 : on ousand	sip	Default:Default:Default	t	×	•	•	x	Thai	x	none	-	x	nicrora@hotmail.com	~	-	Ż
☐ 100	01:two	sip	Default:Default:Default	t	1	1	1	x	Thai	1	snom360	✓	x		1	1	Ż
□ 100 100		sip	Default:Default:Default	t	x	V		x	Thai	x	none	-	x		1	~	Ż
□ 100 100		sip	Default:Default:Default	t	x	V	×	x	Thai	x	none	-	x		1	1	Ż
□ 100 100		sip	Default:Default:Default	t	×	V	×	x	Thai	x	none	-	x		1	-	Ż
□ 100 100		sip	Default:Default:Default	tDefault:Default:Default	x	1		x	Thai	x	none	-	x		~	~	Ż

As the above picture, number 1003 has set mailbox, you can listen your voicemail. But if the mailbox has not set, you can edit or create the new one following this below example

Mailbox:					 	
User Email Address :						
Enable Message Center User Login : Voice Mailbox :	Yes 🔻					
Enable Voicemail Box :	Enabled 🔻	1				
Send Voice Message To Email :	C Yes 🔍 No					
Voice Mailbox Description :						
Voicemail Password (fix) :						
Voice Mailbox Size (messages) :						
Fax Mailbox					 	
Enable Fax Mailbox :		No	•			
Send Fax Message Notification To Em	ail :	Yes	•			
Attached Fax File To Notification Ema	il :	Yes	-			
Send Voicemail Notification For Incom	ing Fax :	Yes	-			
Fax Mailbox Size :		20		-		

This picture show the part of configuration about the extension that you can set voicemail configuration. This part has not shown if you don't be enable mailbox.

Phone Lock You can lock your phone automatic by refer to your mailbox password for each phone. When user is not at his/her seat, it can protect another person use his/her phone. You can press 99 to lock your phone. Then system will ask your password. After you send the correct password, you can choose between press 1 to lock your phone and press 2 to unlock. After you hang up. System will be set the phone as you choose.

Whisper you can join to talk with someone that is talking with another person but the person who call from external line will not hear. Only the internal line can hear. It can help an agent who cannot answer the customer's question. You can press \*97 and the destination number that you want to join that call. For example, number 1002 is talking with the customer, you'll press \*97 1002 to join and advice agent.

Private Whisper you can join to advice and tell something to agent only, but the person who use internal line cannot reply to you. You can press \*98 and the destination number. For example, number 1004 is talking with the customer. You want to tell something to number 1004. Then you press \*98 1004 and tell him/her. Number 1004 will hear you, but customer doesn't.

Channel Spy You can listen another person talks without the speaker's knowledge. You can press \*99 and destination number. Then system will ask you the password following the below picture.



Agent Login Number this feature is to be enable agent that I set on menu Call Features -> Agent

Roaming Station Register/ Dial-out Roaming is a general term referring to the extension of connectivity service in a location that is different from the home location where the service was registered. For example, your phone number is 1001 but you must move to another place that has the phone with number 1002. You can roam your number (1001) to this place (1002) by using the feature. You can press \* \*\* and your phone number (in this case is 1001). When it has call to number 1001, it'll ring on phone number 1002

Before you use roaming station register/Dial-out, you must set permission to be roaming in extension menu.

And Edit the number that you allow to use roaming feature following this below picture.

w All   Close All	Display 1 - 30 From	m 30		Exte	ensi	on In	npor	rt 🕅			Bro	owse	Data Import	Extensi	on Expo	rt
Status	20	-						•					bata inport			_
Report	Page 1 🔻															
Group Manager	Page 1 •	22 21														
Extensions	🗌 Number Ty		-	BLFI/	RO	)/RI	NTL	ang	AP	Phone Type	DIDFollown	eRoaming	) User Email	Vmail	FAX mailbo	×E
Fax	Thousand	Default:Default:Defaul	t	< •	1	1	×	Thai	x	none	-	x	nicrora@hotmail.com	~	-	1
Call Control	1001 : two sip	Default:Default:Defaul	t	1.	1	1	x	Thai	1	snom360	~	x		1	~	(
Call Features	□ 1002 : sip 1002	Default:Default:Defaul	t	x	1	1	x	Thai	x	none	-	×		~	~	
Sounds	1003 : sip	Default:Default:Defaul	t	x	1	1	×	Thai	x	none	-	×		~	~	
Incoming Call Outgoing Call	☐ 1004 : sip 1004	Default:Default:Defaul	t		1	1	x	Thai	x	none	-	×		~	-	
Schedules	□ 1005 : sip 1005	Default:Default:Defaul	tDefault:Default:Default	x	1		x ·	Thai	x	none	-	x		1	Ľ	
IVR	☐ 1006 : sip 1006	Default:Default:Defaul	t	x	1	1	×	Thai	x	none	-	x		1	~	
Site to Site Setup	□ 1007 : sip 1007	Default:Default:Defaul	tDefault:Default:Default	x	1	1	×	Thai	x	none	-	×		~	~	
Manual Config	☐ 1008 : sip	Default:Default:Defaul	t	x	,	1		Thai	x	none	-	×		1	-	
Voice Interface	1008			×		~	×	Inal	×	none		×				
Log	□ 1009 : sip 1009	Default:Default:Defaul	τ	x	0	1	x I	Thai	1	yealinkt28	-	×		~	-	

Choose yes @ allow roaming station feature

Call Features:	
----------------	--

Ring Timeout	30 🔻
Pickup Call from	Default:Default:Default poise:eng:test1234
Record Incoming Calls	
Record Outgoing Calls	⊙ Yes C No
Support Intercom	
Allow BARGE Call	
Default Language	Thai 🔫
Allow Roaming Station Feature	Yes 🔻

#### 3.3. Paging

Paging is the one feature on Plextel System, that is a group of intercommunication. It call one number that set on paging. Then caller can call to that number to talk with other member in paging group

You can create paging number at menu Call Features -> paging as the below picture.

Ρ	lextel System: 🧕
Show	All   Close All
≥ St	atus
≥ Re	eport
Gi	roup Manager
Ex	tensions
Fa	х
Ca	all Control
∿ Ca	II Features
	Conference
	Features Code / System
	Setup
	Paging
	New Features

Next, go to Create Paging tab. You'll see this part.

Call Paging

[ View Paging ] [ Create Paging ]

Call Paging [View Paging] [Create Paging] Add Paging Paging Number	
Phone Members (add Multiple)	
Select Phone	9006 -
APPLY	Agent Members

Then you set number to use paging and select the extensions that want to be a member in paging

Call Paging [ View Paging ] [ Create Paging ] Add Paging Paging Number	6000 9000 9001 9002 9003	
Phone Members (add Multiple) Select Phone	9004 9005 9006	
APPLY	9007 9008 lembers 9009	

After you add member finish, you'll see paging member list as the example below.

Call Paging [View Paging] [Create Paging] Add Paging

Paging Number

Name	Member	Delete		
4444	9005	Del		
4444	9006	Del		
4444	9007	Del		

4444

APPLY

For example, we set number 4444 is the paging number and members of this paging are number 9005, 9006, 9007

#### Call Paging

	Number	Member	Edit
4444	9005 9006 9007	5	Ø.
Page 1:	Delete Selected		

Before using paging, you must set the permission of user at call control menu

#### Call Control

Group Name	DefaultDefaultDefault	poiseengtest1234	pstn	kkk	sip	SATSCORE	virtual-eng_	virtual-33	virtual-abcd	agent_blf_status	conference	parkcalls	testttttttttttt	test_ivr	Paging-4444	test_site	xxx(SIP)	dsdsd(SIP)
All Group				Г	Г		☑		Г	◄		Г	~	Г	F	Г		
Default:Default:Default	V	V	~	Г	Γ	Γ	◄	•		◄	◄	◄	~	7	┍			
poise:eng:test1234	☑	$\square$	☑				~			◄			☑	内	◄	1		
test_site	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	P	ব		
xxx(SIP)																	$\overline{\mathbf{N}}$	
dsdsd(SIP)																		$\square$
hello_world							◄											
testttt																		
111																		
aaa	Г										Г					Г		

Page 1: APPLY

Extensions Group Outgoing Call Call Features Site-to-Site Queue IVR FAX

#### 3.4 Virtual Number

**56** 

Virtual number is the number that is created to use in other features easily. You can create the virtual numbers

as much as you want. And the virtual number must not be same as the existed number.

You can create a virtual number in menu Call feature -> virtual number as picture is below.

Plextel System: 🧕	Virtual Number [ Virtual Number Group ] [ Add New Group ]
Show All   Close All	
Status	
Report	
Group Manager	
Extensions	
Fax	
Call Control	
Conference	
Features Code / System Setup	
Paging	
New Features	
New Macro	
Agents	
Queue	
CallerID Routing	
Customer Satisfaction Setup	
Virtual Number	

Then you create group of virtual number to set permission at Call Control menu



#### Virtual Number

[Virtual Number Group] [Add New Group]

Group Name : call_cente	er [del]				
Virtual Number	Priority	What to Do	Value	edit	delete
Add					

The below picture shows permission status of group's user that can use this virtual number group

# **Call Control**

#### DefaultDefaultDefault center agent\_blf\_status poiseengtest1234 testtttttttttttttt virtual-abcd virtual-call Group Name virtual-eng Paging-4444 conference dsdsd(SIP) parkcalls test\_site SATSCORE test\_ivr (SIP) XXX pstn sip <u>k</u>k • $\overline{\mathbf{v}}$ $\nabla$ Г Г r r All Group N $\overline{\mathbf{v}}$ $\overline{\mathbf{v}}$ Г Default:Default:Default 7 ľ ſ ľ Γ Г ľ Г Г • $\overline{\mathbf{v}}$ $\checkmark$ Г 7 Г Г Г Г poise:eng:test1234 Г Г Г Г Г Г Г Г test\_site Г Г Г ľ Г Г Г xxx(SIP) Г $\nabla$ dsdsd(SIP) Γ Г Г Г Г Г Г ľ Г Г г Г $\overline{\mathbf{v}}$ Г Г Г Г Г Γ Г Г Г Г Г Г Г Г Г Г hello\_world Г Г Г Г Г testttt г Г Г г г Г Г Г Г Г Г г г Г Г Г Г Г Г Г Г Г Г Г Г Г Г 111 Г Г Г ľ ľ ľ r Г Г Г Г ľ aaa Page 1: APPLY Extensions Group Outgoing Call Call Features Site-to-Site Queue IVR FAX

Next step, Add the virtual number in group



Set the action that what to do. If you click Add what to do, it'll show action list.

> /

500 call_center		
AddWhatToDo		
APPLY		

When Setting action has done. Apply that to finish setting.

500 call_center			
Dial Conference	🚽 🔘 null	🗟 600 👻	📚 👕
Hangup	<b>~</b>		🖹 🗎
AddWhatToDo			
APPLY			

As the below picture, shows virtual number group that consists of number 500 in group. When extension that allows

permission to use this virtual number group, calls number 500. Plextel will send your call to conference room NO.600

Virtual Numb	er				
[Virtual Number Group]	[ Add New Group ]				
Group Name : call_ce	enter [del]				
Virtual Number	Priority	What to Do	Value	edit	delete
500	1	Dial Conference	600	l 🧷	1
	2	Hangup	null		
Add					

All actions in virtual numbers feature will explain later on chapter 4.2 IVR

# **59**

#### **Chapter 4 Incoming Line Configuration**

4.1 Schedule is a list of actions from a set of transactions in database. For example, All calls go to IVR in the office hour and all calls go to voicemail system when It is out of office hour because at out of office hour has no employees receive calls.

Schedule Na	me Description		Tin	ne Range		Edit	
ī all	all	Month	Date	Day	Time	Q.	
		*	*	8	allTime	1000	
		*	*	8	allTime		
		*	*	*	allTime		
		8	8	8	allTime	-	

You can set schedule at menu Schedule tab on left hand side. It show as below picture.

You will see the default setting of schedule, "All". That means this schedule support all times. You can add

new schedule in thi	s page.			
Schedules Manag [ View Schedule ] [				
		Schedule Det	ails	
Name:				
Description:				
			~	
			~	
1				
Add Time Rand	ie -			
APPLY			Ca	ancel
				Plextel Enterprise Version 2

From above picture, it has 2 part to set schedule.

Name is the name of schedule such as working time, closed time.

**Description** is the wording that explain about schedule

Next, you add the range time. You can add many range times as much as you want to set

		ld Sche			Sc	hed	lule I	Det	aile		
lam		_	_		50		uici	Jei	ans		
am	e:										
esc	ription:										
											~
											*
A	dd Time Range										
	Month	Dav	-of-I	Month	Day-of-Week		Hour		Minu	te	
rom	*	/ *	¥	]	*	¥	*	¥	*	¥	Remove
0	* 4	*	~		*	~	*	~	*	~	
	Month			) Month	Day-of-Week		Hour	÷.	Minu	te	L
rom		Day	-01-1		*	v	*	v	*	~	Remove
			*		*		*	-	*		Remove
0			~		-	۲		*		<b>×</b>	
	Month	_	-of-l	Month	Day-of-Week	_	Hour	_	Minu		
- i					*	*	*	*	*	~	Remove
rom	*	*	*					_		_	

Then, you set range time as this example. This example shows office hour, is on Mon-Fri 8:30 - 17:00 and Sat 8:30-

12:30

						Sc	hed	ule [	Det	ails		
lam	e:		W	orkir	ng_Tin	ne	_					
)esc	ription:											
												~
												~
			1									
A	dd Time Ran	qe										
ļ	Month		Day	-of-I	Month	Day-of-Week		Hour		Minut	te	
from	*	~	*	~		Monday	~	8	¥	30	~	Remove
to	*	~	*	~		Friday	~	17	¥	30	¥	
	Month		Day	-of-l	Month	Day-of-Week		Hour		Minut	te	
	*	<	*	~		Saturday	~	8	¥	30	<	Remove
from		_	-	v	1	Saturday	~	12	v	30	~	
I	*	~	*	Y		Saturuay						

Then you click apply to save this configure. It'll be display as below picture.

Schedule Name	Description			Time Ran	ge	Edit
Working_Time		Month	Date	Day	Time	Q.
		8	8	Mon - Fri	8:30-17:30	
		*	8	Sat - Sat	8:30-12:30	

#### 4.2 IVR (Interactive Voice Response) is the system that receive all external calls and manage the call to dial

following a diagram. You can create IVR from IVR menu and click to Create voice Menu as the below picture.

Voice Menu [View Voice Menu] [Create Voice Menu]	
[ view voice mena ] [ create voice mena]	Voice Menu Details
Name	
Description	
Allow direct call from this menu	
Default Language	English 💌
Enable Menu Password	
Intro Sound	Intro.wav 💙
Invalid Sound	Invalid.wav 💙
Timeout Sound	Goodbye 🔽
Absolute Timeout Sound	Goodbye.wav 💌
Exit Sound	Goodbye.wav 💌
Music On Hold	Default 💙
Absolute Call Timeout (sec)	D
Wait For Response (sec)	15
Wait For Additional Digit (sec)	3
No Input Max Repeat Times	p
Invalid Input Max Repeat Times	p
Direct Call Menu	None 💙
Automatic Dial Number	None 💌
APPLY	
	Plextel Enterprise Version 2

Name: is the name of IVR

Description: is the detail of IVR to explain how to ivr works

Allow direct call from this menu: when you select this box, you will allow the external line to call the extensions directly if caller knows the extension. The first sound that plays in IVR is from Intro Sound. This sound should tell all details how to call extensions. For example, "Welcome to Poisetechnology, please enter your destination number that you know or press 0 to connect operator."

If you want the external call can dial to extension directly, you must set permission in Call Control.

### Call Control

roup Name	DefaultDefaultDefault	poiseengtest1234	pstn	kkk	sip	SATSCORE	virtual-eng_	virtual-call_center	virtual-abcd	agent_blf_status	conference	parkcalls	testtttttttttttt	test_ivr	Paging-4444	test_site	×××(SIP)	dsdsd(SIP)
All Group		$\overline{\mathbf{N}}$					☑			$\overline{\mathbf{v}}$		Γ	☑					
Default:Default:Default			•							☑				☑				
poise:eng:test1234	☑									☑				☑				
test_site																		
xxx(SIP)																	$\overline{\mathbf{v}}$	
dsdsd(SIP)	Γ													Γ	Γ			
hello_world	Г						☑											
testttt	Г																	
111	Г													Г	Г			
main	V	Г	Г	Г	☑	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	
a	Г	Г										F						

As this example, oranage tab is the ivr that has name is main. It set permission that allow default group and sip

outgoing call. That means caller can call directly to extension in default group and call pass through sip outgoing call.

Default Language: you can select basic sound language to use with IVR

Enable Menu Password: If this is enable, IVR must require password to access menu to edit rule. After you enable

this, it will show details about flow diagram.

Enable Menu Password	
Password Sound	Intro 💌
Menu Password	
Invalid Password Sound	Intro 💌
On Invalid	Hangup 💌

Password sound: You can select sound file to play sound to ask password

Menu Password: You can set password for this menu

Invaid Password Sound: This sound will play when user send wrong password

On Invalid: You can select what ivr to do when user enter wrong password. It has 3 choices to choose.

- 1. Hangup: it hangup when you enter wrong password.
- 2. Repeat Menu: it ask your password again
- 3. Goto Previous Menu: it's back to previous menu

Chapter: Chapter 3 System Features

**Intro sound:** this sound is the first one to play when it has call into ivr. For example, "Welcom to Poise Technology, Please press the extension number or press 0 to call operator."

(If you want to add new sound, you can do that in menu sound -> create voice)

**Invalid Sound**: this sound will be played when user enter wrong number such as "You enter wrong number, please enter the number again"

Timeout Sound: This sound will be played when user does not enter the number in time.

Absolute Timeout Sound: This sound will be played when it reach limit time.

Exit Sound: This sound will be played when user enter wrong number and reach the limitation of enter

Music on Hold: Select this to use in IVR

Absolute Call Timeout(sec): If you set this time, it will cut off a call when it reaches time limitation.

Wait for Response(sec): when you set this. IVR will repeat itself if caller don't enter number in time.

Wait for additional digit(sec): during time that IVR get the number and it has no response. IVR will repeat itself when it's out of time.

No Input Max Repeat Times: the number of repeating when user doesn't enter the number in time.

Invalid Input Max Repeat Times: the number of repeating when user enter wrong number.

Direct Call Menu: If you want to jump to another IVR when IVR finish playing sound, you can select in this IVR list.

Automatic Dial Number: You set that ivr can automatic dial to assigned number when it plays intro sound finish.

After that you apply configure and save change. IVR will save configure and then you can edit again to assign each number has what action to do.

IVR	Descripti	on Intro Sound	Edit
5546	545	Intro.wav	2
111	11111	Intro.wav	I I I I I I I I I I I I I I I I I I I
main	main	Intro.wav	<u>→                                    </u>

Plextel Enterprise Version 2

From the above example, edit IVR to add number what to do.

Voice Menu [View Voice Menu] [Create V	Voice Menu]				64
	Voic	e Menu Details			
Name	main				
Description	main				
Allow direct call from this	s menu 🔽				
Default Language	English 💌				
Enable Menu Password					
Intro Sound	Intro.wav 💙				
Invalid Sound	Invalid.wav 🐱				
Timeout Sound	Goodbye 💙				
Absolute Timeout Sound	Goodbye.wav 💙				
Exit Sound	Goodbye.wav 💙				
Music On Hold	Default 🗸				
Absolute Call Timeout (se					
Wait For Response (sec)	15				
Wait For Additional Digit	(sec) 3				
No Input Max Repeat Tin					
Invalid Input Max Repea	,				
Direct Call Menu	None 🗸				
Automatic Dial Number	None 💌				
Number Priority	WhatToDO	Value	Edit	Delete	
1 1	Dial Group Dial Extension (serial)	Default%%Default%%Default 1000	¢.	<b></b>	
2	Diar Extension (senar)	1000			
			A	dd Number	
APPLY					

When you add number in Menu edit IVR, it has new window appear to you for setting what to do

🖉 ADD NUMBER - Windows Internet Explorer	
http://192.168.0.124/libs/ivr_nong_addnumber.php?name=main	
	^
APPLY	

Assigned number is in box for example, assign number 3 for do some action. It'll show following below

picture.

3 main	
AddWhatToDo	
APPLY	

When you add what to do once time, it shows display like the below picture.

3 main			
Hangup	~	8	
AddWhatToDo			
APPLY			

The select box has a drop down list. You can select many actions in list.



Each option is the action that you must assign to IVR. Before setting this, you must learn about details of all action because some actions need to use together. In this version, you can set many actions to relate with sound file. All action can be explain following this.

1. Dial Group:

3 main					
Dial Group	20	Ō	Default%%Default%%Default 💌	$\geq$	<b>`</b> ₩
AddWhatToDo					
On No Answer AddOnAnswer					
On Unavailable	1				
AddOnUnavailable	]				

This action is set to dial group by refer to group list. That time is the ringing timeout and the last box is to select group . Notice that it has 2 options. There are On No Answer(when no person receive call) and On Unavailable (When phone is disconnected). You can assigned it what to do when 2 events have occurred.

#### 2. Dial Extension(serial)

3 main		
Dial Extension (serial) 🔽 🎱 20	🗟 1000 🛩	🛛 🛅
AddWhatToDo	1000	
Addititation	1001	
	1002	
On No Answer	1003	
AddOnAnswer	1004	
Addonanarici	1005	
	1007	
On Unavailable	1008	
AddOnUnavailable	1009	
Addononavailable	1010	
APPLY	1119	
AFFLI	1144	
	2000	
	3000	
	3001 3002	
	3002	
	3004	
	6000	
	9000	
	9001	
	9002	
	9003	
	9004	
	9005	
	9006	
	9007	
	9008 9009	
	9009	

This action is to dial to specified extension that you assigned.

#### 3. Dial Extension (all)

3 main		
Dial Extension (all) 🛛 🖌 🙆 20	1000 🔽	😒 💼
AddWhatToDo	1000	
	1001	
	1003	
On No Answer	1004	
AddOnAnswer	1005	
	1006	
On Unavailable	1007 1008	
	1009	
AddOnUnavailable	1010	
APPLY	1119	
APPLT	1144	
	2000 3000	
	3001	
	3002	
	3003	
	3004	
	6000	
	9000 9001	
	9002	
	9003	
	9004	
	9005	
	9006	
	9007 9008	
	9008	

O O Chapter: Chapter 3 System Features

This action is liked a Dial Group but it assign the number and not refer group. For example, you assign number

1000, 2000, 3000 to ring at the same time. But those numbers are in different group.

3 main		$\frown$		
Dial Extension (all)	✓ ② 20	a 1000 💌		<b>D</b>
Dial Extension (all)	✓ ② 20	2000 🗸		Ē
Dial Extension (all)	✓ ② 20	3000 🗸		Ē
AddWhatToDo		$\smile$		
On No Answer				
AddOnAnswer				
On Unavailable				
AddOnUnavailable				
APPLY				
4. Go to Que	ue			
3 main				
Go to Queue	🗸 🎱 pull	🕾 Queue_A	gent 💌	🛛 🛅
AddWhatToDo				

on Quei	
AddOni	Finish
APPLY	
7	This actic

This action is to transfer call to queue. On right hand side will show queue list for selection. And another part that is On Queue Exit, it is to assigned action what to do when no person receive call in queue.

 Re-Dial: This action is to repeat dial this number again. Then it must follow the Dial Group or Dial Extension

3 main Dial Group	✓ ② 20	Default%%Default%%Default	✓ 🕅 🛅	68
Re-Dial 🔶	*		 [1]	
AddWhatToDo				
On No Answer				
AddOnAnswer				
On Invalid AddOnUnavailable				
	rence: This action is	to dial to conference room		
3 main				
Dial Conference	🗸 ビ null	¯(☎ 600 ᢦ ) 🛛 🖻		

AddWhatToDo	
On Finish AddOnFinish	
APPLY	

- 7. Leave Voicemail: This action is to leave your message or speech on system. Message will be in voicemail
  - box

3 main		
Leave Voicemail 🔽 🎯 null 🚳	1000 🛩	🛛 🔽
AddWhatToDo	1000	
Addwhatrobo	1001	
	1002	
On Finish	1003 1004	
AddOnFinish	1004	
	1006	
APPLY	1007	
	1008	
	1009	
	1010	
	1119 1144	
	2000	
	3000	
	3001	
	3002	
	3003	
	3004	
	6000 9001	
	9002	
	9003	
	9004	
	9005	
	9006	
	9007	
	9008 9009	
	3003	

8. Playback Sound: this action is to play sound file. You must record your sound(sound -> Create voice) in

plextel before set this.

3 main				
Playback Sound	🖌 🎯 null	ō	1000 💌	🛛 🛅
AddWhatToDo			1000 Intro Goodbye	
On Finish			Dialing Invalid	
AddOnFinish			NoInput test	
APPLY				

- 9. Hangup: This action is to hang up or drop call.
- 10. Repeat Menu: This action is to repeat the last action again.
- 11. Goto Menu: This action is to jump to another IVR. The right hand side shows IVR list.

3 main			
Goto Menu	🔽 🎱 pull	5546 ⊻	🛛 🛅
AddWhatToDo		5546	
		111 main	
APPLY			

- 12. Goto Previous Menu: This action is back to the previous menu
- 13. Set Language: This action is to switch languages to set as default

3 main				
Set Language	🔽 🙆 pull	🙆 Ei	nglish 💌	🛛 💼
AddWhatToDo			nglish	
			hai Jstom1 -	
On Finish			ustom2	
On Finish				
AddOnFinish				
APPLY				

- 14. Announce Call Number : This action is to announce number that caller enters it before do the next step. For example, you set number 3 for this action, system will play sound "You press number 3"
- 15. Wait: this action is to wait the user's number who cannot receive call at this time. When it reach a timeout, it will do the next action.

3 main			
Dial Extension (serial)	✓ ○ 20	🗟 1000 🚩	🛛 🛅
Wait	✓ ∅ 20		🖄  🛅
AddWhatToDo			
On Finish			
Re-Dial	▼		
110 511			
AddOnFinish			

16. Follow me: this action is to set system jump to follow me that is created in extension menu (Extension ->

Add Follow Me Extension). It has a select box to select the extension to set follow me action.

Follow Me 🖌 🎯 pull AddWhatToDo	none v	🖂 🛅
AddWhatToDo		
	1000	
	1001	
	1002	
n Finish	1003	
Re-Dial 💌 💟	1004	
AddOnFinish	1005	
	1006	
APPLY	1007	
	1009	
	1010	
	1119	
	1144	
	2000 🔳	
	3000	
	3001 3002	
	3002	
	3003	
	6000	
	9000	
	9001	
	9002	
	9003	
	9004	
	9005	
	9006	
	9007	

17. Transfer (PABX) : this action is to transfer call to PABX from IVR. In this case, you must connect PABX

with plextel server and assign destination number on the right-hand side tab as the below picture.

1001

18. Call Custom App: this action is to use the additional function. You can add new feature at Menu Call *Features -> New Features* and select feature on right drop down box.

3 main Call Custom APP AddWhatToDo	V 🕑 pull	👋 test ivr	<b>V</b> 🛛 🖬	
On Finish Re-Dial AddOnFinish	<b>v</b>			

- 19. Goto Custom App: this action is liked as Call Custom App but it's different at the call that go to custom app, it cannot return to IVR .
- 20. **Dial External Line:** this action is to dial to the external line. You must assign trunk to call out as the below picture.

3 main	
Dial External Line 🛛 🎱 🙆 20	🖬 trunk%%sip%%asdf 🔽 📓 💼
AddWhatToDo	test ivr trunk%%sip%%asdf trunk%%dahdi%%test
On Finish Re-Dial 💌 💟	
AddOnFinish	
APPLY	

21. Read DTMF: this action is to read the number that caller enter.

To Do														
Read DTMF	• т	VAR1	•	Playback	Intro	<ul> <li>MaxDigit</li> </ul>	unlimited 🔻	Attempts	0 -	Timeout	default	•	8	
AddWhatToDo														

Select "To Do" as a Read DTMF

To: You select the variable to keep value

**Playback:** is the sound to announce when caller into this menu such as "please enter your member's number and press # when you finish"

Max Digit: it is the maximum digit that caller can enter

Attempts: is the number of time that callers can enter their number. IVR will repeat this menu to prompt record

your number again.

Timeout: is the during time that IVR will count when playback sound finish playing

#### 22. Check Database

Description: 1		$\mathbf{)}$
To Do     Check Database     NONE      ARG1 NONE     ARG2 NONE     ARG3 NONE     Variable     VAR1       AddWhatToDo	8	
- select your database		
- ARG1-3 are the data that you want to compare with database		

- Variable is the value that it returns

I.

23. Verify Variable (not exist) is to verify the value that you want to check it

To Do	
Verify Variable (not exist)  Variable (not exist)  Variable (not exist)  Variable (not exist)	To-do Hangup   RetryBeforeExit 0   Playback(exit) NONE   To-do(exit) Hangup
AddWhatToDo	

Name: is the name of variable that you want to check it

Playback: is the sound file that be played when the value of variable is null

To-Do: is the next action will do when finish checking value of variable if the value is not null.

RetryBeforeExit: is the time of checking value

Playback(exit): is the sound file that IVR will play when it check value completely.

To-do(exit): After finish checking value, you can set IVR to do next action at this.

23. Playback Variable: IVR will play value that keep in variable

To Do						_	_
Playback Variable	Name	VAR1	<b>•</b>	Туре	Say Digits 🔹	$ \otimes $	Ē
AddWhatToDo					Say Digits		
Addimacrobo					Say Number		
					Playback Sound		
					Say Alpha		
					Say Phonetic		

Name: is the variable that you want to announce the value

Type: is the type of value. For example, if the value is number you must select "Say Number" if the value is digits, you must select "Say Digits".

24. Label Marking: is the marking position in IVR. You can mark position that you want the IVR return to

play at this step.

To Do Label Marking	<ul> <li>Nam</li> </ul>	Label1	•	$\otimes$	<b></b>	
AddWhatToDo						
Nama: is the name of	fLabal					

Name: is the name of Label

25. Goto Label Condition is the action that jump to Label if the condition is true or false.

To Do										
Goto Label condition 🔹	When VARIABLE	VAR1 -	Equalto	Yes-Goto	Label1	• N	o-Goto	Label1	•	💟 🛅
AddWhatToDo										
When Variable: choose the variable that you want to check in condtion

Equal to: is the value that you want to compare with variable

Yes-Goto: if the condition is true, IVR will go to Label that you assigned

No-Goto: if the condition is false, IVR will go to Label that you assigned

26. SayDate Time: IVR will play date and time

- To Do					
				1	
SayDateTime	•	Туре	Current time 🔻	Format aebY	
AddWhatToDo			Current time		
			VAR1		
			VAR2		
			VAR3		
			1404	1	

Type: is the variable that you want to play

Format: is the date and time format

a = day, e = date, b = month, Y = year

27. **Background Sound:** IVR will play selected sound file. The background sound is different from the playback sound in the during time that you can enter the number. As background sound, you can press number while the background sound is playing but in playback sound you cannot press the number until playback sound finish playing.

Background Sound	▼	🗟 Goodbye 🔻	🛛 🛅
AddWhatToDo			
28. FAX: send fa	x to the selected numbe	r	

To Do					
FAX	•	To number	NONE -	$\gg$	<u>ت</u>
AddWhatToDo					

**4.3 Incoming Call Setting:** incoming call setting is an action that manage what to do when it has incoming call come to system. In menu incoming call at left hand side, it shows as the below picture.

ncoming Cal		ng Call ]					
isplay 1 - 1 from 1		-					
Trunk		Des	tination	DID	Description	Edit	
🗌 trunk:dahdi:test	0 time	queue	Queue_Agent	-		Ż	
age 1:							
Delete Selected	Cancel						

Select Add New Incoming Call, you will see this page.

Incoming Call [View Incoming Call ] [ Add New	v Incoming Call ]
Add Incoming Call	
Trunk	trunk:dahdi:test 💌 🗖 PABX-link
Description	
Support DID	yes 🕶
Incoming DID	
Replace CallerID	
Extensions Ring Timeout(s	ec) 40 💙
Add time based handler	Actions
Value	e 🗸 Destination* Select
Remove	
APPLY Cancel	

Trunk : is shown the trunk list to select trunk that has incoming call pass through as the below picture.

Add Incoming Call		
Trunk	trunk:dahdi:test 💟 🗖 PABX-link	
Description	trunk:dahdi:test	
Support DID	in:trunk:sip:asdf gateway:sip:test_sys	
Incoming DID		

From Example, You will see trunk list and you can select only one trunk.

PABX-link: when you check in check box in this PABX-link, Plextel must connect to PABX.

Description: is to describe the details of this incoming call

Support DID: if you select "yes", you must have a number to match with incoming call.

**Replace Caller ID:** enter the number in this box for replace the number that show on system or don't want to show the called number.

Extension Ring Timeout(sec): is the ring timeout of Extension (sec)

Add time based handler: when you press this, the time base handler will appear in this page. It use this function when you want to do different action in the different period of time. For example, Incoming call will go to IVR on office hour and goes to voicemail on out of office hour.

When: is to select the schedule list to set action

75

**Destination and Value:** two parameters are related each other. When you select different destination, the value will be shown the different destination value.

After that, you press add incoming call. WUI will be refreshed to the main page of incoming call interface and show your incoming call.

## Chapter 5 System Management

5.1 Manage System Administrator is System of Management to set the username and password of all user. The person who can access to create and edit this part of system, must be the Administrator only. And You should set the permission of user cannot allow to edit configuration. The below picture will shows login page

E	Username ::
77-11-	
	Password ::
	Login Clear

If you login incorrect username and password, you cannot access to configure it. The default username and

### password are

Username: admin

Password: plextel

### 5.1.1 user Manager

You can set or edit the new username and password at menu Advance->Manage Admin like as this example

### picture.

Web Username	Phone Number	Phone CallerID	Extensions	Web User type	Enable CRM	CRM User Type	Edit
monitor				Monitor			2
user				User			Ż
admin				Administrator	Enable	Administrator	2
1008	1008	1008	SIP/1008	User			Ż
1007	1007	1007	SIP/1007	User			2
1006	1006	1006	SIP/1006	User			Ż
1005	1005	1005	SIP/1005	User			Ø.
1004	1004	1004	SIP/1004	User			Ż
2000	2000	2000	SIP/2000	User			4
1003	1003	1003	dahdi/5	User			Ż
1002	1002	1002	IAX/1002	User			4
1001	1001	aaaa	SIP/1001	User			Ż
1010	1010	1010	SIP/1010	User			Į.
1009	1009	1009	SIP/1009	User			Ż

If you want to edit the username and password, you click at command. Then you can edit that user as below ure.

16

### **Users Manager**

[View users] [Add us Add user	ser ] [ Copy users from extensions ]
Select Level	User 💌
Web Username	
Password	
Re-entry Password	
Phone Number	None 💌
Enable CRM	No 💌
CRM User Type	Normal User Type 💌
APPLY	ncel

Select Level: it has 3 level to setting for each user – Admin, Monitor, and User. Each level can access different level web page. Admin can access all pages, and user can access some page only such as voicemail.

Web Username: is the name that use to login web page. You should set username same as extension to easy manage.

Password: Password to login web page

Re-entry Password: enter same password again

Phone Number: the extension number of user

Enable CRM: allow user to use CRM

CRM User Type: Select CRM level

### 5.1.2 Change Password

When you login to plextel by user level, it shows some menu only. But you can edit your password at menu Change Password (on Top tab bar) and you can also change your voicemail's password.

[]]

192.168.0.125/libs/change 192.168.0.125/libs/change	
Web Access Password	
Current Password	:
New Password	:
Confirm Password	:
Voicemail Password	
Password	
	APPLY Cancel

If you want to add new users, you can do it at menu Add Users

User Manager [View users] [Add users] [Copy users from extensions] Add users							
Select Level	User 💌						
Users							
Password							
Re-entry Password							
APPLY Cancel							

When you add new user. System will show the new one as this below picture.

78

### Users Manager

79

٢	View users ]	[ Add	user 1	Copy	users	from	extensions	1
۰.				L /				

Web Username	Phone Number	Phone CallerID	Extensions	Web User type	Enable CRM	CRM User Type	Edit
monitor				Monitor			Ż
🔲 user				User			Ż
admin				Administrator	Enable	Administrator	Ż
1000	1000	1000	SIP/1000	User	Enable		Ø.
1001	1001	1001	SIP/1001	User	Enable		4
1002	1002	1002	SIP/1002	User	Enable		Ø.
1003	1003	1003	SIP/1003	User	Enable		Ż
1004	1004	1004	SIP/1004	User			Ż
1005	1005	1005	SIP/1005	User			Ż
1006	1006	1006	SIP/1006	User	Enable		Ż
1007	1007	1007	SIP/1007	User			Ż
1008	1008	1008	SIP/1008	User	Enable		Ż
1009	1009	1009	SIP/1009	User			Ż
1010	1010	1010	SIP/1010	User			Ż
5000	5000	5000	SIP/5000	User			Ż
5001	5001	5001	SIP/5001	User			Ż
5002	5002	5002	SIP/5002	User			2
5003	5003	5003	SIP/5003	User			Ż
5004	5004	5004	SIP/5004	User			Ż
2010	2010	2010	SIP/2010	User			Ø,

Add user in menu user manager, is to set the permission to edit the data in the future. At this time, the user can allow to set the value as the admin only.

**5.2 LAN Network Setup** is to set the IP Address of server. You need to set it as static IP because all IP Phone must be registered to this server. You can set IP Address at menu *Advance->Network Setting* as the below picture.

Network Setting [LAN] [WAN] [Static Routing Table [eth0] [eth1 (no device)] TCP/IP SETUP (eth0)	] [DNS ] [DDNS ] [DHCP Service ] [Firewall Service ] [NTP Service ]
Mode	Static Setting 💌
IP Address	192 🗙 . 168 🗙 . 0 🗙 . 124 🗙
Subnet Mark	255 🗙. 255 🗙. 255 🗙. 0 😵
Gateway	192 🗙 . 168 🗙 . 0 🗙 . 22 🗙
APPLY	

After you finish setting or editing IP address. It has the red tab on screen to restart network.

Please Click here to restart Network

**5.3 WAN Network Setting** is to set the WAN IP address for the phone that is on the other place or different network to register to this server. You must set the real IP address that is given by ISP (Internet Service Provider). The advantage of the real IP address is the transferring voice signal is working most efficiency because it does not pass through NAT or Firewall. Then you must set real IP address at server and set the ADSL Modem as the bridge mode to forward packet to server. You can set username and password that ISP give it to up at this menu *Advance -> WAN Network Setting* 

		-OO
Network Setting		
[LAN] [WAN] [Static Routing Table]	[DNS][DDNS][DHCP Service][Firewall Service][NTP Service]	
WAN Network Setup		
ISP Login Name (ADSL Account)	plextel@truehisp	
Password	•••••	
Start at Boot	C Yes 🖲 No	
APPLY Cancel Start WAN	Stop WAN	

From this example, It set the username from TRUE(ISP). Then you enter the username, password and apply

this. It will activate immediately.

5.4 HA Setup is to backup system of server to have more security. It's necessary to have the another server to reserve

when the main server has something wrong. The reserved server must has same configuration as the main server.

High-Availability Configu	uration
Server Type	Master      Slave
Virtual Server IP-Address	22 . 22 . 22 . 22
Optional Configuration	
Broadcast Interface	eth0 💌
Check Interval Time (ms)	200
Declare Dead Time (Second)	2
Warm Time (Second)	1
Init Dead Time (Second)	120
Auto Failback	○ On ○ Off
Enable Debug log	○ on ○ off
Enable log	• On C Off
APPLY Cancel Start HA St	top HA Default

You can set this HA at menu Advance -> HA Setup

From this example, You can assign the virtual server IP-Address and select type of server. For main server is selected as Master and reserved server is selected as slave. Then you must enable and apply it. You must reboot both of them before HA setup works.

5.5 Firewall/NAT Setup is the setting about the security of system. You should set this part when you allow clients or IP-Phone can register to server. You set this part at *Advance->Network Setting-> Firewall Service* 

	work Se					
		tic Routing Table ] [ DN	S][DDNS][DHCPS	Servi	te ] [ Firewall Service	] [NTP Service ]
		/ NAT Setup				
	ABLE Firewal					
		AN (PPPOE)				
	w Access S					
	ow Access I/	4X2				
	w Access R	TP				
	w Access N	TP				
	w Access W	/EB-Interface				
	w Access T	erminal				
		BX Manager				
Servic	e from LA	N (Ethernet)				
	w Access S	IP				
	w Access I/	AX2				
	w Access R	TP				
	w Access N	ТР				
		/EB-Interface				
	w Access To					
		BX Manager				
	Access Fi	barranager				
NAT S	etup (Sou	rce NAT)				
	ble Outgoin					
	-	-				
Interr	nal Server	Setup (Destination	on NAT)			
Enable	TCP/UDP	WAN Port-Number	Server IP-Addres	55	Server Port-Num	ber
	•/C			1		
	$\circ$ / $\circ$		1		1	
	$\circ$ / $\circ$					
	0/0					
	010					
	$\circ$ / $\circ$		1			
	010			1		
	0/0					
	0/0		1		1	
	0/0					
	010					
	010					
	$\circ$ / $\circ$		1			
	010					
	0/0					
	0/0		1			
	010			1		
	0/0					
	0/0	1	1			
	010					
	010			1		
_						
	$\circ$ / $\circ$					
	010					
	0/0			1		
Apply	Cancel	Default				

Description of these values are following this:

- Enable Firewall: click this to enable firewall
- Service from WAN(PPPoE): service list that allow to access from outside
- Service from LAN(Ethernet): service list that allow to access from inside
- NAT Setup: is the NAT setting
- Internal Server Setup: is to set up port forward to login to another server from outsite

5.6 DNS Setup: DNS setup can set at menu Advance -> Networking Setting -> DNS

Network Setting								
[LAN][WAN][Static Routing Table][DNS][DDNS]								
DNS Setup								
Name Server 1 203.144.207.29								
Name Server 2 202.6.100.1								
Name Server 3								
Name Server 4								
APPLY								

T 8 Chapter: Chapter 3 System Features

ΧI

5.7 Dynamic DNS: Dynamic DNS will use when your office has no real IP Address. You must register with Dynamic DNS Provider before you set this part. Then you will get the username and password and you can set in this menu *Advance -> Networking Setting -> DDNS* 

Network Setting [LAN][WAN][Static Routing Table]	[DNS][DDNS][DHCP Service
Dynamic DNS	
Active	
Service Provider	www.dyndns.org 🛛 💙
Host Name	
E-mail Address	
User	
Password	
🗆 Enable Wildcard	
APPLY Cancel	

5.8 DHCP Server: is to set server to auto configure IP Address to clients or other devices. You can set DHCP at this

menu Advance -> Network Setting -> DHCP Service

Network Setting [LAN] [WAN] [Static Routing Table] [DNS] [DDNS] [DHCP Service] [Firewall Service] [NTP S [DHCP Server Configuration] [DHCP Leased Status] DHCP Server Configuration Image: ENABLE								
Subnet Value	192.168.0.0							
Domain Name	plextel							
SubnetMask Value	255.255.255.0							
Gateway IP Address	192.168.0.22							
Primary DNS Server	203.144.207.29							
Secondary DNS Server	203.144.207.49							
NTP Server	192.168.0.22							
Option tftp-server-name	192.168.0.124	* Required For AutoProvision						
Option bootfile-name	autoprovision/snom.htm							
Client Start IP Address	192.168.0.180							
Client End IP Address	192.168.0.240							
APPLY Cancel								

- Subnet Value: IP Address Network
- Domain Name:
- Subnet Mask Value: Subnet Mask
- Gateway IP Address: You must set your IP Address of Gateway same as IP Address of Server IP-PBX
- Primary DNS Server
- Secondary DNS Server
- NTP Server: set it same as IP Address of Server IP-PBX to synchronize time to server.
- Option TFTP-server-name: the computer that share file to client (set IP Address f server IP-PBX)

- Client Start IP Address: Start IP Address to automatic configure
- Client End IP Address: Last IP address to automatic configure

5.9 Backup & Restore: This feature is to save and restore configuration of system. You can backup or restore at menu

Advance -> Backup & Restore

Backup & Restore Manager [Backup][Restore][Restore Factory Default] Last backups									
backup-2010-06-2908-25.tar.gz 29/06/2010 20:25 Download   Delete									
Create new backup Cancel Comment									

From the example, when you click the create new backup button, it'll create the last backup file and file name is the date

and time of file when it is created. You can download file to save as another location on your computer for using restore

when system has problem or something wrong.

Backup & Restore	e Manager
[Backup] [Restore] [Restore Last backups	Factory Default ]
	gz 29/06/2010 20:25 Download   Delete
Create new backup	Cancel
Comment	
	<b>v</b>

You can reset all configuration by click at Restore Factory Default. All configuration will be deleted.

5.10 License Management: is to add license into system. 1 license is 1 extension. At first license will be add following

customer's requirement. If customer wants to add license, you can contact to sale. You can add license at menu Advance-

> License Management

License Setup Serials Number: 00P0-27L0-0EX0-T013-L047-P028
License Details : 16/06/2010 Client : 100 Remark: demo
License Details : 16/06/2010 Client : Remark:
License Details : 14/06/2010 Client : Remark:
Total License: 100 License File Browse APPLY

From the above picture, it shows number of license is 100 licenses. If you want to add more license, you contact to sale of poise technology and they will send license to you.

## 5.11 Call Details Record(CDR)

CDR is from Call Details Record, that is the vital part of VoIP telephone system because it records all details about the telephone system such as incoming call, outgoing call. You can see this menu at upper tab of plextel software

Page 1 👻 😥 🕽														
lo Call Date	Description	Direction	Through	Value	Last App	CallerID	From Number	To Number	Channel	Duration	Billseo	Disposition	Disposition (details)	Group/Cor
2010-0 1 30 11:33:	Normal_Call	Internal			Hangup	"1003" <1003>	1003	1002		50	19	ANSWERED	NOANSWER	Default%% Default%% Default
2010-0 2 30 11:32:	Normal_Call	Internal			Dial	"1006" <1006>	1006	1009	SIP/1006-00000008	3	0	NO ANSWER	CANCEL	Default%% Default%% Default
2010-0 3 30 11:32:	Normal_Call	Internal			Hangup	"1006" <1006>	1006	100	SIP/1006-00000007	12	12	ANSWERED		Default%% Default%% Default
2010-0 4 30 11:31:	Normal_Call	Internal			Dial	"1006" <1006>	1006	1009	SIP/1006-00000005	1	0	NO ANSWER	CANCEL	Default%% Default%% Default
2010-0 5 30 11:30:	Normal_Call	Internal			Dial	"1002" <1002>	1002	1003	SIP/1002-00000002	10	8	ANSWERED	CANCEL	Default%% Default%% Default
2010-0 6 30 11:30:	<sup>6-</sup> Normal Call Leave <sub>35</sub> Voicemail	Internal	Features	Voicemail	VoiceMail	"1002" <1002>	1002	1002	SIP/1002-00000000	6	4	ANSWERED		Default%% Default%% Default
2010-0 7 30 11:11:	6- Attended_Transfe	Internal			Hangup	1002	SIP/1001	SIP/1003	Local/1003@Default% %Default%%Default- 020f,2		3	ANSWERED	CHANUNAVAII	Default%% Default%% Default
2010-0 8 30 11:11:	Attended_Transfer	Internal			Dial	1002	SIP/1001	SIP/1002	Local/1002@Default% %Default%%Default- 3142.2		0	NO ANSWER	CANCEL	Default%% Default%% Default
2010-0 9 30 11:10:	Normal_Call	Internal			Hangup	"1002" <1002>	1002	1001	SIP/1002-0000001b	73	19	ANSWERED	NOANSWER	Default%% Default%% Default
	<sup>6-</sup> Normal Call Leave <sub>39</sub> Voicemail		Features	Voicemail	VoiceMail	"two" <1001>	1001	1002	SIP/1001-00000019	50	19	ANSWERED		Default%% Default%% Default
2010-0	<sup>6-</sup> Normal Call Leave <sub>49</sub> Voicemail	Internal	Features	Voicemail	VoiceMail	"two" <1001>	1001	1002	SIP/1001-00000017	41	36	ANSWERED		Default%% Default%%

From this above picture, is shows all details. If you want to specify some detail you can use data filter

a Filter						
Time Range	۲	Jun-2010		🗸 - Jun-2010	~	
	$^{\circ}$	Select Date		- Select Date		
Description	AL	L	*			
Direction	AL	L	*			
Group	AL	L	*			
Incoming Type	NC	DNE	*			
Outgoing Trunk	NC	ONE	*			
Source Number						
Destination Number						
Call Duration	AL	L	*		Seconds	
Disposition	AL	ALL				
Disposition(details)	AL	ALL				
Last App	AL	ALL				
Context						

Time Range is the range of time that you want to check details of record

Time Range	January-2010	*	- Jun-2010	*
	C 2010-06-02		. 2010-06-15	
Description	ALL	*		
Direction	ALL	*		
Group	ALL	*		
Incoming Type	NONE	*		
Outgoing Trunk	NONE	*		
Source Number				
Destination Number				
Call Duration	ALL	~		Second
Disposition	ALL	*		
Disposition(details)	ALL	*		
Last App	ALL	*		
Context				

You can set description at this



## Set Direction

Data Filter	
Time Range	<ul> <li>January-2010</li> <li>Jun-2010</li> <li>2010-06-02</li> <li>2010-06-15</li> </ul>
Description	ALL
Direction	ALL 🕑 🔶
Group	ALL IN
Incoming Type	OUT
Outgoing Trunk	INTERNAL
Source Number	
Destination Number	
Call Duration	ALL Seconds
Disposition	ALL 💌
Disposition(details)	ALL 💌
Last App	ALL 💌
Context	
	APPLY

Set Group

ta Filter			
Time Range	G January-2010	🗸 - Jun-2010	*
	C 2010-06-02	_ 2010-06-15	
Description	ALL	*	
Direction	ALL	*	
Group	ALL	✓	
Incoming Type	ALL Default%%Default%	26.06	
Outgoing Trunk	eng%%poise%%te		
Source Number			
Destination Number			
Call Duration	ALL	▼	Seconds
Disposition	ALL	*	
Disposition(details)	ALL	*	
Last App	ALL	*	
Context			

## Set Incoming Type



Set Outgoing Trunk

81

Data Filter			
Time Range	January-2010	🗸 - Jun-2010	*
	C 2010-06-02	. 2010-06-15	
Description	ALL	*	
Direction	ALL	*	
Group	ALL	*	
Incoming Type	NONE	*	
Outgoing Trunk	NONE	✓	-
Source Number	NONE pstn		
Destination Number	kkk		
Call Duration	sip		Seconds
Disposition	ALL	*	
Disposition(details)	ALL	*	
Last App	ALL	*	
Context			
		APPLY	

Menu CDR setup is to set the result of CDR detail. You can select to show some value from this menu

								V
lome   View CDR   Data F	ilter   Setup	Logout						
	Select field t	o show on	CDR windo	ws				
		~	~	~	•	~		
	Call Date	Direction	Description	Through	Value	IVR Details	Last App	CallerID
		•			~			
	From Number	To Number	Channel	Duration	Billsec	Disposition	Disposition(details)	Group/Context
								APPLY
	CDR Databas	se Manage	ment					
	Select Time	e Range	_					
	C Date F	Range Sta	rt Date: S	Select Da	ate		End Date:	
	Select I	Date						
	Last bac	kups						
								APPLY

Call Date: shows date and time

Direction: shows calling status. There are 3 typex

- Internal call between extension,
- Incoming when receive call from external line
- Outgoing Call out by external line

 $\infty$   $\infty$  Chapter: Chapter 3 System Features

Description: shows details of direction

Through: shows working type

Value: shows through details

IVR Details: shows values of IVR system

Last App: show the last calling

Caller ID: shows incoming number

From number: shows the caller number

To number: shows the destination number

Channel: shows call routing

Duration: shows time duration from you enter the number to hang up call

Billsec: shows time duration from destination receive call to hang up call for calculating cost

Disposition: shows call status

Disposition(Details): show call status details

Group/Context: shows group name list

The low part of CDR database management is to backup database of CDR. You can select all or select by time

duration.

**5.12 Auto Provision:** You can set auto provision at menu *Advance -> Auto Provision* 

age 1	- 33 51				
Enable	Extensions	MAC-Address	PhoneType	Allow FW- Update	
-	1000 on Thousand		none	-	📝 🚳 X
1	1001 two	000413235C51	snom360	1	📝 🚳 🕽
-	1002 1002		none	-	2 🗟 🕽
-	1003 1003		none	-	📝 🚳 🕽
1	1004 1004	0004132F9BB3	snom300	✓	📝 😡 🕽
-	1005 1005		none	-	📝 🚳 🕽
-	1006 1006		none	-	📝 👰 🕽
-	1007 1007		none	-	📝 🐼 >
-	1008 1008		none	-	📝 🐼 🕽
1	1009 1009	001565115C69	yealinkt28	1	📝 🖗 🕽
1	1010 1010	0123456789BB	yealinkt26	1	📝 🐼 🕽
1	1119 1119	0123456AAABB	yealinkt22	1	📝 🐼 🕽
1	1144 1144	AAA111222BBB	yealinkt20	1	📝 🐼 >
1	2000 Mr. TTwo Thoudsand	AAAAAAAAAAA	yealinkt18	1	2 🗟 🕽
1	3000 3000	BBBBBBBBBBBBB	yealinkt12	1	2 🗟 🕽
-	3001 3001		none	-	2 🐼 🔊
-	3002 3002		none	-	2 🐼 🕽
-	3003 3003		none	-	200

Auto Provision is to configure IP-Phone by using plextel register. This feature supports some IP-Phone series. For this page, you can upload firmware for updating IP-Phone (look at 1.2.1 add sip extension)

From above picture, it shows phone list that activate auto-provision. If you have new firmware and you want to update to your IP-Phone. You go to advance setup and it shows display as below picture

90	9
----	---

Advanced Setup		
[ Phone Provisioning ] [ Advanced Se	etup ]	
Firmware Selection		
SNOM300	Browse	Upload
SNOM320	Browse	Upload
SNOM360	Browse	Upload
SNOM370	Browse	Upload
SNOM820	Browse	Upload
SNOM821	Browse	Upload
SNOM870	Browse	Upload
Yealink T12	Browse	Upload
Yealink T18	Browse	Upload
Yealink T20	Browse	Upload
Yealink T22	Browse	Upload
Yealink T26	Browse	Upload
Yealink T28	Browse	Upload
SNOM Phone Update policy		auto_update 👻
SNOM Setting Refresh Timer mins	120	
Phone Username For SNOM		admin
Phone Password For SNOM		0000
Phone Interface Password For SNO	0000	
Yealink Phone update policy		Power on + Repeatedly -
Yealink Setting Refresh Timer min	s (for Repeatedly rules)	1
Admin Password for Yealink		0000
APPLY Note: This features required functional set to http://< ipaddress >/autoprovis		on external DHCP server, DHCP option 66 must be

You can browse firmware for each IP-Phone series that you want and upload it. After you reboot your IP-

Phone, it automatic update firmware.

### 5.13 SSH Terminal

This feature is in menu Advanced -> SSH Terminal





This menu uses for remote to server by using SSH to run linux command. The PC or Notebook that open this

page, must have Java Runtime on theirs. After open this page, you must input username and password to access this.

### CLI

This menu is at Advanced -> CLI

• /	Advanced
	PBX Advanced Setting
	Network Setting
	Manage Admin
	Screen pop-up Account
	HA Setup
	SSH Terminal
	CLI
	Auto Provision

This menu uses asterisk command to manage it. This menu for user who have asterisk knowledge but cannot login to system. You can run command by using this menu. For example, You want to show which extensions are online in plextel. You run command "sip show peers" to show that.

# Chapter: Chapter 3 System Features

plextel*CLI > sip sh								
Name/username	Host		Dvp	Nat	ACL Por	t Status		
4000	(Unspecified)	D	Dyn	0		nonitored		
sipsite_dsdsd/dsdsd	(Unspecified)			U	5060	UNKNOWN		
sipsite_ususu/ususu sipsite_xxx/xxx	(Unspecified				5060	UNKNOWN		
trunk%%sip%%asdf				N			ad	
9009/9009	192.168.0.20		D	IN	5060	OK (87 ms)	su	
9008/9008	192.168.0.21		D		5060	OK (32 ms)		
9007/9007	192.168.0.21		D		41056	OK (106 ms)		
9006	(Unspecified)	D		0		(NOWN		
9005	(Unspecified)	D		ŏ		KNOWN		
9004	(Unspecified)	D		õ		KNOWN		
9003/9003	192.168.0.21		D		5062	OK (54 ms)		
9002	(Unspecified)	D		0		KNOWN		
9001	(Unspecified)	D		0		KNOWN		
3004	(Unspecified)	D		0	UNI	KNOWN		
3003	(Unspecified)	D		0		KNOWN		
3002	(Unspecified)	D		0	UNI	KNOWN		
3001	(Unspecified)	D		0	UNI	KNOWN		
3000/3000	192.168.0.21	16	D		24671	OK (106 ms)		
1144	(Unspecified)	D		0	UNI	KNOWN		
1119	(Unspecified)	D		0	UNI	KNOWN		
1010	(Unspecified)	D		0	UNI	KNOWN		
1009/1009	192.168.0.19	90	D		5062	OK (67 ms)		
1008/1008	192.168.0.18	34	D		38671	OK (108 ms)		
1007	(Unspecified)	D		0	UNI	KNOWN		
1006/1006	192.168.0.23	36	D		5060	OK (14 ms)		

## 5.14 Asterisk CLI

Go to menu Advanced →Asterisk CLI

~/	Advanced
	PBX Advanced Setting
	Network Setting
	User Manager
	Screen pop-up Account
	HA Setup
	SSH Terminal
1	Asterisk CLI
	Auto Provision
	External Storage
	Update Version

This menu uses for the system administrator to access the shell from webpage. User can use to this shell and input the command and getting display from this shell. Eg. Input "sip show peers" for display the online extension.

## Chapter: Chapter 3 System Features

92

plextel*CLI >								execut
plextel*CLI > sip sh								
Name/username	Host		Dyn		ACL Por			
4000	(Unspecified)	D		0		nonitored		
sipsite_dsdsd/dsdsd	(Unspecifi				5060	UNKNOWN		
sipsite_xxx/xxx	(Unspecified				5060	UNKNOWN		
trunk%%sip%%asdf				Ν			red	
9009/9009	192.168.0.20		D		5060	OK (87 ms)		
9008/9008	192.168.0.2		D		5060	OK (32 ms)		
9007/9007	192.168.0.2		D		41056		)	
9006	(Unspecified)	D		0		KNOWN		
9005	(Unspecified)	D		0		KNOWN		
9004	(Unspecified)	D		0		KNOWN		
9003/9003	192.168.0.2		D		5062	OK (54 ms)		
9002	(Unspecified)	D		0		KNOWN		
9001	(Unspecified)	D		0		KNOWN		
3004	(Unspecified)	D		0		KNOWN		
3003	(Unspecified)	D		0	UNI	KNOWN		
3002	(Unspecified)	D		0	UNI	KNOWN		
3001	(Unspecified)	D		0	UNI	KNOWN		
3000/3000	192.168.0.2		D			OK (106 ms	)	
1144	(Unspecified)	D		0	UNI	KNOWN		
1119	(Unspecified)	D		0	UNI	KNOWN		
1010	(Unspecified)	D		0		KNOWN		
1009/1009	192.168.0.19		D		5062	OK (67 ms)		
1008/1008	192.168.0.1	34	D		38671	OK (108 ms	)	
1007	(Unspecified)	D		0	UNI	KNOWN		
1006/1006	192.168.0.23	36	D		5060	OK (14 ms)		
1005	(Unspecified)	D		0	UNI	KNOWN		

## 5.15 Screen pop-up Account

94 This menu uses for create the account to use with Screen pop-up API Go to menu Advanced -> Screen pop-up Account then select "Create Screen pop-up Account"

Screen pop-up Accou	nt	
[ View Screen pop-up Accourt Screen pop-up Account I ENABLE	nt ] [ Create Screen pop-u;	o Account ]
Username	ABCD	
Password	1234	
Permit Range	-	
Network ID	0.0.0.0	
Subnet Mask	0.0.0.0	
APPLY Cancel		

CheckENABLE box and specifyUser Name and Password for thisAccount, then click APPLY. You will get an account for theScreen pop-up API. You can set that program and login to use it.

Chapter:

## **Chapter 6 System Monitoring**

This chapter, we will introduce the feature for displaying the status and monitoring the system of Plextel IP-PBX. This feature can facilitate the system administrator to manage system easily form the web-interface.

**6.1 Show Status:** Go to menuStatus ->Show Status. This screen will show the real-time status of PlextelIP-PBX

- There are 3 parts in this page
- 1. Network Usage: Shows the status of data transfer (sent/receive/lost) of the network
- 2. Memory Usage: Shows the status of the main memory using in the system
- 3. Mounted Usage: Shows the status of harddisk storage in the system

Network Usage Device		Received	Sent		Err/Drog
lo		980.78 MB	980.78 MB		0/0
eth0		1.24 GB	3.04 GB		0/0
w1g1		3.16 GB	3.16 GB		0/0
sit0		0.00 KB	0.00 KB		0/0
Memory Usage					
Туре	Percent Capac	ity	Free	Used	Size
Physical Memory		77%	228.37 MB	766.11 MB	994.48 MI
- Kernel + applications	30'	%		295.93 MB	
- Buffers	<b>—</b> 16%			160.99 MB	
- Cached	31	%		309.19 MB	
Disk Swap	0%		2.00 GB	128.00 KB	2.00 Gł
Mounted Filesystems					
Mount	Type Partition	Percent Capacity		Free Us	ed Size
/	ext3 /dev/sda2	10% (1%)	24.0	3 GB 2.88 I	GB 28.38 GE
/var	ext3 /dev/sda5	0 1% (1%)	106.8	5 GB 1.06 ·	GB 113.78 GB
/boot	ext3 /dev/sda1	5% (1%)	258.0	8 MB 15.65 I	MB 288.63 MB
/dev/shm	tmpfs tmpfs	0% (1%)	497.2	4 MB 0.00	KB 497.24 MB
/var/spool/asterisk/monite	orext2 /dev/ram0	0 1% (1%)	15.3	5 MB 140.00	KB 15.49 MI
/ext_hdd	vfat /dev/sdb1	- 1 /0 (1 /0)	∃ 66% 1.2	5 GB 2.47	GB 3.72 GI
/mnt/smb	cifs //192.168.0.25/	/temp		8 GB 1205.42	GB 1394.10 GI
		(1%)	00%		

If you would like to shutdown or reboot properly, you can do it by go to menu Status > Power Management. It will show these 3 buttons.

Power M	anagement	
Reboot	Restart Service	Shutdown

**Reboot**: This button uses for restart the system when it happen some fault from any running process.

**Restart Service**: This button uses for restart every services which using with IP-PBX (eg. Interface card)

Shutdown: This buttons uses for shutdown the system.

## 6.2 Show Graph

Go to menuStatus -> System Statistic. This screen uses for displaying the status of hardware using in this system. It can show in graph that makes you compare the status easily.You can see somevariables;Traffic Analysis, Active CPU Load, Free Memory, New TCP Connection, Established TCP Connections andServer Load Average.



O G Chapter: Chapter 6 System Monitoring

## **6.3 Service Status**

Go to menu Status -> Service Status, This screen uses for display the status of the service using in the IP-PBX whether it is running or not. You can turn the service on or off by clicking on the start/stop button. And you can also set it to turn on automatically at every booting time.

- SSH: Allow server to remote via secure shell
- NTP: Service to update the time from Internet time server
- WEB: Allow to do the configuration from website
- FIREWALL: Service to enable the network security of this system
- IP-PBX: Service of telephony system

			Service Status
Name	Status		On-Boot
SSH NTP WEB FIREWALL IP-PBX	running running running running running	StartStopStartStopStartStopStartStop	© Yes ⊂ No © Yes ⊂ No © Yes ⊂ No © Yes ⊂ No Apply
			Network Tools
Network Too	<b>is</b> Select	Tool : ping 👻	URL : Show Status

We can do the network testing by ping, check DNS by nslookup and trace the router hop from traceroute from the Network Tools part.

## 6.4 List DID Number

Go to menuStatus -> List DID Number, This screen uses for display the list of DID for the digital telephony in the system.

Trunk	DID Number	Extension
gateway:sip:test_sys	202	none
trunk:dahdi:test	343	none
trunk:dahdi:test	908	none
trunk:dahdi:test	555	none

## 6.5 Phone's Connection Status

**Phone's Connection Status** 

Go to menuStatus -> Phone's Connection Status, This screen uses for checking the IP-Phone connection, PSTN connection, IAX connection, VoIP provider connectionto the system separated in type and display the color labeledof each unit.

	Number	Name (Callerid)	IPaddress	Natsupport	VideoSupport	Status	Туре
L	4000		-none-	eno	eno	Unmonitored	SIP
2	sipsite_wwwww		-none-	eno	eno	UNKNOWN	SIP
3	trunk%%sip%%asdf		0.0.83.97	yes	eno	Unmonitored	SIP
1	3004	3004	-none-	eno	eno	UNKNOWN	SIP
5	3003	3003	-none-	eno	eno	UNKNOWN	SIP
5	3002	3002	-none-	no	eno	UNKNOWN	SIP
7	3001	3001	-none-	no	eno	UNKNOWN	SIP
3	3000	3000	-none-	eno	eno	UNKNOWN	SIP
)	1144	1144	-none-	eno	eno	UNKNOWN	SIP
10	1119	1119	-none-	eno	eno	UNKNOWN	SIP
11	1010	1010	-none-	eno	eno	UNKNOWN	SIP
12	1009	1009	192.168.0.190	eno	eno	OK (56 ms)	SIP
13	1008	1008	192.168.0.214	eno	eno	OK (109 ms)	SIP
14	1007	1007	192.168.0.204	eno	eno	OK (85 ms)	SIP
15	1006	1006	-none-	eno	eno	UNKNOWN	SIP
16	1005	1005	-none-			UNKNOWN	SIP
17	1004	1004	192.168.0.220	eno	eno	OK (112 ms)	SIP
18	1003	1003	-none-			UNKNOWN	SIP
19	1002	1002	-none-	eno	eno	UNKNOWN	SIP
20	1001	two	192.168.0.217	eno	eno	OK (18 ms)	SIP
21	1000	on Thousand	192.168.0.216	yes	eno	OK (105 ms)	SIP
(A) No	Number	Name (Callerid)	IPaddress	Natsupport	VideoSupport	Status	Туре
1	server_qqqq/use 6000	6000	(Unspecified)				IAX2
2 3	2000	Mr. TTwo Thoudsand	(Unspecified) (Unspecified)			UNKNOWN UNKNOWN	IAX2 IAX2
1	FaxDSP2/FaxDSP2		127.0.0.1			OK	IAX2

From the picture above, there are the colored labels at the status column. You can identify from red colored whether it is not available and green color with the response time whether it is available.

## 6.6 Phone Status Panel:

Go to menuStatus -> Phone Status Panel, This screen uses for display the status of every phone in the system from the color labeled.

Red-That phone is in use.

Blue- That phone is calling

Orange – That phone is ringing

Green– That phone is available.

Grey- That phone is not available

Brown– That phone is waiting from hold.



## 6.7 Active Call Status

Go to menuStatus -> Active Call Status, This screen uses for monitor the phone call in the system immediately (real-time). It displays the source, destination, application and duration. You can click to hung-up or transfer the call from this page.

Active Call S We have total 1 active				
Source	Destination	Application	Duration	Command
1004	1009	Dial	00:00:06	<b>P</b>

## Chapter 7 System's Sound Files (Manage the sound file)

## 7.1 Create Voice

This function uses to manage the sound prompt in the system. You can use this sound prompt to be an IVR system. You can do it by go to menu Sounds -> Create Voice. The screen will show as picture below.

Filename	Description	Th	En	C1	C2	Edit
🗖 Intro	Introduction to Plextel System Sound (Default)	1	💶 🖾 x			Ż
Goodbye	Goodbye Sound (Default)	💽 💟 🗸	💶 🖂 🗙			Ż
🗌 Dialing	Dialing Sound (Default)	💽 🖂 🗸	💶 🖂 🗙			Ż
🗌 Invalid	Invalid Sound (Default)	💽 🔽 🖌	💶 🖂 🗙			Ø,
NoInput	NoInput Sound (Default)	1	💶 🖾 🗙			Ż

From the picture above, you can add maximum 4 sound files for 4 languages.(There are 2 default sound file; Thai and English; in the system.)

- Filename: Specify the filename
- Description : Specify the description of this prompt
- Listen: Click at <sup>I</sup>button to play the voice
- Download: Click at button to save the voice to machine
- Edit: Click to change the voice of that prompt

'Default Voice' is the main voice which will play that language at the first time. For example, you set English to be the default voice. It will play English voice first. However, you can change it later.

Filename	Description	Th	En	C1	C2	Edit
ask_trunkpassword	Ask Trunk password before dial out	🚺 🖾 🗸	💶 🖂 🗙			Ø,
channel_full	Tell Outgoing Channel is Full	🚺 🖾 🖌	💶 🖂 🗙			2
incorrect_password	Used before dial out, if user input wrong password	💽 🛛 🗸	💶 🖾 x			Ż
invalid_number	Used in IVR when user input wrong number	💶 🖂 🗸	💶 🖾 x			Ż
lock_invalid	Used in phonelock feature when user input wrong nu	T 🖸 🖉 🗸	💶 🖾 x			Ż
lock_nowlock	Used in phonelock feature when phone is locked	💶 🖂 🗸	💶 🖾 x			Ż
lock_select	Used in phonelock feature, ask user to lock or unlock	💶 🖂 🗸	💶 🖾 x			Ż
lock_status	Used in phonelock feature, not allow to call out	💶 🖂 🗸	💶 🖾 🗴			Ż
lock_terminate	Used in phonelock feature, when finish lock menu	💶 🖂 🗸	💶 🖾 x			Ż
press	Used to tell pressing status	💽 😒 🖌	💶 🖂 🗙			Ż
transferred_operator	Transfered Operator	💽 💟 🖌	💶 🖂 🗙			I I
now_unlock	Used in phonelock feature when phone is unlocked	💽 🛛 🗸	💶 🖾 x			Ż
ivrdial-unavailable	Used in IVR when dial to extension with unavailable	💶 🖂 🗸	💶 🖾 x			Ż
ivrdial-noanswer	Used in IVR when dial to extension with no answer	1	💶 🖾 x			Ż
from_IVR_userselect_novm	Direct call from IVR with noanswer status	💽 😒 🖌	💶 🖂 🗙			Ż
from_IVR_userselect_vm	Direct call from IVR with voicemail	💽 💟 🖌	💶 🖂 🗙			1
menu_error	Direct call-wrong option	🚺 💟 🖌	💶 🖂 🗙			Ż
roaming select	Used in roaming features	🚺 💟 🧹	💶 🖂 🗴			1

If we click on the edit button, the screen will show as picture below.

[ View Voice prompt ] [ Creat Create Voice prompt	te Voice Prompt ] [ Default Voice ] [ Clear All Record Dialplan ]	
Filename	Intro	
	Introduction to Plextel System Sound (Default)	*
Description		Ŧ
Default sound	Thai 👻	
Thai Sound file	Browse 🔯 🚺	
English Sound file	Browse 📧 🚺	
Custom1 Sound file	Browse	
Custom2 Sound file	Browse	

- Filename: Change the filename of this voice prompt
- Description: Change the description of this voice prompt
- Default Sound: Change the default voice
- Listen: Click at subtron to play the voice.

- Record :Click at button to record the new voice from the phone. You can upload the voice file to the system separately to each language by clickingBrowse button at each field. Then click apply button to upload.

At the 'Create Voice Prompt' menu, The screen will show as picture below.

Filename		
		*
Description		
		-
Default sound	Thai 👻	
Thai Sound file		Browse
English Sound file		Browse
Custom1 Sound file		Browse
Custom2 Sound file		Browse

This screen shows that you have 2 waysto add the voice to the system.

- 1. Upload from your computer
- 2. Record from your phone

You can record the voice from your computer by using 'Sound Recorder'programat your computer. Go to Start -> Program File -> Accessory ->Entertainment -> Sound Recorder

	P2	Programs •	m	Accessories	•	m	Entertainment	Þ	Ø,	Sound Recorder
	L	Programs •	6	EditPlus 2	Þ		Calculator	Т	1	Volume Control
	Ò	Documents •	m	Nero	Þ	¥	Paint		Θ	Windows Media Player
6	1	Settings •		Startup	Þ	Ē	Communications	۲		
l i		-		Tennis Pro	•	Ļ	*			
ese		Search 🕨		The Weather Channel Microsoft Word	•					
Professional	?	Help and Support	W 3	Outlook Express						
۲ ۲		Run	9	*						
Windows	$\mathbf{P}$	Log Off admin								
Wind	0	Turn Off Computer								
4	🋃 sta	nt 📄 🕴 🔘 🗏 🔘 💈 🖉								

The program is opened. You should click button and record your voice from microphone.



After finished recording, you should click 'File -> Save as' and change some properties of that voice by click atChange button as picture below.

Save As			
Save in: 🗀 Sound File		- 🖬 📩 –	
<ul> <li>voice1.wav</li> <li>voice2.wav</li> <li>voice3.wav</li> </ul>			
voice4.wav			
File name:		▼ Save	
Save as type: Sounds (*.wa	/)	Cance	el
Format: PCM 22.050 k	Hz, 16 Bit, Mor	no Change	
	Sound Sele	ection	? 🔀
	Name:		
	[untitled]	▼ Save A	As Remove
	Format:	PCM	•
	Attributes:	8.000 kHz, 8 Bit, Mono	7 kb/sec 🔍
		8.000 kHz, 8 Bit, Mono 8.000 kHz, 8 Bit, Stereo 8.000 kHz, 16 Bit, Mono	7 kb/sec 15 kb/sec 15 kb/sec
		8.000 kHz, 16 Bit, Stereo 11.025 kHz, 8 Bit, Mono	31 kb/sec 10 kb/sec

At the 'Sound Selection' window, you have to choose the format of this sound to be 'PCM' and Attributes is 8.000 KHz, 16 Bit, Mono as picture above. Then clickOK and Save this file. You can bring this voice file to use with the system.

Another method is recording from your phone. Go to menu Sounds -> Voice Prompt - > Create Voice Prompt

Voice Prompt [View Voice prompt] [Create Create Voice prompt	Voice Prompt] [Default Voic	e ] [ Disable Recording from Phone ]
Filename		
Description		* ~
Default sound	Thai 🔻	
Thai Sound file		Browse
English Sound file		Browse 🔊
Custom1 Sound file		Browse
Custom2 Sound file		Browse
**NOTE: WAV file must be in f	10no 16-bit 8KHz format**	
APPLY Cancel		

You can record directly from yourtelephone by these following methods.

- 1. Specify the filename of this voice prompt
- 2. Click on the button on the right hand side of that language

		$\frown$
Thai Sound file	เลือกไฟล์ ไม่ได้เลือกไฟล์ใด	
English Sound file	เลือกไฟล์ ไม่ได้เลือกไฟล์ใด	
Custom1 Sound file	เลือกไฟล์ ไม่ได้เลือกไฟล์ใด	<b>(</b>
Custom2 Sound file	เลือกไฟล์ ไม่ได้เลือกไฟล์ใด	

3. There is a pop-up window appeared. It will show the phone number for dial to record the sound at 'Recording Number' and the time to stop and disconnect when silent (represented in second). If you ready to record press Activate button.

📀 192.168.0.124/libs/rec	orsdsound.ph 🗖 🗖 🗮 🗙
() 192.168.0.124/libs/rec	orsdsound.php?filename=⟨=:
Recording Number	2
Silence Disconnect	Activate Done & Apply
*note: finish record	ing by press #, or silence for 2 seconds

Now, you have to dial to that phone number as shown in the Recording Number field. (Eg. 105) and talk to the phone. After finish, you can stop and silent for a 2 seconds (as shown in the Silence Disconnect) or press # to save. The system will give 'Beep' sound to confirm. Then hung up and click at the nDone & Apply button as picture below.

📀 192.168.0.124/libs/rec	corsdsound.ph 🗖 🗖 💌
S 192.168.0.124/libs/red	corsdsound.php?
**Update completed!	
Recording Number	<b>1</b> 05
Silence Disconnect	2
*note: finish record	Activate Done & Apply ling by press #, or silence for 2 seconds

## 7.2 Call Record Files

You can listen to the recorded voice call from every trunk (eg.Zap Channels andSip Channels from this menu. Go to Sounds -> Call Record Files. The screen will show as picture below.

Start date	Select Date	End date Select Date				
FILTER	NONE	•				
)isplay 1 - 2	1 6					
Files			Date Time	Size	Listen	Download
	JT-from-1000-to-:	11@2010-06-25-14-47-48.gsm	2010-06-25 14:47:48	1188 byte		$\overline{\otimes}$
		L1@2010-06-25-14-48-17.gsm	2010-06-25 14:48:17	858 byte		×
_		06@2010-06-25-15-52-24.gsm	2010-06-25 15:52:24	36432 byte		$\sim$
		@2010-06-25-15-52-39.gsm	2010-06-25 15:52:39	8184 byte		$\sim$
	-from-1006-to-10	 06@2010-06-25-15-57-17.gsm	2010-06-25 15:57:17	3300 byte		~
	-from-1006-to-10	05@2010-06-25-16-49-02.gsm	2010-06-25 16:49:02	19041 byte		$\sim$
	om-1006-to-1005	@2010-06-25-16-49-04.gsm	2010-06-25 16:49:04	13563 byte		~
	/T-from-1009-to-:	11@2010-06-25-17-15-29.gsm	2010-06-25 17:15:29	12276 byte		$\geq$
	JT-from-1009-to-:	11@2010-06-25-17-15-49.gsm	2010-06-25 17:15:49	3960 byte		$\geq$
	JT-from-1009-to-:	11@2010-06-25-17-46-10.gsm	2010-06-25 17:46:10	8415 byte		$\geq$
	-from-1008-to-10	07@2010-06-28-12-03-43.gsm	2010-06-28 12:03:43	4884 byte		$\otimes$
USR_IN	-from-1005-to-10	01@2010-06-28-13-24-56.gsm	2010-06-28 13:24:56	36993 byte		$\geq$
USR_IN	-from-1007-to-10	08@2010-06-28-13-30-24.gsm	2010-06-28 13:30:24	4917 byte		$\geq$
	-from-1007-to-10	08@2010-06-28-13-30-54.gsm	2010-06-28 13:30:54	25113 byte		$\geq$
USR_IN	-from-1008-to-10	07@2010-06-28-13-31-36.gsm	2010-06-28 13:31:36	29865 byte	<b>A</b>	$\geq$
USR_IN	-from-1000-to-10	09@2010-06-28-15-42-37.gsm	2010-06-28 15:42:37	85239 byte		$\geq$
USR_IN	-from-1000-to-10	09@2010-06-28-15-43-42.gsm	2010-06-28 15:43:42	103356 byte	<b>A</b>	$\geq$
USR_IN	-from-1007-to-10	09@2010-06-28-15-44-33.gsm	2010-06-28 15:44:33	19074 byte		$\geq$
USR_IN	-from-1004-to-10	09@2010-06-29-09-28-50.gsm	2010-06-29 09:28:50	49467 byte		$\geq$
USR_IN	-from-1004-to-10	09@2010-06-29-09-31-04.gsm	2010-06-29 09:31:04	26895 byte	<b>A</b>	$\geq$
USR_IN	-from-1009-to-10	04@2010-06-29-09-55-19.gsm	2010-06-29 09:55:19	9669 byte	<b>A</b> 1	$\geq$

You can set the date range and condition from filter menu. NONE : Select all SIP Trunk : Select only the call recorded from SIP Trunk ZAP Trunk: Select only the call recorded from ZAP Trunk (PSTN) USER : Select only the call recorded of specific extension QUEUE : Select only the call recorded in queue CONFERENCE: Select only the call recorded from conference room AGENT : Select only the call recorded from specific agent in queue Then selectFROM, TO orBOTH and specify the number. And press select button, It will appear the result as picture below.

Start date	Select	End date Select				
FILTER	USER	BOTH V 1004 SELECT				
	CHOOSE					
Display 1 - 3						
Files	SIP TRUNK		Date Time	Size	Listen	Download
USR_IN-f	ZAP TRUNK USER	2010-06-29-09-28-50.gsm	2010-06-29 09:28:50	49467 byte		$\sim$
USR_IN-f		2010-06-29-09-31-04.gsm	2010-06-29 09:31:04	26895 byte		$\geq$
USR_IN-f		2010-06-29-09-55-19.gsm	2010-06-29 09:55:19	9669 byte		$\geq$
Page 1	CONFERENCE AGENT					

Start date	Select	End date Select				
FILTER	USER	BOTH V 1004 SELECT				
isplay 1 - 3	from 3					
Files			Date Time	Size	Listen	Download
USR_IN-	from-1004-to-1009	@2010-06-29-09-28-50.gsm	2010-06-29 09:28:50	49467 byte		$\sim$
USR_IN-f	from-1004-to-1009	@2010-06-29-09-31-04.gsm	2010-06-29 09:31:04	26895 byte		$\geq$
USR_IN-	from-1009-to-1004	@2010-06-29-09-55-19.gsm	2010-06-29 09:55:19	9669 byte		
Page 1	<b>•</b>					

## 7.3 Music On Hold

You can set the voice to play during waiting the call. The voice file should be only mp3 orwav only. Go to menuSounds -> Music On Hold, The screen will appear as picture below.

Group Name : Default			Random : yes		
Voice Prompt		Date	Time	Size	Listen
55533.wav		23/06/2010	00:25	27944 byte	
macroform-the_sim	plicity.wav	15/09/2009	02:39	4464220 byte	
song.wav		25/06/2010	09:25	2918198 byte	
Delete Selected Group Name : eng	Cancel	Random : yes			Į.
	Cancel				🖉 Listen

You can create the group of music-on-hold by click at menu 'Add new Music-on-hold Group'. The screen will show as picture below.

Music On Hold					
[ View Music-on-hold ] [ Add new Music-on-hold Group ] [ Upload Music ] [ Restart Service ]					
Add new Music-on-hold Group					
Music-Group					
Random	Yes 🔻				
APPLY Cancel					

You can set the music-on-hold group separately to each group in the system whether each group has their own sound (eg. music, announcement, etc.).

Music On Hold [View Music-on-hold Group][Upload Music][Restart Service]
Upload recorded files
Select Group Default 👻
File to upload Browse
Upload Cancel

After added new group, You should click 'Upload Music ->Browse' and select the file. You can change the filename by fill in the 'File Name' field. Then click 'Upload' button to upload to the system.

## 7.4 Backup Voice Record

There are 2 choices to back up the voice record; Manual Backup and Schedule Backup. Go to menu 'Sounds -> Manual Backup or Schedule Backup'. These 2 choices have similar methods.

## **Manual Backup Voice Record**

	<b>ce Record</b> ecord files will take a lot of resources on the IP-PBX and it may uality/performance on the system, please do it after working hour**
Backup Type	сору 🔻
Backup Location	Local Harddisk 💌
Create Backup Timer	Create Backup Timer
From Day 🔻	Select Date Create New Backup

From the picture above, these are the description of each fields.

- **Backup Type:**Choose the backup method to CopyorMove (If you selected Move, It will delete file out of the system. On the other hand, Copy method willnot delete the file.)
- **Backup Location:**Chosse the destination of backup file Local Harddisk : Keep theBackup on thatServer, You can download that file from the this page
  - USB Harddisk: Keep the file to externallUSB Harddisk (You have to setup the External Storage to mount the USB Disk first)
  - SMB Drive: Keep the file to the SMB Drive (You have to setup the network drive at External Storage first)
- Create Backup Timer: Set the time schedule toBackup automatically
- From:Specify the date and time periodBackup for filter the voice file

The picture below is the example of Backup Voice record. You can choose the date from "From" then click "Create New Backup". The backup file will be created and display at "Last backups". You can download to your computer by click on Download link of each backup file.

Backup Voice Record **warning: backup record files will take a lot of resources on the IP-PBX and it may effect on the voice quality/performance on the system, please do it after working hour**		
Backup Type copy 🔻		
Backup Location Local Harddisk 👻		
Create Backup Timer Create Backup Timer		
From Day 🔻 2010-07-15 Create New Backup		
Last backups		

8 0 Chapter: Chi

es (Manage the sound file)
This is the example of schedule backup. You have to click "Create Backup Timer". It will display the popup screen as picture below. This screen allows you to specify the backup time; eg. Everyday at 14:40.

🍘 http://192.168.0.124/libs/createbackuptimer.php?t	
bttp://192.168.0.124/libs/createbackuptimer.php?type=copy&location	on:
Create Backup Timer:	^
APPLY	
	-
😜 Internet   Protected Mode: On 🛛 🖓 👻 🔍 100% 🔻	æ

### **Schedule Backup**

[schedule backup	p] [backup now] [LOG]
Backup Location	USB Harddisk 🔻
Status	OFF 🔻
backup every	00 • 15 • 31 • * • * help
Backup type	full backup 👻
File older than	day 👻
Compress	yes 👻
	APPLY reset

- Backup Location: Chosse the destination of backup file
   USB Harddisk: Keep the file to externalUSB Harddisk (You have to setup the External Storage to mount the USB Disk first)
   SMB Drive: Keep the file to the SMB Drive (You have to setup the network drive at External Storage first)
- Status:Select to enable or disable
- **Backup every:**Specify the date and time toBackup. You can go to help menu for example as the picture below.

Minute	help Hour	Date	Month	Day
set minute from 00 to 59	set hour from 00 to 23	set date of the month from 01 to 31	set month of the year January to December	set day of week Sunday to Saturday
Example				-
Config			_	Execution time
backup every	00 🗸 02 🗸 *	*	*	02:00 am of every day
backup every	00 - 15 - 1	10 💌 *	*	▼ 03:00 pm on the 10th of every month
backup every	30 - 18 - 1	15 - February	*	06:30 pm on the 15th in February
backup every	30 🔹 🔭 🎽	July	Monday	very hour at minute = 30 on every Monday in Ju

- Backup type:Choose the method toBackup
   full backup Backup all files
   increment backup -Backup only the new file which never backup in the past
   move Backup the files and delete those files in the system
- File older than -Backup only the older than selected day.
- **Compress**Enable the compression for backup file. It will represent in zip file.
- **backup now**Click to createBackup file from the specified backup properties

After finished setup the external storage, you can go back to voice record backup, select the destination andclick the backup button. It will display the popup screen as the picture below. This screen will show the status eg. Filename, quantity of the files, size, location, etc.



### 7.4 External Storage

This menu uses for setup the external HardDisk Drive for keeping voiceBackup (eg. USB Flash drive, USB External Harddisk and Network shared drive). There are 2 types.

### 1. SMB Network Drive

You can backup and store the files to the network shared drive by setup the drive at the Plextel IP-PBX first. Go to menu Advanced -> External Storage ->choose "SMB Network Drive"

		1	1 1
External Storage			
[SMB Network Drive ] [USB Han	ldisk Mount ]		
SMB Network Drive			
Check Current Mount		refresh	
Server IP address	List Serve	er	
Mount at boot	Yes 👻		
Required Authentication	No 🔻		
	Username Password		
Shared Name **	List Shared		
Local Mount-Name APPLY	/mnt/smb		

Click at 'List Server' button to find the Computer which sharing theDrive on the network. It will display the pop-up window and list the name of shared drive's computer. Click on the "List Shared". It will display the pop-up window to specify the destination folder. Then clickAPPLY button to finish.

External Storage [SMB Network Drive][USB Hard SMB Network Drive	ldisk Mount ]
Check Current Mount	//192.168.0.25/temp on /mnt/smb type cifs (rw,mand) refresh
Server IP address	192 168 0 25 List Server
Mount at boot	Yes 👻
Required Authentication	No 💌
	Username Password
Shared Name **	temp List Shared
Local Mount-Name	/mnt/smb
APPLY	

#### 2. USB Harddisk Drive

This is the method to setup theExternal HDD Drive to use with the backup system on Plextel IP-PBX. Go to menuAdvanced -> External Storage -> choose "USB Harddisk Drive"

External Storage		
[SMB Network Drive ] [USB Ha	rddisk Mount ]	
USB Harddisk Mount		
*external harddisk must be form	nat as FAT32	
Current Mount		
Harddisk device	Select Device	•
Partition		
Detect harddisk Mou	nt harddisk Umount harddisk	

Plug-in the USB Storage device (eg. USB Flashdrive or USB External Harddisk), ClickDetect harddisk, selectHarddisk device and Partition then clickMount harddisk to mount that device to the system.

External Storage	
[SMB Network Drive ] [USB Ha	rddisk Mount ]
USB Harddisk Mount	
*external harddisk must be forn	nat as FAT32
Current Mount	
Harddisk device	Disk /dev/sdb: 255 MB, 255327744 bytes 👻
Partition	/dev/sdb1 1 31 248976 6 FAT16 🔻
Detect harddisk Mour	nt harddisk Umount harddisk

1

After click 'Mount harddisk'. I will display as picture below.

## 112

External Storage	)
[SMB Network Drive ] [USB Har	arddisk Mount ]
USB Harddisk Mount	
*external harddisk must be form	mat as FAT32
Current Mount	/dev/sdb1 on /ext_hdd type vfat (rw,uid=5060,gid=5060)
Harddisk device	Select Device 🗸
Partition	
Detect harddisk Mour	unt harddisk Umount harddisk

You can see the status of bothSMB andUSB Drive at Status -> System Status. It will display as picture below.

Network Usage							
Device		R	eceived	Sen	-		Err/Drop
lo eth0		-	5.24 MB 15.46 MB	5.24 M 47.67 M			0/I 0/I
w1q1			12.75 MB	47.67 M 312.75 M	-		0/1
sitO			0.00 KB	0.00 K	3		0/0
Memory Usage							
Туре		Percent Capacity			Free	Used	Siz
Physical Memory	(		83%	173.		320.46 MB	994.45 MI
<ul> <li>Kernel + applications</li> </ul>	(	24%				237.70 MB	
- Buffers	(	15%			-	L53.73 MB	
- Cached	(	43%			4	129.03 MB	
Disk Swap	(	0%		2.	00 GB	0.00 KB	2.00 G
Mounted Filesystems							
Mount		Partition	Percent Capacity		Free		Siz
/		/dev/sda2	<b>—</b> 11% (1%)		23.78 GB		28.38 G
/var		/dev/sda5	🔵 3% (1%)		104.90 GB	3.02 GB	113.78 G
/boot	ext3	/dev/sda1	🚍 5% (1%)		258.08 MB	15.65 MB	288.63 M
/dev/shm	tmpfs	tmpfs	0% (1%)		497.23 MB	0.00 KB	497.23 M
/var/spool/asterisk/monitor	ext2	/dev/ram0	2% (1%)		93.04 MB	1.51 MB	94.56 M
/mnt/smb	cifs	//192.168.0.25/temp	(	78% (1%)	304.40 GB	1089.69 GB	1394.10 G
/ext_hdd	vfat	/dev/sdb1	0%		242.54 MB	360.00 KB	242.89 MI

### **Chapter 8 Call Center**

### 8.1 Queue and Agent Management

### How to CreateCall Center orQueue

Before start using Call Center feature in the PEXTEL IP-PBX system, You should design the flow of call. Eg.Set the incoming call and IVR, Type of call distribution.

Type of queue.

- 1. Agents login permanently and wait for the call with their headset until the call has come. Agents will listen to the music-on-hold all the time.
- 2. Agents can login/off by their phone. If there is any call to the call center system, It will ring at their phone.

### How to create call-center agent

Go to menuCall Features >> Agents >> Create Agents

[View Agent] [Create / Add Agent	Agents ] [ Advanced Config
Agent ID	
Agent Admin Password	
Agent Name	
Agent Group Number	

Agent ID: Number to identifyagentfor logging into the system. Agent Admin Password:Password forlogging into the system.

Agent Name: Specify theagent's name. (Optional)

Agent Group Number: Specify the group number of agent.

Then clickAPPLY to createagent

### How to createQueue

Go to menu Call Features >> Queue >> Create Queue

Queue Name	
Announce	None 👻
queue-thereare	None 👻
queue-callswaiting	None 👻
queue-holdtime	None 👻
queue-seconds	None 👻
queue-youarenext	None 👻
queue-thankyou	None 👻
Allow direct call From this queue	no 👻
Music On-Hold	Default 👻
Ring Strategy	ringall 👻
Auto Pause Agent ?	no 👻
Service Level Statistics (sec) ?	o
Agent Ring Timeout (sec) ?	15
Enable Ring Timeout Reset **BETA** ?	no 🔻
Enable Ring when SIP Agent in "INUSE" state ?	no 🔻
Retry all agents ring wait time (sec) ?	5
Queue Weight (higher the better)	o
Agent Rest Time (sec) ?	o
Maximum Queue Size (0-infinity)	<b>o</b>
Report Hold time to Agent ?	no 👻
Announce Queue Information Every (sec)	60
Announce Estimate Hold Time To Caller	once 👻
Recording Format	gsm 👻
Allow user to enter queue when no AGENT	● Yes C No C Strict
Allow user to enter queue when no Adem	C Yes  No

Queue Name: Specify the Queue name

Announce: Play the sound prompt before coming to the queue

Allow direct call from this queue:Enable to allow dialing to specific extension from user's dial pad while waiting the call. (This menu has to setup to 'Incoming call' and 'Call control' first.)

**Music On-Hold:** Specify the sound prompt to be the music-on-hold while waiting the call in the Queue

**Ring Strategy**: Type of call distribution to agents.

Ringall : Ring every active agents at the same time.

Roundrobin: Ring each agent as loop.

Leastrecent: Ring the agent who has longest free-from-receive period first.

Fewest calls: Ring the agent who has fewest received the callfirst.

Random: Ring randomly.

Rrmemory: Ring as loop and remember the last position of agent then go ahead.

**Auto Pause Agent:**Pause Agent automatically if he doesn't answer the call in time. **Agent Ring Timeout**: Set the expire time for ringing

**Enable Ring when SIP Agent in "INUSE" state**: Allow call waiting during call **Retry all agent ring wait time**:Set the time for waiting until skip to other agent. **Agent Rest Time**: Set a break time foragent in case that agent receives too many

calls. (Specify in second)

Maximum Queue Size (0-infinity): Set the maximum incoming call to this queue in the same time

**Report Hold time to Agent:**Enable system to say number of waiting call in the queue **Announce Queue Information Every (sec)**: Enable system to say information of the queue every minute during waiting call.

Announce Estimate Hold Time To Caller: Enable to tell the estimate time for waiting the call for people who waiting in the queue

**Recording Format**: Set the file format of voice recording

Allow user to enter queue when no AGENT: Enable incoming call to the queue incase the agent has not logged in.

**Remove Caller from queue when no AGENT:**Enable to decline every call in the queue in case every agent logged out.

**Agent Members (Add Multiple):** Add member or agent to this queue. Agents can be 3 types.

### Type of agents

1. Specific SIP extension: This type suits for the system which has not too much agent in the system.

2. Agent login: This type suits for the system which has too much agent in the system. Agents have not to sit at their phone permanently. Agents can login/off by their phone.

3. External line: Agent can be the people who are not sit in the office (may be mobile phone or other sites)

To setup priority of agent is the easy way to manage the skill of agent who has their specific skill (eg. Language or performance of that agent)

### Agent and Queue management and monitoring

Manual Agent	t CallBack L	ogin										
Select Agent :	102 👻 s	elect Extensio	on: 9009 🔻	Ackcall :	false	✓ Wrate	apupTime:	5	login			
Queue	Agent	Agent Name	LoginChannel	Status	Talk to	Penalty	Call Taken	Last State (sec)	Last Call	LoggedInT	imePause	d
Queue_SIP	Sip/9009			Not in use	2	1	0	0 secs	00:00:00		YES	
Queue_SIP	Sip/9008		-	Not in use	1	1	0	0 secs	00:00:00		NO	00000
Queue_Agent	Agent/102	John	n/a	Unavailable	n/a	1	1	55 secs	13:42:48	00:00:00	NO	0000
Queue Agent	Agent/101	Peter	n/a	Unavailable	n/a	1	0	0 secs	00:00:00	00:00:00	NO	0000

### Go to menuReport >> Realtime Agent Status

From the picture above, there are 2 queues using in this system.

Queue\_Agent has 2 members (101 and 102) and Status isUnavailable. The status tab shows in grey colored. It means that agent has not logged in to this queue. Queue\_SIP has 2 members (9008 and 9009). We have not use agent with this queue. But we fix the extension to this queue. The green colored tab shows online status of that extension and ready to receive a call. If the tab shows red colored, it means unavailable.

Administrator can do 'Manual Agent CallBack Login' for the agent at this page by selecting agent number and extension number. Then click 'login' button.

**Login Channel:** shows the extension number that agent logged in (only agent login type)

**Status:** There are 4 types of status.

- Not in use (green colored)
- In use (or Busy)
- Take To : shows talking agent number
- Priority : shows the importance of this call

**Call Taken:** amount of incoming call (It will be resetto be zero when reloaded) **Last State (sec) :**free time gap from last call (start with 1 when hung-up) **Last Call:** Call duration of last call

LoggedInTime : Last agent login time(Only agent login, not include SIP agent)

**Pause:** You can do 'Pause' when you would like to avoid receive the call no matter you are logged in until you do 'unpause'. There are 2 ways to do.

1. Agents pause his number by himself. (See how to press the dial pad at menu Call Features>>Feature Code/System Setup )

2. Pause from web interface (individual agent pause)



- 3. Pause all agent from each queue
- 4. UnPause: cancel pause individual agent
- 5. UnPause All: cancel pause agent frome ach queue
- 6. Logoff : click to Logoff that agent

### At menuReport >> Realtime Queue Status

This screen shows the real time status of queue. You can selected the queue from dropdown box above.

Queue Summaryshows basicstatus of theQueue

Queue Details shows agentstatus of theQueue (both logged-in and not logged-in)

Sei	ect Queues :	Queue_	_Agent 🔻						
Queue Name :	Queue Ag	ent							
Call Limits	Waiting Ca		wage Hold Call	Complete Aban	doned Call Ring	Strategy Servi	ice Level S	ervice Level Perf	Weig
0	1		17	0	0 lea:	strecent	0	0.0	0
Queue Member	'S								
Name	Agent N	ame	LoginChannel	Membership	Penalty	Calls Taken	Lastcall	Status	Pause
Agent/102	John		SIP/9009-0000000e	static	1	0	00:00:00	Busy	NO
Agent/101	Peter	:	n/a	static	1	0	00:00:00	Unavailable	NO
Current Waitin	g Calls		CallerID	CallerIDNAME					

### Inbound Queue Report

[ View ] [ Setting ]						
Start date 2010-09-14	End date 2010	0-09-15				
Select Queue Queue_SIP	▼ View					
Queue Name <b>Queue_SIP</b> Total Incoming Calls <b>1</b> Bangkok <b>0</b> Mobile <b>0</b> Others <b>1</b>		CONNECT ABANDON EXIT 0	<b>1</b> 0			
				<b>6 – 1</b>	 1- 6	 

Agent Number	Total Receive Call From Queue	Total Transfer Calls	Internal Tranfer	External Tranfer
SIP/9007	1	1	0	1
SIP/9008	0	0	0	0

List of "Internal & External Transferred To" extensions

Extensions	Total
DAHDI/3	1

Internal Tranfer

External Tranfer

### At menu Report >> Queue Statistic Report

This menu uses for display the statistic of incoming call toeach queue. You can filter the time for each period. It can report as monthly, daily and display in graph for each agents.

#### Queue Statistic Report



### 8.2 Customer Satisfaction Report

This feature uses for evaluate the quality of service of agents. The concept of this feature, Agents have to transfer a customer's call to this feature. It will play the sound to advice customer to give the score in range. After customer press number, the system will play goodbye sound and hung-up.

Go to menu Report  $\rightarrow$  Customer Satisfaction Report  $\rightarrow$  Setup. The screen will show as picture below.

Customer Satisfaction Report	
[ Summary ] [ Details ] [ Setup ]	
Score Range: 5 V Apply	
Clean All Data: Clean All Data	

From the picture above, we can set the score range (eg. 1-5 or 1-9)

### 119 e screen will

Next, Goto menu Call Feature  $\rightarrow$  Customer Satisfaction Report Setup, The screen will show as picture below.

Customer Satisfactio		epo
Number	999	
Greeting Sound	Intro	
Goodbye Sound	Goodby	/e ,
Confirmation Sound	Dialing	
Language	th 👻	
Required Confirmation		$\bigcirc$ N
Repeat this menu for NO-input (times)	2 🔻	
*note* only blind transfer is supporte APPLY	d	

**Number:**Specify the number to use with this system. (Beware of duplication with extension or call-feature)

Greeting Sound: Select the greeting sound prompt to use with this system. (Eg. "Please give the score for this service. Excellent press 1. Good press 2.....)However, you can create the sound prompt at topic 7.1 Create Voice
Goodbye Sound: Select goodbye sound prompt to play after press button. (Eg." Thank for participation, Thank you")
Confirmation Sound: Select sound prompt for confirmation (Eg. "You choose 5. Press 1 to confirm")
Language: Choose the language (It will choose at sound prompt)
Required Confirmation: Enable to use confirmation for each selection.
Repeat this menu for NO-input (times):Specify the repeat time whether customer has not press the button in time.

After we set up this feature, You can see the score statistic of each agent by go to menu Report  $\rightarrow$  Customer Satisfaction Report  $\rightarrow$  Detail. The score will show in GPA.

### CHAPTER 9 VirtualFAX System

This chapter will describe about the fax system in the Plextel IP-PBX. The fax feature in the Plextel IP-PBX uses virtual fax technology. You can send, receive, backup, forward and manage easily from web-interface in the PDF format.

### 9.1 Add virtual fax

You have to setup the virtual fax machine in the system first. Go to menu FAX>add virtual fax

Plextel System: 🥵		FAX Dev FAX Device [Ad		J <b>D</b> Device] [Fax Status	]			
Show All   Close All	Fax	Fax group name	Fax phone number	Local	Number of ring before answer	Fax	Default fax email address	Edit
Status	Page (	Delete Selected	l		diswei	type	auuress	
Report	Fage	Delete Selected		2				
Group Manager							Plextel Enterprise	version 2
Extensions								
Fax 4								
Call Features								
Sounds								
Incoming Call								
Outgoing Call								
Schedules								
IVR								
Site to Site Setup								
Manual Config								
Voice Interface								
Log								
Advanced								

Click at 'Add virtual FAX Device', the screen will show as picture below.

### Virtual FAX Device Setup

[Edit virtual FAX Device] [Add virtual FAX D	evice] [Fax S	Status]					
Fax Properties							
0. Fax name	Fax_test						
1. Fax Group Name (for Outgoing FAX)	new 🔻	internal		Add			
2. Fax Phone Number	1009						
3. Local Identifier (Company Name)	poise						
4. Number of Rings before answer	1 🔻						
5. Fax Type	internal	•					
7. Forward fax to Internal Users	✓ 1002	✓ 1003	1006	9001	1009	3000	3001
	3002	3003	□ 3004	1007	1005	9009	9008
	9007	9006	9005	9004	9003	9002	

Save

**Fax name:**Specify the name of this virtual fax device.

**Fax Group Name (for Outgoing FAX):**Select the user group of this fax device. However, you have to create the group at the first time by select new in the dropdown box and specify the group name.

1. Fax Group Name (for Outgoing FAX)	new 🔻	internal	Add
(ior outgoing rang			

Then click add, the new group name will be added to the dropdown box. So, you have to select it again.

1. Fax Group Name (for Outgoing FAX)	internal	▼ del
(for outgoing r Avy		

**FAX phone number:**Specify the phone number of this virtual fax device **Local Identifier (company name):**Specify the company name

**Number of ring before answer:** Select the number of ring time for receive the fax **Fax Type:**Select the fax type

Internal: This is the virtual fax device using in within the system.

External: This is the virtual fax device using with other system. Eg.Sending fax outside by PSTN or VoIP.

In case of selected External fax, It will appear the textbox to enter default fax email address.

5. Fax Type	external	-
Fax Routing		
6. Default Fax email address		

orwards fax to internal users: Select the extension to forward the fax whenever the system received the fax.

After finished setup virtual fax device, click at view virtual fax. You will see the detail of that virtual fax device. (normally, it will show 'running and idle')

Virtual FAX [Edit virtual FAX Dev			ax Status]			
Fax name	Fax group name	Fax phone number	Local identifier	Number of ring before answer	Fax type Default fax email address	Edit
Fax_test::2	external	1008	poise	1	external	2
	Selected					<u>-</u>

You can see the real-time status of the fax by click at 'Fax Status'

### Virtual FAX Device Setup

[Edit virtual FAX Device] [Add virtual FAX Device] [Fax Status]

#### Fax Status

test_inte Fax_test	ernal (1009):	dem :	Sta Running Running	and idle				
JID	User	File	Destination	Page	Dials	TTS	Status	[x]

In case of using internal fax, you have to know the fax prefix of that number by go to menu call features>feature codes.

Agent Login (permanent)					
Local College A. College Ve	*	•	9		
Agent Calback Automatic Login/Logoff	*	·	45		
Agent CallBack Login	*	٠	40		
Agent CalBack Logoff	*	٠	41		
Pause Agent Prefix	*	•	42		
UnPause Agent Prefix	*	•	43		
Transfer to Agent Prefix	44	_			
Whisper	•	•	97		
Private Whisper	*	•	98		
Channel Spy	*	•	99		Password: 1234
Call Parking			-		
Parking Number	700	)			
Parking Position	701	l		- 720	
Max Parking Time	120	)			
Transfer digit timeout	3				
Features Key Mapping					
Features digit timeout(ms)	300	0			
Call Pickup	*	Ŧ	8		
Extensions Pickup	*	Ŧ	٠		
Bind Transfer	#	Ŧ	1		
	#	Ŧ	2	_	
Attend Transfer					
Attend Transfer Disconnect	٠	Ŧ	D		
	•	•	р 3		
Disconnect	* * 100	•	-		
Disconnect One Touch Record	_	•	-		
Disconnect One Touch Record Voicemail	100 99	•	-	]	
Disconnect One Touch Record Voicemail Phone Lock Roaming Station Register / Dial-	100 99	• • •	β	1	
Disconnect One Touch Record Voicemail Phone Lock Roaming Station Register / Dial- Out Prefix	100 99	•	3	]	
Disconnect One Touch Record Voicemail Phone Lock Roaming Station Register / Dial- Out Prefix Fax Prefix	100 99	• • •	3		Enable
Disconnect One Touch Record Voicemail Phone Lock Roaming Station Register / Dial- Out Prefix Fax Prefix Features Key Mapping	100 99	• • •	3	-	Enable

You would like to send fax to number 1003 and fax prefix is '33'. You have to press '331003' to get fax to of 1003)

123

In case of using external fax, you have to set the outgoing call first. And change the call control to enable sending out.

### Call Control

Group Name	DefaultDefaultDefault	poiseengtest1234	pstn	kkk	sip	SATSCORE	virtual-eng_	virtual-call_center	virtual-abcd	agent_blf_status	conference	parkcalls	testttttttttttt	test_ivr	Paging-4444	test_site	xxx(SIP)	dsdsd(SIP)
All Group																		Γ
Default:Default:Default	$\square$											$\mathbf{\nabla}$	$\mathbf{\nabla}$	$\mathbf{\nabla}$				
poise:eng:test1234		$\square$																
test_site																$\overline{\mathbf{N}}$		
xxx(SIP)																	$\overline{\mathbf{N}}$	Γ
dsdsd(SIP)	Г		Γ								Γ	Γ	Г	Γ				$\overline{\mathbf{M}}$
111			Г	Г														Γ
main			Г	Г														Γ
2Queue		Г	F	5														Γ
external	Г	П	7	ŋ	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г		
internal	Г		F	1	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г		
programmer			Г															
Page 1 : APPLY	ng C	all	Ca	l Fea	ture	s	Site	-to-S	ite	Qu	ieue	I	VR	FA	x			

From the picture above, you have to change the permission of external fax (identified in yellow tab)to allow call out via PSTN.

Internal user who would like to use fax, they have to enable the fax mailbox by go to menu extension and click edit that extension.

Plextel System: 🥵			Manager [ Add SIP Extension ] [ Add	dd Multiple SIP ] [ Add I/	AX Ext	ensi	ion]	[ Ad	d Ana	alog	Extension	] [View A	ll,Add Follow-N	le Extension]			
Show All   Close All	Display 1 - 20	30 From	30		Ext	tens	ion Ir	npor	't				Browse	Data Import	Extens	ion Expo	rt
Report Group Manager	Page 1	•	0														
Extensions	🗖 Numb	er Typ	eGroup	Pickup Call From	BLFI	/RO	)/R1	NTL	.ang	AP	Phone Type	DIDFollo	wmeRoamir	g User Email	Vmai	FAX mailbo	x Edit
Fax	☐ 1000 : Thouse	on sip and	Default:Default:Defaul	t	1	1	✓	x	Thai	x	none		- x	nicrora@hotmail.com	1	-	Ż
Call Control	□ 1001:	two sip	Default:Default:Defaul	tDefault:Default:Default poise:eng:test1234	1	V	•	x	Thai	1	snom360		×	email@company.com	1	×	Ż
Conference	□ 1002 : 1002	sip	Default:Default:Defaul	tDefault:Default:Default poise:eng:test1234	x	1	1	x	Thai	x	none		- x		×.		2
Features Code / System	□ 1003: 1003	sip	Default:Default:Defaul	Default:Default:Default poise:eng:test1234	x	1	•	x	Thai	x	none		- x		Ľ	~	2
Setup	□ 1004 : 1004	sip	Default:Default:Defaul	t –		1	<	x	Thai	x	snom300		- x		×	-	2

### Go to 'enable the fax mailbox' and change to 'Yes'

Fax Mailbox	
Enable Fax Mailbox :	Yes 👻
Send Fax Message Notification To Email :	Yes 🔻
Attached Fax File To Notification Email :	Yes 👻
Send Voicemail Notification For Incoming Fax :	Yes 👻
Fax Mailbox Size :	20 👻

To send fax, you can do it by logged in to system and go to menu 'Message 125 Center>FAX'

		_						V	/elcon	ie Admin A	dministra	ator Lev
me   Phone Books   Call Detail	s Record	Mess	age Center	CRM   Logout								
		S		<b>~</b> 1								
Plextel System:		_	Cente	[Fax] [ SMS ]								
iow All   Close All	[ ÷	Fax ] [ 👻	Send fax ]	[ FAX Status] 2								
Status												
Show Status	Se	nder(TSI	-	▼ Start Date	2:		End Date :				SEARCH	1
System Statistic		Delete Se	elected									
Power Management	Page	1:										
Service Status		Туре	Device	Filename / Status	Date	From (CallerID)	Sender (TSID)	То	Size (Kb)	Downloa	d View I	Fowor
List DID Number		Outgoing	test_internal	test_page_03957-from	- 2010-	55555	Admin	15	3.24	$\geq$	Q	>>
Phone's Connection Status				55555.pdf status::No local dialtone; t many attempts to dial<333						_	Ň	
Phone Status Panel		Internal		hahaha_6V04Z-from-	2010-	1006	Admin	1005	3.24	$\geq$	Q	$\gg$
Active Call Status				55555_6LQ91_5OWUI.p	df 06-28 13:41:48							
Report		Internal		fax000000041-from-	2010-	55555	1234567890	1006				
Group Manager				1234567890_RY0BG.pdf	06-28 13:41:46							
Extensions		Internal		hahaha_6V04Z-from- 55555_6LQ91.pdf	2010- 06-28	55555	Admin	1006	3.24	$\geq$	Q	<b>&gt;&gt;</b>
Fax				33333_0EQ31.pu	13:40:23							
Call Control		Internal		fax000000046-from- 1234567890_1CEYR.pdf	2010-	55555	1234567890	1006				
Call Features					13:39:24							
Conference		Internal	test_internal	fax000000046-from- 1234567890.pdf	2010- 06-25	unknown	1234567890	1007	27.06	$\geq$	Q	$\gg$
Features Code / System					15:13:55							
Setup		Internal	test_internal	fax000000045-from- 1234567890.pdf	2010- 06-25	unknown	1234567890	1001	27.52	$\geq$	Q	$\gg$
Paging	_	Outering	teet intered		15:11:51		Adacia	1000	2.24			
New Features		Outgoing	test_internal	wd_PS4YG-from- 55555.pdf	2010- 06-25	55555	Admin	1009	3.24	$\geq$	Q	$\gg$
New Macro				status::No local dialtone; t many attempts to dial<333								
Agents		Outgoing	Fax_test	hahaha_6V04Z-from-	2010-	55555	Admin	11	3.24	$\geq$	Q	$\gg$
Queue				55555.pdf status::Complete<0>	06-25 14:55:54							
CallerID Routing		Outgoing	test_internal	test_E53VY-from-	2010-	55555	Admin	11	31.01	<b>S</b>	Q	>>
Customer Satisfaction				55555.pdf status::**** Warning<34	06-25 7> 14:53:26							
Setup		Outgoing	test_internal	test_03EST-from- 55555.pdf	2010- 06-25	55555	Admin	11	31.01	$\sim$	Q	$\gg$
Virtual Number				status::**** Warning<34	7> 14:52:54							
Sounds		Incoming	Fax_test	fax000000042-from- 1234567890_6R5KH.pdf	2010-	unknown	1234567890	Admin	26.74	$\geq$	Q	$\gg$
Incoming Call					14:51:12					_		
Outgoing Call		Incoming	Fax_test	fax000000042-from- 1234567890_2IU1T.pdf	2010- 06-25	unknown	1234567890	1002	26.74	8	Q	<i>»</i>
Schedules	_	Internal	test interest	fax000000041-from-	14:51:12 2010-	1009	1234547900	1002	27.41		0	
IVR		internal	test_internal	1234567890.pdf	06-25		1234567890	1002	27.41	$\geq$	Q	>>
Site to Site Setup		Internal	test internal	fax000000027-from-	11:51:18 2010-	1009	1234567890	1000	23.79	$\geq$	Q	>>
Manual Config		internal	cest_internal	1234567890.pdf	06-24		220 1007090	1000	20.79		~	
Voice Interface		Outgoing	All	2223_D74KH-from-	16:39:56 2010-	55555	Admin	777	230.8		Q	
Log				55555.pdf	06-23						~	
Advanced		Outgoing	All	status::Unknow<> 2223_D74KH-from-	04:40:38 2010-	55555	Admin	666	230.8	$\geq$	Q	>>
PBX Advanced Setting				55555.pdf status::Unknow<>	06-23 04:40:34						Ň	
Network Setting		Outgoing	All	2223_D74KH-from-	2010-	55555	Admin	555	230.8	$\geq$	Q	>>
Manage Admin				55555.pdf	06-23							

You can see the detail of fax pages at this page and also download to your machine or forward that page to other user (also outside)

To send fax by virtual fax device, you can go to menu 'Send fax'. The screen will show as picture below.

Message Center	
[Voicemail] [Call Record] [Fai	
[ 💻 Fax ] [ 🐑 Send fax ] [ 🖁	FAX Status ]
Sending fax	
Select file to fax :	C: \Users \intania86 \Documer Browse Upload
New file name (English Only) :	manual
	Upload file Preview Del
Destination fax number :	12
Fax device :	Fax_test::ttyIAX2 -
Use cover page :	no 👻
Priority :	normal 127 💌
Retries (times) :	1 -
Resolution :	normal 👻
SEND CANCEL	

Select file fax:Browse the file from your computer (only in PDF format) New file name(English Only): Specify the filename (only in English)

After that, you have to click upload button and wait until the table show the filename of that document.

Select file to fax :	C: \Users \intania86 \Documen	Browse	Upload
New file name (English Only) :	manual		
	Upload file	Preview	Del
	manual_43STP.pdf	Preview	Del

**Destination Fax Number:**Specify the phone number of destination (you have to put some prefix or dial-pattern here)

Fax Device:Select virtual fax device

**User cover page:** Enable to use fax cover. If you selected 'yes', it will appear some form to edit fax cover as picture below.

Use cover page :	yes 👻	
To :		
Company :		
From(CallerID) :		
Subject :		
Note :		*
		~
		-

Priority:Select the level of importance of this fax. There are 4 levels; Junk / Low / Normal / High. Normally, We use normal level.
Retries (times):Number of retry time whether it occurs error.
Resolution:Select the resolution of document.

During sending fax, you can see the real-time status of that fax by go to menu 'Fax status'. It will open a pop-up window as picture below.

#### Fax status

JID	Us You can forward the fax Flo other receiver by go	to <b>Destination</b>	var <b>Pag</b> an	d Select The	Status	[x]
43	Admin manual 3RDVQ-from-55555.pdf extension (receiver) then click forward button.	12	0:1	0:12		<u>[X]</u>
	extension (receiver) then click forward button.					

Forward fax to Inte	ernal Users				
1001	1002	1003	1006	9001	
1009	3000	3001	3002	3003	
3004	1007	1005	9009	9008	
9007	9006	9005	9004	9003	
9002					
Foward					

### **Chapter 10 CallerID Routing & Call Back**

**10.1 CallerID Routing**is the menu to manage the incoming call to any destination. This feature will detect the Caller-ID whether it matched to the condition. You can set the condition by go to menuCall Features >> CallerID Routing.

CallerID Routing [View Routing Rules] [Add New Rules] [Default Rule]								
	e : default 🖉 n: noneselected ::							
APPLY	Cancel							

First, you have to createnew rue by click at 'Add new rule' and specify the rule name.From the picture below, we specify the rule name to be'test\_CID'. The destination is extension 1000. That means, this rule will be route to extension number 1001. Then click apply button.

CallerID Rout	ting	
[ View Routing Rules ] [ /	Add New Rules ] [ Default Rule ]	
Add New Rules		
Group Name	test_CID	
Rules Name	test_CID	
Destination	Extension Value SIP/1000	•
APPLY Cancel		

After click finish, the screen will show as picture below.

Group Name : test	CID			
Rule Name : test_CI Destination: extens				
			DataImport	CallerID Routing Export
No.	CallerID	Note	Edit	Delete
[Add New CallerID]				

Then specify the CallerID number to this rule by click Add New CallerID'.

Group Name : test_CID					
Rule Name : tes Destination: exte					
				DataImport	CallerID Routing Export
No.	CallerID		Note		Edit Delete
1	026568598	poise			ali
2	026568597	poise			add cancel
[Add New CallerID]					
APPLY Cancel					

You can specify the CallerID and click Add button to add number into system.when finished, you have to click Apply button. The screen will show as picture below.

After finished these settings, the incoming call; which CallerID matched to the settings; will be route to the specified destination (Extension 1000).

You can apply this feature for the agent who is the steady customer. He will have the hotline to this agent.

CallerID Routing [View Routing Rules] [ Add New Rules ] [ Default Rule ]					
Rule Name : default Destination: noneseled					
Group Name : test_					
Rule Name : test_CID					
Destination: extensi	ion :: 1000				
				DataImport	CallerID Routing Export
No.	CallerID	Note		Edit	Delete
1	026568598	poise	I I I I I I I I I I I I I I I I I I I		<b></b>
2	026568597	poise	I I I I I I I I I I I I I I I I I I I		<b></b>
[Add New CallerID]					
APPLY Cancel					

Then configureIncoming Call

Incoming Call								
[ View Incoming Call ] [ Add New Incoming Call ]								
Add Inco	ming Call							
Trunk		trunk:d	ahdi:test	•	PABX-link			
Description	on							
Support [	DID					no 🔹	•	
Replace (	allerID							
Extension	ns Ring Timeout(s	ec)				40 🔻		
Enable Ca	llerID-Based Rout	ting Serv	vice			test_(	CID 🔻	1
Ad	d time based handler							
		Ac	tions					
When *	Working_Time	•	Destination*	ivr		•		
Value ttttttt								
Remove	e							
APPLY	Cancel							

From the picture above, Every incoming call will check the rule from CallerID Routing. If it matched to any rule, It will go to that destination. If not, it will follow the rule at incoming call(Actions).

### **10.2 Call Back**

Call Back is a function for automatically call back. This feature will check the CallerID of the customer whether it matched to any rule in the CallerID routing. It will hang –up the customer call and call the customer back automatically; to reduce customer's payment.

To use call back feature, You have to set up CallerIDRouting , Outgoing Call, Incoming Call, andCall Control.

1. CreateCall Back by go to menuCall Feature>CallBack > Add New Rules

### Call Back

[Call Back Rule] [ Ad	d New Rule ] [ Add Web Call Back ] [ Call Back Properties ]
Add New Rule	
Rule Name	test_callb
Description	call back testing
Select IVR to Play	business time 💌
APPLY Cancel	]

Specify the Call Back name and select IVR. Then click APPLY button. The screen will show as picture below.

Call Back						
[Call Back Rule ] [ Add	New Rule ] [ Add Web Call Back ] [ Call Back Properties ]					
	Rule Name/Web Rule Name	Description	IVR to Play	WEB	Web URL	Edit
test_callb		call back testing	business time	-	-	Ż
Delete Selected	Cancel					

- \*\*This Features must be used in combination with the Incoming Call rules or CallerID Routing rules and Outgoing Route on Call Control Page\*\*
- Create Caller ID Routing to check CallerID from incoming call whether it matched to the group of Call Back feature by go to menuCall Feature > Caller ID Routing >Add New Rules

CallerID Routing	
[ View Routing Rules ] [ Add New	w Rules ] [ Default Rule ]
Add New Rules	
Group Name	test_cb 🖌 del
Rules Name	
Destination APPLY Cancel	Select  Select Value Select NONE (go to next priority) Group Extension IVR Voicemail Conference Queue Custom Application Call Back Fax

Then go to AddNew Rulesand select function for destination. When theCallerID matched to the rule, It will transfer that call to the destination. In this case, we set the destination to Call Back feature. Then clickAPPLYbutton. You will get the table ofCallerIDRouting as picture below.

Group Name : tes	st_cb			
Rule Name : pstn_ Destination: call	_ 🖉 🛅 back ∶: test_callb			
		DataIm	CallerID Routing	Export
No.	CallerID	Note	Edit	Delete
1	10001		I de la companya de l	Ē
2	10002		I de la companya de l	Ē
3	10003		I I I I I I I I I I I I I I I I I I I	<b>T</b>
4	10004		I I I I I I I I I I I I I I I I I I I	<b></b>
5	10005		I de la companya de l	面
6	10006		I.	<b></b>
_	10007		I.	一一一一
7	10007			

APPLY Cancel

From the picture above, we have added CallerID number 10001 to 10008. Whenever number 10001 - 10008 call to the system. It will hung-up and enter Call back feature.

3. SetupIncomingCallandEnable CallerID-Based Routing Serviceto check the CallerID and rules. It that caller ID follow the rule. It will hung-up and call back automatically. From this example, we enter the call back feature with IVR.

Incoming can			
[ View Incoming Call ] [ Add New	w Incoming Call ]		
Add Incoming Call			
Trunk	trunk:dahdi:pstn_test 💌	PABX-link	
Description			
Support DID		no 💌	
Replace CallerID			
Extensions Ring Timeout(s	ec)	40 💌	
Enable CallerID-Based Rou	ting Service	test_cb 💌	

Add time based handler

Actio	ns
When * All_Time   Des	tination* Custom Application 💌
Value aaa test 💌 aaa	
Remove	

T Chapter: Chapter 10 CallerID Routing & Call Back

 SetupOutgoing Call for call out. Egcall to02-xxx-xxxx or08-xxxx-xxxxfrom CallerID Routingfeature. We can set theOutgoing Call at menu Outgoing call > Add New Outgoing Call

OutGoing Call		
[ View Outgoing ] [ Add New Out Going Call ] Outgoing Route Information		
Route Name	pstn	
Route Description		
Route Password		
Time Based Call Routing		
Default Route		
Enable No. Trunk / Group Defau	Outgoing Number Dialing Prefix Digit to Strip dialing option	1
1 trunk%%dahdi%%pstn_test	т	
Add Fnable Trunk: ZAP/trunk:dahdi:pstn Dialing Option: T Call Patterns	est 💌 Default Outgoing Number: Dialing Prefi	x: Digit to Strip:
Call Prefix		
Destination Pattern	02.	
Destination Pattern2(optional)	08.	
Destination Pattern3(optional)		
Dial Timeout	40	
Concurrent Call Limit for this trunk	100	
Strict Time Routing	no 💌	
Support DID With This Route	no 💌	
APPLY		

5. Set the permission at Call Control to allowOutgoingCallforCallBackas picture below Call Control

Group Name	DefaultDefaultDefault	SALESALESALE	TESTTEST	pstn	SATSCORE	virtual-engineer	agent_blf_status	conference	parkcalls	test	Paging-555
All Group	V	V	V					<b>V</b>			
Default:Default:Default	V	☑			•		<b>V</b>	V	<b>V</b>		
SALE:SALE:SALE	☑	V	☑					✓			
TEST:TEST:TEST	V	V	V		V		☑	<b>V</b>	✓		
call_center											
business time											
fax_test											
test_callb											

We can also use Call Back feature from Web User Interface for enter the phone number and call back to that number.

To SetupWeb Call Back, go toWeb Call Back at menuCall Feature > Call Back > Add Web Call Back. Then specifyWeb Rules Name. The system will create the URLLink. You should select the IVR to use when call back. Then click Apply button.

Call Back						
[ Call Back Rule ] [ Add N	ew Rule ] [ Add Web Call Back ] [ Call Back Properties ]					
Add Web Call Back						
Web Rules Name	web_call_b					
Web URL	http://192.168.0.126/callback/web_call_b.php					
Select IVR to Play	business time 💌					
Enable CAPTCHA	Yes 💌					
APPLY Cancel						

From this example, We get URL Link

:<u>http://192.168.0.126/callback/web\_call\_b.php</u>We can go to this page and enter our phone number for automatic call back.



### 10.3 External Database Connector

Go to menu Call Features >> External Database Connectorand choose New Connector

### External Database Connector

[List connector ] [	New Connector J	
New Connector		
Connector Name		
Description		
		1.
Database Type	Mysql 💌	
Server IP Address		]
Username		
Password		
		APPLY

Connector Name : Specify name

Description : Specify the description

Database Type : Select database type

Server IP Address : Specify IP-Address of database server (or you can fill 'localhost' to

connect internal database)

Username :Specify username of the databaseserver

Password : Specify password of the database server

### Then click apply button.

There 2 types to query database
1. Automatic SQL Query
Database Name : Choose database
Table Name : Choose database
Request Field Name : Choose field name
Compare Field Name : Choose field to compare
Request Value : Input the test value (for testing)

Automatic SQL Query HOST : localhost					1	3
Database Name:	test		•			
Table Name:	ch_db	•				
Request Field Name:	ID_Card	•				
Compare Field Name:	ID_Card	•				_
Request Value:				AND		
					TEST	•

After select database, table and field, you can test by input the data into Request Value and pressTEST button. The system will bring those Request Valueto compare with field name and return the rquest field nameto show on the screne.

2. Manual SQL Query Command - Input SQL query string by yourself.

### Manual SQL Query Command

Request Value 1	%CMP1%	
Request Value 2	%CMP2%	
Request Value 3	%CMP3%	TEST

### Chapter11: Help

on the top right of the

You can take a quick look for key feature code at Help menu
---

Features Code	
Call Center Code Agent Login (permanent) :	
Agent Login (permanent).	( <b>#</b> ) (wxvz9)
Agent Callback Automatic Login/Logoff :	(#) (ari 4) (ari 5)
Agent CallBack Login :	(#) (and 4) (0)
Agent CallBack Logoff :	(#) (are 4) (1)
Pause Agent Prefix :	(#) (are 4) (ABC 2)
UnPause Agent Prefix :	(#) (GHI 4) (DEF 3)
Transfer to Agent Prefix :	(GHI 4) (GHI 4)
Whisper:	(*** * (****29) (*PORS 7)
Private Whisper :	(*** (********************************
Channel Spy :	(exvw) (exvx) (****
Call Parking Parking Number :	(PORS 7) (0) (0)
Parking Position :	
Max Parking Time :	1) (ABC 2) (0)
Transfer digit timeout :	DEF 3
Features Key Mapping	
Features digit timeout(ms) :	(1) (1) (1) (1) (1) (1) (1) (1) (1) (1)
Call Pickup :	(*** *) (TUY 8)
Extensions Pickup :	(*** ** (**** ***
Blind Transfer :	<b>#</b> 1
Attend Transfer :	<b>#</b> (ABC <b>2</b> )
Disconnect :	<b>#</b>
One Touch Record :	<b>#</b> (DEF <b>3</b> )
Voicemail :	100
Phone Lock :	(exxw)
Roaming Station Register / Dial-Out Prefix :	(
Fax Prefix :	(

O C Chapter: Chapter 10 CallerID Routing & Call Back