
MultiVOIP®

Voice/Fax over IP Gateways

Configuration Guide for
MultiVOIP Units deployed with
Avaya™ Communication Manager

Models:

MVP-130/210/410/810-AV



Configuration Guide

S000299G

Analog MultiVOIP Units Models MVP-130/210/410/810-AV for use with Avaya™ Communication Manager

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Record of Revisions

Revision	Date	Description
A	04/25/03	Initial Release.
B	06/19/03	Manual revised to include name change to Avaya Communication Manager.
C	08/20/03	Manual revised to include IP Phone Survivability. Software Vs 9.05.28
D	12/26/03	Manual revised to correct IP Phone and FXS port configuration. Software Vs 9.05.28 for MVP210/410/810-AV and software Vs 2.04.17 for MVP130.
E	05/10/04	Manual revised for software version 9.06.xx. Added support for additional voice coders, DID, Caller ID remote ping feature, off-hook alerting, and audible message waiting.
F	12/07/04	Manual revised to include editorial comments.
G	06/26/08	Manual revised to include support for 16xx and 96xx IP Phone Modules, in addition to the 46xx series. This update documents VOIP Software Version 9.06.DV or higher.
H	02/23/09	Manual revised to include support for IP phone 4622 and VOIP software version 2.06 for MVP130-AV

Patents

This Product is covered by one or more of the following U.S. Patent Numbers: *6151333, 5757801, 5682386, 5.301.274; 5.309.562; 5.355.365; 5.355.653; 5.452.289; 5.453.986.* Other Patents Pending.

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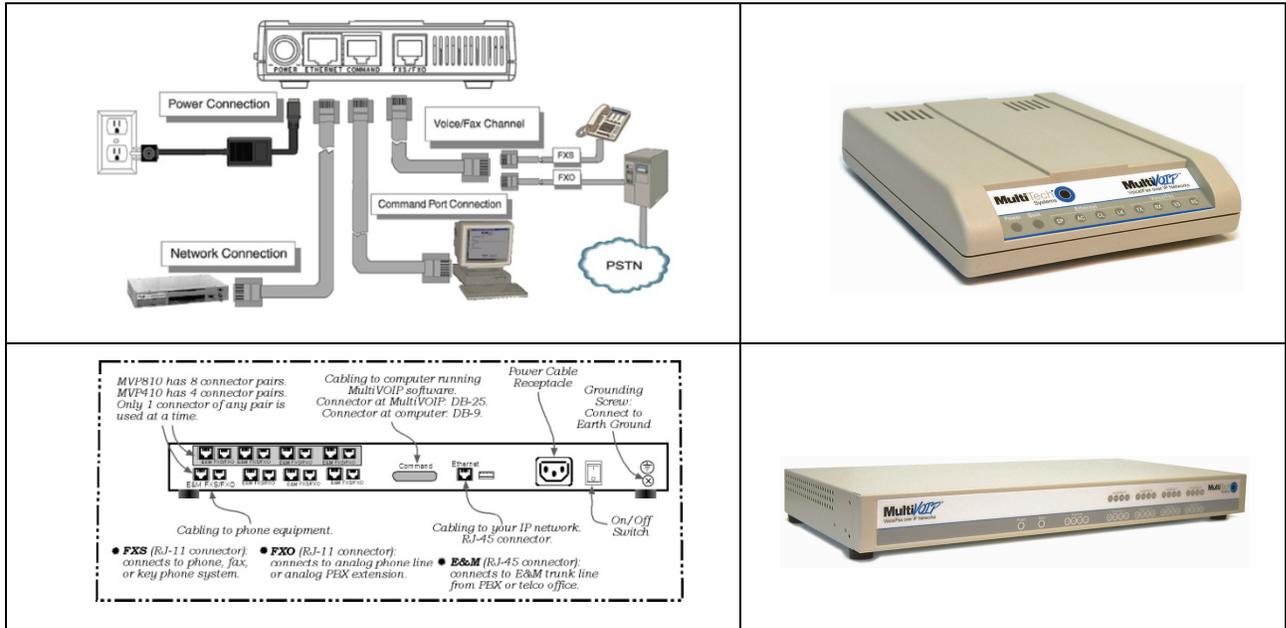
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Chapter 1 - Introduction

About This Manual

MultiVOIP models (MVP130-AV, MVP130-AV-FXS, MVP210-AV, MVP410-AV, & MVP810-AV) being used in systems containing Avaya PBX units and controlled by Avaya™ Communication Manager software differ only slightly from those being used in general voice-over-IP applications. Because this is so, comprehensive information on the general setup of these MultiVOIP units can be found in the MultiVOIP User Guide. However, certain MultiVOIP parameters are unique to operation in Communication Manager systems. This manual describes these parameters.



Beyond general setup information (like Mechanical Installation, & Cabling, LED definitions, and other general info which are presented in the MultiVOIP User Guide & Quick Start Guide), this software guide presents all the information you need to configure your MultiVOIP for use in a Communication Manager system. It presents information on the MultiVOIP software screens as well as the related Communication Manager screens. The MultiVOIP User Guide remains valuable as a reference source.

MultiVOIPs in Avaya Communication Manager Systems

The Communication Manager build of the MultiVOIP software allows for use of MultiVOIP units in conjunction with IP-equipped Avaya PBX units. These PBXs are equipped with H.323 gatekeeper functionality and support endpoints such as IP phones and MultiVOIP gateways.

In addition to functioning as a gateway, the MultiVOIP also contains an integrated gatekeeper, except MVP130 Models. This allows the MultiVOIP to serve as it's own alternate gatekeeper should Communication Manager be unavailable. When serving as an alternate gatekeeper, the MultiVOIP processes registration and call routing for it's local FXS/FXO/DID ports as well as local Avaya IP phones. This is the MultiVOIP's "gatekeeper survivability" mode.

Each MultiVOIP voice channel can be registered with a primary Communication Manager gatekeeper or one of two optional alternate gatekeepers. The optional gatekeepers can be other Communication Manager gatekeepers or the integrated MultiVOIP gatekeeper.

The diagram below shows a common function for the MultiVOIP unit within the Communication Manager network: the MultiVOIP allows selected callers at a site remote from the Avaya PBX units to have access to the PBX units and all of their extensions without tolls. Voice and fax calls are supported.

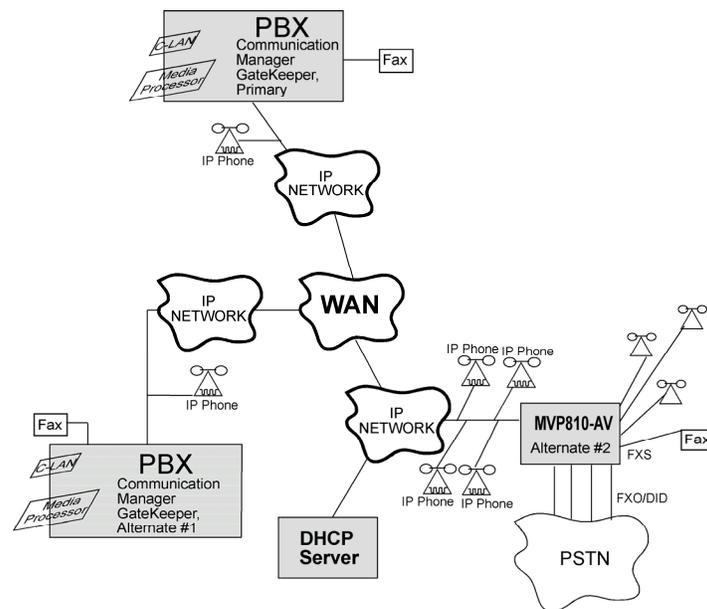


Figure 1. MultiVOIP in Avaya Communication Manager System

The MVP810-AV shown in the diagram on the previous page functions as a gateway registered to one of the Communication Managers in the IP network. The MVP810-AV provides survivable calling between the IP Phones, stations (FXS) and trunk (FXO/DID) ports at the remote site should the WAN link go down.

The MultiVOIP will not serve as a survivable gatekeeper for IP phones on the other side of the wan link.

If the WAN link goes down, the MultiVOIP will try to register six times with the Primary Communication Manager Gatekeeper at 3-second intervals. If the Primary gatekeeper does not respond, the MultiVOIP will try to register with the Alternate #1 Communication Manager Gatekeeper six times at 3-second intervals. If the alternate #1 gatekeeper does not respond, the MultiVOIP will register with the Alternate #2 gatekeeper, which is the integrated MultiVOIP gatekeeper. If the MultiVOIP registers to its own gatekeeper, this is gatekeeper survivability mode where the MultiVOIP will process registration and call routing for its local FXS/FXO/DID ports as well as local Avaya IP phones.

While in gatekeeper survivability mode, the MultiVOIP checks for the availability of the primary Communication Manager gatekeeper every 3-minutes. The MultiVOIP will re-register and discontinue its survivability mode automatically when the primary Communication Manager gatekeeper becomes available. Any calls that were established during the "survivable mode" which are still active while the MultiVOIP gateway returns to "subtending mode" to the CM gatekeeper, shall be preserved by the MultiVOIP gatekeeper. When these calls become inactive (idle), the MultiVOIP gatekeeper shall force these endpoints to unregister with the MultiVOIP gatekeeper and re-register with the CM gatekeeper.

If the MultiVOIP is not configured as an alternate gatekeeper, and none of the Primary, Alternate #1, or Alternate #2 CM gatekeepers respond, the MultiVOIP will enter gateway survivable mode (if enabled) and will process calls according to its inbound and outbound phone books. In gateway survivable mode, the MultiVOIP will check for the availability of the primary Communication Manager gatekeeper according to the Switching Time Interval programmed into the Communication Manager Parameters screen of the MultiVOIP configuration. The MultiVOIP will exit gateway survivability mode if it is able to register with the Primary CM gatekeeper. Gateway survivability mode is enabled by checking the "Enable Survivability Mode" checkbox in the Communication Manager Parameters screen. If IP phone survivability is required, you must enable gatekeeper survivability mode instead by configuring the MultiVOIP as one of the alternate #1 or alternate #2 gatekeepers.

The following Avaya IP Phone models are supported in gatekeeper survivable mode:

4601	4602	4606	4610	4612	4620	4621	4622	4624	4625
9610	9620	9630	9640	9650					
1603	1608	1616							

Chapter 2 - Configuration

Configuring MultiVOIP for Avaya Communication Manager Use

To configure the MultiVOIP unit within such a system (similar to that in Figure 1), use the **Communication Manager Parameters** screen within the MultiVOIP program. If “Register with Communication Manager Gatekeeper” is selected, then many parameters must assume default values and those will be grayed out on the screen. In setting up the MultiVOIP unit for Communication Manager use, bear in mind that you may configure the MultiVOIP to operate with any combination of stations (FXS) and/or trunks (FXO/DID).

1. Configure IP Parameters

- 1a. Configure the IP address, gateway address, and mask used by the MultiVOIP unit in the **Configuration | IP Parameters** screen of the MultiVOIP program. The IP address must match that used in the Communication Manager “Change Node Names IP” form (dialog box).
- 1b. DiffServ PHB (Per Hop Behavior) values pertain to a differential prioritizing system for IP packets as handled by DiffServ-compatible routers. There are 64 values, each with an elaborate technical description. These descriptions are found in TCP/IP standards RFC2474, RFC2597, and, for present purposes, in RFC3246, which describes the value 34 (34 decimal; 22 hex) for Assured Forwarding behavior (default for Call Control PHB) and the value 46 (46 decimal; 2E hexadecimal) for Expedited Forwarding behavior (default for Voip Media PHB).

If you wish to use just one DiffServ PHB value, Avaya recommends the value 46 (46 decimal; 2E hex) for both Call Control and VOIP Media.

- 1c. If you require that the MultiVOIP get its IP address from a DHCP server, select the “Enable DHCP” option. The DHCP server must be located at the same site as the MultiVOIP so it is available in the event of wan link failure. The DHCP server must issue an IP address that is statically defined on the DHCP server so that the MultiVOIP gets the same IP address all the time.
- 1d. **TDM Routing Option.** Calls placed between ports on the same MultiVOIP voice channel board will normally be routed over the MultiVOIP’s internal TDM (Time Division Multiplexed) bus. TDM calls have less delay than calls routed over IP. On MVP410/810-AV models, calls between channels 1-4 or channels 5-8 will be TDM routed. Calls between voice boards (for example, channel 1 calling channel 5) will be IP routed. If you require all calls to be IP routed, disable the “use TDM Routing for Intra-Gateway Calls” option. This should not normally be required, so it is recommended that you leave TDM Routing enabled.

The screenshot shows the 'IP Parameters' configuration window. It is divided into several sections:

- Diff Serv Parameters:**
 - Call Control PHB: 34
 - VoIP Media PHB: 46
 - Frame Type: TYPE-II (dropdown menu)
- IP Parameters:**
 - Enable DHCP:
 - IP Address: 192 . 168 . 25 . 48
 - IP Mask: 255 . 255 . 255 . 0
 - Gateway: 192 . 168 . 25 . 1
- DNS:**
 - Enable DNS:
 - DNS Server IP Address: [Empty field]
- FTP Server:**
 - Enable:
- TDM Routing Option:**
 - Use TDM Routing For Intra-Gateway calls:

Buttons for 'OK', 'Cancel', and 'Help' are located on the right side of the window.

```

Telnet - 192.168.25.10
Connect Edit Terminal Help
change node-names ip Page 1 of 1
IP NODE NAMES
Name IP Address Name IP Address
MultiVOIP 192.168.25.48 . . .
MultiVOIP2 192.168.25.49 . . .
MultiVOIP3 192.168.25.50 . . .
clan1 192.168.25.30 . . .
cuctserver 192.168.25.20 . . .
default 0.0.0.0 . . .
medpro1 192.168.25.40 . . .

```

- 1e. **TCP/UDP port requirements.** To ensure proper functioning of the MultiVOIP, the TCP/UDP network connection must not be blocked on the network connection between the MultiVOIP and CM.

The UDP ports used by the MultiVOIP are as follows:

Channel	RTP Port	RTCP Port	T38 Port
1	5006	5007	5008
2	5016	5017	5018
3	5026	5027	5028
4	5036	5037	5038
5	5046	5047	5048
6	5056	5057	5058
7	5066	5067	5068
8	5076	5077	5078

The TCP ports used by the MultiVOIP are as follows:

H.225 Listen Port – User configured for each channel in the CM Settings screen (default 1721 – 1728)

H.225 Client Port – Dynamic (16000 – 20000)

H.245 Listen Port – Dynamic (16000 – 20000)

RAS Client Port – Dynamic (16000 – 20000)

- 1f. **Emergency Transfer.** Emergency transfer allows local analog telephones to access the local CO and to answer telephone calls during a power failure.

For the various products the features operate as follows:

MVP210-AV

Port 1 is connected to Port 2

MVP410-AV

Port 1 is connected to Port 2

Port 3 is connected to Port 4

MVP810-AV

Port 1 is connected to Port 2

Port 3 is connected to Port 4

For example, suppose ports 1 and 3 of MVP410-AV are configured as FXS with telephones attached. Ports 2 and 4 are configured as FXO trunks with CO lines attached. If there is a power failure, the telephones on ports 1 and 3 would automatically be connected to the CO lines attached to ports 2 and 4 allowing inbound and/or outbound call to be make.

Note: This feature shall only be available on “AV” models of the gateway that are:

Rev B: For the MVP410 and MVP810 models

Rev B: For the MVP210 model

2. Configure Interface Parameters.

- 2a. Determine which voice channels will be stations (FXS), trunks (FXO/DID), or disabled. Configure them as such in the **Configuration | Interface** screen of the MultiVOIP GUI.

- 2b. If you are configuring a DID-DPO (Direct Inward Dial – dial-pulse originating) interface for DID incoming calls, you need to choose DID-DPO from the pull down menu for Interface Type. You also have to set the DID Options Start Modes; Immediate Start, Wink Start, or Delay Dial.

Immediate Start allows the MultiVOIP to detect the off-hook condition at the originating end of the call and becomes ready to receive the digits immediately.

Wink Start allows the MultiVOIP to detect the off-hook condition at the originating end of the call. The MultiVOIP then reverses the battery polarity for a specified time (140-290 ms; a “wink”) and then becomes ready to receive the dialed digits. You can select the Wink Timer duration (140 to 290 ms).

Delay Dial allows the MultiVOIP to detect the off-hook condition at the originating end of the call. Then the MultiVOIP reverses battery polarity for a specified time (reverse polarity duration has a wider acceptable range than Wink Start) and then becomes ready to receive dialed digits.

- 2c. Enable the “Message Waiting Light” feature if you have FXS Interface selected and are using an Avaya analog telephone with Message Waiting indicator or enable “Stutter Dial Tone” if you are

using a non-Avaya analog telephone. This station must be properly administered in the Audix Voice Mail System for the message waiting feature to work.

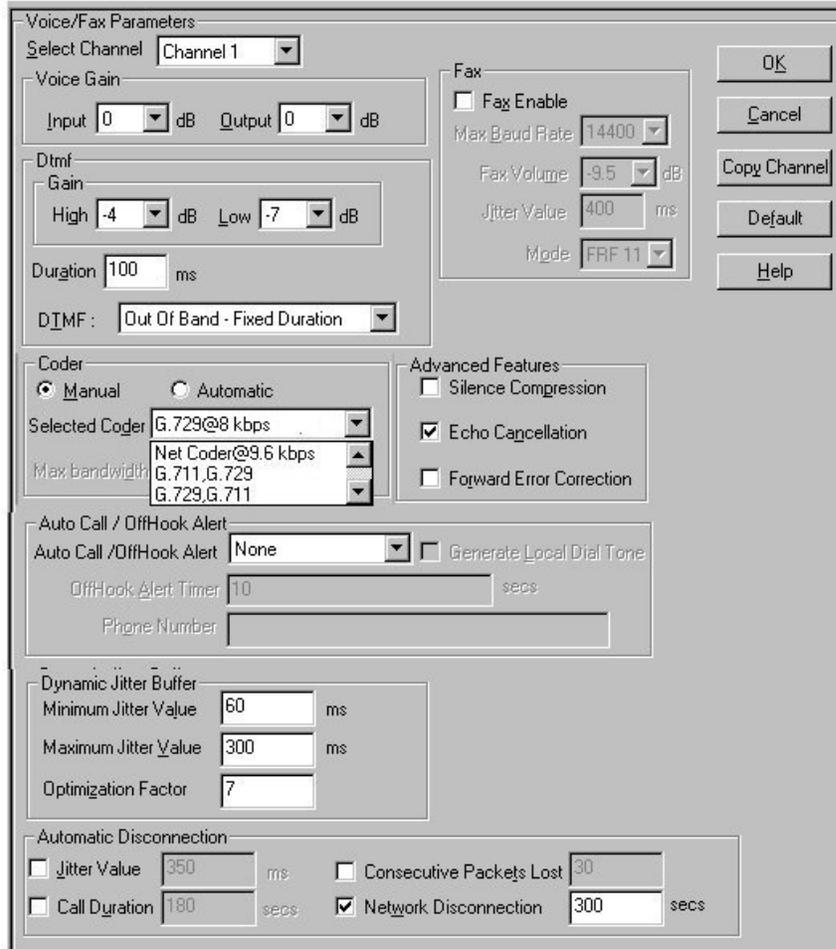
- 2d. Activate the “Caller ID” feature if you have FXS Interface selected and are using an analog telephone with caller ID display. The MultiVOIP will output Caller ID information, if received, between the first and second rings.
- 2e. Activate “Caller ID” on FXO interfaces if the attached line supports it. The MultiVOIP will capture the Caller ID information, if provided, between the first and second rings and will forward the information to CM.

Note: If the CM customer is operating on CM release 2.0.1 (or newer), remember that for the ‘h323’ specified station, the following CM admin fields must be set appropriately:

- a) The ‘Message Waiting’ field must be enabled and set to “led” if this feature is intended to be supported on the MultiVOIPs analog station port.
- b) Similarly, the following features must be enabled as desired:
 - Switchhook flash
 - Call Waiting Indication
 - Attendant Call Waiting Indication

3. Configure Voice/Fax Parameters.

- 3a. In the MultiVOIP’s **Configuration | Voice/Fax Parameters** screen, configure the voice channels for a voice coder that is supported by Communication Manager.



The voice coder selected must match the voice coders listed in the ip-codec-set for the network region the MultiVOIP resides in. If only G.729 is listed in the ip-codec-set, select G.729 on the MultiVOIP. If G.711 is listed before G.729 in the ip-codec-set, select G.711, G.729 on the MultiVOIP. If G.729 is listed before G.711 in the ip-codec-set, select G.729, G.711 on the MultiVOIP.

There is one exception, if only G.711 is listed in the ip-codec-set, you must select G.711, G.729 on the MultiVOIP. This will allow support for G.711 for CM managed calls, and will allow support for G.729 for survivable mode calls. The MultiVOIP only supports G.729 in survivable mode.

The table below shows the bandwidth required per voice channel based on the codec selected and the CM “Frames per Packet” setting in the “Change IP-Codec-Set” screen.

Bandwidth per Channel Voice Frames per IP Packet				
Coder	1	2	3	4
G.729	40K bps	24K bps	18K bps	16K bps
G.711	96K bps	80K bps	74.7K bps	72K bps

- 3b. If the CM customer desires that a particular FXS port is to operate as a “hot extension”, set the “auto call/off-hook alerting” feature to auto call and enter the desired destination number in the “phone number” field. Enabling this feature allows an analog phone user to be able to lift his/her

handset (and without dialing the number) immediately call the far-end party. To prevent toll fraud on FXO trunk ports, set the “autocall enable” feature and enter the phone number for a receptionist or auto-attendant.

- 3c. The MultiVOIP supports off-hook alerting on FXS ports. To enable the feature, set the “auto call/off-hook alert” feature to “off-hook alert” and enter the destination number in the “phone number” field. If the user lifts the handset, but does not dial a number, dial tone will be heard for 10 seconds, followed by intercept tone for the number of seconds configured in the “off-hook alert timer” field. If this timer expires, a call will automatically be placed to the destination number in the “phone number” field.
4. Configure the station (FXS) channels in the **Communication Manager Parameters** screen.

Communication Manager MultiVOIP: FXS Channels

Communication Manager Parameters

Select Channel: Channel 1

Interface Type is FXS (Loop Start)

Q.931 Parameters:

Use Fast Start

Call Signaling Port: 1721

Delay before Dial: 400 ms

Register with Communication Manager

Gatekeeper RAS Parameters:

Gateway Name: MultiVoIP

Gateway Prefix:

Gateway H323 ID:

Phone Number / Extension: 6001

Security Code: xxxxx

OK

Cancel

Default

Help

Unregister

Register

	GateKeeper IP Address	RAS Port	GateKeeper Name
Primary	192 . 168 . 25 . 30	1719	
Alternate GK 1	0 . 0 . 0 . 0	1719	
Alternate GK 2	0 . 0 . 0 . 0	1719	

RAS TimeToLive: 60 secs

Enable Survivability Mode

Switching Time Interval: 60 secs

H323 Version 4 Options

Q.931 Multiplexing [Mux] H.245 Tunneling [Tun]

Parallel H.245 [FS+Tun] Annex -E [AE]

- 4a. Make sure the “Register with Communication Manager Gatekeeper” option is checked.
- 4b. Configure Call Signaling Port. This needs to be unique on each registered FXS channel and must fall within the range of port values supported by Communication Manager. Typically, it is OK to use the default values 1721-1728 on channels 1-8 respectively.

- 4c. Configure Gateway Name (optional). This is the name that gets displayed when viewing Call Progress statistics for this channel. If used, the MultiVOIP's Gateway Name must match the name used in the Communication Manager "Node-Names IP" form.

```

Telnet - 192.168.25.10
Connect Edit Terminal Help
change node-names ip Page 1 of 1
IP NODE NAMES
Name IP Address
MultiVOIP 192.168.25.48
MultiVOIP2 192.168.25.49
MultiVOIP3 192.168.25.50
clan1 192.168.25.30
cvctserver 192.168.25.20
default 0.0.0.0
medpro1 192.168.25.40
. . .
. . .
. . .
. . .
. . .
. . .
. . .
. . .
. . .
. . .
( 7 of 7 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
F1=Cancel F2=Nxt Page F3=Submit F4=Help Esc p=Prv Page Esc r=Refresh
    
```

- 4d. Configure Phone Number / Extension. This must match the extension configured for this station in the Communication Manager "station" form. If "Message Waiting Indicator" is required for this extension, set this option to "led" in the CM "station" form.

```

Telnet - 200.2.9.100
Connect Edit Terminal Help
change station 6001 Page 1 of 4
STATION
Extension: 6001 Lock Messages? n BCC: 0
Type: 323 Security Code: * TN: 1
Port: 300037 Coverage Path 1: 1 COR: 1
Name: MultiVOIP CH1 Coverage Path 2: COS: 1
Hunt-to Station: Tests? y
STATION OPTIONS
Loss Group: 19 Time of Day Lock Table:
Message Waiting Indicator: led
Message Camp EXT: 6001
Survivable COR: internal
Survivable Trunk Dest? y
DTMF over IP: in-band IP Video? n
F1=Cancel F2=Nxt Page F3=Submit F4=Help Esc p=Prv Page Esc r=Refresh
    
```

Note that, in page 2 of the Communication Manager “Station” form, the “Direct IP-IP Audio Connection” field must be set to “y” and the “IP Audio Hairpinning” field must be set to “n.”

```

Telnet - 200.2.9.100
Connect Edit Terminal Help
change station 6001                                     Page 2 of 4
STATION
FEATURE OPTIONS
  LWC Reception: Spe
  LWC Activation? y
  LWC Log External Calls? n
  CDR Privacy? n
  Redirect Notification? y
  Per Button Ring Control? n
  Bridged Call Alerting? n
  Switchhook Flash? y
  H.320 Conversion? n
  Per Station CPN - Send Calling Number?
  HMI Served User Type: qsig-mwi
  Coverage After Forwarding? s
  Direct IP-IP Audio Connections? y
  IP Audio Hairpinning? n
Emergency Location Ext: 6001
F1=Cancel F2=Nxt Page F3=Submit F4=Help Esc p=Prv Page Esc r=Refresh

```

- 4e. Configure Security Code. This must match the security code configured in the Communication Manager “Station” form (see first graphic in step 4d).
- 4f. Configure Primary Gatekeeper IP Address. This must be configured to match the IP address of the CLAN or PROCR card in the “IP Node Names” form of the primary CM gatekeeper. Do not change the RAS port from the default 1719 and leave the Gatekeeper Name field blank.

```

Telnet - 192.168.25.10
Connect Edit Terminal Help
change node-names ip                                   Page 1 of 1
IP NODE NAMES
Name      IP Address  Name      IP Address
MultiVOIP 192.168.25 .48
MultiVOIP2 192.168.25 .49
MultiVOIP3 192.168.25 .50
CLAN1     192.168.25 .30
cuciserver 192.168.25 .20
default   0 .0 .0
medpro1   192.168.25 .40
. . .
. . .
( 7 of 7 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
F1=Cancel F2=Nxt Page F3=Submit F4=Help Esc p=Prv Page Esc r=Refresh

```

- 4g. Configure IP addresses for Alternate Gatekeeper 1 and Alternate Gatekeeper 2 (optional). These must be configured to match the IP address of the respective CLAN or PROCR card in the PBX unit serving as the Alternate Gatekeeper. This field appears in IP Node Names form of the Communication Manager software for that PBX. If you are using the MultiVOIP Gatekeeper as an alternate gatekeeper, enter the IP address of the MultiVOIP. The alternate MultiVOIP gatekeeper must appear after any Communication Manager alternate gatekeepers.

5. Configure trunk (FXO) channels in the Communication Manager Parameters screen.

Communication Manager MultiVOIP: FXO Channels

Communication Manager Parameters

Select Channel: Channel 5

Interface Type is FXO

Q.931 Parameters

Use East Start

Call Signaling Port: 1725

Delay before Dial: 400 ms

Register with Communication Manager

Gatekeeper RAS Parameters

Gateway Name: MultiVoIP

Gateway Prefix:

Gateway H323 ID:

Phone Number / Extension: 1725

Security Code: ****

OK

Cancel

Default

Help

Unregister

Register

	GateKeeper IP Address	RAS Port	GateKeeper Name
Primary	192 . 168 . 25 . 30	1719	
Alternate GK 1	0 . 0 . 0 . 0	1719	
Alternate GK 2	0 . 0 . 0 . 0	1719	

RAS TimeToLive: 60 secs

Enable Survivability Mode

Switching Time Interval: 60 secs

H323 Version 4 Options

Q.931 Multiplexing [Mx] H.245 Tunneling [Tun]

Parallel H.245 [F8+Tun] Annex-E [AE]

- 5a. Make sure the “Register with Communication Manager Gatekeeper” option is checked.

- 5b. Configure Call Signaling Port. Unlike station (FXS) ports, which require a unique value for each channel, trunk (FXO) ports require that the same Call Signaling Port value be used on all trunk (FXO) ports that are registered to the same Communication Manager PBX. Configure this to a unique value that is not being used by another station (FXS) or trunk (DID) port on this MultiVOIP unit. This port value must fall within the range of port values supported by Communication Manager and should match the “Far-end Listen Port” value configured in the Communication Manager “Signaling Group” (“Display Signaling Group”) form (dialog box). All FXO ports on this MultiVOIP will use the same Signaling Group for communication with the CM Server.

```

Telnet - 200.2.9.100
Change signaling-group 20
Page 1 of 5

SIGNALING GROUP

Group Number: 20
Group Type: h.323
Remote Office? n
SBS? n
Max number of NCA TSC: 0
Max number of CA TSC: 0
Trunk Group for NCA TSC:

IP Video? n
Trunk Group for Channel Selection: 20
TSC Supplementary Service Protocol: a
1303 Timer(sec): 10

Near-end Node Name: procr
Near-end Listen Port: 5000

ARQ Required? n
RRQ Required? y
Media Encryption? n

DTMF over IP: out-of-band
Link Loss Delay Timer(sec): 90
Enable Layer 3 Test? n
H.323 Outgoing Direct Media? n

Far-end Node Name: MultiVOIP
Far-end Listen Port: 1725
Far-end Network Region:
Calls Share IP Signaling Connection? y
Bypass If IP Threshold Exceeded? n
H.235 Annex H Required? n
Direct IP-IP Audio Connections? y
IP Audio Hairpinning? n
Interworking Message: PROGRESS
DCP/Analog Bearer Capability: 3.1kHz

F1=Cancel F2=Next Page F3=Submit F4=Help Esc p=Prv Page Esc r=Refresh

```

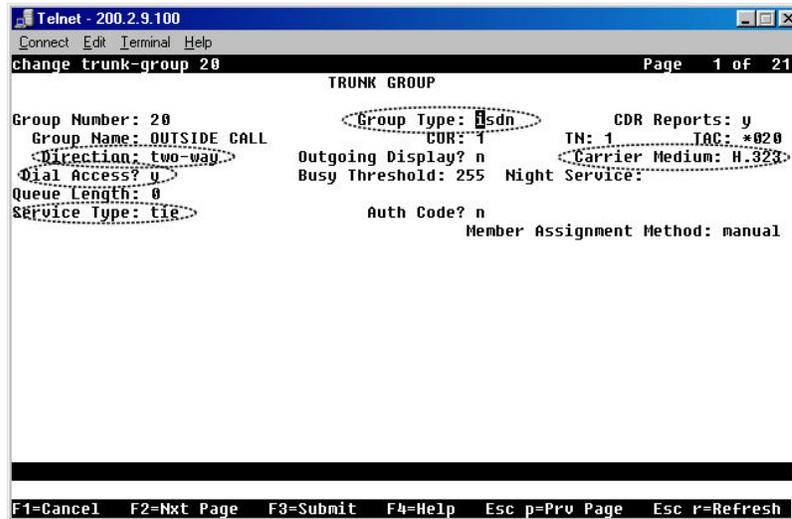
Note that several other fields in the Communication Manager “Signaling Group” form (dialog box) must be set to accommodate operation with the MultiVOIP:

Group Type	= h.323
Remote Office?	= n
RRQ Required?	= y
Direct IP-IP Audio Connection?	= y
IP Audio Hairpinning?	=n
Calls Share IP Sig Connection	=y

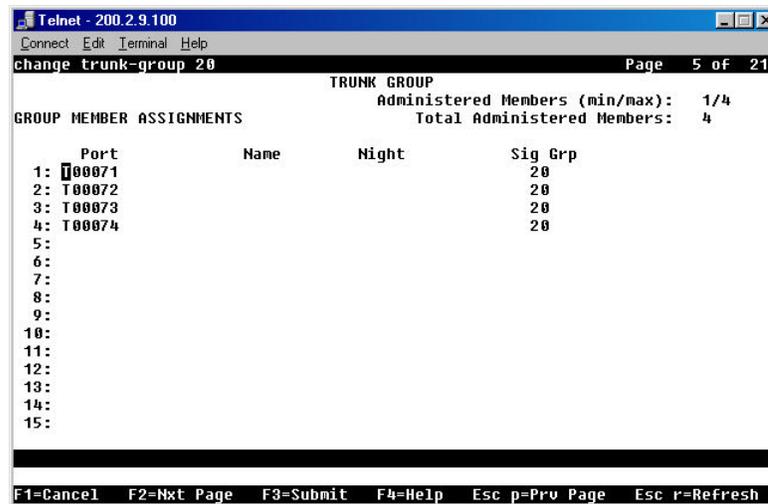
- 5c. Configure Delay before Dial value. This is the delay after the MultiVOIP goes off hook before it dials a number to the attached PSTN line. Some PSTN lines may require more than the default 400 ms value. Configure as appropriate for the PSTN lines you are using. If you are not sure, leave the default value in place.
- 5d. Configure Gateway Name. This should match the value configured for this MultiVOIP unit in the Communication Manager “IP Node Names” and “Signaling Group” forms.
- 5e. Configure Primary Gatekeeper IP Address. This must be configured to match the IP address of the CLAN or PROCR card in the “IP Node Names” form of the primary Communication Manager Gatekeeper unit. Do not change the RAS Port and leave the Gatekeeper name field blank.
- 5f. Configure IP addresses for Alternate Gatekeeper1 and Alternate Gatekeeper 2 (optional). These must be configured to match the IP address of the respective CLAN or PROCR card in the PBX unit serving as the Alternate Gatekeeper. This field appears in IP Node Names form of the Communication Manager software for that PBX. If you are using the MultiVOIP Gatekeeper as an alternate gatekeeper, enter the IP address of the MultiVOIP. The alternate MultiVOIP gatekeeper must appear after any Communication Manager alternate gatekeepers.

- 5g. To accommodate the MultiVOIP unit, the following fields in the Communication Manager “Trunk Group – Page 1” form must be set as shown:

Dial Access?	= y
Service Type	= tie
Group Type	= ISDN
Carrier Medium	= IP/H.323



- 5h. To accommodate the MultiVOIP unit, all of the MultiVOIP’s FXO channels must be listed as trunk-group members in the Communication Manager Trunk Group form (“Change Trunk-Group” dialog box).



For example, if four channels of an MVP810-AV are configured as FXO, four trunk group numbers must be listed, as above.

6. Configure trunk (DID) channels in the Communication Manager Parameters screen.

Communication Manager MultiVOIP: DID Channels

Communication Manager Parameters

Select Channel: Channel 1

Interface Type is DID-DPD

Q.931 Parameters:

Use Fast Start

Call Signaling Port: 1726

Delay before Dial: 400 ms

Register with Communication Manager

Gatekeeper RAS Parameters:

Gateway Name: MultiVoIP

Gateway Prefix:

Gateway H323 ID:

Phone Number / Extension: 3201

Security Code:

	GateKeeper IP Address	RAS Port	GateKeeper Name
Primary	192 . 168 . 25 . 30	1719	
Alternate GK 1	0 . 0 . 0 . 0	1719	
Alternate GK 2	0 . 0 . 0 . 0	1719	

RAS TimeToLive: 60 secs

Enable Survivability Mode

Switching Time Interval: 60 secs

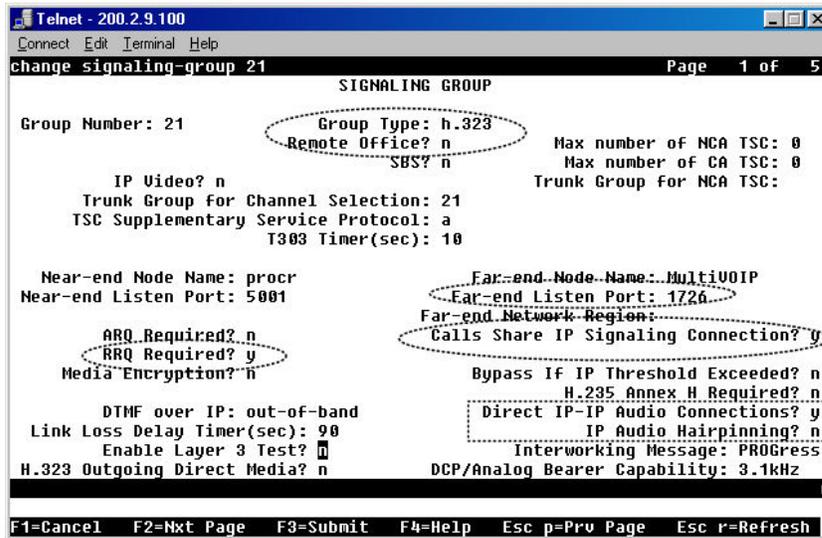
H323 Version 4 Options:

Q.931 Multiplexing [Mx] H.245 Tunneling [Tun]

Parallel H.245 [P6+Tun] Annex-E [AE]

- 6a. Make sure the "Register with Communication Manager Gatekeeper" option is checked.

- 6b. Configure Call Signaling Port. Unlike station (FXS) ports, which require a unique value for each channel, trunk DID ports require that the same Call Signaling Port value be used on all DID trunk ports that are registered to the same Communication Manager PBX. Configure this to a unique value that is not being used by another station (FXS) or trunk (FXO) port on this MultiVOIP unit. This port value must fall within the range of port values supported by Communication Manager and should match the “Far-end Listen Port” value configured in the Communication Manager “Signaling Group” (“Display Signaling Group”) form (dialog box). All DID trunk ports on this MultiVOIP will use the same Signaling Group for communication with the CM Server.



Note that several other fields in the Communication Manager “Signaling Group” form (dialog box) must be set to accommodate operation with the MultiVOIP:

Group Type	= h.323
Remote Office?	= n
RRQ Required?	= y
Direct IP-IP Audio Connection?	= y
IP Audio Hairpinning?	= n
Calls Share IP Sig Connection	=y

- 6c. Configure Gateway Name. This should match the value configured for this MultiVOIP unit in the Communication Manager “IP Node Names” and “Signaling Group” forms.
- 6d. Configure Primary Gatekeeper IP Address. This must be configured to match the IP address of the CLAN or PROCR card in the “IP Node Names” form of the primary Communication Manager Gatekeeper unit. Do not change the RAS Port and leave the Gatekeeper name field blank.
- 6e. Configure IP addresses for Alternate Gatekeeper1 and Alternate Gatekeeper 2 (optional). These must be configured to match the IP address of the respective CLAN or PROCR card in the PBX unit serving as the Alternate Gatekeeper. This field appears in IP Node Names form of the Communication Manager software for that PBX. If you are using the MultiVOIP Gatekeeper as an alternate gatekeeper, enter the IP address of the MultiVOIP. The alternate MultiVOIP gatekeeper must appear after any Communication Manager alternate gatekeepers.

- 6f. To accommodate the MultiVOIP unit, the following fields in the Communication Manager “Trunk Group – Page 1” form must be set as shown:

Dial Access?	= n
Service Type	= tie
Group Type	= ISDN
Carrier Medium	= IP/H.323

```

Telnet - 200.2.9.100
Change trunk-group 21
TRUNK GROUP
Group Number: 21          Group Type: isdn          CDR Reports: y
Group Name: OUTSIDE CALL  COR: 1                   TN: 1                   IAC: *021
Direction: incoming      Outgoing Display? n      Carrier Medium: H.323
Dial Access? n           Busy Threshold: 255      Night Service:
Service Type: tie
Auth Code? n
Member Assignment Method: manual
  
```

- 6g. To accommodate the MultiVOIP unit, all of the MultiVOIP’s DID channels must be listed as trunk-group members in the Communication Manager Trunk Group form (“Change Trunk-Group” dialog box).

```

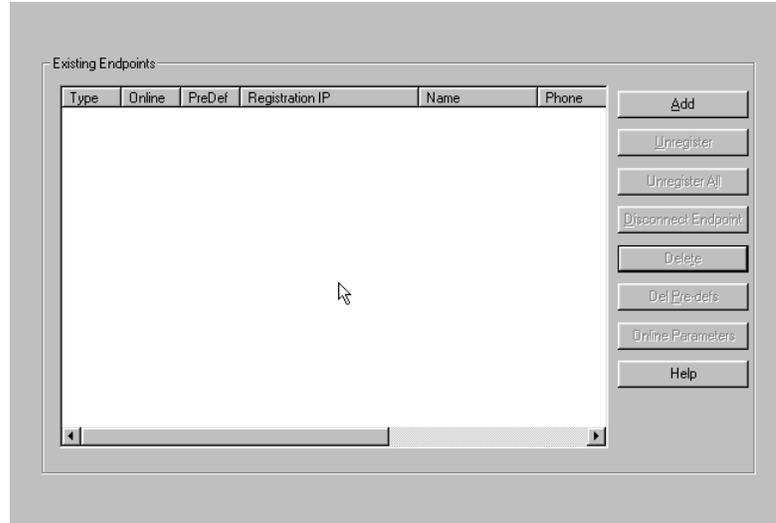
Telnet - 200.2.9.100
Change trunk-group 21
TRUNK GROUP
Administered Members (min/max): 1/4
Total Administered Members: 4
GROUP MEMBER ASSIGNMENTS
Port      Name      Night      Sig Grp
1: T00075
2: T00076
3: T00077
4: T00078
5:
6:
7:
8:
9:
10:
11:
12:
13:
14:
15:
  
```

For example, if four channels of an MVP810-AV are configured as DID, four trunk group numbers must be listed, as above.

7. Configure MultiVOIP gatekeeper (except for the MVP130 Models) to provide survivability to IP phones and local FXS/FXO/DID ports.

If you are using Avaya IP phones in addition to the MultiVOIP, you must configure the MultiVOIP gatekeeper to provide survivability for the IP phones and local FXS/FXO/DID ports in the event of WAN link failure.

- 7a. In the MultiVOIP’s **Gatekeeper | Endpoints** screen, add predefined endpoints to the gatekeeper.



One endpoint entry is required for each FXS port or IP phone. Only one entry is required for one or more FXO ports per MultiVOIP gateway. Only one entry is required for one or more DID ports per MultiVOIP gateway.

- 7b. To add an entry for a MultiVOIP FXS port, configure **Endpoint Type**, **Product ID**, **Names**, **Phones**, **Registration IP**, and **Security code** fields as indicated below. Configure the Names field with a unique descriptive identifier. The information contained in the Names and Phones fields will be used for Caller ID purposes in survivable mode. Configure **Phones** and **Security Code** to match the values configured for this station in Communication Manager. The **Registration IP** should match the IP address of the MultiVOIP where the FXS port is located. This could be the IP address of this very same MultiVOIP or another MultiVOIP. Leave the other fields blank. One entry is required for each MultiVOIP FXS port that is configured to register with Communication Manager. Click **OK** to accept this entry. Repeat the above steps to add additional FXS station port entries.

If the autocal feature is being used to route incoming FXO trunk calls to an extension (1000, etc.,) at the main site, you can program a second entry in the “Phones” field to 1000 to route incoming FXO trunk calls to a given FXS station to cover the autocal situation in Gatekeeper Survivable mode.

- 7c. To add an entry for MultiVOIP FXO ports, configure **Endpoint Type**, **Product ID**, **Names**, **Registration IP**, and **Call Signalling Port** fields as indicated below. Configure the **Names** field with a unique descriptive identifier for the FXO trunk. This same name will be used later to map Outbound Phone Book entries with specific digit patterns (area codes, etc.) to this FXO trunk, and for Caller ID purposes in survivable mode. The **Registration IP** should match the IP address of the MultiVOIP where the port is located. This could be the IP address of this very same MultiVOIP or another MultiVOIP. Leave the other fields blank. Only one entry is required for one or more ports on a given MultiVOIP. For example, if the MultiVOIP has four FXO ports, only one gatekeeper entry should be added. Click OK to accept this entry.

Predefined Properties

Predefined Properties for Endpoint Allowed Distance

Endpoint type: Gateway Product Id: MultiVolP Force this Endpoint to Online Status

Names: Minneapolis
 Forward To:

Phones:
 Forward To:

Other Aliases: URL
 Email
 Transport
 Party Number Type

Registration IP: 192.168.25.48 Port
 Call Signalling Port: 1725 Security Code:

- 7d. To add an entry for MultiVOIP DID ports, configure **Endpoint Type**, **Product ID**, **Names**, **Registration IP**, and **Call Signalling Port** fields as indicated below. Configure the **Names** field with a unique descriptive identifier for the DID trunk. This name will be used for Caller ID purposes in survivable mode. The **Registration IP** should match the IP address of the MultiVOIP where the DID port is located. This could be the IP address of this very same MultiVOIP or another MultiVOIP. Leave the other fields blank. Only one entry is required for one or more ports on a given MultiVOIP. For example, if the MultiVOIP has four DID ports, only one gatekeeper entry should be added. Click OK to accept this entry.

Predefined Properties

Predefined Properties for Endpoint Allowed Distance

Endpoint type: Gateway Product Id: MultiVolP Force this Endpoint to Online Status

Names: Minneapolis DID
 Forward To:

Phones:
 Forward To:

Other Aliases: URL
 Email
 Transport
 Party Number Type

Registration IP: 192.168.25.48 Port
 Call Signalling Port: 1726 Security Code:

- 7e. To add an entry for an IP phone, configure **Endpoint Type**, **Product ID**, **Names**, **Phones**, and **Security code** fields as indicated below. Configure **Phones** and **Security Code** to match the values configured for this IP phone in Communication Manager. Leave the other fields blank. One entry is required for each IP phone located at the same site as this MultiVOIP. Do not create entries for IP phones located at other sites. They will not be able to register with this MultiVOIP in the event of a wan link failure. Click **OK** to save this entry. Repeat the above steps to add additional IP phone entries. The MVP210, MVP410, and MVP810 support 5,10, or 15 IP phones respectively.

The screenshot shows the 'Predefined Properties' dialog box. At the top, 'Predefined Properties for Endpoint' is set to '3' and 'Allowed Distance' is '0'. The 'Endpoint type' is 'Terminal' and 'Product Id' is 'AvayaPhone-4612'. There is a checkbox for 'Force this Endpoint to Online Status'. Below are three main sections: 'Names' with a text field containing 'John Smith' and a 'Forward To:' field; 'Phones' with a text field containing '2110' and a 'Forward To:' field; and 'Other Aliases' with fields for 'URL', 'Email', 'Transport', 'Party Number', and 'Type'. At the bottom, there are fields for 'Registration IP', 'Port', 'Call Signalling Port', and 'Security Code' (masked with asterisks).

The IP phones must be configured to use DHCP and the DHCP server must be located at the same site as the MultiVOIP and IP phones. This is the only means by which the IP phones are made aware of alternate gatekeepers, such as the MultiVOIP gatekeeper when the CM is unavailable due to WAN link failure.

Refer to the following Avaya documents for instructions on how to configure the DHCP server.

- 4600 Series IP Telephone Lan Administrator's Guide (Avaya document number 555-233-507). Read the section titled "DHCP" under "Server Administration".
- Avaya one-x Deskphone Edition for 9600 Series IP Telephones Administrator's Guide Release 1.5 (Avaya document number 16-300698). Read the section titled "DHCP and File Server" under "Server Administration".
- Avaya one-x Deskphone Value Edition 1600 Series IP Telephones Administrator's Guide Release 1.0 (Avaya document number 16-601443). Read the section titled "DHCP Server Administration".

Your DHCP server must utilize site-specific options (SSON) #176 (46xx models) and/or #242 (16xx, 96xx models) which are IP Telephone-specific DHCP options specifying information such as TFTP server and Definity CLAN IP address. The MultiVOIP IP address must be listed at the end of the string. MCIPADD=xxx.xxx.xxx.xxx,yyy.yyy.yyy.yyy where xxx.xxx.xxx.xxx is one or more IP addresses used for Definity/CM CLAN IP boards and yyy.yyy.yyy.yyy is the MultiVOIP's IP address.

- 7f. Configure the “Survivable GK Node Name” field with the node name previously given to the MultiVOIP in the “Node-Names IP” form.

Avaya IP phones running firmware version 2.0 or higher require CM 2.0.1 or higher to work in survivable mode with the MultiVOIP. When these phones register with the CM, they replace the list of gatekeepers acquired from the DHCP server (including the MultiVOIP) with a new list of gatekeepers acquired from CM. This differs from 1.x IP phone firmware where the gatekeeper list sent to the IP phone from CM was appended to the list of gatekeepers acquired from the DHCP server. When using CM 2.0.1 or higher, configure the “Survivable GK Node Name” field in the station form for the IP phone. By doing this, the MultiVOIP will be included in the list of gatekeepers sent to the IP phone by CM.

Note: IP phone models 4610SW and 4620SW only support 2.0 or higher firmware, therefore these phones cannot be used with the MultiVOIP unless you are using CM 2.0.1 or higher.

```

add station 2110                                     Page 1 of 4
STATION
Extension: 2110                                     Lock Messages? n      BCC: 0
Type: 4620                                         Security Code: *      TN: 1
Port: IP                                           Coverage Path 1:      COR: 1
Name:                                               Coverage Path 2:      COS: 1
                                                    Hunt-to Station:

STATION OPTIONS
Loss Group: 19                                     Personalized Ringing Pattern: 1
                                                    Message Lamp Ext: 2110
Speakerphone: 2-way                               Mute Button Enabled? y
Display Language: english                         Expansion Module? n
Survivable GK Node Name: MultiVOIP                Media Complex Ext:
                                                    IP SoftPhone? n

F1=Cancel  F2=Nxt Page  F3=Submit  F4=Help  Esc p=Pro Page  Esc r=Refresh

```

- 7g. The completed Endpoints screen is shown below:

Type	Online	PreDef	Registration IP	Name	Phone
Terminal	+		192.168.25.48:0	Mike Jones	6011
Gateway	+		192.168.25.48:0	Minneapolis FXO	
Gateway	+		192.168.25.48:0	Minneapolis DID	
Terminal	+			John Smith	2110

Existing Endpoints

Buttons: Add, Unregister, Unregister All, Disconnect Endpoint, Delete, Del Pre-Def, Online Parameters, Help

- 7h. Click on Save Setup | Save GK Parameters to save the gatekeeper settings.

8. Create Outbound Phone Book entries to allow the mapping of digit patterns (area codes, 911 calls, etc) to FXO trunks during gatekeeper survivable mode.

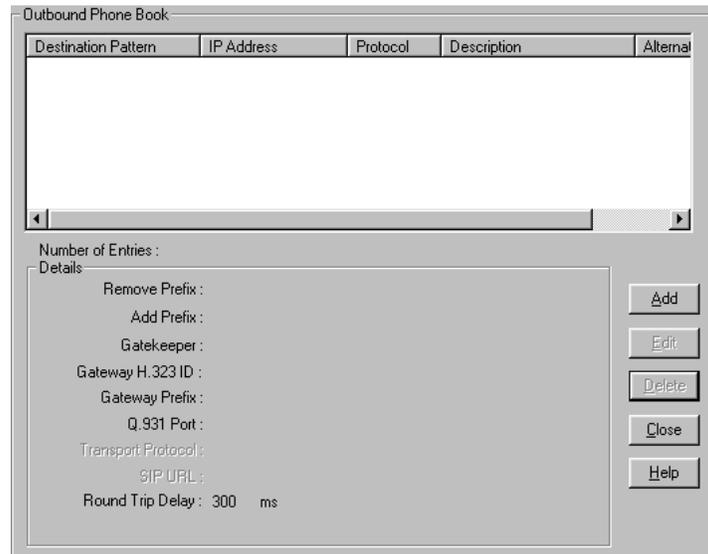
When a survivable call comes in from the IP network to the MultiVOIP gatekeeper, the MultiVOIP will search through the Outbound Phone book entries in an attempt to find an entry whose Destination Pattern matches the first few digits of the destination phone number for the call.

If a match is found, the MultiVOIP will manipulate the digits according to the Remove Prefix and Add Prefix fields of the Outbound Phone book entry. The call with the manipulated digits will be routed to the predefined gatekeeper endpoint whose **Names** entry matches the **H.323 ID** configured in the Outbound Phone Book entry. This is the normal sequence for calls to FXO trunks.

If there is no match in the Outbound Phone book, the MultiVOIP gatekeeper will try to find a matching number in the **Phones** field of one of the MultiVOIP gatekeeper predefined endpoints. This is the normal sequence for calls to FXS ports or IP phones.

Suppose the MultiVOIP resides in the 763 exchange of the Minneapolis area with its FXO trunk connected to PSTN lines. This area has an overlaid area code system where calls to 763, 612, 952, and 651 exchanges are all local calls. In this example, we will add Outbound Phone book entries to allow calls to phone numbers within these exchanges as well as 911 calling for emergencies.

- 8a. Go to the MultiVOIP’s Phone Book | Phone Book Modify | Outbound Phone Book | List Entries screen as seen below.



- 8b. Click the **Add** button to add an Outbound Phone Book entry. The Add/Edit Outbound Phone Book screen will be displayed.

- 8c. Enter the digit pattern 9763 in the **Destination Pattern** field. Enter the digit 9 in the Remove Prefix field. Enter a description in the **Description** field for information purposes only. Enable the **Use Gatekeeper** option and enter **Minneapolis-FXO** in the **Gateway H.323 ID** field. The Gateway H.323 ID is essentially a “trunk group name” that matches the **Minneapolis-FXO** entered in the **Names** field of the Gatekeeper Predefined Endpoint for the FXO trunk. Note, If your analog trunk line requires that all outgoing calls must dial a “9” or some other access code as part of a CO-Centrex service, you would enter this in the **Add Prefix** field. Leave the **Total Digits** and Q.931 Port Number fields set to 0. Leave all other fields blank. Click **OK** to accept the changes.

- 8d. Go to the MultiVOIP's **Phone Book | Phone Book Modify | Outbound Phone Book | List Entries** screen and click the **Add** button to add another Outbound Phone Book entry for the 612 area code. Enter the digit pattern of 9612 in the **Destination Pattern** field. Enter the digit 9 in the Remove Prefix field.

The screenshot shows the 'Add/Edit Outbound Phone Book' dialog box. The 'Phone Number Details' section includes a checkbox for 'Use as default entry' (unchecked), a 'Destination Pattern' field with '9612', a 'Total Digits' field with '0', a 'Remove Prefix' field with '9', and an empty 'Add Prefix' field. Below this is an empty 'IP Address' field and a 'Description' field with 'Local calls, 612 exchange'. The 'Protocol Type' section has radio buttons for 'SIP' (unchecked) and 'H.323' (checked). The 'H.323' section includes a checked 'Use GateKeeper' checkbox, a 'Gateway H.323 ID' field with 'Minneapolis FXD', an empty 'Gateway Prefix' field, and a 'Q.931 Port Number' field with '0'. On the right side, there are buttons for 'OK', 'Cancel', 'Help', and 'Advanced'.

- 8e. Add additional Outbound Phone Book entries for calls to the 651 and 952 exchanges as indicated in the screens below:

The screenshot shows the 'Add/Edit Outbound Phone Book' dialog box. The 'Phone Number Details' section includes a checkbox for 'Use as default entry' (unchecked), a 'Destination Pattern' field with '9651', a 'Total Digits' field with '0', a 'Remove Prefix' field with '9', and an empty 'Add Prefix' field. Below this is an empty 'IP Address' field and a 'Description' field with 'Local calls, 651 exchange'. The 'Protocol Type' section has radio buttons for 'SIP' (unchecked) and 'H.323' (checked). The 'H.323' section includes a checked 'Use GateKeeper' checkbox, a 'Gateway H.323 ID' field with 'Minneapolis FXD', an empty 'Gateway Prefix' field, and a 'Q.931 Port Number' field with '0'. On the right side, there are buttons for 'OK', 'Cancel', 'Help', and 'Advanced'.

Add/Edit Outbound Phone Book

Phone Number Details

Use as default entry

Destination Pattern: 9952

Total Digits: 0

Remove Prefix: 9

Add Prefix:

IP Address:

Description: Local calls, 952 exchange

Protocol Type

SIP H.323

H.323

Use GateKeeper

Gateway H.323 ID: Minneapolis FX0

Gateway Prefig:

Q.931 Port Number: 0

OK

Cancel

Help

Advanced

- 8f. Add an Outbound Phone Book entry to handle the routing of 911 emergency calls to the FXO trunk. Since 911 must be dialed out the FXO trunk, leave the Remove Prefix field blank.

Add/Edit Outbound Phone Book

Phone Number Details

Use as default entry

Destination Pattern: 911

Total Digits: 0

Remove Prefix:

Add Prefix:

IP Address:

Description: 911 emergency calls

Protocol Type

SIP H.323

H.323

Use GateKeeper

Gateway H.323 ID: Minneapolis FX0

Gateway Prefig:

Q.931 Port Number: 0

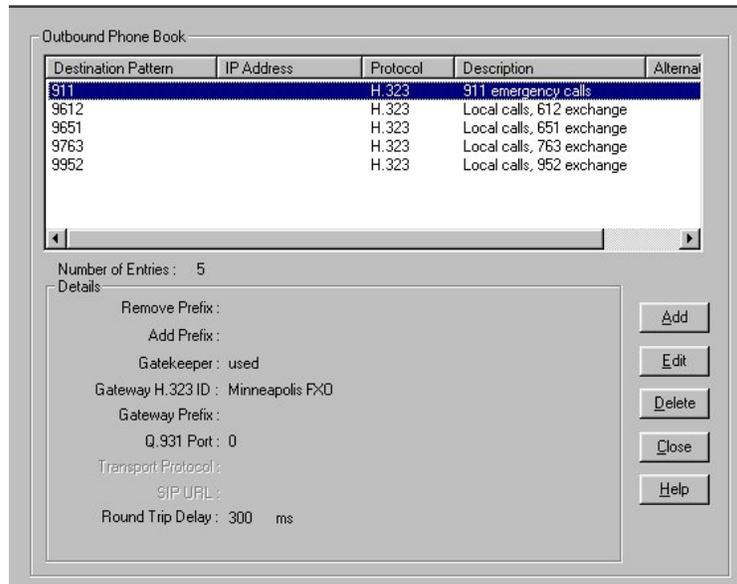
OK

Cancel

Help

Advanced

- 8g. The completed Outbound Phone book now includes the entries added in the earlier steps.



Note: For gatekeeper survivability mode, the Inbound Phone book is not used.

- 8h. Click on **Save Setup | Save and Reboot** to save the Outbound Phone Book entries.
9. Dialing examples for gatekeeper survivability mode.
- 9a. **Gatekeeper Survivability dialing example #1:** A local station to local IP phone call.
- The station at FXS port #1 (6011) goes off hook and receives the survivable dial tone from the MultiVOIP gateway. The client then proceeds to dial 2110.
- The MultiVOIP gatekeeper searches the Outbound Phone book for a match to 2110. There is no match.
- The MultiVOIP gatekeeper searches the **Phones** fields of the Predefined Endpoints and finds a match for 2110.
- The MultiVOIP gatekeeper routes the call to the IP phone registered with the phone number 2110. When IP phone 2110 answers, the call is completed.
- 9b. **Gatekeeper Survivability dialing example #2:** IP phone 2110 dials outward on the local analog trunk to gain access to the local central office.
- IP phone 2110 goes off hook and receives survivable dial tone from the MultiVOIP gatekeeper. The client then proceeds to dial 97637174321.
- The MultiVOIP gatekeeper searches the Outbound Phone book for a match to 97637174321. It finds a match with **Destination Pattern** 9763 and **Gateway H.323 ID = Minneapolis-FXO**. The 9 is removed and the remaining digits 7637174321 remain.
- The MultiVOIP gatekeeper searches the Predefined Endpoints for an entry with Minneapolis-FXO in the **Names** field. It finds a match.
- The call with digits 7637174321 is routed to the FXO predefined endpoint registered with **Name = Minneapolis-FXO**.
- The FXO trunk port activates its tip/ring loop-start interface and the digit string "7637174321" is dialed out towards the network.
- When the called party of 7637174321 answers, the call is completed.

- 9c. **Gatekeeper Survivability dialing example #3:** Incoming Central Office call to station 6011. In this case, the Autocall feature has been enabled in the Voice/Fax screen of the MultiVOIP for the FXO trunk channels. The Autocall Phone Number field has been set to 6011.

A call from the Central Office line rings in to the MultiVOIP's FXO trunk.

The MultiVOIP goes off hook on its FXO tip/ring loop-start interface.

The MultiVOIP automatically place a call to 6011 which has been enabled for autocall in the Voice/Fax screen for the FXO channel.

The MultiVOIP gatekeeper searches the Outbound Phone book for a match to 6011. There is no match.

The MultiVOIP gatekeeper searches the **Phones** fields of the Predefined Endpoints and finds a match for 6011.

The MultiVOIP gatekeeper routes the call to the FXS station port registered with the phone number 6011. When 6011 answers, the call is completed.

Note: If Autocall was not enabled for the FXO port, the incoming caller would have been presented with survivable dial tone from the MultiVOIP gateway. At this point they could manually dial a phone number to complete the call.

- 9d. **Gatekeeper survivability dialing example number 4:** Incoming central office DID call to IP phone 2110. The central office DID line seizes the MultiVOIP DID trunk and dials 2110.

The MultiVOIP gatekeeper searches the Outbound phone book for a match to 2110. There is no match.

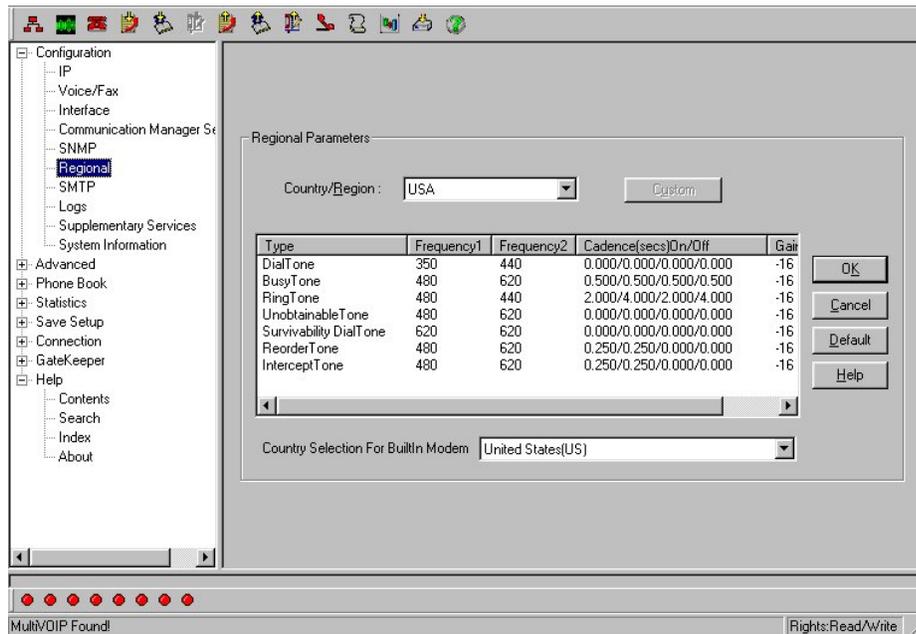
The MultiVOIP gatekeeper searches the Phones fields of the predefined Endpoints and finds a match for 2110.

The MultiVOIP gatekeeper routes the call to the IP phone registered with the phone number 2110. When IP phone 2110 answers, the call is completed.

Regional Parameters screen

The main distinguishing feature of the Regional Parameters screen for Avaya Communication Manager MultiVOIPs is the addition of the survivability dial tone.

1. Set the Survivability Tone in the **Configuration|Regional Parameters** screen.
2. The **Regional Parameters** screen will appear. For the country selected, the standard set of frequency pairs will be listed for dial tone, busy tone, ‘unobtainable’ tone (fast busy or trunk busy), and ring tone.



3. In each field, enter the values that fit your particular system. For most applications, it should be sufficient to select the appropriate country/region and leave the settings at their default values.

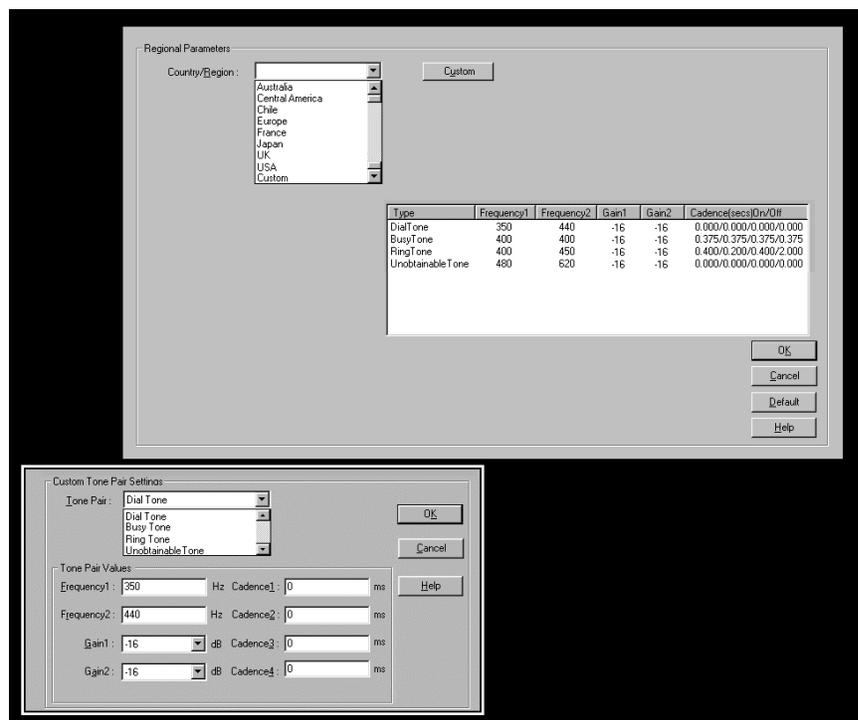
The **Regional Parameters** fields are described in the table below.

“Regional Parameter” Definitions		
Field Name	Values	Description
Country/Region	Australia, Central America, Chile, Europe, France, USA, Japan, UK, Custom Note: “Survivability” tone indicates a special type of call-routing redundancy & applies to Comm Mgr VOIP units only.	Name of a country or region that uses a certain set of tone pairs for dial tone, ring tone, busy tone, and ‘unobtainable’ tone (fast busy tone), survivability tone (tone heard briefly, 2 seconds, after going offhook denoting survivable mode of VOIP unit) and re-order tone (a tone pattern indicating the need for the user to hang up the phone). In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The “Custom” option (button) assures that any tone-pairing scheme worldwide can be accommodated.
Type column	dial tone, ring tone, busy tone, unobtainable tone (fast busy), survivability tone, re-order tone	Type of telephony tone-pair for which frequency, gain, and cadence are being presented.
Frequency 1	freq. in Hertz	Lower frequency of pair.
Frequency 2	freq. in Hertz	Higher frequency of pair.
Gain 1	gain in dB +3dB to -31dB and “mute” setting	Amplification factor of lower frequency of pair. This applies to the dial, ring, busy and ‘unobtainable’ tones that the MultiVOIP outputs as audio to the FXS, FXS, or E&M port. Default: -16dB

"Regional Parameter" Definitions Continued		
Field Name	Values	Description
Gain 2	gain in dB +3dB to -31dB and "mute" setting	Amplification factor of higher frequency of pair. This applies to the dial, ring, busy, and 'unobtainable' (fast busy) tones that the MultiVOIP outputs as audio to the FXS, FXO, or E&M port. Default: -16dB
Cadence (msec) On/Off	n/n/n/n four integer time values in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable (fast busy), dial tone ("0" indicates continuous tone), survivability, and re-order. Default values differ for different countries/regions. Although most cadences have only two parts (an "on" duration and an "off" duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.
Custom (button)	--	Click on the "Custom" button to bring up the Custom Tone Pair Settings screen. (The "Custom" button is active only when "Custom" is selected in the Country/Region field.) This screen allows the user to specify tone pair attributes that are not found in any of the standard national/regional telephony toning schemes.

4. **Set Custom Tones and Cadences** (optional). . The **Regional Parameters** dialog box has a secondary dialog box that allows you to customize DTMF tone pairs to create unique ring-tones, dial-tones, busy-tones or "unobtainable" tones (fast busy signal) or "re-order" tones (telling the user that she must hang up an off-hook phone) or "survivability" tones (an indication of call-routing redundancy in Communication Manager systems only) for your system. Regional Parameters can only be adjusted to change tones played on FXS or FXO ports. Tones for IP phones are fixed and cannot be changed.

This screen allows the user to specify tone-pair attributes that are not found in any of the standard national/regional telephony toning schemes. To access this customization feature, click on the **Custom** button on the **Regional Parameters** screen. (The "Custom" button is active only when "Custom" is selected in the **Country/Region** field.)



The **Custom Tone-Pair Settings** fields are described in the table below.

Custom Tone-Pair Settings Definitions		
Field Name	Values	Description
Tone Pair	dial tone, busy tone, ring tone, 'unobtainable' tone, survivability tone, re-order tone	Identifies the type of telephony signaling tone for which frequencies are being specified.
TONE PAIR VALUES		About Defaults: US telephony values are used as defaults on this screen. However, since this dialog box is provided to allow custom tone-pair settings, default values are essentially irrelevant.
Frequency 1	frequency in Hertz	Frequency of lower tone of pair. This outbound tone pair enters the MultiVOIP at the input port.
Frequency 2	frequency in Hertz	Frequency of higher tone of pair. This outbound tone pair enters the MultiVOIP at the input port.
Gain 1	gain in dB +3dB to -31dB and "mute" setting	Amplification factor of lower frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the input port. Default = -16dB
Gain 2	gain in dB +3dB to -31dB and "mute" setting	Amplification factor of higher frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the input port. Default = -16dB
Cadence 1	integer time value in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, dial tone ("0" indicates continuous tone) survivability and re-order. Cadence 1 is duration of first period of tone being "on" in the cadence of the telephony signal (which could be ring-tone, busy-tone, unobtainable-tone, or dial-tone).
Cadence 2	duration in milliseconds	Cadence 2 is duration of first "off" period in signaling cadence.
Cadence 3	duration in milliseconds	Cadence 3 is duration of second "on" period in signaling cadence.
Cadence 4	duration in milliseconds	Cadence 4 is duration of second "off" period in the signaling cadence, after which the 4-part cadence pattern of the telephony signal repeats.

Chapter 3 – Related MultiVOIP Setup Parameters

For your convenience, the Communication Manager Settings, MultiVOIP Voice/Fax, Interface, and Gatekeeper parameters are detailed in this section.

Communication Manager Settings

In the Communication Manager Settings screen, if the “Register with Communication Manager Gatekeeper option is selected, then many of the parameters must assume default values and those will be grayed on the screen.

Communication Manager Parameters

Select Channel: Channel 5

Interface Type is FXO

Q.931 Parameters

Use Fast Start

Call Signaling Port: 1725

Delay before Dial: 400 ms

Register with Communication Manager

Gatekeeper RAS Parameters

Gateway Name: MultiVoIP

Gateway Prefix:

Gateway H323 ID:

Phone Number / Extension: 1725

Security Code: XXXX

	GateKeeper IP Address	RAS Port	GateKeeper Name
Primary	192 . 168 . 25 . 30	1719	
Alternate GK 1	0 . 0 . 0 . 0	1719	
Alternate GK 2	0 . 0 . 0 . 0	1719	

RAS TimeToLive: 60 secs

Enable Survivability Mode

Switching Time Interval: 60 secs

H323 Version 4 Options

Q.931 Multiplexing [Mux] H.245 Tunneling [Tun]

Parallel H.245 [P8+Tun] Annex-E [AE]

The **Communication Manager Parameter** fields are described in the table below.

Communication Manager Parameter Definitions		
Field Name	Values	Description
Select Channel	Channel <i>n</i>	Denotes the VOIP channel to which settings in the current screen apply.
“Interface Type is ...”	FXS Loop Start; FXO; DID	Indicates the telephony interface used on the selected VOIP channel.

Communication Manager Parameter Definitions Continued		
Field Name	Values	Description
Q.931 Parameters		
Use Fast Start	Y/N	Enables the H.323 Fast Start procedure. If “ Register with Communication Manager Gatekeeper ” is enabled, this field defaults to “N” and cannot be changed.
Call Signaling Port	port number	This value must be unique for each of the MultiVOIP’s FXS ports. This value must be the same for all of the MultiVOIP unit’s FXO/DID ports that are managed by CM.
Delay Before Dial	in milliseconds	Applicable to FXO ports only. It indicates the amount of delay that will be introduced before a dialed sequence is sent.
GateKeeper RAS Parameters		
Gatekeeper Name	<i>alpha-numeric string</i>	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register.
Gateway Prefix		If “ Register with Communication Manager Gatekeeper ” is enabled, this field is not used. This number becomes registered with the GateKeeper. Call requests sent to the gatekeeper and preceded by this prefix will be routed to the VOIP gateway.
Gateway H.323 ID		The H.323 ID is used to register FXO trunks with the MultiVOIP GateKeeper.
Phone Number/ Extension	numeric	This phone number (PBX extension) must match the extension configured on the Communication Manager “station” form (dialog box). It is only relevant for the MultiVOIP’s FXS channels.
Security Code	alpha-numeric	This field value must match the security code configured on the Communication Manager “station” form (dialog box). It is only relevant for the MultiVOIP’s FXS channels.
Buttons		
Default		Applies default values to all fields in the Communication Manager Parameters screen.
Unregister		Allows you to unregister the registered station or trunk with respect to the Communication Manager Gatekeeper. This command has an immediate effect. This is a very useful diagnostic command.
Register		Allows you to register the unregistered station or trunk with respect to the Communication Manager Gatekeeper. This command has an immediate effect. This is a very useful diagnostic command.
Columns		
GateKeeper IP Address		The IP address of the Communication Manager Gatekeeper.
RAS Port		Typically this is 1719.
GateKeeper Name		This field must be left blank
Rows		
Primary [gatekeeper]		One PBX in the system must serve as the primary gatekeeper.
Alternate GK 1		It is optional for the system to have a PBX that serves as a first alternate gatekeeper.
Alternate GK 2		It is optional for the system to have a PBX that servers as a second alternate gatekeeper.

Communication Manager Parameter Definitions (cont'd)		
Field Name	Values	Description
RAS Time to Live	Numeric (in seconds)	The MultiVOIP sends this value to CM in the Registration Request (RRQ). However, the MultiVOIP will send "Keepalive" RRQs based on the TTL received from CM during Registration Confirm (RCF)
Enable Survivability Mode	Y/N	Enable/disable MultiVOIP's Gateway Survivable mode wherein MultiVOIP manages its own IP traffic using its inbound and outbound phonebooks in the event of failure of the primary, alternate #1 and #2 gatekeepers (if alternates are present).
Switching Time Interval	numeric (in seconds)	When MultiVOIP is in "gateway" survivable mode meaning the Communication Manager gatekeeper to which it is registered failed (along with any alternate gatekeepers providing redundancy), the MultiVOIP manages IP phone traffic with its own phonebooks. It will seek to return to being controlled by the Communication Manager gatekeeper. The Switching Time Interval determines how often it will seek to restore its registration with the primary Communication Manager gatekeeper.
H.323 Version 4 Parameters		
Q.931 Multiplexing (Mux)	Y/N	Signaling for multiple phone calls can be carried on a single port rather than opening a separate signaling port for each call. This conserves bandwidth resources. If " Register with Communication Manager Gatekeeper " is enabled, this field defaults to "Y" for FXO/DID ports and to "N" for FXS ports and it cannot be changed.
H.245 Tunneling (Tun)	Y/N	H.245 messages are encapsulated within the Q.931 call-signaling channel. Among other things, the H.245 messages let the two endpoints tell each other what their technical capabilities are and determine who, during the call, will be the client and who the server. Tunneling is the process of transmitting these H.245 messages through the Q.931 channel. The same TCP/IP socket (or logical port) already being used for the Call Signaling Channel is then also used by the H.245 Control Channel. This encapsulation reduces the number of logical ports (sockets) needed and reduces call setup time. If " Register with Communication Manager Gatekeeper " is enabled, this field defaults to "Y" for FXS and FXO ports and cannot be changed.
H.323 Version 4 Parameters		
Parallel H.245 (FS + Tun)	Y/N	FS (Fast Start or Fast Connect) is a Q.931 feature of H.323v2 to hasten call setup as well as 'pre-opening' the media channel before the CONNECT message is sent. This pre-opening is a requirement for certain billing activities. Under Parallel H.245 FS + Tun, this Fast Connect feature can operate simultaneously with H.245 Tunneling (see description above). If " Register with Communication Manager Gatekeeper " is enabled, this field defaults to "N" for FXS and FXO ports and cannot be changed.
Annex -E (AE)	Y/N	Multiplexed UDP call signaling transport. Annex E is helpful for high-volume VOIP system endpoints. Gateways with lesser volume can afford to use TCP to establish calls. However, for larger volume endpoints, the call setup times and system resource usage under TCP can become problematic. Annex E allows endpoints to perform call-signaling functions under the UDP protocol, which involves substantially streamlined overhead. (This feature should not be used on the public Internet because of potential problems with security and bandwidth usage.) If " Register with Communication Manager Gatekeeper " is enabled, this field defaults to "N" for FXS and FXO ports and cannot be changed.

Voice/Fax Parameters

In each field, enter the values that fit your particular network.

Voice/Fax Parameters

Select Channel Channel 1

Voice Gain

Input 0 dB Output 0 dB

Dtmf Gain

High -4 dB Low -7 dB

Duration 100 ms

DTMF : Out Of Band - Fixed Duration

Fax

Fax Enable

Max Baud Rate 14400

Fax Volume -9.5 dB

Jitter Value 400 ms

Mode FRF 11

OK

Cancel

Copy Channel

Default

Help

Coder

Manual Automatic

Selected Coder G.723.1 @ 6.3 kbps

Max bandwidth 10 kbps

Advanced Features

Silence Compression

Echo Cancellation

Forward Error Correction

Auto Call / OffHook Alert

Auto Call /OffHook Alert Auto Call Generate Local Dial Tone

OffHook Alert Timer secs

Phone Number

Dynamic Jitter Buffer

Minimum Jitter Value 60 ms

Maximum Jitter Value 300 ms

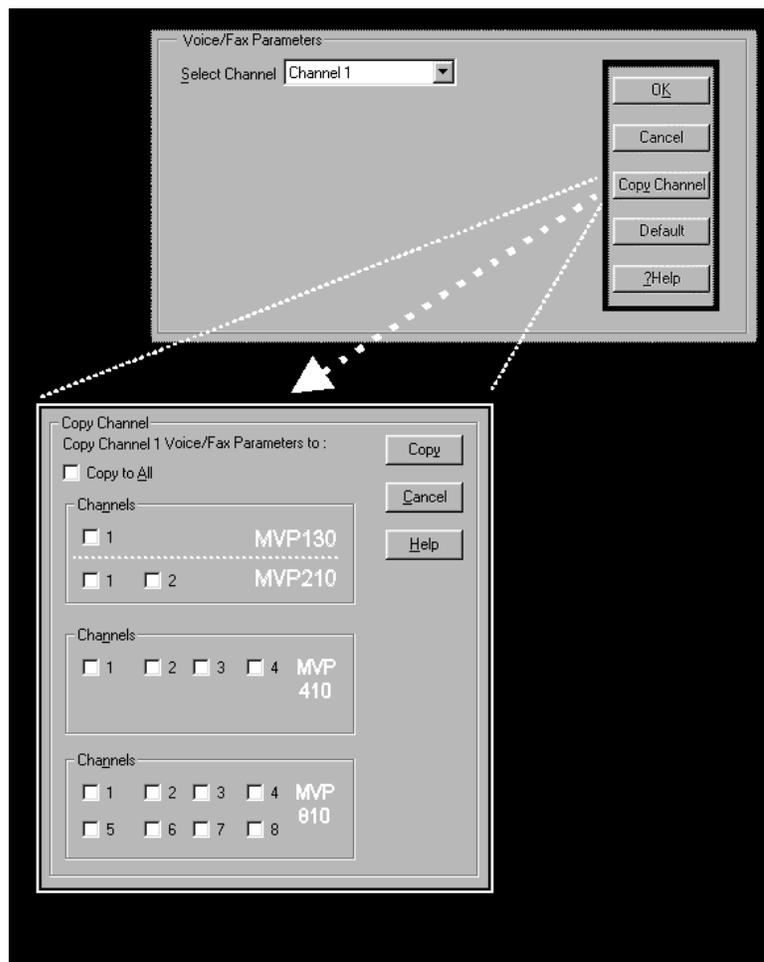
Optimization Factor 7

Automatic Disconnection

Jitter Value 350 ms Consecutive Packets Lost 30

Call Duration 180 secs Network Disconnection 300 secs

Note that Voice/FAX parameters are applied on a channel-by-channel basis. However, once you have established a set of Voice/FAX parameters for a particular channel, you can apply this entire set of Voice/FAX parameters to another channel by using the **Copy Channel** button and its dialog box. To copy a set of Voice/FAX parameters to all channels, select “Copy to All” and click **Copy**.



The **Voice/FAX Parameters** fields are described in the tables below.

Voice/Fax Parameter Definitions		
Field Name	Values	Description
Default	--	When this button is clicked, all Voice/FAX parameters are set to their default values.
Select Channel	1-2 (210) 1-4 (410) 1-8 (810)	Channel to be configured is selected here.
Copy Channel	--	Copies the Voice/FAX attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.
Voice Gain	--	Signal amplification (or attenuation) in dB.
Input Gain	+31dB to -31dB	Modifies audio level entering voice channel before it is sent over the network to the remote VOIP. The default & recommended value is 0 dB .
Output Gain	+31dB to -31dB	Modifies audio level being output to the device attached to the voice channel. The default and recommended value is 0 dB .

Voice/Fax Parameter Definitions Continued		
Field Name	Values	Description
DTMF Parameters		
DTMF Gain	--	The DTMF Gain (Dual Tone Multi-Frequency) controls the volume level of the DTMF tones sent out for Touch-Tone dialing.
DTMF Gain, High Tones	+3dB to -31dB & "mute"	Default value: -4 dB . Not to be changed except under supervision of MultiTech's Technical Support.
DTMF Gain, Low Tones	+3dB to -31dB & "mute"	Default value: -7 dB . Not to be changed except under supervision of MultiTech's Technical Support.
Duration (DTMF)	60 – 3000 ms	When DTMF: Out of Band is selected, this setting determines how long each DTMF digit 'sounds' or is held. Default = 100 ms.
DTMF In/Out of Band	Out of Band, or Inband	When DTMF Out of Band is selected, the MultiVOIP detects DTMF tones at its input and regenerates them at its output. When DTMF Inband is selected, the DTMF digits are passed through the MultiVOIP unit as they are received.
FAX Parameters		
Fax Enable	Y/N	Enables or disables fax capability for a particular channel.
Max Baud Rate (Fax)	2400, 4800, 7200, 9600, 12000, 14400 bps	Set to match baud rate of fax machine connected to channel (see Fax machine's user manual). For CM managed fax calls it is recommended to use 9600. Default = 14400 bps.
Fax Volume (Default = -9.5 dB)	-18.5 dB to -3.5 dB	Controls output level of fax tones. To be changed only under the direction of Multi-Tech's Technical Support.
Jitter Value (Fax)	Default = 400 ms	Defines the inter-arrival packet deviation (in milliseconds) for the fax transmission. A higher value will increase the delay, allowing a higher percentage of packets to be reassembled. A lower value will decrease the delay allowing fewer packets to be reassembled.
Mode (Fax)	FRF 11; T.38	FRF11 is frame-relay FAX standard using these coders: G.711, G.728, G.729, G.723.1. T.38 is an ITU-T standard for storing and forwarding FAXes via email using X.25 packets. It uses T.30 fax standards and includes special provisions to preclude FAX timeouts during IP transmissions. Must use T.38 when faxing to/from CM. Note: When the MultiVOIP stops receiving T.38 packets, it will wait ten seconds before switching to voice mode.
Coder Parameters		
Coder	Manual or Auto-matic	Determines whether selection of coder is manual or automatic. Must be set to manual for CM managed calls. When Automatic is selected, the local and remote voice channels will negotiate the voice coder to be used by selecting the highest bandwidth coder supported by both sides without exceeding the Max Bandwidth setting. G.723, G.729, or G.711 are negotiated.

Voice/Fax Parameter Definitions Continued		
Field Name	Values	Description
Coder Parameters (cont'd)		
Selected Coder	G.711 a/u law 64 kbps; G.726, @ 16/24/32/40 kbps; G.727, @ nine bps rates; G.723.1 @ 5.3 kbps, 6.3 kbps; G.729, 8kbps; Net Coder @ 6.4, 7.2, 8, 8.8, 9.6 kbps G.711/G.729 G.729/G.711	Select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling must have the same voice coder selected. Default = G.723.1 @ 6.3 kbps, as required for H.323. Here 64K of digital voice are compressed to 6.3K, allowing several simultaneous conversations over the same bandwidth that would otherwise carry only one. To make selections from the Selected Coder drop-down list, the Manual option must be enabled.
Max bandwidth (coder)	11 – 128 kbps	This drop-down list enables you to select the maximum bandwidth allowed for this channel. The Max Bandwidth drop-down list is enabled only if the Coder is set to Automatic. If coder is to be selected automatically (“Auto” setting), then enter a value for maximum bandwidth.
Advanced Features		
Silence Compression	Y/N	Determines whether silence compression is enabled (checked) for this voice channel. With Silence Compression enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. Default = off.
Echo Cancellation	Y/N	Determines whether echo cancellation is enabled (checked) for this voice channel. Echo Cancellation removes echo and improves sound quality. Default = on.
Forward Error Correction	Y/N	Determines whether forward error correction is enabled (checked) for this voice channel. Forward Error Correction enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off

Voice/Fax Parameter Definitions Continued		
Field Name	Values	Description
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall, Offhook Alert	<p>The AutoCall option enables the local MultiVOIP to place a call without the user having to dial a number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote endpoint identified in the Phone Number box of this option.</p> <p>If the “Pass Through Enable” field is checked in the Interface Parameters screen, AutoCall must be used.</p> <p>The Offhook Alert option applies only to FXS channels.</p> <p>The Offhook Alert option works like this: if a phone goes offhook and no number is dialed within a specific period of time (as set in the Offhook Alert Timer field), then that phone will automatically dial the Alert phone number for the voip channel. (The Alert phone number must be set in the Voice/Fax Parameters Phone Number field). One use of this feature would be for emergency use where a user goes off hook but does not dial, possibly indicating a crisis situation. The Offhook Alert feature uses the Intercept Tone, as listed in the Regional Parameters screen. This tone will be output on the phone that was taken off hook but that did not dial. The other end of the connection will hear audio from the “crisis” end as it would during a normal phone call.</p> <p>Both functions apply on a channel-by-channel basis. It would not be appropriate for either of these functions to be applied to a channel that serves in a pool of available channels for general phone traffic.</p>
Generate Local Dial Tone	Y/N	<i>Used for AutoCall only.</i> If selected, dial tone will be generated locally while the call is being established between gateways. The capability to generate dial tone locally would be particularly useful when there is a lengthy network delay.
Offhook Alert Timer	0 – 3000 seconds	The length of time that must elapse before the offhook alert is triggered and a call is automatically made to the phone number listed in the Phone Number field.
Phone Number	--	Phone number used for Auto Call function or Offhook Alert Timer function.
Dynamic Jitter		
Dynamic Jitter Buffer		Dynamic Jitter defines a minimum and a maximum jitter value for voice communications. When receiving voice packets from a remote MultiVOIP, varying delays between packets may occur due to network traffic problems. This is called Jitter. To compensate, the MultiVOIP uses a Dynamic Jitter Buffer. The Jitter Buffer enables the MultiVOIP to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An Optimization Factor adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MultiVOIP gateways.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
Dynamic Jitter (cont'd)		
Minimum Jitter Value	60 to 400 ms	The minimum dynamic jitter buffer of 60 milliseconds is the minimum delay that would be acceptable over a low jitter network. Default = 150 msec
Maximum Jitter Value	60 to 400 ms	The maximum dynamic jitter buffer of 400 milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 msec
Optimization Factor	0 to 12	The Optimization Factor determines how quickly the length of the Dynamic Jitter Buffer is changed based on actual jitter encountered on the network. Selecting the minimum value of 0 means low voice delay is desired, but increases the possibility of jitter induced voice quality problems. Selecting the maximum value of 12 means highest voice quality under jitter conditions is desired at the cost of increased voice delay. Default = 7.
Auto Disconnect		
Automatic Disconnection	--	The Automatic Disconnection group provides four options which can be used singly or in any combination.
Jitter Value	1-65535 milli-seconds	The Jitter Value defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. The default is 300 milliseconds. A higher value means voice transmission will be more accepting of jitter. A lower value is less tolerant of jitter. Inactive by default. When active, default = 300 ms. However, value must equal or exceed Dynamic Minimum Jitter Value.
Call Duration	1-65535 seconds	Call Duration defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. Inactive by default. When active, default = 180 sec. This may be too short for most configurations, requiring upward adjustment.
Consecutive Packets Lost	1-65535	Consecutive Packets Lost defines the number of consecutive packets that are lost after which the call is automatically disconnected. Inactive by default. When active, default = 30
Network Disconnection	1 to 65535 seconds; Default = 30 sec.	Specifies how long to wait before disconnecting the call when IP network connectivity with the remote site has been lost.

Interface Parameters

In each field, enter the values that fit your particular network.

Interface Parameters

Select Channel: Channel 1

Interface

Interface Type: FXD
 FXS (Loop Start)
 E & M
 DID-DPO

Dialing Options

Regeneration:
 Pulse
 DTMF

Inter Digit Timer: 2 secs
 Message Waiting Indication: None
 Inter Digit Regeneration Timer: 100 ms

E&M Options

Signal:
 Dial Tone
 Wink

Wink Timer: 250 ms
 Type: TYPE I
 TYPE II
 TYPE III
 TYPE IV
 TYPE V
 Mode:
 2Wire
 4Wire

FXD Disconnect On

Current Loss
 Tone Detection
 Silence Detection: None
 Disconnect Tone Sequence: '0' + None
 Silence Timer: 15 secs
 Current Loss Detect Timer: 500 ms

Ring Count

FXS: 8
 FXD: 2

FXS Options

Current Loss

Disconnect On Call Progress Tone

Enable

Pass Through

Enable

DID Options

Start Modes:
 Wink Start
 Immediate Start
 Delay Dial
 Wink Timer: 200

Caller ID

Caller ID Type: BellCore
 Caller ID

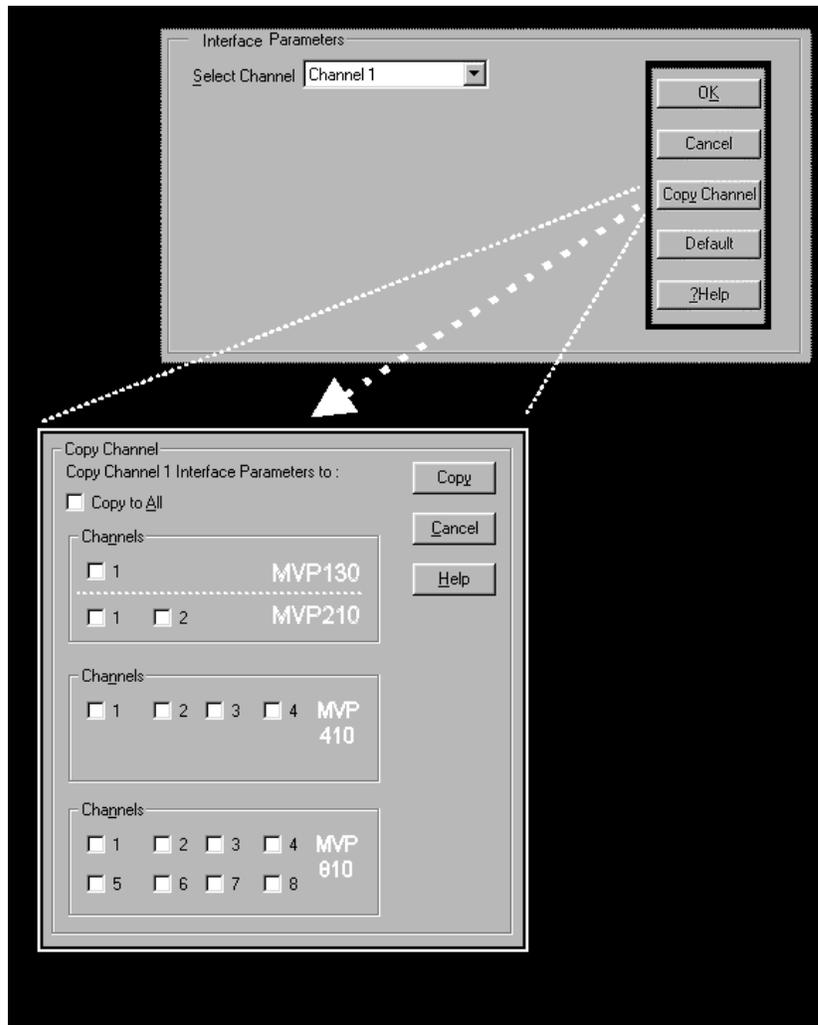
Flash Hook Options

Generation: 600 ms
 Detection Range:
 Min: 500 ms
 Max: 1000 ms

OK
 Cancel
 Copy Channel
 Default
 Help

The kinds of parameters for which values must be chosen depend on the type of telephony supervisory signaling or interface used (FXO, E&M, etc.). We present here the various parameters grouped and organized by interface type.

Note that Interface parameters are applied on a channel-by-channel basis. However, once you have established a set of Interface parameters for a particular channel, you can apply this entire set of Voice/FAX parameters to another channel by using the **Copy Channel** button and its dialog box. To copy a set of Interface parameters to all channels, select “Copy to All” and click **Copy**.



FXS Loop Start Parameters. The parameters applicable to FXS Loop Start are shown in the figure below and described in the table that follows.

The screenshot displays several configuration panels for the FXS Loop Start interface:

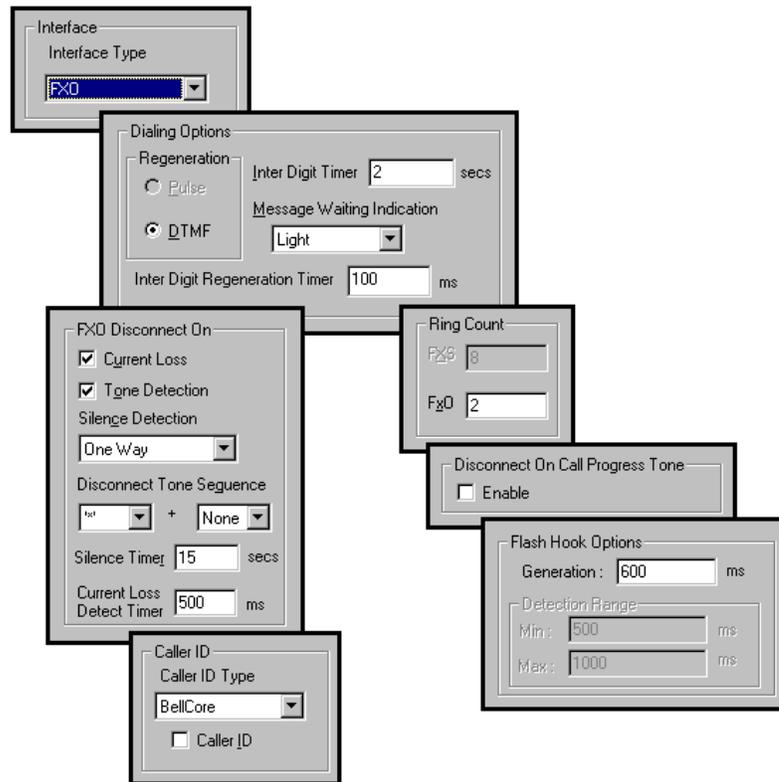
- Interface:** Interface Type is set to **FXS (Loop Start)**.
- Dialing Options:**
 - Inter Digit Timer: **2** secs
 - Message Waiting Indication: **Light**
 - Inter Digit Regeneration Timer: **100** ms
- Pass Through:** Enable
- Ring Count:** FXS is set to **8**
- Caller ID:** Caller ID Type is **BellCore**, and Caller ID is unchecked.
- FXS Options:** Current Loss
- Flash Hook Options:**
 - Generation: **500** ms
 - Detection Range:
 - Min: **500** ms
 - Max: **1000** ms

FXS Loop Start Interface: Parameter Definitions		
Field Name	Values	Description
FXS (Loop Start)	Y/N	Enables FXS Loop Start interface type.
Dialing Options fields		
Inter Digit Timer	1 - 10 seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will place the call. Default = 2.
Message Waiting Indication	--	If "Register with Communication is enabled", this field enables message waiting indication on FXS channels. Choices are "Light" or "Stutter Dial Tone".
Inter Digit Regeneration Time	in milliseconds	The length of time between the outputting of DTMF digits. Default = 100 ms.
Ring Count, FXS	1-99	Maximum number of rings that the MultiVOIP will issue before giving up the attempted call.
FXS Options, Current Loss	Y/N	When enabled, the MultiVOIP will interrupt loop current in the FXS circuit to initiate a disconnection. This tells the device connected to the FXS port to hang up. The Multi-VOIP cannot drop the call; the FXS device must go on hook.

FXS Loop Start Interface: Parameter Definitions (cont'd)		
Field Name	Values	Description
Flash Hook Options fields		
Generation	--	<i>not applicable to FXS interface</i>
Detection Range	<i>for Min. and Max.,</i> 50 - 1500 milliseconds	For a received flash hook to be regarded as such by the MultiVOIP, its duration must fall between the minimum and maximum values given here.
Pass Through Enable	Y/N	Not used if "Register with Communication Manager" is enabled
Caller ID fields		
Caller ID Type	Bellcore	The MultiVOIP currently supports only one implementation of Caller ID. That implementation is Bellcore type 1 with Caller ID placed between the first and second rings of the call.
Caller ID enable	Y/N	Caller ID information is a description of the remote calling party received by the called party. The description has three parts: name of caller, phone number of caller, and time of call. The 'time-of-call' portion is always generated by the receiving MultiVOIP unit (on FXS channel) based on its date and time setup.
Caller ID enable (cont'd)	Y/N	The forms of the 'Caller Name' and 'Caller Phone Number' differ depending on the IP transmission protocol used (H.323, SIP, or SPP) and upon entries in the phonebook screens of the remote (CID generating) voip unit. The CID Name and Number appearing on the phone at the terminating FXS end will come either from a central office switch (showing a PSTN phone number), or the phonebook of the remote (CID sending) voip unit.

The Caller ID feature has dependencies on both the telco central office and the MultiVOIP phone book. See discussion after the FXO Parameters section below.

FXO Parameters. The parameters applicable to the FXO telephony interface type are shown in the figure below and described in the table that follows.



FXO Interface: Parameter Definitions		
Field Name	Values	Description
Interface, FXO	Y/N	Enables FXO functionality
Dialing Options		
Regeneration	Pulse, DTMF	Determines whether digits generated and sent out will be pulse tones or DTMF.
Message Waiting Indication	--	Not applicable to FXO interface.
Inter Digit Regeneration Time	50 to 20,000 milliseconds	The length of time between the outputting of DTMF digits. Default = 100 ms.
FXO Disconnect On		There are three possible criteria for disconnection under FXO: current loss, tone detection, and silence detection. Disconnection can be triggered by more than one of the three criteria.
Current Loss	Y/N	Disconnection to be triggered by loss of current. That is, when Current Loss is enabled ("Y"), the MultiVOIP will hang up the call when it detects a loss of current initiated by the attached device.
Current Loss Detect Timer	integer values (in milliseconds)	The minimum time required for detecting the current loss signal on the FXO interface. In other words, this is the minimum length of time the current must be absent to validate 'current loss' as a disconnection criterion. Default = 500 ms.
Tone Detection	Y/N	Disconnection to be triggered by a tone sequence.

FXO Interface: Parameter Definitions (cont'd)																																
Field Name	Values	Description																														
FXO Disconnect On (cont'd)																																
Disconnect Tone Sequence	1 st tone pair + 2 nd tone pair	These are DTMF tone pairs. Values for first tone pair are: *, #, 0, 1-9, and A-D. Values for second tone pair are: none, 0, 1-9, A-D, *, and #. The tone pairs 1-9, 0, *, and # are the standard DTMF pairs found on phone sets. The tone pairs A-D are "extended DTMF" tones, which are used for various PBX functions.																														
	<table border="1"> <thead> <tr> <th colspan="4">DTMF Tone Pairs</th> <th>Low Tones</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>2</td> <td>3</td> <td>A</td> <td>697Hz</td> </tr> <tr> <td>4</td> <td>5</td> <td>6</td> <td>B</td> <td>770Hz</td> </tr> <tr> <td>7</td> <td>8</td> <td>9</td> <td>C</td> <td>852Hz</td> </tr> <tr> <td>*</td> <td>0</td> <td>#</td> <td>D</td> <td>941Hz</td> </tr> <tr> <td>High Tones</td> <td>1209Hz</td> <td>1336Hz</td> <td>1447Hz</td> <td>1633Hz</td> </tr> </tbody> </table>		DTMF Tone Pairs				Low Tones	1	2	3	A	697Hz	4	5	6	B	770Hz	7	8	9	C	852Hz	*	0	#	D	941Hz	High Tones	1209Hz	1336Hz	1447Hz	1633Hz
DTMF Tone Pairs				Low Tones																												
1	2	3	A	697Hz																												
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7	8	9	C	852Hz																												
*	0	#	D	941Hz																												
High Tones	1209Hz	1336Hz	1447Hz	1633Hz																												
Silence Detection	One-Way or Two-Way	Disconnection to be triggered by silence in one direction only or in both directions simultaneously.																														
Silence Timer in seconds	integer value	Duration of silence required to trigger disconnection.																														
Disconnect on Call Progress Tone	Y/N	Allows call on FXO port to be disconnected when a PBX issues a call-progress tone denoting that the phone station on the PBX that has been involved in the call has been hung up.																														
Ring Count, FXO	1-99	Number of rings required before the MultiVOIP answers the incoming call.																														
Flash Hook Options fields																																
Generation	50 - 1500 milliseconds	Length of flash hook that will be generated and sent out when the remote end initiates a flash hook and it is regenerated locally. Default = 600 ms.																														
Detection Range	--	Not applicable to FXO.																														
Caller ID fields																																
Caller ID Type	Bellcore	The MultiVOIP currently supports only one implementation of Caller ID. That implementation is Bellcore type 1 with caller ID placed between the first and second rings of the call.																														
Caller ID enable	Y/N	Caller ID information is a description of the remote calling party received by the called party. The description has three parts: name of caller, phone number of caller, and time of call. The 'time-of-call' portion is always generated by the receiving MultiVOIP unit (on FXS channel) based on its date and time setup. The forms of the 'Caller Name' and 'Caller Phone Number' differ depending on the IP transmission protocol used (H.323, SIP, or SPP) and upon entries in the phonebook screens of the remote (CID generating) voip unit. The CID Name and Number appearing on the phone at the terminating FXS end will come either from a central office switch (showing a PSTN phone number), or the phonebook of the remote (CID sending) voip unit.																														

The Caller ID feature has dependencies on both the telco central office and the MultiVOIP phone book. See discussion below.

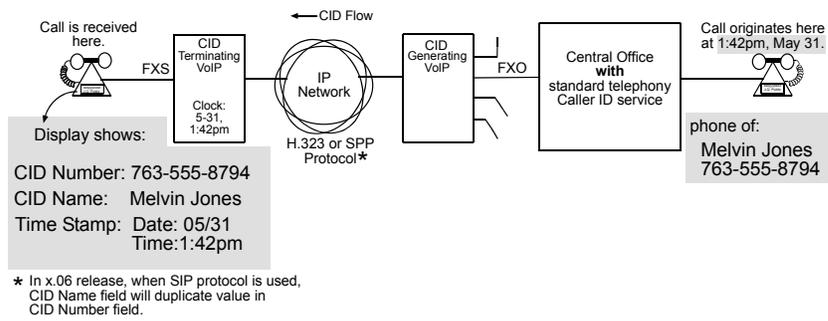


Figure 1: Voip Caller ID Case #1 – Call, through telco central office *with* standard CID, enters voip system

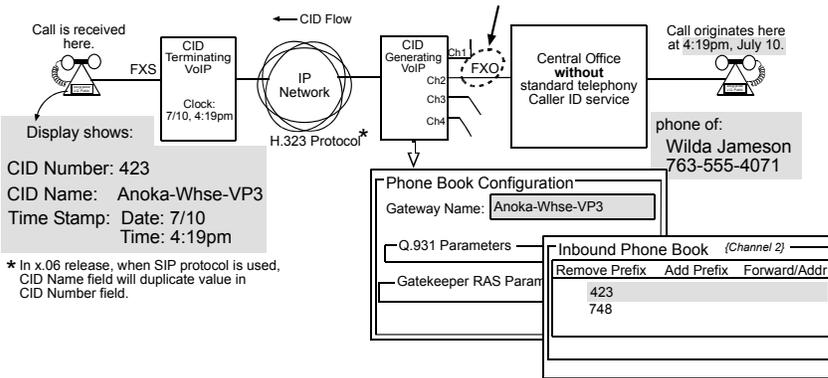


Figure 2: Voip Caller ID Case #2 – Call, through telco central office *without* standard CID, enters H.323 voip system

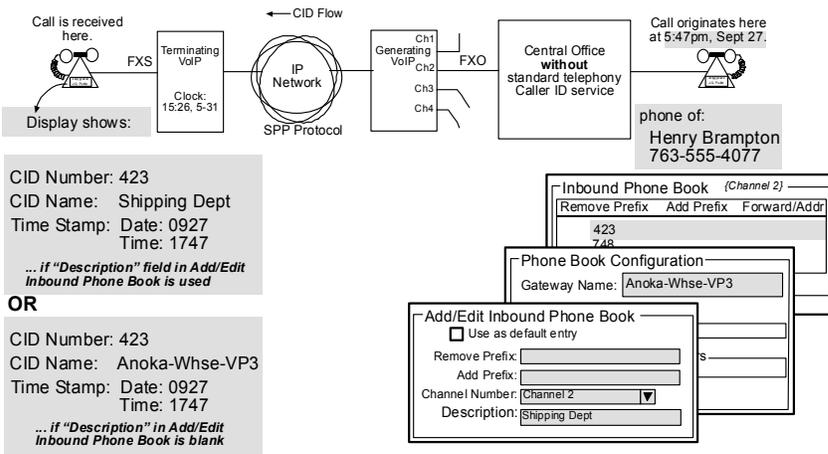


Figure 3: Voip Caller ID Case #3 – Call, through telco central office *without* standard CID, enters SPP voip system

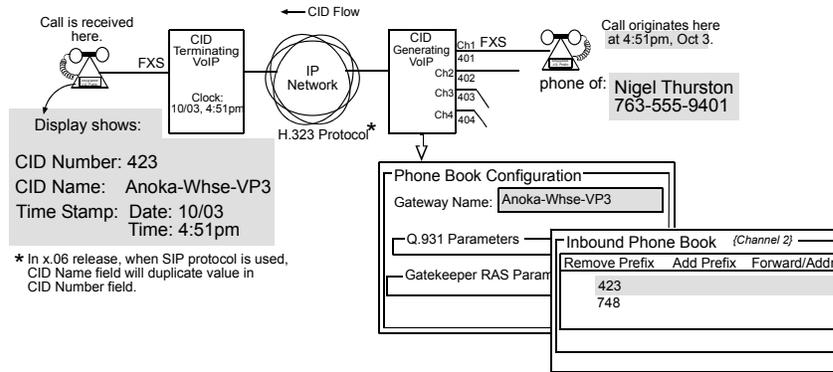


Figure 4: Voip Caller ID Case #4 – Remote FXS call on H.323 voip system

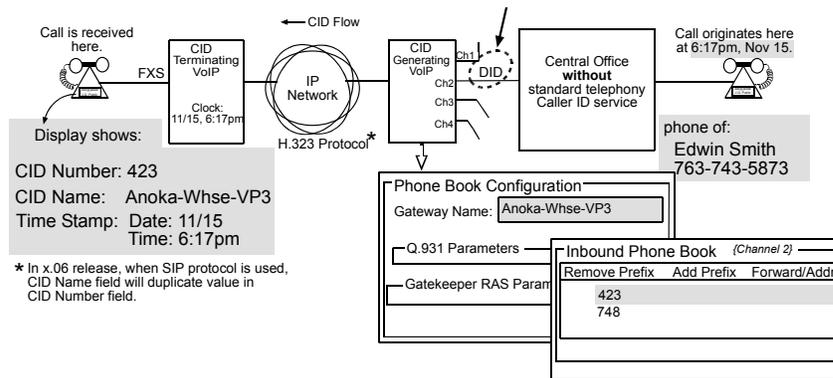
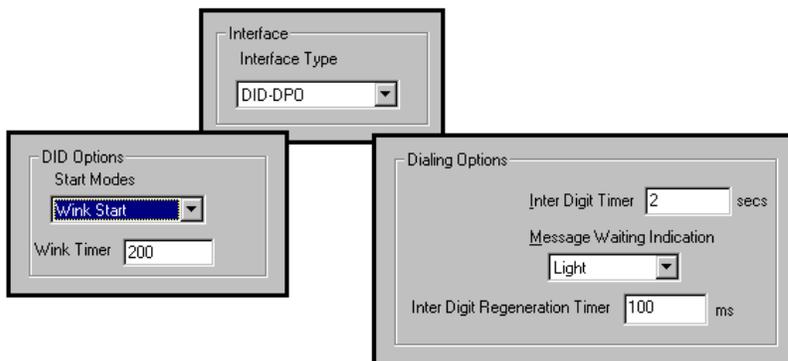


Figure 5: Voip Caller ID Case #5 – Call through telco central office without standard CID enters DID channel in H.323 voip system

E&M Parameters. E&M is not supported if “Register with Communication Manager” is enabled.

DID Parameters. The parameters applicable to the Direct Inward Dial (DID) telephony interface type are shown in the figure below and described in the table that follows. The DID interface allows one phone line to direct incoming calls to any one of several extensions without a switchboard operator. Of course, one DID line can handle only one call at a time. The parameters described here pertain to the customer-premises side of the DID connection (DID-DPO, dial-pulse originating); the network side of the DID connection (DID-DPT, dial-pulse terminating) is not supported.



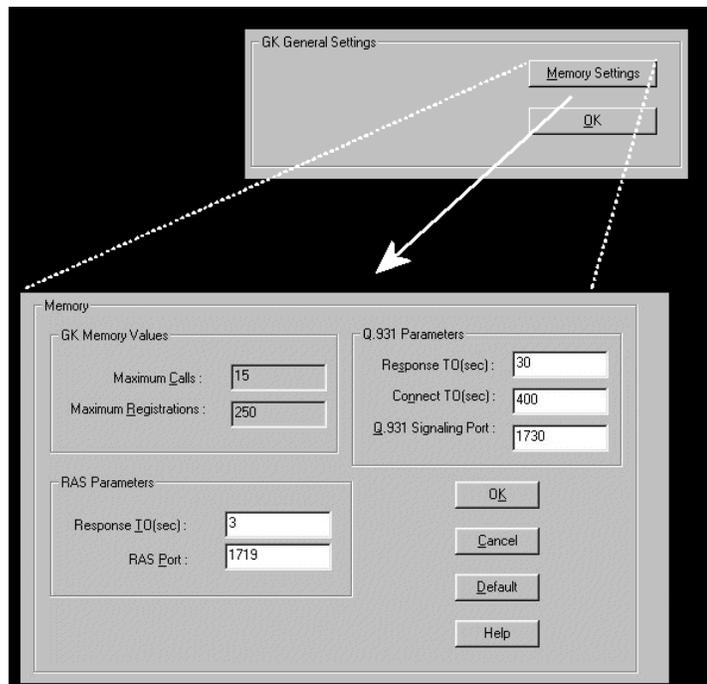
DID Interface Parameter Definitions		
Field Name	Values	Description
Interface	DID-DPO	Enables the customer-premises side of DID functionality
DID Options		MultiVOIP's use of DID applies only for incoming DID calls. The Start Mode used by the MultiVOIP must match that used by the originating telephony equipment, else DID calls cannot be completed.
Start Modes	Immediate Start, Wink Start, Delay Dial	For Immediate Start , the voip detects the off-hook condition initiated by the telco central-office call and becomes ready to receive dial digits immediately. For Wink Start , the voip detects the off-hook condition. Then the voip reverses battery polarity for a specified time (140-290 ms; a "wink") and then becomes ready to receive dial digits. For Delay Dial , the voip detects detects the off-hook condition. Then the voip reverses battery polarity for a specified time (reverse polarity duration has wider acceptable range than for Wink Start) and then becomes ready to receive dial digits.
Wink Timer (in ms)	integer values, in milliseconds	This is the length of the wink for Wink Start and Delay Dial signaling modes.. Applicable only when Start Mode parameter is set to "Wink Start" or "Delay Dial."
Dialing Options		
Inter Digit Timer	integer values, in seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the phonebook for the number entered. Default = 2.
Message Waiting Indication	--	Not applicable to DID-DPO interface.
Inter-Digit Regeneration Timer	integer values, in milliseconds	This parameter is applicable when digits are dialed onto a DID-DPO channel after the connection has been made. The length of time between the outputting of DTMF digits. Default = 100 ms.

GK General Settings

The fields in the main gatekeeper screen, the **GK General Settings** screen, must be configured as shown below.

GK General Settings Definitions		
Field Name	Values	Description
Registration Policy		
No Endpoints	Y/N	When selected, sets a policy whereby the Gatekeeper accepts no registrations.
Predefined Endpoints	Y/N	When selected, sets a strict zone policy, in which the Gatekeeper accepts only registrations that arrive from predefined endpoints. A strict zone policy controls network resources and services more tightly than an open zone policy.
All Endpoints	Y/N	When selected, sets an open zone policy, in which the Gatekeeper accepts any legal registration. Under this policy, the Gatekeeper can operate in “plug-and-play” mode.
Activity Configuration		
Accepts Calls	Y/N	When checked, the voip unit will accept calls.
GK Active	Y/N	When checked, the voip unit’s gatekeeper function is active.
Debug Level	0-100	The higher the value, the greater the details in Syslog or Console reports.
Buttons		
Memory Settings		Launches secondary screen on Memory issues. (See next table.)

Click on the **Memory Setting** button to access the **Memory** screen. The configurable fields must be configured as shown below.

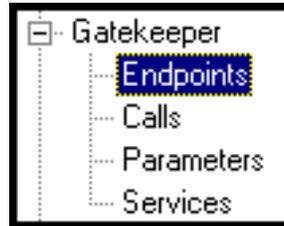


GK General Settings Definitions (cont'd)		
Field Name	Values	Description
GK Memory Values		
Maximum Calls	10, 20, 30	The maximum number of concurrent calls. MVP210AV support 5 calls; MVP410 AV supports 10 calls; MVP810 AV supports 15 calls.
Maximum Registrations	2 - 250	Maximum number of endpoints that can be registered on the gatekeeper-controlled network. Note: IP phones registrations are limited to 15 regardless of this setting.
RAS Parameters		In H.323, RAS parameters pertain to Registration, Admission, and Status in the H.225 Call Signaling Protocol.
Response TO		The timeout (in seconds) before re-transmission of a RAS message that had previously fetched no response.
RAS Port		The RAS port for gatekeeper communication with endpoints. Default value = 1719
Q.931 Parameters		In H.323, Q.931 parameters are those that pertain to the set-up and tear- down of connections between H.323 endpoints.
Response TO (sec)		The timeout (in seconds) waiting for the TCP reply.
Connect TO (sec)		The timeout (in seconds) waiting for the Connect message of a call.
Q.931 Signaling Port		Logical port through which Q.931 protocol messages are handled. Default value = 1730
Buttons		
Default		Invokes default values for all parameters on the GK General Settings screen.

Gatekeeper – Endpoints

When an endpoint registers with the gatekeeper, the endpoint is activated. That is, it becomes an acknowledged participant on the network (or on a particular zone of a network). Registration tells the gatekeeper that the endpoint is active and ready to receive calls. An endpoint's registration can be static (essentially permanent) or dynamic (timed or conditional).

The fields of the **Existing Endpoints** screen are described in the table below.

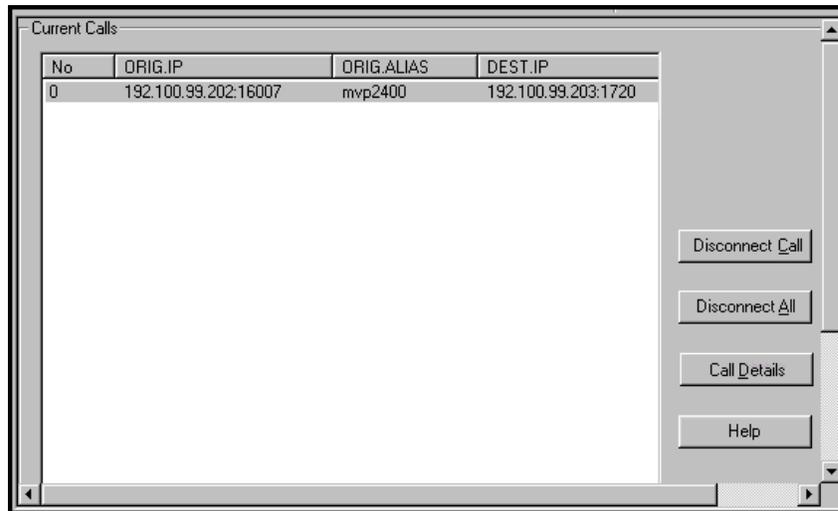


Type	Online	PreDef	Registration IP	Name	Phone	Other Aliases	Msg	TTL
Gateway	+		192.100.99.203:16001	mvp810			GRQ	48
Gateway	+		193.100.99.203:16001	mvp210G.K			GRQ	48
Gateway	+		192.100.99.202:16001	mvp2400			GRQ	10

Existing Endpoints Parameter Definitions		
Field Name	Values	Description
Type	Gatekeeper, Gateway, MCU, Terminal, or Undefined.	The endpoint type . When an endpoint attempts to register with the Gatekeeper, the Gatekeeper compares the endpoint type with the predefined value. If the Gatekeeper detects a discrepancy, the registration is not accepted. If you are not sure of the endpoint type, select Undefined , which allows any endpoint of any type to register with the Gatekeeper. (Multipoint Control Units, MCUs, are used to facilitate conference calls.)
Online	+ or [blank]	When “+” appears, the endpoint’s registration is dynamic or “online.”
PreDef	+ or [blank]	When “+” appears, the endpoint’s registration is static or “predefined.”
Registration IP	n.n.n.n 0-255	The RAS address and RAS port of the endpoint.
Name		The H.323 ID alias of the endpoint.
Phone		The e164 alias number (conventional PSTN phone number)of the endpoint.
Other Aliases		Additional aliases for the endpoint: URL, e-mail address, transport address, party.address, or private network number (per ISO/IEC 11571). Alias addresses must be unique within a zone. Gatekeepers themselves cannot have aliases.
Msg	LRQ, RRQ, URQ, or AppURQ	The type of message sent by the endpoint when the mode for processing registration is manual. This can be an LRQ, RRQ, URQ, or AppURQ (which is a URQ sent by the Gatekeeper).).
TTL	seconds	The time remaining in seconds before the TimeToLive timer expires. If the endpoint fails to reregister within this time, the endpoint is unregistered.
Command Buttons		
Add	--	Opens an empty Predefined Properties dialog box where you can predefine a new registration.
Unregister	--	Sends a URQ message to the selected endpoint, deleting the online (or dynamic) registration properties and unregistering the endpoint.
Unregister All	--	Sends a URQ to all the online endpoints in order to unregister them.
Disconnect Endpoint	--	Disconnects all calls with which the endpoint is involved.
Delete	--	Deletes the endpoint from the Gatekeeper database. A URQ will not be sent to the endpoint.
Del Pre-def	--	Deletes the predefined (static) properties of the endpoint.
Online Properties	--	Opens the Online properties screen or the selected endpoint whereupon are shown details of that endpoint’s configuration.

Gatekeeper - Calls

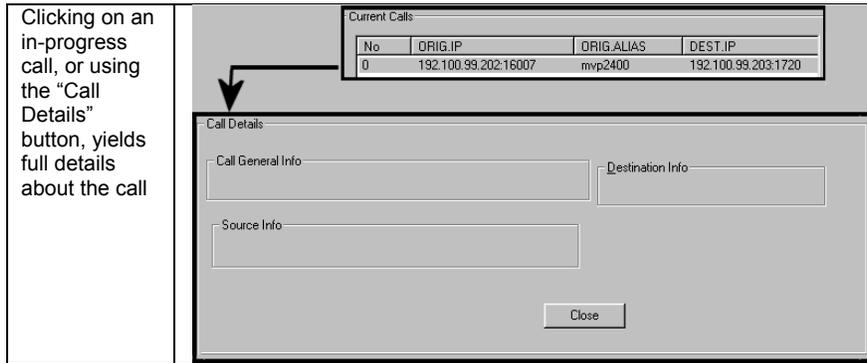
The fields of the **Current Calls** screen are described in the table below.



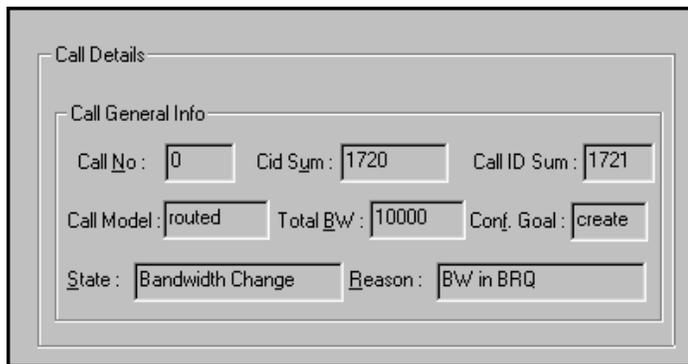
The **Calls** window displays a list of all the calls currently taking place and the basic details of the calls:

Current Calls Field Definitions		
Field Name	Values	Description
No	numeric	Number. A sequential number for identification in the list.
ORIG IP	n.n.n.n 0-255	Originating IP Address. IP Address of endpoint originating the call.
ORIG ALIAS	???	Originating Alias. The first alias given by the call's origin. The H.323 ID alias of the endpoint originating the call.
Disconnect Call (button)		Disconnects the selected call.
Disconnect All (button)		Causes all current calls to disconnect.
Call Details		Launches Call Details screen that presents technical particulars of an ongoing call.

A Call Details screen for a call in progress can be launched either by clicking on the “Call Details” button for a selected call in the Current Calls screen, or by double-clicking on a selected call listed in the Current Calls screen. The Call Details screen contains general information about the call, as well as details about the call’s source endpoint and destination endpoint.



The Call Details screen consists of three panes: Call General Info, Destination Info, and Source Info. We describe the fields for each of these panes in a separate table below.



Call Details Field Definitions		
Field Name	Values	Description
Call General Info		
Call No.		Call Number. Accession number identifying a call in progress.
Cid Sum		The conference ID number (CID) is a unique non-zero value created by the calling endpoint and passed in various H.225.0 messages. The CID identifies the conference with which the message is associated. Therefore, messages from all endpoints participating in the same conference will have the same CID.
Call ID Sum		The call ID number is a globally unique non-zero value created by the calling endpoint and passed in various H.225.0 messages. The Call ID identifies the call with which the message is associated.
Call Model	direct OR routed	Indicates whether the call is direct or routed. For direct -mode calls, the gatekeeper gives each endpoint involved in the call the destination address of the other and establishes a common call-signaling channel for them to use during the call. Then the two endpoints conduct the call without further gatekeeper involvement. For routed -mode calls, the gatekeeper establishes a connection between the two endpoints but keeps itself involved in call signaling for the duration of the call. In routed mode, the gatekeeper keeps a call-signaling channel open for the entire duration of the call. As a call-management service, the gatekeeper can change the routing of the call (by line hunting) while the calls is in progress. If the gatekeeper is to implement supplementary (H.450) services, it must operate in routed mode.

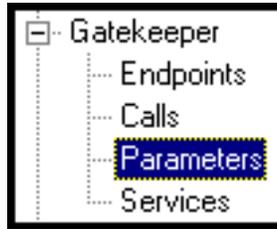
Call Details Field Definitions (Cont'd)		
Field Name	Values	Description
Call General Info (cont'd)		
Total BW		The total amount of bandwidth used by the call.
Conf. Goal		The type of conference request: create , invite or join .
State		The last reported state of the call.
Reason		The reason associated with the last state of the call.

Call Details Field Definitions (Cont'd)		
Field Name	Values	Description
Source Info fields		
Names		The H.323 alias name(s) for the originating endpoint.
Phone Numbers		The e164 alias phone number(s) of the originating endpoint.
Other Aliases: Email		An e-mail address of the originating endpoint.
OtherAliases: Trans. Name		Transport Name. An alias of the originating endpoint consisting of an IP address and port number.
Other Aliases: URL		A Internet-type address of the originating endpoint.
Call Signaling IP		The call signaling transport address of the originating endpoint.
Req. Bandwidth		Requested Bandwidth. The bandwidth requested by the calling endpoint for this call.
App. Bandwidth		Approved Bandwidth. The bandwidth the Gatekeeper made available to the calling endpoint.

Call Details Field Definitions (Cont'd)		
Field Name	Values	Description
Destination Info fields		
Names		The H.323 alias name used to make the call.
Phone Numbers		The e164 alias phone number used to make the call.
Other Aliases: Email		An e-mail address used to make the call.
Other Aliases: Trans. Name		A transport name alias used to make the call, consisting of an IP address and port number.
Other Aliases: URL		A URL alias used to make the call.
Call Signaling IP		The call signaling transport address of the called endpoint.
Reg. Bandwidth		Requested Bandwidth. The bandwidth the called endpoint requested for the call, as it appears in the ARQ/BRQ messages.
App. Bandwidth		Approved Bandwidth. The bandwidth the Gatekeeper made available to the called endpoint for the call.
Additional Phone Numbers		These allow calling with more than one B-channel.
Remote Extension Phone		This is the phone number of the called endpoint on the remote LAN. It is used for calls between multiple gateways.
Remote Extension Name		This is the identifier (name) of the called endpoint on the remote LAN. It is used for calls between multiple gateways.

Gatekeeper – Network Parameters

The fields of the **Network Parameters** screen must be configured as shown below.



Network Parameter Definitions		
Field Name	Values	Description
Status Information		
Use Update button to refresh the Status Information fields.		
Ongoing Calls	number	The number of current calls with the Gatekeeper.
Currently Registered	number	The number of endpoints registered with the Gatekeeper.
Current BW Usage	number	The current bandwidth usage of the ongoing calls in Kbps.
Configuration Options		
Alias Giving	Y/N	<p>When an endpoint sends an RRQ message, the Gatekeeper uses the additional aliases that were predefined for the endpoint as online aliases. This enables the Gatekeeper to assign terminal alias names through which the terminal can be accessed by others. The following are two examples of how this option can be used:</p> <ul style="list-style-type: none"> Example of Alias Giving for a Terminal. To make a terminal accessible by dialing 100, add the alias 100 to the terminal's predefined information, and select the Alias Giving option. When the terminal sends an RRQ message, the 100 alias becomes a dynamic (online) alias, and all calls to 100 will be directed to the terminal. Example of Alias Giving for Gateways. To make all Gateways supply Service 80, add Service 80 to the Service Table, add the 80 alias as predefined information to all registered gateways, and select the Alias Giving option. When the gateways register, they will support Service 80.

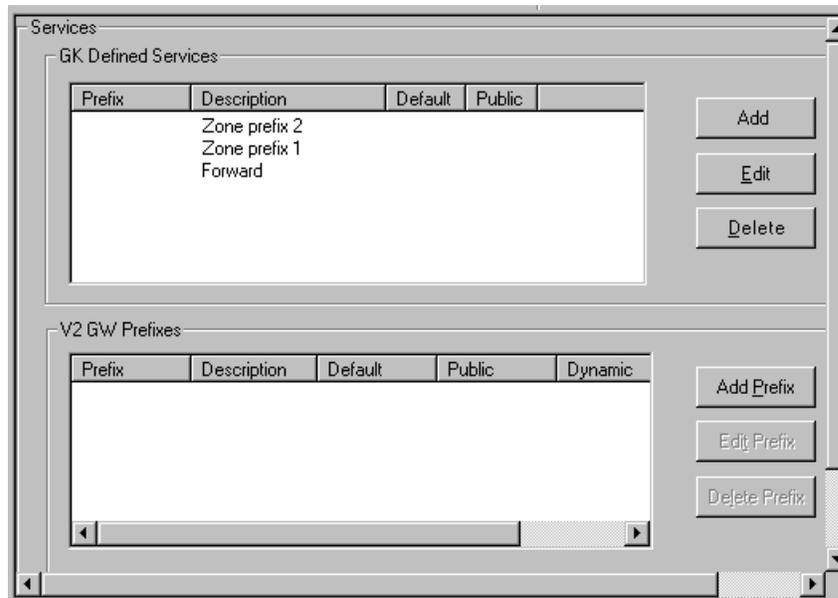
Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Pre-Granted ARQ		
PreGrant ALL	Y/N	Select to cause the Gatekeeper to send a pregrantedARQ permission in the RCF message for each endpoint that wishes to register. The pregranted ARQ permission is given to both makeCall and answerCall with routed mode. When an endpoint receives the permission, it may start the call with a Setup message or directly answer the call with a Connect message.
Line Hunting Information		
Call to Out-of-Service Supplier	Y/N	“Y” enables the sending of RAI messages. In a normal scenario, the gatekeeper will hunt among all the available endpoints that have been registered using the same tech-prefix. Each endpoint can inform the gatekeeper about its resource availability using an RAI (Resource Available Indication) message. Upon receiving an RAI message from an endpoint, the gatekeeper would consider that endpoint as an Out-of-Service Supplier. The ‘Almost Out of Resources’ configuration would allow the gatekeeper to hunt such Out-of-Service Supplier endpoints for routing the calls.
Remove H.245 Addr in Call Hunt	Y/N	When selected, the gatekeeper will not convey in its outgoing setup message the H.245 address received in an incoming setup message. This prevents H.323 terminals from establishing a channel for a call only to refuse the call later.
Service Configurable Properties	Y/N	When “Y” is selected, the gatekeeper will perform a Priority Based Line Hunting among those destinations registered using the same tech-prefix.
Call Proceeding		
Send Immediately	Y/N	Immediate return of call-proceeding message to originating endpoint. When selected, the gatekeeper will send the Q.931 call –proceeding message to the originating endpoint immediately after receiving that endpoint’s call setup request.
With H.245 Addr	Y/N	When enabled, gatekeeper supplementary services will remove the H.245 address from the outgoing setup in order to prevent early H.245 establishment to the call’s destination. This destination can be changed during Forward on Busy or during Forward on No Response (CFNR).
After Overlapped Sending	Y/N	Delayed return of call-proceeding message to originating endpoint. When selected (in routed mode), the gatekeeper will send a Q.931 call-proceeding message to the originating endpoint after it receives a return call-proceeding message back from the destination endpoint.

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Call Mode		
Direct Mode		Sets the call mode to direct. In this mode, terminals send ARQ messages to the Gatekeeper, but pass the call signaling and media control signaling directly between them.
Routed Mode		Sets the call mode to routed. In this mode, terminals pass admission requests and call signaling through the Gatekeeper. Media control information is sent directly between the terminals. Note: Though direct calls consume fewer Gatekeeper resources, call control is better for indirect (or routed) calls.
Configuration Parameters		
Max Number of Calls		The maximum number of concurrent calls allowed in the zone. This number is fixed to 5, 10, or 15 call for MVP210-AV, MVP410-AV, and MVP810 models respectively.
Max Total BW (KBps)		The amount of bandwidth in Kbps that call traffic can consume at any given time.
Registration TO (hrs)		Registration Timeout. Sets the number of hours of inactivity after which the dynamic registration of a terminal expires. Only the dynamic (online) properties will be unregistered. If the endpoint is also static (predefined), the static properties remain valid.
IRQ Interval (sec)		The interval, in seconds, between IRQ messages sent by the Gatekeeper. IRQ messages are sent to all online endpoints registered as dynamic in order to verify that the endpoints are online. The number you set determines the delay between two IRQ messages to the same endpoint. Choosing the desired delay should take into account the following factors: <ul style="list-style-type: none"> • IRQ messages add to the traffic already present over the network, and the shorter the delay, the more IRQ messages are sent. However, the longer the delay, the longer it takes for the Gatekeeper to detect dynamic registrations that have ceased to be online. • The delay parameter relates to the interval between two IRQ messages per one endpoint, so the actual number of the IRQ messages the Gatekeeper creates during this interval should be multiplied by the number of endpoints registered dynamically. • To disable the IRQ polling, set this value to zero. • The effective IRQ interval cannot fall below three times the RAS timeout. • IRQ messages will not be sent at a rate exceeding 20 per second.

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Configuration Parameters (Cont'd)		
Call IRQ Interval		<p>The interval, in seconds, between IRQ messages sent by the Gatekeeper to query the status of calls. IRQ messages are sent to all online endpoints registered as dynamic and having ongoing calls in order to verify that the calls are still ongoing. The number you set determines the delay between two IRQ messages to the same endpoint regarding the same call. Choosing the desired delay should take into account the following factors:</p> <p>IRQ messages add to the traffic already present over the network, and the shorter the delay, the more IRQ messages are sent. However, the longer the delay, the longer it takes for the Gatekeeper to detect calls that are stale.</p> <p>The delay parameter relates to the interval between two IRQ messages per one call, so the actual number of the IRQ messages the Gatekeeper creates during this interval should be multiplied by the number of ongoing calls registered dynamically.</p> <p>To disable the IRQ polling, set this value to zero.</p> <p>The effective IRQ interval cannot fall below three times the RAS timeout.</p> <p>IRQ messages will not be sent at a rate exceeding 20 per second.</p>
Default Distance		<p>The “distance” (number device-to-device hops that a call must traverse between endpoints) allowed for endpoints which are only dynamically registered, such as an endpoint with no predefined values. This distance is compared to the distances of the neighbor gatekeepers and to the multicast distance in order to determine if an LRQ can be sent on behalf of the requesting endpoint.</p> <p>NOTE: The neighboring gatekeeper feature is not supported in the current software version.</p>
Out-of-Zone Distance		<p>The “distance” (number device-to-device hops that a call must traverse between endpoints) allowed for an out-of-zone endpoint that is making a call through the Gatekeeper. This distance is compared to the distances of the neighbor gatekeepers and to the multicast distance in order to see if an LRQ can be sent on behalf of the requesting endpoint.</p> <p>NOTE: The neighboring gatekeeper feature is not supported in the current software version.</p>
Multicast Distance		<p>The “distance” (number device-to-device hops that a call must traverse between endpoints) associated with sending an LRQ by multicast.</p> <p>NOTE: The neighboring gatekeeper feature is not supported in the current software version.</p>
GK-ID		<p>The name of the Gatekeeper. The terminals identify the Gatekeeper by this name during the discovery process. The Gatekeeper responds only to Discovery requests that either contain a matching Gatekeeper identifier or have no Gatekeeper identifier.</p>
Update (button)	--	<p>Click to update information in the “Status Information” fields of the Network Parameters screen.</p>

Gatekeeper - Services

The fields of the **Services** screen are described in the table below.



Services Screen Definitions		
Field Name	Values	Description
GK Defined Services		
Prefix		A prefix that identifies the service.
Description		A description of the service that is accessible by dialing the prefix. See “ <i>GK Defined Service Types</i> ” section on following pages.
Default		For any GK-defined service being used, the user must select either “Default” or “Public.” When Default is selected, the service is accessible to all endpoints that are not predefined in the zone.
Public		For any GK-defined service being used, the user must select either “Default” or “Public.” When Public is selected, the service is accessible to all endpoints that are not part of the zone.

Services Screen Definitions (cont'd)		
Field Name	Values	Description
V2 GW Prefixes		<p>H.323 Version 2 enables the gateway to specify prefixes that the user should dial before the WAN number in order to make a call using a certain medium. E.g., the user could dial the prefix 3 for voice calls or 77 for H.320 video calls. The prefixes are defined in the RRQ message at registration. Prefix can be any H.323 alias, including an H.323 ID & mail address.</p> <p>When a terminal places a LAN to WAN call, it should add one of the prefixes to the dialed number. The Gatekeeper identifies the prefix & routes the call to the appropriate gateway. If more than one gateway supplies the same prefix, line hunting is possible between the gateways.</p>
Prefix		Identifies the service. The prefix can be a numeric code, alphanumeric string, name, or phone number that the user dials. Per H.323 Vers. 2, prefixes can also be of URL and e-mail type. Also for H.323 Vers. 2, the type must precede the prefix. For example, TEL: 3 or NAME: John.
Description		A description of the service that is accessible by dialing the prefix.
Default		Select to make the service accessible to all endpoints that are not predefined in the zone.
Public		Select to make the service accessible to all endpoints that are not part of the zone.
Dynamic	Y/N	Indicates whether the service is static (essentially permanent) or timed & conditional (dynamic). This field indicates whether the service has been added manually (non-dynamically; field value =N) or dynamically (field value = Y) as part of registration from endpoints.
Buttons		These buttons allow you add, edit, or delete a selected service or prefix.

Chapter 4 - Miscellaneous

Gateway Survivability Mode

A customer may wish to have some ports on the MultiVOIP gateway not be managed by CM, but rather have them directly routed over a private IP WAN to a second MultiVOIP gateway at the side of the WAN network. An example of this would be to send a T.38 fax call over the private IP WAN network to a fax port on a second MultiVOIP gateway.

Create phonebook entries for optional gateway survivable mode (if chosen). In this mode, the MultiVOIP provides survivability to local FXS/FXO ports; IP phones are not supported. If IP phone survivability is desired, it is recommended that you configure gatekeeper survivability in Section 6 and skip this section.

MultiVOIP Phonebook: A Communication Manager Example

1. An MVP810 (eight-channel) MultiVOIP is used
2. The MultiVOIP's first six ports are used for analog stations with extensions numbers (6001-6006).
3. The MultiVOIP's last two ports are used for analog FXO trunks that may be reached by dialing 9 plus the local area codes of "303" or "720".
4. The IP address of this MVP810 is 204.26.122.1.
5. The incoming PSTN trunk calls from either port #7 or port #8 will use the "AutoCall" feature and will be routed to the first station port. This port will serve as an "attendant" during survivability. NOTE. Normally the "AutoCall" register will have the PBX extension of 8001 that will be forwarded up to the MultiVantage server. But in this example of survivable mode operation, the 8001 will be routed to local station port 6001 with the help of the Outbound Phonebook table.
6. The Q.931 Signaling Port value of each Outbound Phonebook entry must match the Call Signaling port value configured for the destination voice channel in the Communication Manager Settings screen. The MultiVOIP's Outbound PhoneBook will be configured as illustrated in the table below.

MultiVOIP Phonebook: A Comm Mgr Example			
Outbound Destination Pattern	Remove Prefix	Add Prefix	IP Address
6001			204.26.122.1
6002			204.26.122.1
6003			204.26.122.1
6004			204.26.122.1
6005			204.26.122.1
6006			204.26.122.1
8001	8001	6001	204.26.122.1
9303			204.26.122.1
9720			204.26.122.1

The Inbound Phonebook will be configured as illustrated in the table below.

Inbound Destination Pattern: Filter & Remove this Prefix	Add Prefix	Channel to direct call toward
6001		1
6002		2
6003		3
6004		4
6005		5
6006		6
9303	303	7
9303	303	8
9720	720	7
9720	720	8

Comm Mgr Dialing Example #1: A local station-to-station call

1. The station at FXS port #1 goes off hook and received the survivable dial tone from the MultiVOIP gateway. The client then proceeds to dial “6002”.
2. The dialed digits are conveyed, per the instructions of the “Outbound Phonebook,” to IP address 204.25.122.1 (which is the self-same MultiVOIP unit).
3. The “Inbound Phonebook” filters on the dialed string “6002” and sees that it is destined for port #2, and ringing tone is applied to station 6002.
4. When station 6002 answers, the call is completed.

Comm Mgr Dialing Example #2: A local analog station dials outward on the local analog trunk to gain access to the local central office.

1. The station at FXS port #1 goes off hook and receives the survivable dial tone from the survivable gateway. The client then proceeds to dial “93035380000”.
2. These dialed digits are forwarded, per the instructions of the “Outbound Phonebook,” to IP address 204.25.122.1 (which is the self-same MultiVOIP unit).
3. The “Inbound Phonebook” filters the dialed digits because it detects the string “9303.” Acting per its configuration, it removes “9303,” adds “303” and directs that the call be dialed out to either FXO trunk port #7 or #8. It would have been possible to direct calls with these dialed sequences simply to one port (for example, to port #7). Because two ports are listed, the gateway will hunt for an available port.
4. The selected trunk port activates its tip/ring loop-start interface and the digit string “3035380000” is dialed out toward the network.
5. When the called party of 5380000 answers the call, the call is completed.

Remote Configuration

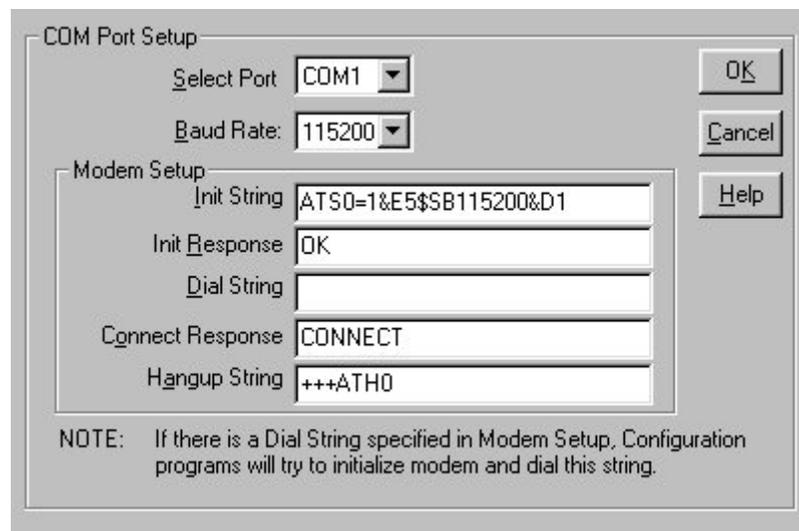
Remote configuration provides procedures for viewing or changing the configuration of a remote MultiVOIP. Remote configuration requires the MultiVOIP software to be loaded on the local PC. The local PC then controls the remote MultiVOIP.

To accomplish this, a MT5634ZBA modem is included with your MultiVOIP, except on models MVP130-AV and MVP130-AV-FXS. Two serial cables are also included, the DB25 male to RJ45 cable is used with the MultiVOIP (MVP210 model) or the DB25 male to DB25 male (larger connectors) is used with the MultiVOIP (MVP410 and MVP 810 models).

Note: On some MVP410 and MVP810 models, the modem is built in. On these models, connect the dial-up line to the RJ11 modular connector labeled Command Modem.

To remotely configure a MultiVOIP, a local PC needs to be connected to a dial-up line and the MultiVOIP software configured to call the remote MultiVOIP. The remote MultiVOIP needs to have the MT5634ZBA modem connected to a dial-up line and the Command Port. Once the connection to the remote unit is made, you can change the configuration as required. Once the configuration is changed, you can download the new configuration to the remote MultiVOIP.

1. At the remote site, verify that the "Console Messages" in the Configuration Logs dialog box is disabled before you power down the remote MultiVOIP.
2. At the remote site, remove the serial cable from the PC to the Command Port connector on the back panel of the MultiVOIP.
3. At the remote site, connect one of the serial cables provided from the Command Port connector on the back panel of the MultiVOIP to the RS-232 connector on the modem.
 - 3a. Connect the modem to your local telephone line.
 - 3b. Provide your telephone number to the person verifying your configuration.
 - 3c. Power On the modem first to allow it to initialize. Then power On the MultiVOIP.
4. At the main site, connect your local PC to a modem that is connected to a dial-up line.
5. Install the MultiVOIP software on the local PC. When installed, click **Start | Programs | MultiVOIP 9.06.XX | Configuration Port Setup**.
6. The **Comm Port Setup** dialog box displays.



Verify that the **Select Port** box is set for the COM port of your local PC and that the **Baud Rate** is set to **115200**.

In **Modem Setup**, the modem Initialization String, Initialization Response, and Connect Response should be configured as shown in the above dialog box.

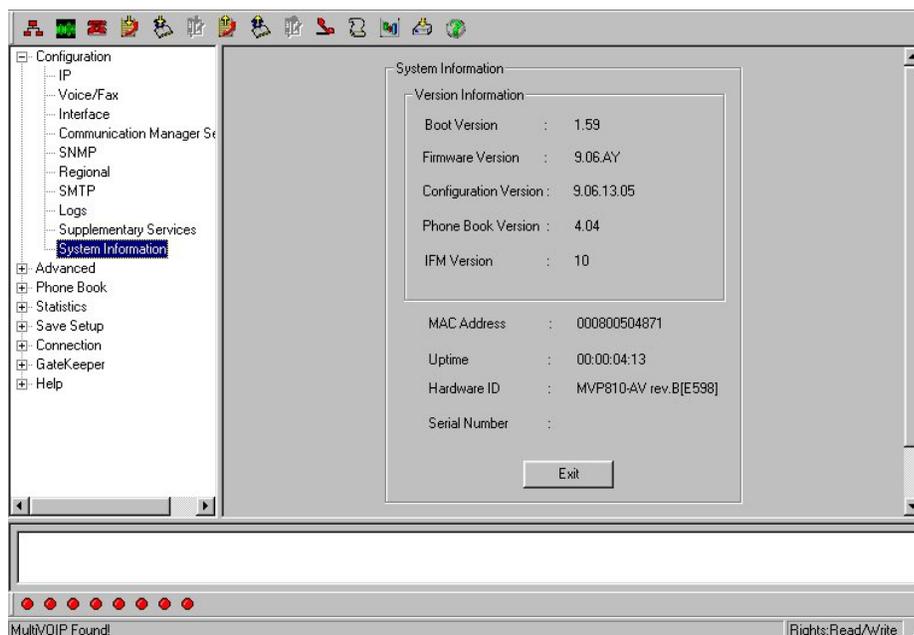
In the **Dial String** box, enter the AT command for dialing (atdt) plus the phone number of the remote MultiVOIP.

Click **OK** when you are satisfied with your selections.

7. Run the MultiVOIP Configuration program. Click **Start | Programs | MultiVOIP 9.06.XX Configuration**.
8. The **Dialing** dialog box displays while software is dialing the remote MultiVOIP.
9. The **Reading Setup** dialog box displays.
10. The **MultiVOIP** main menu displays. This is the main menu of the remote MultiVOIP.
11. Change the configuration of the remote MultiVOIP and then click **Save and Reboot** to update the configuration. The remote MultiVOIP will be brought down, the new configuration written to the unit, and the unit will reboot.
12. Close the configuration program when the downloading is complete.
13. If the same telephone number is not going to be used again in the immediate future, you may want to remove it from the **Port Setup** dialog box.

Information

The System Information screen displays version and addressing information on your MultiVOIP.



The System Information screen is a display information screen with such information as Firmware Version level, MAC Address, and Serial Number which can be helpful in identifying your unit when contacting Technical Support.

Boot Version: This field displays the version of the boot code running on the MultiVOIP. The boot code operates during initial power-up and during the firmware upgrades.

Firmware Version: Displays the firmware version running on the MultiVOIP. The MultiVOIP receives operating instructions from firmware during normal operation.

Configuration Version: Displays the version of the configuration file being used by the MultiVOIP.

Phone Book Version: Displays the version of the Phone Book file being used by the MultiVOIP.

MAC Address: Displays the MAC address assigned to the MultiVOIP's Ethernet port.

Uptime: Displays how long the MultiVOIP has been operating since the last power-up or Save and Reboot operation.

Hardware ID: The hardware ID determines the MultiVOIP model and features supported.

Displayed Hardware ID	Interfaces Supported	Configuration Modem
MVP130-AV Rev 0	FXS/FXO	No
MVP130-AV Rev A	FXS/FXO/DID	No
MVP130-AV-FXS Rev A	FXS	No
MVP210-AV Rev A	FXS/FXO	No
MVP210-AV Rev B	FXS/FXO/DID	No
MVP410-AV Rev A	FXS/FXO	No
MVP410-AV Rev B	FXS/FXO/DID	Yes
MVP810-AV Rev A	FXS/FXO	No
MVP810-AV Rev B	FXS/FXO/DID	Yes

Note: It is possible for the hardware revision printed on the packaging label and PCB (printed circuit board) to be higher than what is displayed in the **Hardware ID** field in the System Information screen. In these cases, the features supported are the same as listed for the displayed Hardware ID above.

For example, a MVP810-AV may have Revision C printed on the packaging label and PCB, but display “MVP810-AV Rev B” in System Information. This means no new features were added in Rev C and it supports the same features as the Rev B (FXS/FXO/DID).

Link Management

The Link Management screen allows you to ping a remote system by entering the IP address of the remote unit in the **IP Address to Ping** field. The Link Status window displays the results of the ping.

Link Management

Monitor Link

IP Address to Ping: 0 . 0 . 0 . 0

Pings per Test: 4

Ping Size in Bytes: 32

Response Timeout: 1000 ms

Time Interval between Tests: 0 min

Start Now Clear

Link Status

IP Address	No of Pings Sent	No Of Pings Received

Abort

Each ping test sends a user configurable number of pings to the remote unit. Configure the Time Interval between Tests field to the number of minutes (0-30) to wait between each ping test. If this field is set to 0, only one ping test is performed. The Ping Size in Bytes and Response Timeout values can be changed to accommodate various packet sizes or network delay conditions. The results are displayed under Link Status and consist of IP Address, No of Pings Sent, No of Pings Received, Round Trip Delay (Min/Max/Avg), and last Error. You can compare the number of pings sent/received and Round Trip Delay to determine the condition of the network between the two units. Ping tests are discontinued when you exit the Link Management screen.

Feature Issues

1. Calling Party Number is not displayed on the Caller ID for FXS ports after receiving a call via CM. This is due to CM not sending the Calling Party Number to the MultiVOIP as part of the Q.931 setup message.
2. “Unknown Name” is displayed on the Caller ID for FXS ports after receiving a call from a MultiVOIP FXO port that was routed by CM. This is due to CM sending “Unknown Name” in the Display field of the Q.931 setup message instead of the actual Caller ID Name received by the MultiVOIP FXO port.

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