MultiVOIP®

Voice/Fax over IP Gateways

MVP130

MVP130-FXS

User Guide

User Guide

S000386D

Analog MultiVOIP Units, Models MVP130, Models MVP130-FXS

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Record of Revisions

Revision	Date	Description
A	09/26/05	Doc re-organization. Follows S000249K. Describes 1.08 software release.
B	04/25/07	Update tech support contact list & revise warranty.
C	02/08/08	Format revision and software version 1.11 update.
D	04/30/09	Add link to Multi-Tech website for warranty update.
E	10/25/13	Removed references to product CD. Updated RoHS, safety, and other regulatory information.
F	12/17/13	Added UL translations.

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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Warranty

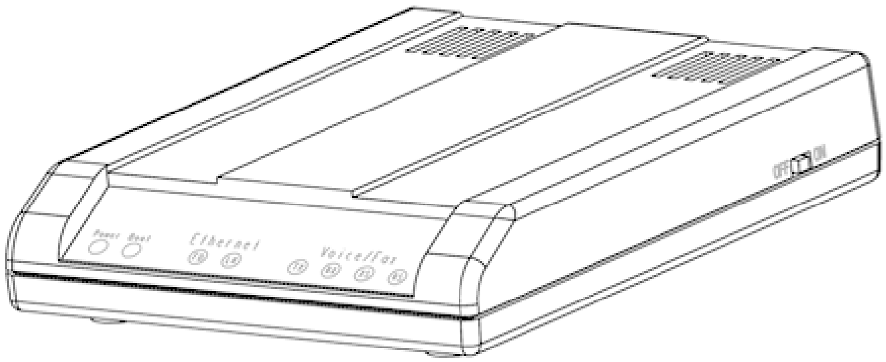
To read the warranty statement for your product, please visit: http://www.multitech.com.

Contents

Chapter 1 – Product Overview

Introduction

The MultiVOIP gateway provides toll-free voice and fax communications over the Internet or Intranet. The MVP130 and MVP130FXS models are a single-channel units. The MVP130FXS supports the FXS telephony interface only. Both units have a 10/100Mbps Ethernet interface and a command port for configuration.



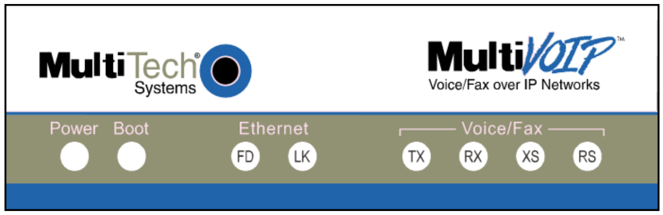
These MultiVOIPs inter-operate with a telephone switch or PBX, acting as a switching device that directs voice and fax calls over an IP network. The MultiVOIPs have “phonebooks,” directories that determine to who calls may be made and the sequences that must be used to complete calls through the MultiVOIP. The phonebooks allow the phone user to interact with the VOIP system just as they would with an ordinary PBX or telephone company (telco) switch. When the phonebooks are set, special dialing sequences are minimized or eliminated altogether. Once the call destination is determined, the phonebook settings determine whether the destination VOIP unit must strip off or add dialing digits to make the call appear at its destination to be a local call.

Interface

There are two options for accessing your MultiVOIP, one is the Windows software that is included and is necessary for the initial setup, and the other is a web-based interface that uses your web browser to access the unit. While the web interface appears differs slightly, its content and organization are essentially the same as that of the Windows interface (except for logging). These will be addressed in the following chapters.

Front Panel LEDs

On both the MVP130 and MVP130-FXS models, there are eight LEDs. These are explained in the table below.



LED	Description
Power	Indicates presence of power
Boot	After power up, the Boot LED will be on briefly while the MultiVOIP is booting. It lights whenever the MultiVOIP is booting or downloading a setup configuration data set
Ethernet	**FD.** LED indicates whether Ethernet connection is half-duplex or full-duplex and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions. **LK**. Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (that is, when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
TX	**Transmit**. This indicator blinks when voice packets are being transmitted to the local area network.
RX	**Receive**. This indicator blinks when voice packets are being received from the local area network.
XS	**Transmit Signal**. This indicator lights when the FXS-configured channel is off-hook or the FXO-configured channel (MVP130 only) is receiving a ring from the Telco or PBX.
RS	**Receive Signal**. This indicator lights when the FXS-configured channel is ringing or the FXO-configured channel (MVP130 only) has taken the line off-hook.

Computer Requirements

The computer used to configure the MultiVOIP:

- Must have Windows operating system
- Must have an available COM port

This computer is only required for local configuring and monitoring. After initial setup, most configuring and monitoring can be done remotely via the IP network.

Specifications

	MVP130 & MVP130-FXS
Operating Voltage/Current	100-240VAC / 1.0 A
Mains Frequencies	50/60 Hz
Power Consumption	4.5 watts (9.7 watts with phone off hook)
Mechanical Dimensions	1.0” H x 4.3” W x 5.6” D (2.5 cm H x 10.9 cm W x 14.2 cm D)
Weight	8 oz. (23 g)
Ambient temperature range	Maximum: 14060 degrees Celsius (140 degrees Fahrenheit) @ 20-90% non-condensing relative humidity. Minimum: 0 degrees Celsius (32 degrees Fahrenheit).
Warranty	2 years

Chapter 2 – Installing and Cabling the MultiVOIP

Introduction

MVP130 MultiVOIP models must be installed by qualified service personnel in a restricted-access area, in accordance with Articles 110-16, 10-17, and 110-18 of the National Electrical Code, ANSI/NFPA 70.

Safety Warnings

Lithium Battery Caution

- A lithium battery located within the product provides backup power for the timekeeping. This battery has an estimated life expectancy of ten years.
- When this battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for battery replacement.
- Lithium cells and batteries are subject to the Provisions for International Transportation. Multi-Tech Systems, Inc. confirms that the Lithium batteries used in the Multi-Tech product(s) referenced in this manual comply with Special Provision 188 of the UN Model Regulations, Special Provision A45 of the ICAO-TI/IATA-DGR (Air), Special Provision 310 of the IMDG Code, and Special Provision 188 of the ADR and RID (Road and Rail Europe).

CAUTION: Risk of explosion if this battery is replaced by an incorrect type. Dispose of batteries according to instructions.

ATTENTION: Risque d'explosion si cette batterie est remplacée par un type incorrect. Jetez les batteries conformément aux instructions.

Safety Warnings Telecom

Before servicing, disconnect this product from its power source and telephone network. Also:

- Never install telephone wiring during a lightning storm.
- Never install a telephone jack in wet locations unless the jack is specifically designed for wet locations.
- Use this product with UL and cUL listed computers only.
- Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
- Use caution when installing or modifying telephone lines.
- Avoid using a telephone during an electrical storm. There may be a remote risk of electrical shock from lightning.
- Do not use a telephone in the vicinity of a gas leak.

CAUTION: To reduce the risk of fire, use only 26 AWG or larger UL Listed or CSA Certified telecommunication Line cord.

Avertissements de sécurité télécom analogique

Avant de l'entretien, débrancher ce produit de son réseau d'alimentation et de téléphone. également:

- Ne jamais installer du câblage téléphonique pendant un orage électrique.
- Ne jamais installer de prises téléphoniques à des endroits mouillés à moins que la prise ne soit conçue pour de tels emplacements.
- Utilisez ce produit avec UL et cUL ordinateurs répertoriés seulement.
- Ne jamais toucher fils ou des bornes téléphoniques non isolés à moins que la ligne téléphonique n'ait été déconnectée au niveau de l'interface réseau.
- Faire preuve de prudence au moment d'installer ou de modifier des lignes téléphoniques.
- Éviter d'utiliser le téléphone pendant un orage électrique. Il peut y avoir un risque de choc électrique causé par la foudre.
- N'utilisez pas un téléphone à proximité d'une fuite de gaz.

ATTENTION: Pour réduire les risques d'incendie, utiliser uniquement des conducteurs de télécommunications 26 AWG au de section supérleure.

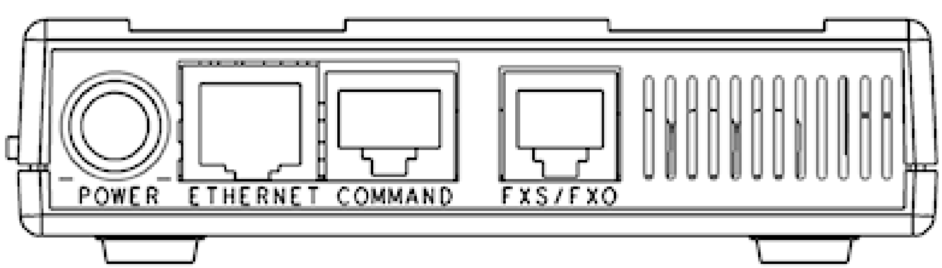
Package Contents

- MVP130 or MVP130-FXS
- DB9 to RJ45 cable
- Power transformer
- Power cord
- RJ-11 phone cord
- Printed Cabling Guide

Cabling Procedure for MVP130

To connect the MultiVOIP to your LAN and telephone equipment:

1. Connect the power cord to the power connector on the back of the MultiVOIP and to a live AC outlet.



Back connections for MVP130

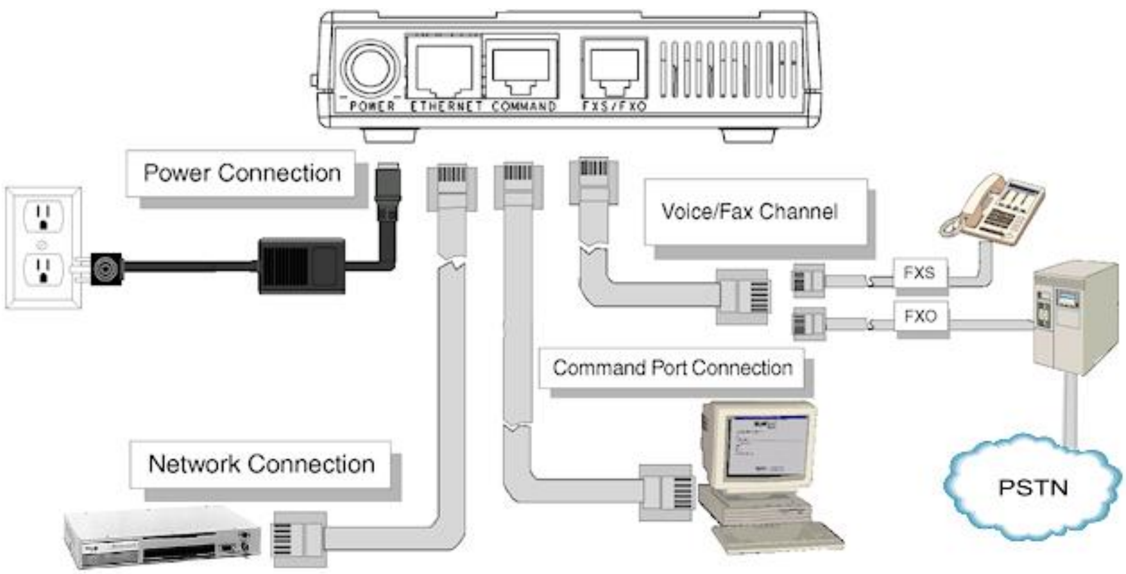
2. Connect the MultiVOIP to a PC by using the RJ-45 (male) to DB-9 (female) cable. Plug the RJ-45 end of the cable into the Command port of the MultiVOIP and the other end into the PC serial port.

3. Connect a network cable to the **ETHERNET** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.

   a. **For an FXS or FXO connection** (*MVP130-FXS*).
   FXS Examples: analog phone, fax machine
   FXO Examples: PBX extension, POTS line from telco central office
   Connect one end of an RJ-11 phone cord to the **FXS/FXO** connector on the back of the MultiVOIP. Connect the other end to the device or phone jack.

   b. **For a DID connection**. (*MVP130*)
   (*DID Example: DID fax system or DID voice phone lines*)
   Connect one end of an RJ-11 phone cord to the **FXS/FXO** connector on the back of the MultiVOIP. Connect the other end to the DID jack.
   NOTE: DID lines are polarity sensitive. If, during testing, the DID line rings busy consistently, you will need to reverse the polarity of one end of the connector (swap the wires to the two middle pins of one RJ-11 connector).

4. Turn on power to the MultiVOIP by placing the ON/OFF switch on the side to the ON position. Wait for the **BOOT** LED on the MultiVOIP to go off before proceeding. This may take a few minutes.



Cable connections

Chapter 3 – Software Installation

Introduction

Configuring software for your MultiVOIP entails three tasks:

Loading the software onto the PC (this is “Software Installation” and is discussed in this chapter).

Setting values for telephony and IP parameters that will fit your system (details are in Chapter 4).

Establishing “phonebooks” that contain the various dialing patterns for VOIP calls made to different locations (a detailed discussion of this is found in Chapter 5).

Installing the Software

To download and install the software:

1. Go to multitech.com/setup/product.go and select your model.
2. Click the Software tab and download the MultiVOIP Manager file.
3. Double-click mvm108d8.exe to extract the files.
4. Double-click on c:\mvm108d8\Disk1\Setup.exe.
5. Click **Yes** to allow the program to make changes and launch the Installation Wizard.
6. Click **Next** and follow the wizard instructions to install your MultiVOIP software.
7. Click **Finish** to exit the wizard. Click **Yes** to launch the software.

Setup Overview

There are a few necessary settings that need to be entered in the configuration software to achieve this and they are noted in the *action* lists for the categories below. The following chapters will cover all aspects in detail, but here we will cover the basic configuration needed to start VOIP communications. Below you will find the list of categories requiring information to be set before VOIP communication will be ready.

⇒ **Ethernet/IP**

⇒ **Voice/Fax**

⇒ **Interface**

⇒ **Call Signaling**

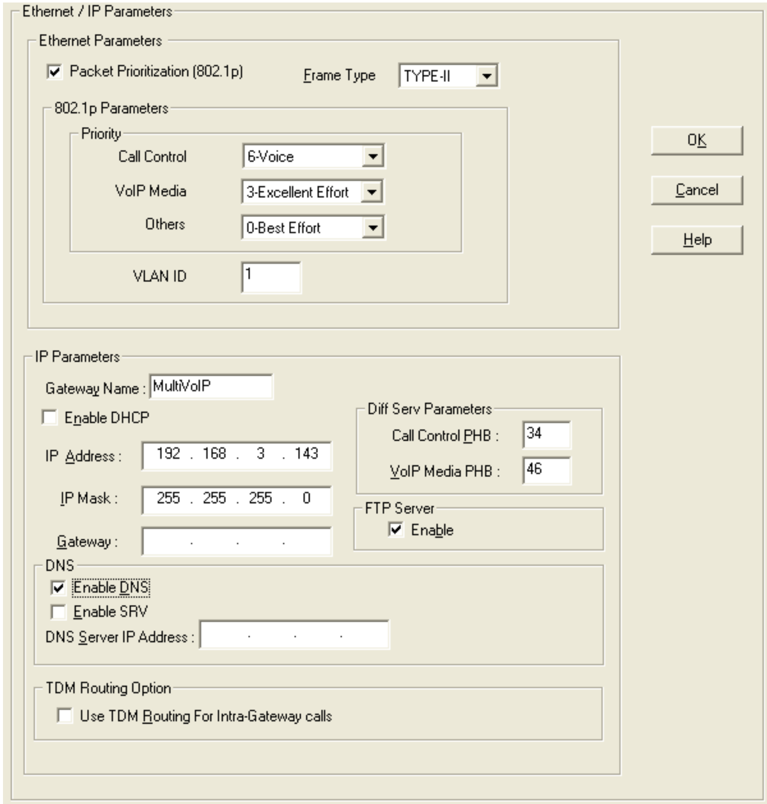
⇒ **Regional**

⇒ **Phone Book**

This setup process is followed by the **Save & Reboot** step which is very important.

Ethernet/IP

For basic operation, you need a unique LAN IP address the MultiVOIP, a subnet mask, and Gateway IP. Other settings in this category pertain to specific features and protocols that can be used, but are not necessary for basic operation. Details for all settings are provided in chapter 4.



IP settings

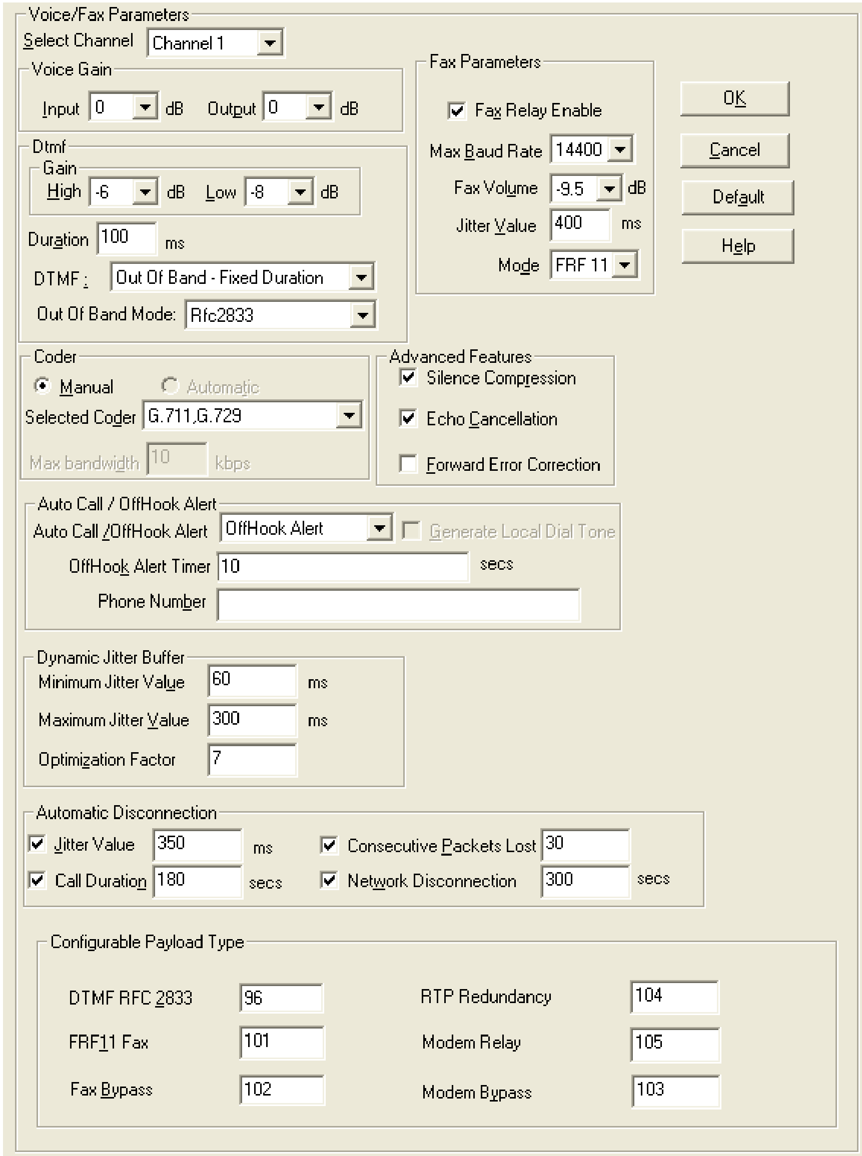
Actions

1. If used, select Packet Prioritization and set 802.1 Priority Parameters as needed.
   - Priority levels can be from 0 – 7, where 0 is lowest priority (details in Chapter 4)
   - VLAN ID identifies a virtual LAN by a number (1 to 4094)
2. Set the Frame Type to match the network. Options are TYPE II or SNAP.
3. Enter Gateway Name and if used, check to enable DHCP.
4. Enter IP Address.
5. Enter Subnet IP Mask.
6. Enter Gateway IP.
7. Enable DNS if desired. If enabled, enter the DNS Server IP Address.
8. Enable SRV support if needed.
9. Diff Serv Parameters are for routers that are Diff Serv compatible.

   Note: Setting both values to 0 effectively disables Diff Serv
10. TDM Routing can be used if necessary

Voice/Fax

The individual channels must be set up before use. The Copy Channel button can save a lot of time during this step if channels are to be set with the same parameters. Some options should be noted for future changes if necessary, but the defaults are likely to work without adjustment.



Voice & Fax settings

Actions:

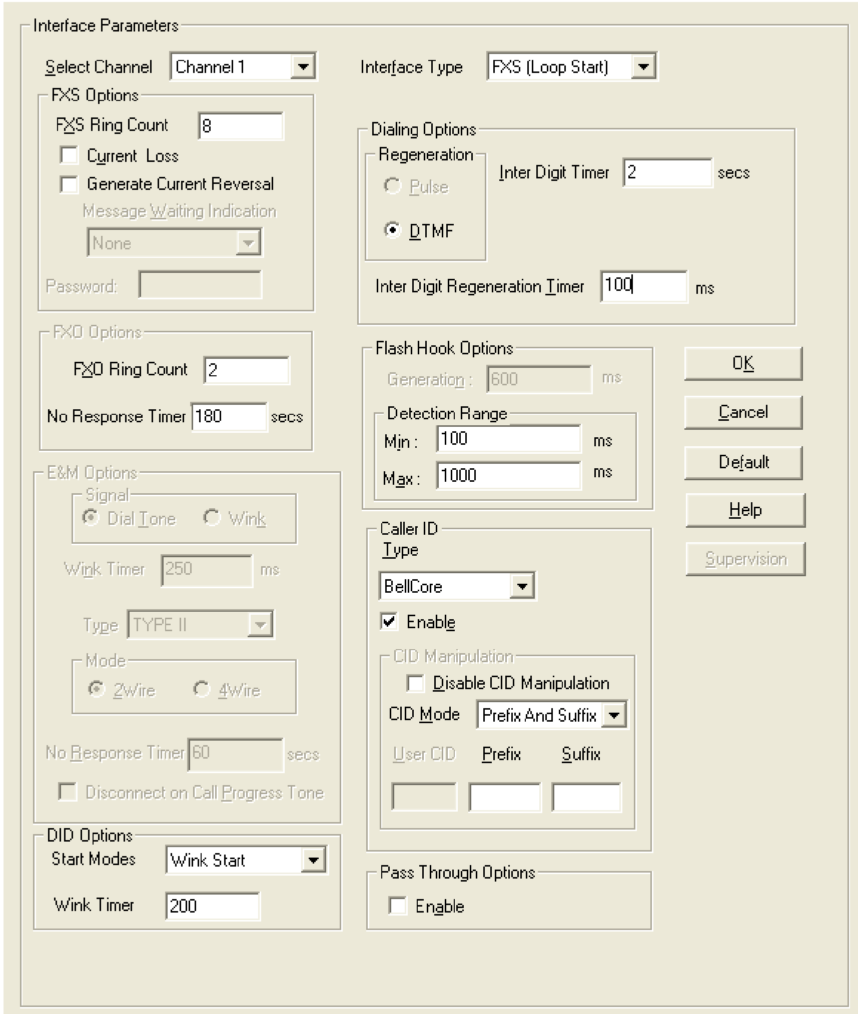
1. Select Channel
   a. Choose channel parameters:
      - Set the Fax parameters to meet your needs
        - Set Max Baud Rate to match fax machine (2400 to 14400 bps)
        - Fax Volume *should not be changed* as it may impair function
        - Jitter Value affects the time for packet reassembly
        - Mode: Select T.38 or FRF 11
      - Modem Relay Enable allows modem traffic through the VOIP system
      - Adjusting Voice Gain and DTMF *should not be done* as it may adversely affect quality
      - Select a Coder or allow Automatic negotiation
      - Advanced Features
        - Silence Compression, when enabled, will not send silence packets
        - Echo Cancellation removes echo to improve voice quality
        - Forward Error Correction allows some bad packets to be recovered
      - Choose Auto Call / OffHook Alert settings
        - For automatically calling a remote VOIP without dialing (details in Chapter 4)
      - Change Dynamic Jitter values if necessary (details in Chapter 4)
      - Select any Automatic Disconnection options needed to ensure lines are not left "open"
      - Configurable Payload Types are best left at their defaults.

   b. The Copy Channel button is available for easily transferring these settings to the other channels
2. Repeat for all channels to be used

Interface

The Interface Parameters are the telephony settings that are to be applied to the MultiVOIP channel.

Note: Feature options are enabled or unavailable depending on the selected interface type. The one option available for all interface types is the inter digit timer option. This option defines the maximum amount of time that the unit will wait before mapping the dialed digits to an entry in the phone book database. If too much time elapses between digits, and the wrong numbers are mapped, you will hear rapid busy signal. If this happens, hang up and dial again.



Interface Parameters

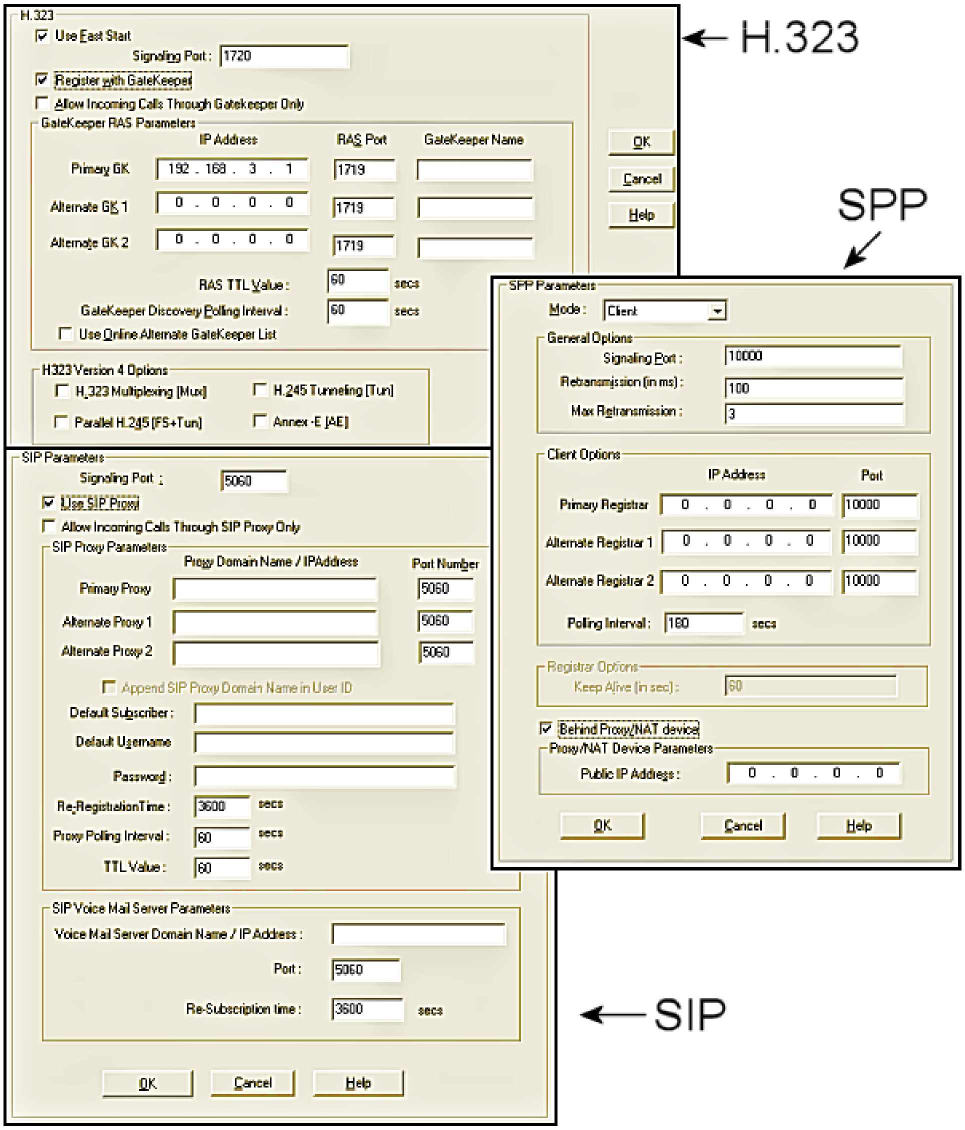
Actions:

1. Select Channel
    a. Select Interface Type: FXS, FXO, or DID (*FXS only for the MVP130-FXS*)
    b. Regeneration
        - Choose how signal is regenerated; as Pulse or DTMF
    c. Inter Digit Timer
        - Time the MultiVOIP waits between digits
    d. Message Waiting Indication is available if desired
    e. Inter Digit Regeneration Timer
        - Length of time between sent DTMF digits
2. Flash Hook Options
    a. Generation (used in conjunction with FXO)
    b. Detection Range (used in conjunction with FXS)
3. Caller ID

   a. Bellcore is the only option available
   b. CallerID Manipulation is available if needed
4. Pass Through (opens an audio path through the MultiVOIP)
5. FXS Options
   a. Set Ring Count (the number of rings allowed before call abandoned; default is 8)
   b. Use Current Loss (MultiVOIP interrupts current to disconnect)
   c. Generate Current Reversal (activates Answer/Disconnect Supervision to FXO)
6. FXO Options (not available for the MVP130-FXS)
   a. Ring Count (set number of rings before MultiVOIP answers)
   b. No Response Timer (set time to attempt call before abandoning)
   c. Supervision Button (for call answering and disconnection settings)
      - Answer Fields:
        - Current Reversal (use current reversal to answer)
        - Answer Delay
        - Answer Delay Timer (in seconds)
        - Tone Detection (allow tone sequence to disconnect)
        - Available Tones
        - Answer Tones (shows current selection from Available Tones)
      - Disconnect Fields
        - Current Reversal (use current reversal to disconnect)
        - Current Loss (loss of current will trigger disconnect)
        - Current Loss Timer (time after current loss to disconnect; in milliseconds)
        - Silence Detection Enable (use silence detection to disconnect)
        - Silence Detection Type (one-way or two-way)
        - Silence Timer (time of silence needed to trigger disconnect; in seconds)
        - DTMF Tone (use tones to disconnect)
        - Disconnect Tone Sequence (select tone pairs to use for disconnecting)
        - Tone Detection (disconnect from termination of tone)
        - Available Tones
        - Disconnect Tones (shows current selection from Available Tones)
7. DID Options (not available for the MVP130-FXS)
   a. Start Modes (Immediate, Wink or Delay Dial)
   b. Wink Timer (in milliseconds)

Call Signaling

There are three choices for Call Signaling: H.323, SIP and SPP. It is best to select one of these as the protocol to be used, rather than mixing them. Single Port Protocol (SPP) is a non-standard protocol created by Multi-Tech that allows dynamic IP allocation. Generally, the default settings will work for most users and the individual parameters may be changed if the need arises. Additional details for all settings are found in Chapter 4.



Signaling Protocols

Actions:

1. Configure your chosen Call Signal type

    a. H.323

    - Use Fast Start (may be needed for third-party vendor compatibility)
    - Signaling Port (default is 1720)
    - Register with Gatekeeper (needed if the VOIP is to be controlled by a gatekeeper)
    - Allow Incoming Calls Through Gatekeeper Only
    - Gatekeeper RAS Parameters
        - Enter parameters for Primary and any Alternate Gatekeepers
        - RAS TTL Value ("Time To Live" in seconds)
        - Gatekeeper Discovery Polling Interval (time between attempts connecting to gatekeepers)
        - Use Online Alternate Gatekeeper List
    - H.323 Version 4 Options (detailed descriptions of these can be found in Chapter 4)

    b. SIP

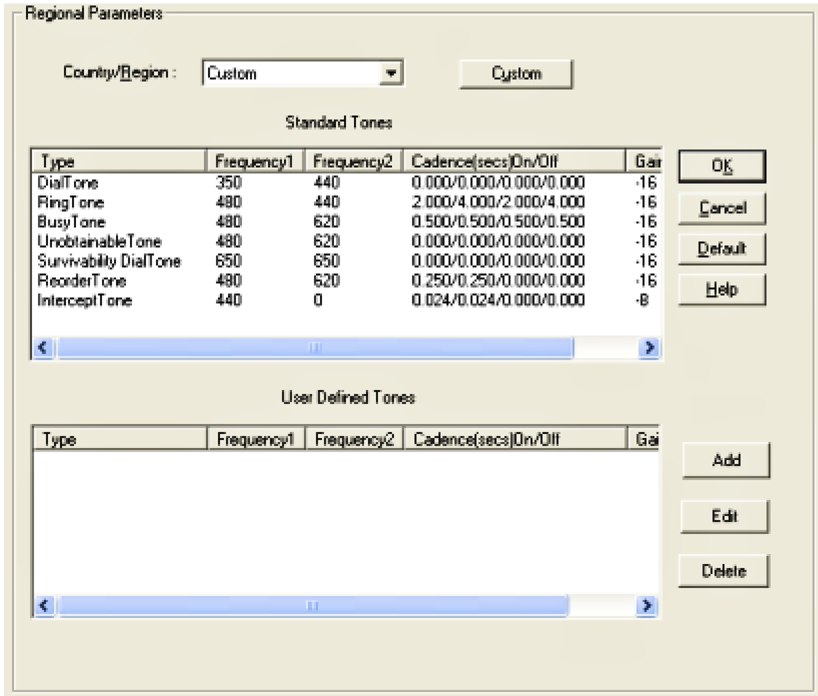
    - Signaling Port (default is 5060)
    - Use SIP Proxy (enable to work with a proxy server)
    - Allow Incoming Calls Through SIP Proxy Only
    - SIP Proxy Parameters
        - Enter information for Primary and any Alternate Proxy servers
        - Append SIP Proxy Domain Name in User ID
        - Enter User Name and Password
        - Re-Registration Time (in seconds)
        - Proxy Polling Interval (time between proxy server connect attempts)
        - TTL Value (in seconds)

    c. SPP

    - Mode (Direct, Client or Registrar)
    - Signaling Port (must be unique for any VOIP unit behind same firewall)
    - Retransmission (time before retransmission of lost packets)
    - Max Retransmission (number of retransmission attempts)
    - Client Options
        - Enter information for the Primary and Alternate Registrars
        - Polling Interval (time between connect attempts)
    - Keep Alive (time out for client un-registering)
    - Behind Proxy/NAT device
        - Enter Public IP of Proxy/NAT server

Regional

Select the country or region that the MultiVOIP unit will operate in, or use the custom option if the available settings are not adequate.



Regional Parameters

Actions:

1. Select the choice that matches the location of the MultiVOIP from the Country/Region field

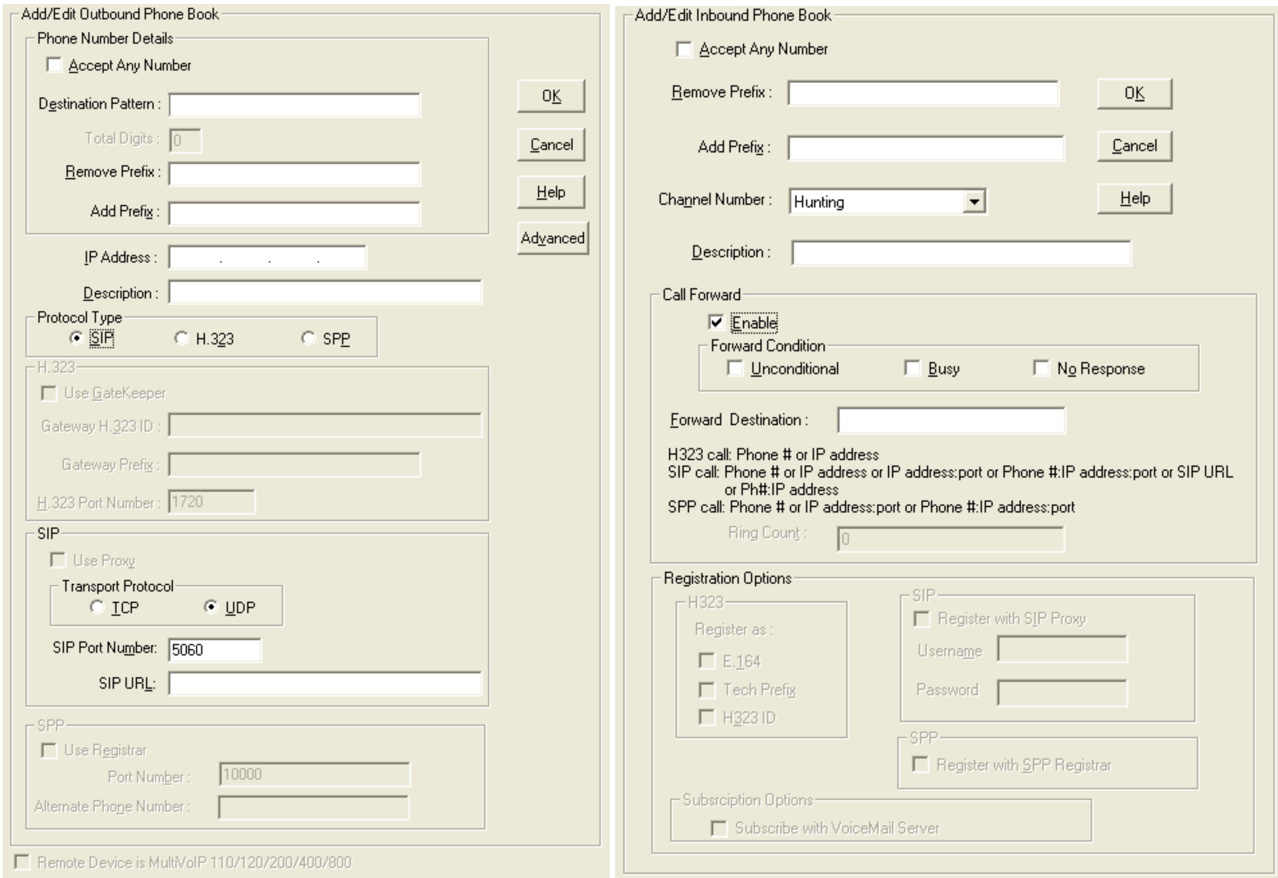
   d. If there is not a selection to fit your needs, you may select Custom and set the tones manually

   e. User Defined tones can be created for use in conjunction with FXO Supervision with the Add button

Phone Book

Without a populated phone book, the VOIP unit is unable to translate call traffic. You will need the information for both a local and any remote sites that are to be used.

Detailed descriptions and examples are available in chapter 5.



Phone Book screens

Actions

1. Select Outbound Phone Book
   a. Select Add Entry
   b. Accept Any Number may be selected to allow unmatched destinations an alternative
   c. Enter the number necessary to get out from the PBX system followed by the calling code of the destination in the Destination Pattern field
   d. Enter the PBX access digit (same number as needed to get out of the PBX system) in the Remove Prefix field
   e. Any digits that need to be added should be put in the Add Prefix field
   f. Enter the IP address of the call destination (add a Description if you like)
   g. Select a Protocol type
      - For H.323:
        - Enter Gateway settings
      - For SIP:
        - Select Transport Protocol, Proxy and URL if needed
      - For SPP:
        - Enter Registrar settings if needed
   h. The *Advanced Button* will allow an Alternate IP Address to be entered for outbound traffic
2. Select Inbound Phone Book
   a. Select Add Entry
   b. Accept Any Number for inbound traffic does not work when external routing devices are used
   c. Enter any access digits followed by the local calling code in the Remove Prefix field
   d. Enter any digits needed to access an outside line in the Add Prefix field
   e. Select Hunting in the Channel Number field to have the VOIP use the next available channel
   f. Add a description if you like
   g. Call Forward may be set up (details available in Chapter 5)
   h. Select Registration Option
3. Repeat the Phone Book steps for any additional entries needed

Save & Reboot

After you change settings on the VOIP unit, choose the **Save & Reboot** option; otherwise all changes made will be lost when the MultiVOIP is reset or shutdown.

Chapter 4 – Configuring Your MultiVOIP

Access your MultiVOIP through either a web browser or Windows software interface. There are eight parameters required for MultiVOIP to operate properly. To configure the device, you need the IP address, IP mask, Gateway IP, DNS, and the telephone interface type. Initially, the MultiVOIP must be configured locally. After initial configuration, make changes locally or remotely. Local configuration requires a connection between the MultiVOIP Command port and the computer's COM port. Use the MultiVOIP configuration software to do this.

Alternatively, MultiVOIP Manager is a Simple Network Management Protocol (SNMP) agent program that extends the capabilities of the MultiVOIP configuration software. MultiVOIP Manager allows the user to manage any number of VOIPs on a network, whereas the MultiVOIP configuration software manages only one. The MultiVOIP Manager can configure multiple VOIPs simultaneously. MultiVOIP Manager may reside on the same PC as the MultiVOIP configuration software.

Navigating the Software

The MultiVOIP software is launched from the Start button and is found in the All Programs area under the title of MultiVOIP *n.nn* (where *n* represents version number). The top option is "Configuration" – choose this.

There are several ways to arrive at the parameter that you want to use: through the left-hand panel, from the drop-down menu, clicking a taskbar icon (if available) or a keyboard shortcut (if available). Once the initial settings are entered, you may choose to configure the MultiVOIP through a Web browser instead.

Web Browser Interface

The MultiVOIP web browser interface gives access to the same commands and configuration parameters as are available in the MultiVOIP Windows interface except for logging functions. When using the web browser interface, logging can be done by email (the SMTP option).

Set up the Web Browser interface (Optional). After establishing an IP address for the MultiVOIP you can use the MultiVOIP web browser to configure the unit. To configure using the web browser interface, you must set it up:

1. Set IP address of MultiVOIP unit using the MultiVOIP Configuration program (the Windows interface).
2. Save Setup in Windows interface.
3. Close Windows interface.
4. If your computer does not have Java, you can download it from multitech.com/setup/product.go. Select your model and click Drivers.
5. Open a web browser.
6. Browse to IP address of MultiVOIP unit.
7. If username and password have been established, enter them when prompted.
8. Set browser to allow pop-ups. The MultiVOIP Web interface makes use of pop-up windows.
9. The configuration screens in the web browser will have the same content as their counterparts in the software; only the presentation differs.

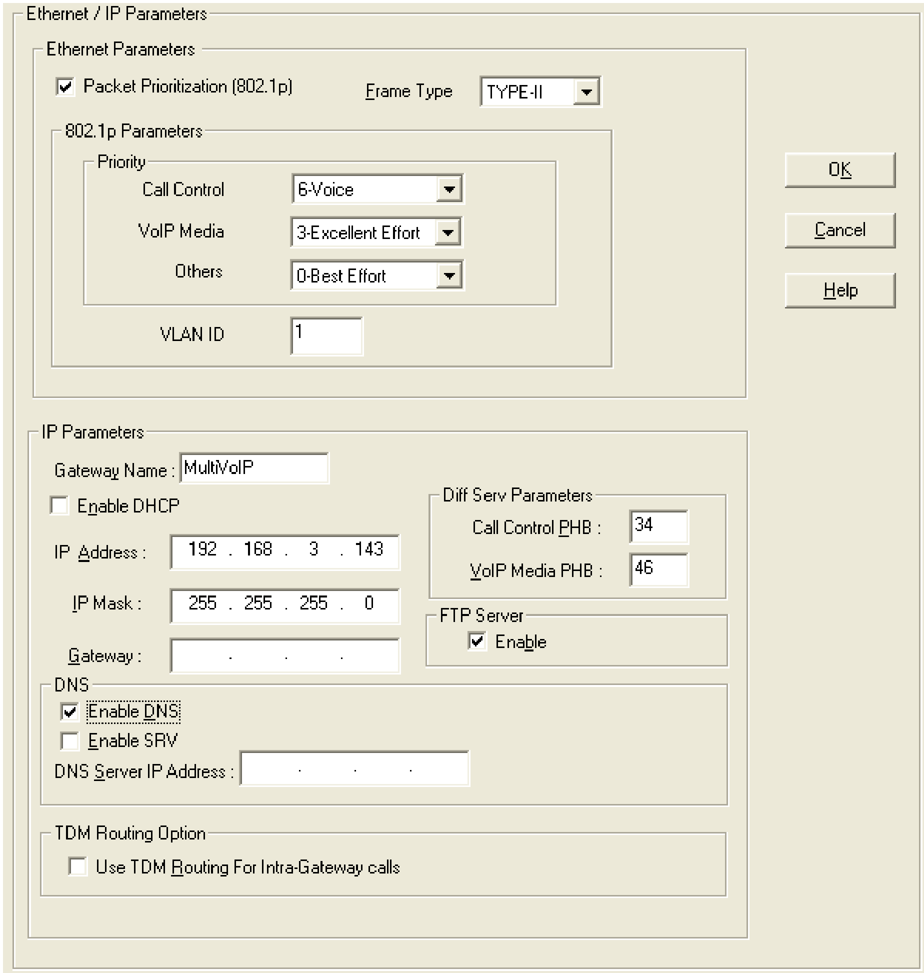
Configuration Information Checklist

To assist with the organization of the information needed, below is a chart summarizing what is necessary.

Type of Configuration Info Gathered:	Configuration screen where info is entered:	Info Obtained? ✔	Info Entered? ✔
IP info for VOIP unit	Ethernet/IP parameters		
• IP address			
• Gateway			
• DNS IP (if used) • 802.1p Prioritization (if used)			
Interface Type • FXS/FXO* • DID-DPO	Interface parameters (*In FXS/FXO systems, channels used for phone, fax, or key system are FXS; channels used for analog PBX extensions or analog telco lines are FXO).		
DID info (only if DID used) • Wink • Immediate • Delay Dial	Interface parameters		
Country code	Regional parameters		
Email address for VOIP (optional)	SMTP parameters		
Reminder: Be sure to **Save Setup** after entering configuration values.			

Ethernet/IP

This section covers the Ethernet settings needed for the MultiVOIP unit. In each field, enter the values that fit the network to which the MultiVOIP will be connected to. For many of the settings, the default values will work best – try these settings first unless you know you definitely need to change a parameter.



Network parameters

The **Ethernet/IP Parameters** fields are described in the tables and text passages below. Note that both Diff Serv parameters (Call Control PHB and VOIP Media PHB) must be set to zero if you enable Packet Prioritization (802.1p). Nonzero Diff Serv values negate the prioritization scheme.

Field Name	Values	Description
Ethernet Parameters		
Packet Prioritization (802.1p)	Y/N	Select to activate prioritization under 802.1p protocol (described below).
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.
802.1p	A draft standard of the IEEE about data traffic prioritization on Ethernet networks. The 802.1p draft is an extension of the 802.1D bridging standard. 802.1D determines how prioritization will operate within a MAC-layer bridge for any kind of media. The 802.1Q draft for virtual local-area-networks (VLANs) addresses the issue of prioritization for Ethernet networks in particular. 802.1p enacts this Quality-of-Service feature using 3 bits. This 3-bit code allows data switches to reorder packets based on priority level. The descriptors for the 8 priority levels are given below.	

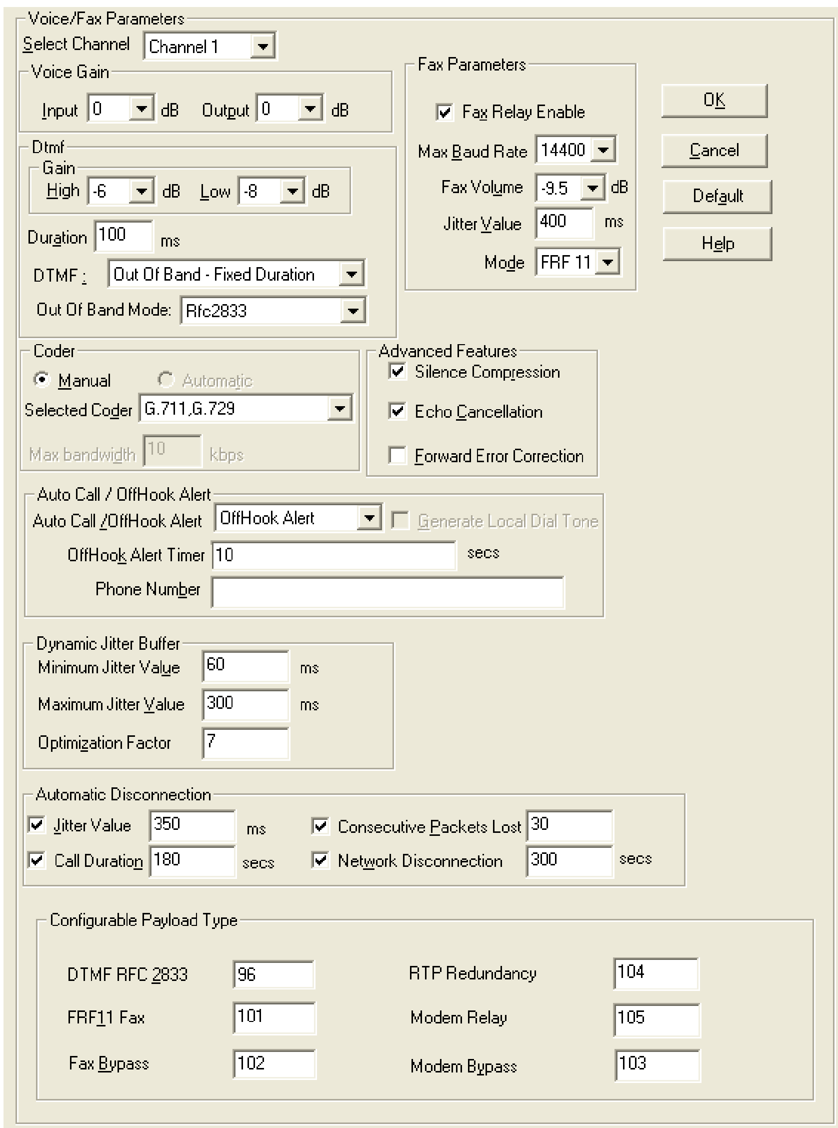
Field Name	Values	Description
	802.1p PRIORITY LEVELS: *LOWEST PRIORITY* 1 – **Background**: Bulk transfers and other activities permitted on the network, but should not affect the use of network by other users and applications. 2 – **Spare**: An unused (spare) value of the user priority. 0 – **Best Effort** (default): Normal priority for ordinary LAN traffic. 3 – **Excellent Effort**: The best effort type of service that an information services organization would deliver to its most important customers. 4 – **Controlled Load**: Important business applications subject to some form of "Admission Control", such as preplanning of Network requirement, characterized by bandwidth reservation per flow. 5 – **Video**: Traffic characterized by delay < 100 ms. 6 – **Voice**: Traffic characterized by delay < 10 ms. 7 - **Network Control**: Traffic urgently needed to maintain and support network infrastructure. *HIGHEST PRIORITY*	
Call Control Priority	0-7, where 0 is lowest priority	Sets the priority for signaling packets.
VOIP Media Priority	0-7, where 0 is lowest priority	Sets the priority for media packets.
Others (Priorities)	0-7, where 0 is lowest priority	Sets the priority for SMTP, DNS, DHCP, and other packet types.
VLAN ID	1 - 4094	The 802.1Q IEEE standard allows virtual LANs to be defined within a network. This field identifies each virtual LAN by number.
IP Parameter fields		
Gateway Name	alphanumeric	Descriptor of current VOIP unit to distinguish it from other units in system.
Enable DHCP	Y/N disabled by default	Dynamic Host Configuration Protocol is a method for assigning IP address and other IP parameters to computers on the IP network in a single message with great flexibility. IP addresses can be static or temporary depending on the needs of the computer.
IP Address	*n.n.n.n*	The unique LAN IP address assigned to the MultiVOIP.
IP Mask	*n.n.n.n*	Subnetwork address that allows for sharing of IP addresses within a LAN.
Gateway	*n.n.n.n*	The IP address of the device that connects your MultiVOIP to the Internet.
Diff Serv Parameter fields	Diff Serv PHB (Per Hop Behavior) values pertain to a differential prioritizing system for IP packets as handled by Diff Serv-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). Before using values other than these default values of 34 and 46, consult these standards documents and/or a qualified IP telecommunications engineer. To disable Diff Serv, configure both fields to 0 decimal.	
Call Control PHB	0 – 63 default = 34	Value is used to prioritize call setup IP packets. Setting this parameter to 0, in conjunction with VOIP Media PHB disables Diff Serv.
VOIP Media PHB	0 – 63 default = 46	Value is used to prioritize the RTP/RTCP audio IP packets. Setting this parameter to 0, in conjunction with Call Control PHB disables Diff Serv.
FTP Parameter fields		
FTP Server Enable	Y/N Default = disabled See "FTP Server File Transfers" in Chapter 6	MultiVOIP unit has an FTP Server function so that firmware and other important operating software files can be transferred to the VOIP via the network.
DNS Parameter fields		
Enable DNS	Y/N Default = disabled	Enables Domain Name Space/System function where computer names are resolved using a worldwide distributed database.

Field Name	Values	Description
Enable SRV	Y/N	Enables 'service record' function. Service record is a category of data in the Internet Domain Name System specifying information on available servers for a specific protocol and domain, as defined in RFC 2782. Newer internet protocols like SIP, STUN, H.323, POP3, and XMPP may require SRV support from clients. Client implementations of older protocols, like LDAP and SMTP, may have been enhanced in some settings to support SRV.
DNS Server IP Address	*n.n.n.n*	IP address of specific DNS server to be used to resolve Internet computer names.

Voice/Fax

Setting the Voice/FAX Parameters. The Voice/Fax section needs to be set for your system. The majority of the settings should be left at their default settings as changes often introduce problems with signal quality. In each field, enter the values that fit your particular setup.

Modem relay is not supported in MVP130 and MVP130-FXS models. Instead, modem bypass is supported automatically when modems are used for communication. It is recommended to disable the FAX relay when doing modem bypass for a higher success rate.



Voice/Fax parameters

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**.
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 Multi-Tech Technical Support.
DTMF Gain, Low Tones	+3dB to -31dB & “mute”	Default value: **-7 dB**. Not to be changed except under supervision of Multi-Tech Technical Support.
DTMF Parameters		
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.
Out of Band Mode	RFC 2833, SIP Info	**RFC2833 method**. Uses an RTP mode defined in RFC 2833 to transmit the DTMF digits. **SIP Info method**. Generates dual tone multi frequency (DTMF) tones on the telephony call leg. The SIP INFO message is sent along the signaling path of the call. You must set this parameter per the capabilities of the remote endpoint with which the VOIP will communicate. The RFC2833 method is the more common of the two methods.
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). Default = 14400 bps.
Fax Volume	-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. Default = -9.5 dB
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 real time faxing of Group 3 faxes over IP networks. It uses T.30 fax standards and includes special provisions to preclude FAX timeouts during IP transmissions.
Coder Parameters		
Coder	Manual or Automatic	Determines whether selection of coder is manual or automatic. 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 is negotiated.
Selected Coder	**G.711** a/u law 64 kbps; **G.726**, @ 16/24/32/40 kbps; **G.727**, @ nine	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 is compressed to 6.3K, allowing several simultaneous conversations over the same bandwidth that would otherwise carry only one.

Field Name	Values	Description
	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	To make selections from the Selected Coder drop-down list, the Manual option must be enabled.
Selected Coder: "Coder Priority"	**G.711, G.729** -or- **G.729, G.711**	Coder Priority has two options (G.711, G.729 or G.729, G711) on the Selected Coder listing of the Coder group on the Voice/Fax screen. If G.711 is the higher priority, that is, G.711 is preferred to G729 on the sending side, then G.711, G.729 option is selected. Similarly, if G.729 has the higher priority, then G.729, G.711 option is selected. It is used whenever a user wants to advertise both G.711 and G.729 coders with higher preference to a particular coder. It is useful when the calls are made from a particular channel on the VOIP to two different destinations where one supports G.711 and the other supports G.729.
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 = on.
Echo Cancellation	Y/N	Determines whether echo cancellation is enabled 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.
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall, Offhook Alert	The **AutoCall** option enables the local MultiVOIP to call a remote MultiVOIP without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote MultiVOIP identified in the **Phone Number** box of this option. If the "Pass Through Enable" field is checked in the Interface Parameters screen, AutoCall must be used. The Offhook Alert option applies only to FXS channels. The **Offhook Alert** option works like this: if a phone goes off hook and yet 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; if the VOIP system is working without a gatekeeper unit, there must also be a matching phone number entry in the Outbound Phonebook.). 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 outputted 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 is it would during a normal phone call. 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. Either function requires an entry in the Outgoing

Field Name	Values	Description
		phonebook of the local MultiVOIP and a matched setting in the Inbound Phonebook of the remote VOIP.
Generate Local Dial Tone	Y/N	*Used for AutoCall only*. 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 off hook 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. This phone number must correspond to an entry in the Outbound Phonebook of the local MultiVOIP and in the Inbound Phonebook of the remote MultiVOIP (unless a gatekeeper unit is used in the VOIP system).
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 affects the voice delay between MultiVOIP gateways.
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 ms
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 ms
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	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	**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 some 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; Default = 30 sec.	Specifies how long to wait before disconnecting the call when IP network connectivity with the remote site has been lost.

Configurable Payload Type

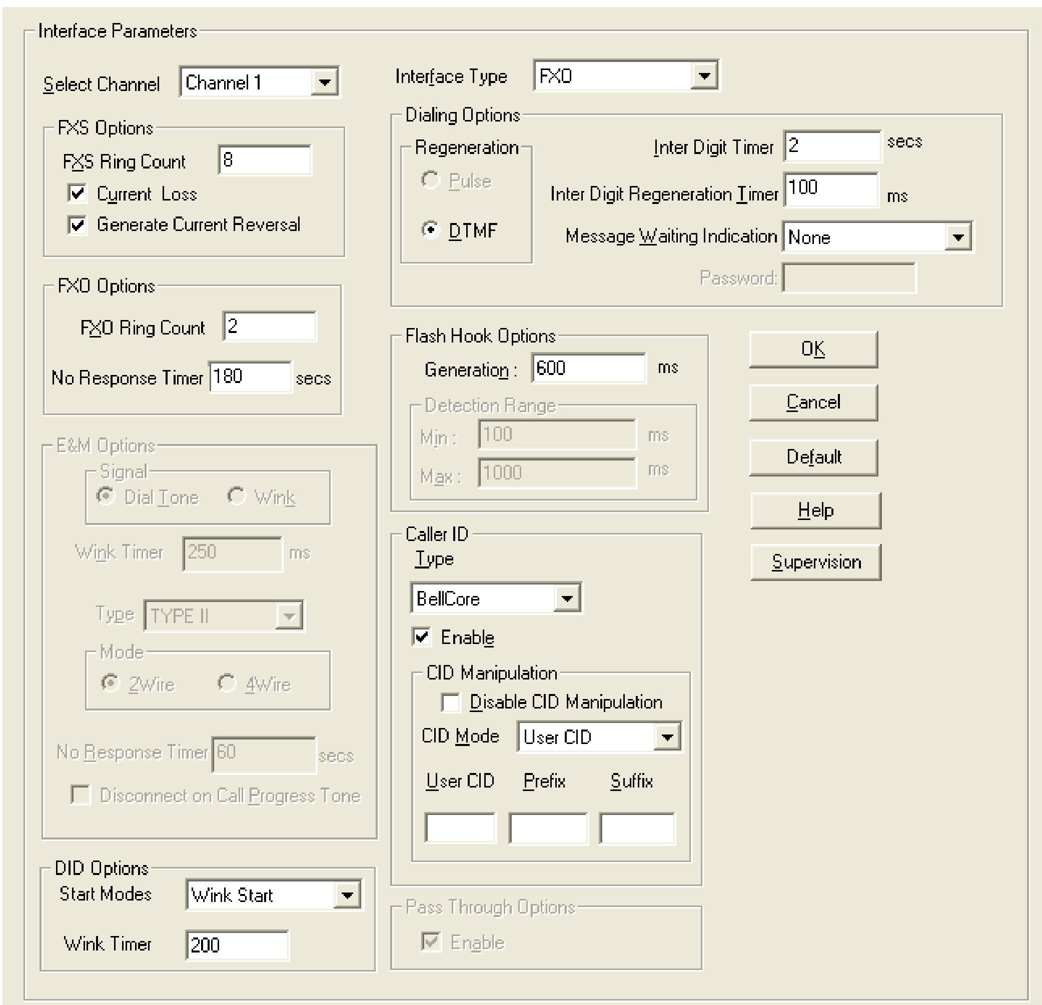
The Configurable Payload Type is located on the bottom of the Voice/Fax screen. The Configurable Payload Type is used when the remote side uses a different payload type for the associated features. In previous firmware versions, MultiVOIP's used 101 for DTMF RFC2833. If the remote side uses some other dynamic payload type such as 110, it will fail. To avoid these failures, the payload types are made configurable.

DTMF RFC2833 Configurable Payload Type is supported only for SIP & SPP but not for H.323.

Whenever you interoperate with older MultiVOIP products (that is, earlier than release *n*.11), for backward compatibility, make sure to configure the payload type values to default ones, which match the values of older MultiVOIP's.

Interface

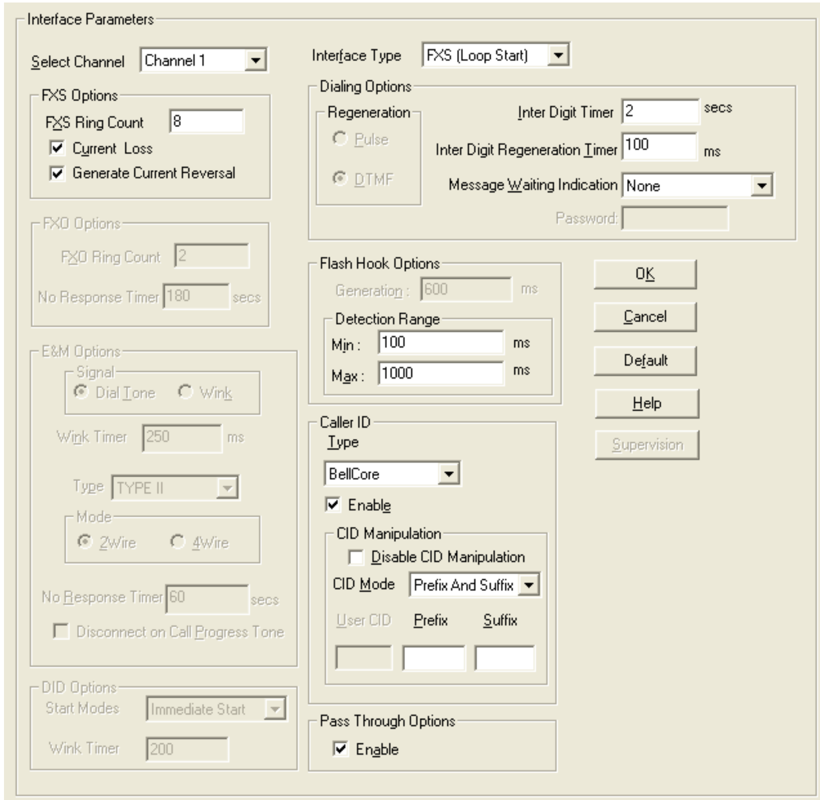
The Telephony Interface parameters are set individually for each channel and include the line types as well as some specific situational settings for those that need them. The kinds of parameters for which values must be chosen depend on the type of telephony supervisory signaling or interface used. Here you will find the various parameters grouped and organized by interface type. In each field, enter the values that fit your particular setup. The screen below shows more options available than are actually used for clarity. Your settings will determine what fields are available.



Telephony parameters

FXS Loop Start Parameters

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



FXS Loop Start parameters

Field Name	Values	Description
Dialing Options fields		
FXS (Loop Start)	Y/N	Enables FXS Loop Start interface type.
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 look in the outbound phonebook for the number entered and place the call accordingly. Default = 2.
Message Waiting Indication	--	*See details below.*
Inter Digit Regeneration Time	in milliseconds	The length of time between the outputting of DTMF digits. Default = 100 ms.
FXS Options fields		
FXS Ring Count, FXS	1-99	Maximum number of rings that the MultiVOIP will issue before giving up the attempted call.
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.
Generate Current Reversal	Y/N	When selected, this option implements Answer Supervision and Disconnect Supervision to the FXO interface using current reversal to indicate events. Applicable only when FXS and FXO interfaces are connected back to back.

Field Name	Values	Description
Flash Hook Options fields		
Generation	--	*Not applicable to FXS interface*
Detection Range	*for Min. and Max.,* 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	When enabled, this parameter creates an open audio path through the MultiVOIP. If the Pass-Through feature is enabled, the AutoCall feature must be enabled for this VOIP channel in the Voice/Fax Parameters screen
Caller ID fields		
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.
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.
CID Manipulation	Enabled by default with Caller ID enable above Disable	Caller ID Manipulation is used whenever the user wants to manipulate the Caller ID before sending it to the remote end. Caller ID Manipulation is activated on the Interface Screen. By enabling Caller ID option, you can set manipulation to Transparent, User CID, Prefix, Suffix, or Prefix and Suffix. Caller ID Manipulation is a feature, where the Caller ID detected from the PSTN line can be changed and then sent to the remote side over IP.
CID Mode	Transparent, User CID, Prefix, Suffix	**The MultiVOIP is not allowed to modify the caller ID info and then send it to the PSTN side. It only allows it to detect the caller ID from the PSTN line, modify it and then send them via IP to the remote end point.** Transparent: the CID received from PSTN will be sent out as such, without any manipulation. User CID: the CID received from PSTN will be replaced by this User CID value. Prefix: the CID received from PSTN will be prefixed with this value. Suffix: the CID received from PSTN will be suffixed with this value.

Message Waiting

Message Waiting Indication is a feature that displays an audible or visible indication that a message available. A type of message waiting is sounding a special dial tone (called stutter dial tone), lighting a light, or indicator on the phone.

When a user enables a subscription for message waiting indication, a subscription is made with the Voice Mail Server (VMS) for that particular event. Whenever the Voice Mail Server finds a change in the state of a corresponding mailbox or some event happens (for example, when a new voice message is recorded or a message is deleted, then the VMS server sends a notification to the gateway. Its indication to the user is a flashing LED or sounding a stutter dial tone.

The message waiting feature is active when the Use SIP Proxy option is selected on the Call Signaling SIP screen, a Primary Proxy IP address is entered in the SIP Proxy Parameters Primary Proxy field, the Voice Mail Server Domain Name or IP Address is entered in the SIP Voice Mail Server Parameters Group, and the Interface Type is set to FXS (Loop start). Then the FXS Options Group becomes active. The Message Waiting Indication options are None, Light, or Stutter Dial Tone.

Message Waiting

To receive messages from the VMS (Voice Mail Server/System), the subscription needs to be enabled and the voice mail server address has to be entered in the SIP Voice Mail Server Parameters Group.

The Voice Mail server IP Address, Port and Re-subscription time are configured on the SIP Call Signaling screen. When this is configured, the “Subscribe with Voice Mail Server” option is activated in the inbound phone book. Only when this option is enabled, the subscribe message will be sent to the VMS.

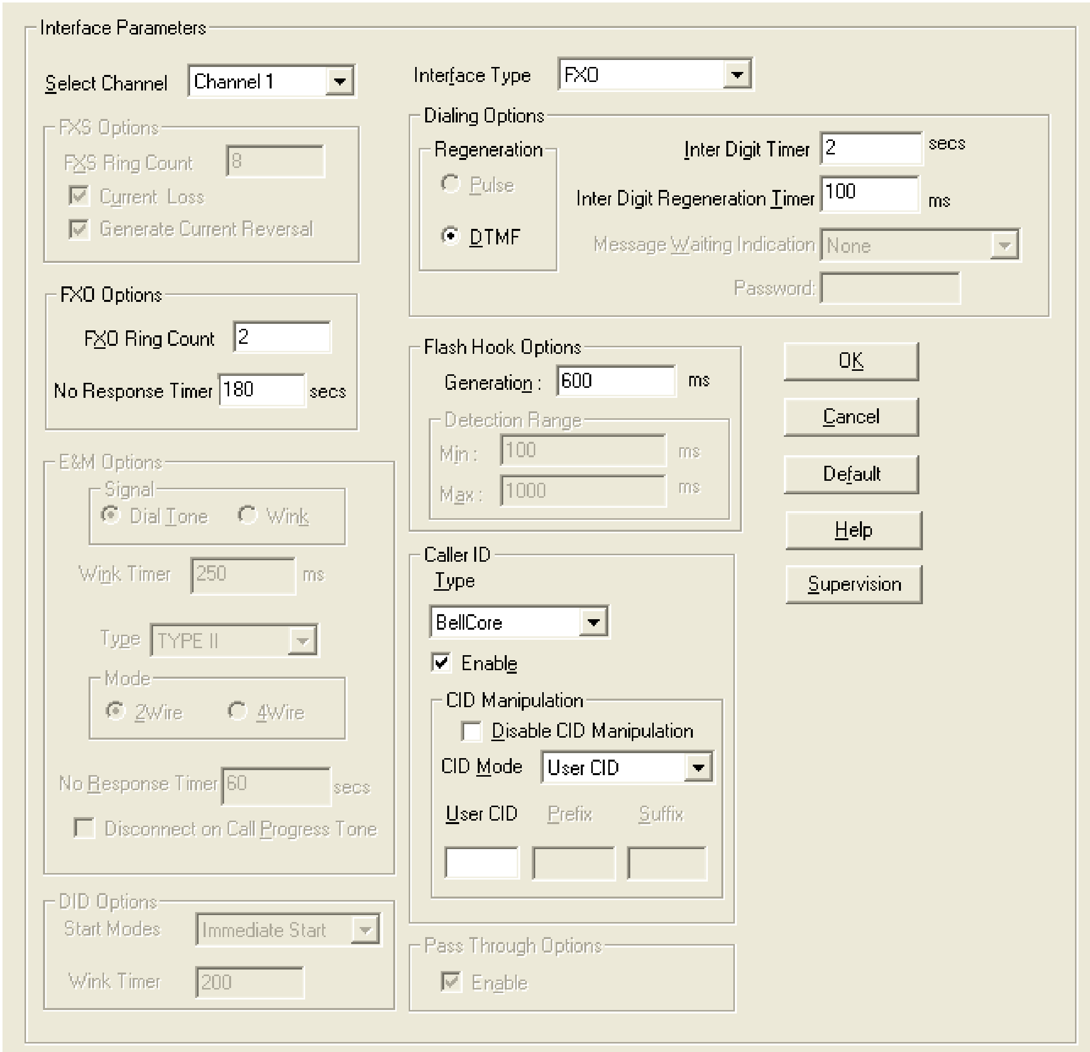
The following sequence needs to be done to enable all of the Message Waiting Features:

1. The "Use SIP Proxy" must be enabled, and the SIP Proxy Parameters and Voice Mail Server Parameters in the SIP Call Signaling Menu must be set, and the Interface Type option must be set to FXS (Loop Start) on the Interface menu's "Message Waiting Indication" options become active.
2. Then the "Message Waiting Indication" options must be set to light or stutter tone for the "Subscribe to Voice Mail Server" option to become available in the Inbound phone book entry with that channel selected.
3. To send Subscriptions for Inbound Phone Book entries, all of the following conditions are met:
   - The user needs to enter a valid voice mail server domain name or IP address in the Voice Mail Server Domain Name/IP Address field on the Call Signaling screen.
   - For an Inbound Phone Book entry, a subscription with Voice Mail Server checkbox is enabled on the Add or Edit Inbound Phone Book entries screen.
   - The Channel type corresponding to that Inbound phone book entry has to be FXS on the Interface screen.
   - The Message Waiting Indication has to be either Light or Stutter Dial Tone on the Interface Parameters screen.

The password on the Interface screen is used for that particular channel when a “SUBSCRIBE” request is sent (that is, if the MultiVOIP gets a 401/407 response from a subscribe request. Then it will take the configured password, calculate the response, and resend the “SUBSCRIBE” request.

FXO Parameters

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



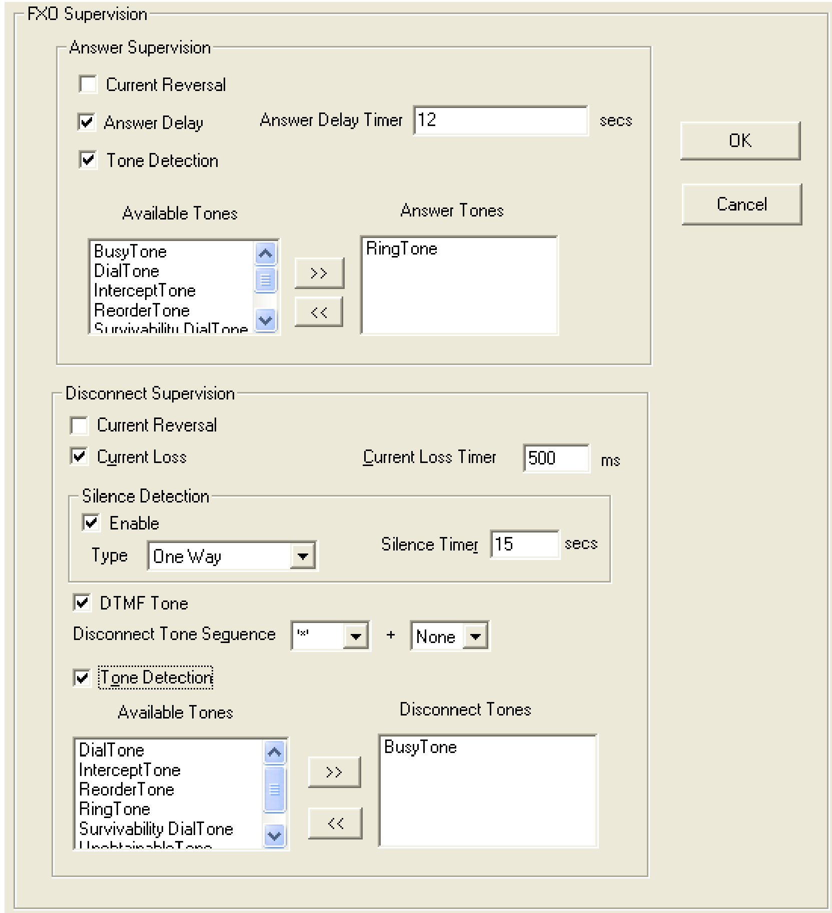
FXO parameters

Field Name	Values	Description
Interface Type	FXO	Enables FXO functionality
Dialing Options		
Regeneration	Pulse, DTMF	Determines whether digits generated and sent out will be pulse tones or DTMF.
Inter Digit Timer	1 to 10 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 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 Options		
FXO Ring Count	1-99	Number of rings required before the MultiVOIP answers the incoming call.

Field Name	Values	Description
No Response Timer	1 – 65535 (in seconds)	Length of time before call connection attempt is abandoned.
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.
CID Manipulation	Enabled by default with Caller ID enable above Disable	Caller ID Manipulation is used whenever the user wants to manipulate the Caller ID before sending it to the remote end. Caller ID Manipulation is activated on the Interface Screen. By enabling Caller ID option, you can set manipulation to Transparent, User CID, Prefix, Suffix, or Prefix and Suffix. Caller ID Manipulation is a feature, where the Caller ID detected from the PSTN line can be changed and then sent to the remote side over IP.
CID Mode	Transparent, User CID, Prefix, Suffix	**The MultiVOIP is not allowed to modify the caller ID info and then send it to the PSTN side. It only allows it to detect the caller ID from the PSTN line, modify it and then send them via IP to the remote end point.** Transparent: the CID received from PSTN will be sent out as such, without any manipulation. User CID: the CID received from PSTN will be replaced by this User CID value. Prefix: the CID received from PSTN will be prefixed with this value. Suffix: the CID received from PSTN will be suffixed with this value.

FXO Supervision

When the selected Interface type is FXO, the **Supervision** button is active. Click on this button to access call answering supervision parameters and call disconnection parameters that relate to the FXO interface type.



FXO Supervision

Field Name	Values	Description
Answer Supervision fields		
Current Reversal	Y/N	When this option is selected, the FXO interface sends notice to make connection upon detecting current reversal from the PBX (which occurs when the called extension goes off hook).
Answer Delay	Y/N	When this option is selected, the FXO interface sends the connection notice to the calling party only when the Answer Delay Timer expires. The connection notice is sent regardless of whether or not the called extension has gone off hook.
Answer Delay Timer	1 – 65535 (in seconds)	When Answer Delay is enabled, this value determines when the FXO interface sends the connection notice.
Tone Detection	Y/N	When selected, call disconnection will be triggered by a tone sequence
Available Tones	dial tone, ring tone, busy tone, unobtainable tone (fast busy), survivability tone, re-order tone	List from which tones can be chosen to signal call answer.
Answer Tones	any tone from Available Tones list	Currently chosen call-answer supervision tone.
Disconnect Supervision fields		There are four possible criteria for disconnection under FXO: current reversal, current loss, tone detection, and silence detection. Disconnection can be triggered by more than one of the three criteria.
Current Reversal	Y/N	Disconnection to be triggered by reversal of current from the PBX.

Field Name	Values	Description
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 at a specified interval after it detects a loss of current initiated by the attached device.
Current Loss Timer	200 to 2000 (in milliseconds)	Determines the interval after detection of current loss at which the call will be disconnected.
Silence Detection Enable	Y/N	Enables/disables silence-detection method of supervising call disconnection.
Silence Detection Type	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 Supervision fields		
DTMF Tone		Enables supervision of call disconnection using DTMF tones.

DTMF Tone Pairs

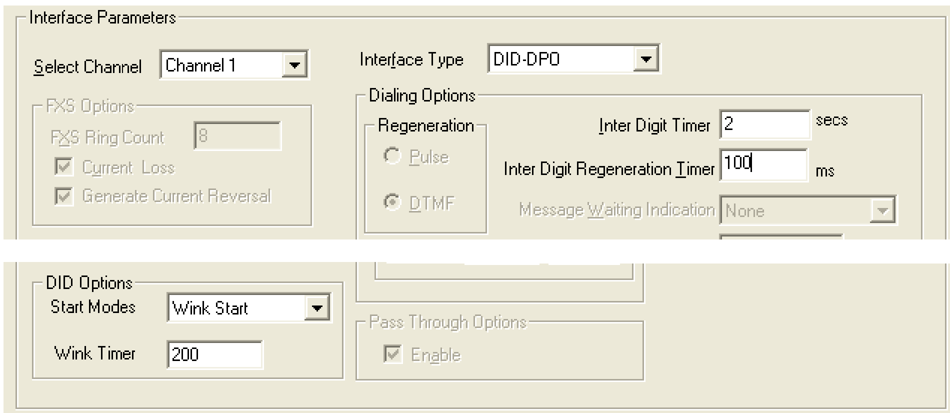
					Low Tones
	1	**2**	**3**	**A**	*697Hz*
	4	**5**	**6**	**B**	*770Hz*
	7	**8**	**9**	**C**	*852Hz*
	*****	**0**	**#**	**D**	*941Hz*
High Tones	*1209Hz*	*1336Hz*	*1447Hz*	*1633Hz*	

Disconnect Tone Sequence	1st tone pair + 2nd 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.
Tone Detection	Y/N	Enables supervision of call disconnection by detecting cessation of a pre-specified tone from the PBX.
Available Tones	dial tone, ring tone, busy tone, unobtainable tone (fast busy), survivability tone, re-order tone	List from which tones can be chosen to signal call disconnection.
Disconnect Tones	any tone from Available Tones list	Currently chosen disconnection supervision tone.

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.

Note: The FXS model does not support DID.



DID parameters

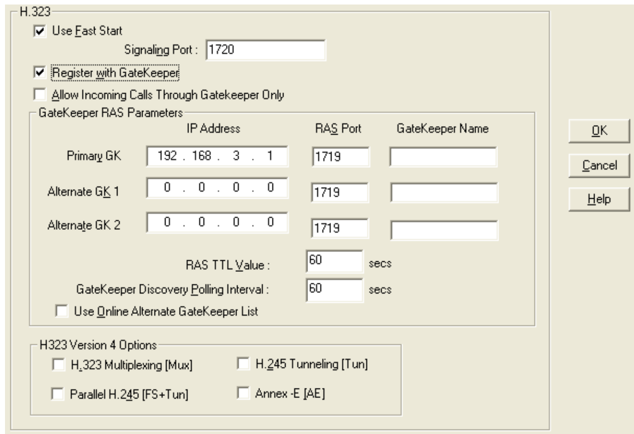
Field Name	Values	Description
Interface	DID-DPO	Enables the customer-premises side of DID functionality
DID Options		MultiVOIP's use of DID applies only for incoming DID calls. The Start Mode used by the MultiVOIP must match that used by the originating telephony equipment; else DID calls cannot be completed.
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 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.

Call Signaling

There are three types of Call Signaling available: H.323, SIP and SPP. Each type has some individual features that may make it more appealing to use than the others, depending on your needs.

H.323

H.323 is an ITU-T recommended set of standards for audio and video communications.



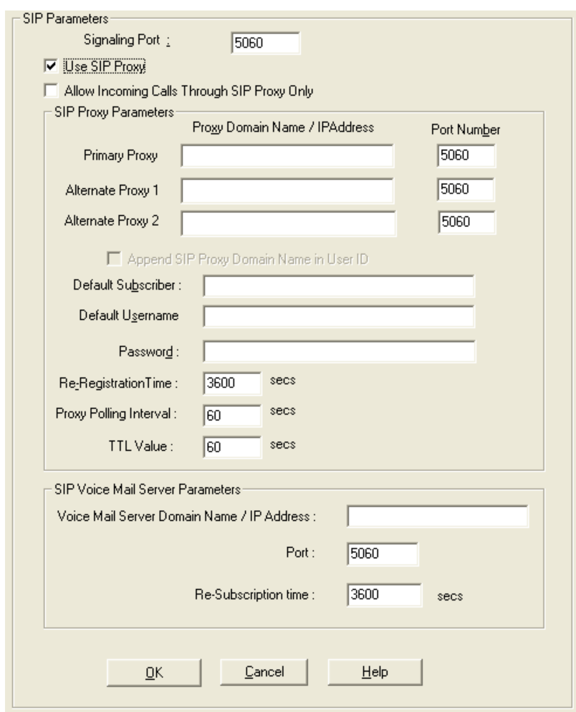
H.323 call signaling

Field Name	Values	Description
Use Fast Start	Y/N	Enables the H.323 Fast Start procedure. May need to be enabled / disabled for compatibility with third-party VOIP gateways.
Signaling Port	*port*	Default: 1720 (H.323)
Register with Gatekeeper	Y/N	Check this field to have traffic on current VOIP gateway controlled by a gatekeeper.
Allow Incoming Calls Through Gatekeeper Only	Y/N	When selected, incoming calls are accepted only if those calls come through the gatekeeper.
GateKeeper RAS Parameters		
Primary GK	--	This is the preferred gatekeeper for controlling the traffic of the current VOIP.
Alternate GK 1 and 2	--	A first and a second alternate gatekeeper can be specified for use by the current VOIP for situations where the Primary GK is busy or otherwise unavailable.
IP Address	*n.n.n.n*	IP address of the GateKeeper.
RAS Port	1719	Well-known port number for GateKeepers. Must match port number (1719).
Gatekeeper Name	*alpha-numeric*	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register. A primary gatekeeper and two alternate units are listed.
RAS TTL Value	*seconds*	H.323 Gatekeeper Time to Live value. When the MultiVOIP gateway registers it starts a gatekeeper a countdown timer. The RAS TTL Value is the interval of the countdown timer. Before the TTL countdown expires, the MultiVOIP gateway needs to register with the gatekeeper to maintain the connection. If the MultiVOIP does not register before the TTL interval expires, the MultiVOIP gateway's registration with the gatekeeper expires and the gatekeeper no longer permits call traffic to or from that gateway. Calls in progress continue to function even if the gateway becomes de-registered
Gatekeeper Discovery Polling Interval	integer 60 - 300	The interval between the VOIP gateway's successive attempts to connect to and be governed by a higher level gatekeeper. The Primary GK is the highest level gatekeeper. Alternate GK1 is second; Alternate GK2 is the lowest.

Field Name	Values	Description
Use Online Alternate Gatekeeper List		When selected, VOIP will seek an alternate gatekeeper (when none of the 3 gatekeepers shown on this screen are available) from a list. The list will reside on the Primary gatekeeper or one of the Alternate gatekeepers. The gatekeeper holding the list would download that list onto the VOIP gateways within the system.
H.323 Version 4 Options		
H.323 Multiplexing	Y/N	Signaling for multiple phone calls can be carried on a single port rather than opening a separate signaling port for each. This conserves bandwidth resources.
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.
Parallel H.245 (FS + Tun)	Y/N	FS (Fast Start) 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.
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 due to potential problems with security and bandwidth usage).

SIP

Session Initiation Protocol is the second option available for application layer control of the MultiVOIP. The fields are detailed in the table below.

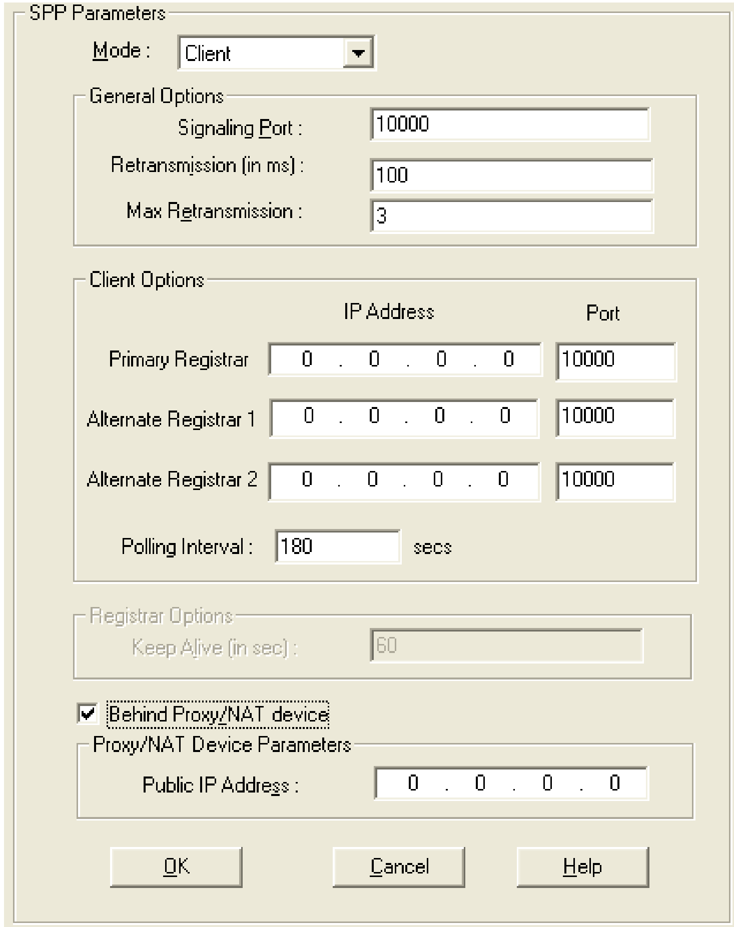


SIP call signaling

Field Name	Values	Description
SIP Proxy Parameters		
Signaling Port	*port*	Port number on which the MultiVOIP UserAgent software module will be waiting for any incoming SIP requests. Default = 5060
Use SIP Proxy	Y/N	Allows the MultiVOIP to work in conjunction with a proxy server.
Allow Incoming Calls Through SIP Proxy Only	Y/N	When selected, incoming calls are accepted only if those calls come through the proxy.
Primary Proxy	--	This is the preferred SIP proxy server for controlling the traffic of the current VOIP.
Alternate Proxy 1 and 2	--	A first and a second alternate SIP proxy server can be specified for use by the VOIP for situations where the Primary proxy server is otherwise unavailable.
Proxy Domain Name / IP Address	*n.n.n.n*	Network address of the proxy server that the VOIP is using.
Append SIP Proxy Domain Name in User ID	Y/N	When checked, the domain name of the SIP Proxy serving the MultiVOIP gateway will be included as part of the User ID for that gateway. If unchecked, the SIP Proxy's IP address will be included as part of the User ID instead of the SIP Proxy's domain name.
Port Number	*port*	Logical port number for proxy communications. Default = 5060
Default Subscriber		This is used as the default end point register with a Proxy.
Default Username	*name*	If the Username is not populated in the Phone Book, this is the Username that will be used. This works the same for the password as well.
Password	*password*	Password for proxy server function. See "Default Username" description above.
Re-Registration Time	10–65535 seconds	This is the timeout interval for registration of the MultiVOIP with a SIP proxy server. The time interval begins the moment the MultiVOIP gateway registers with the SIP proxy server and ends at the time specified by the user in the Re-Registration Time field (this field). When/if registration lapses, call traffic routed to/from the MultiVOIP through the SIP proxy server will cease. However, calls in progress will continue to function until they end.
Proxy Polling Interval	60 - 300	The interval between the VOIP gateway's successive attempts to connect to and be governed by a higher level SIP proxy server. The Primary Proxy is the highest level gatekeeper. Alternate Proxy 1 is second; Alternate Proxy 2 is the lowest order SIP proxy server.
TTL Value	SIP proxy "Time to Live" value. (*in seconds*)	As soon as a MultiVOIP gateway registers with a SIP proxy server (allowing the proxy server to control its call traffic) a countdown timer begins. The TTL Value is the interval of the countdown timer. Before the TTL countdown expires, the MultiVOIP gateway needs to register with the gatekeeper in order to maintain the connection. If the MultiVOIP does not register before the TTL interval expires, the MultiVOIP gateway's registration with the proxy server will expire and the proxy server will no longer permit call traffic to or from that gateway. Calls in progress will continue to function even if the gateway becomes de-registered.

SPP

Single Port Protocol was developed by Multi-Tech to allow for dynamic IP addressing when it is set to Registrar/Client mode. The other choice, Direct mode, has IP addresses assigned to the gateways. The table below describes all fields in the general SPP Call Signaling screen.



SPP call signaling

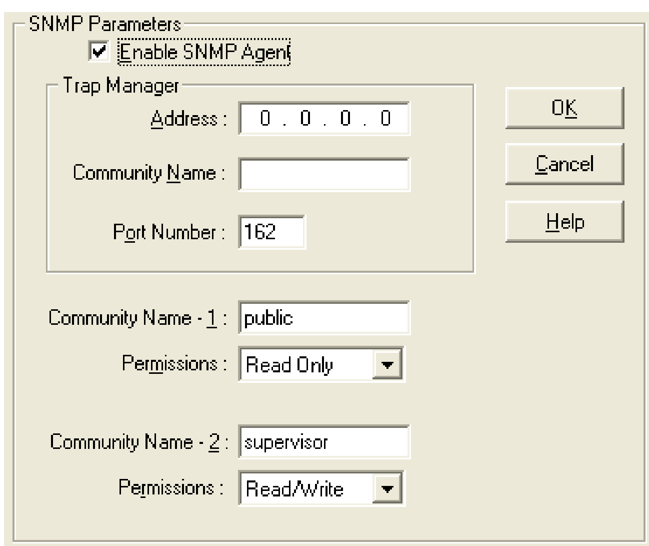
Field Name	Values	Description
Mode	Direct, Client, or Registrar	In **direct mode**, all VOIP gateways have static IP addresses assigned to them. In **registrar/client mode**, one VOIP gateway serves as registrar and all other gateways, being its clients, point to that registrar. The registrar assigns IP addresses dynamically.
General Options		
Port	*port*	The UDP port on which data transmission will occur. Each client VOIP has its own port. If two client VOIPs are both behind the same firewall, then they must have different ports assigned to them. If there are two clients and each is behind a different firewall, then the clients could have different port numbers or the same port number. Default port number = 10000.
Re-transmission	50 - 5000ms	If packets are lost (as indicated by absence of an acknowledgment) then the endpoint will retransmit the lost packets after this designated time duration has elapsed. Default = 2000 milliseconds.
Max Re-transmission	0 - 20	Number of times the VOIP will re-transmit a lost packet if no acknowledgment has been received. Default = 3.
Client Options		Client Option fields are active only in registrar/client mode and only for client VOIP units.

Field Name	Values	Description
Primary Registrar	--	This is the preferred SPP registrar gateway for controlling the traffic of the current VOIP.
Alternate Registrar 1 and 2	--	A first and a second alternate SPP Registrar gateway can be specified for use by the current VOIP for situations where the Primary Registrar gateway is busy or otherwise unavailable.
Registrar IP Address	*n.n.n.n*	This is the IP address of the registrar VOIP to which this client is assigned. Default value = 0.0.0.0; effectively, there is no useful default value.
Registrar Port	10000 or other	This is the port number of the registrar VOIP to which this client is assigned. Default port number = 10000.
Polling Interval	integer 60 - 300	The interval between the VOIP gateway's successive attempts to connect to and be governed by a higher level SPP registrar gateway. The Primary Registrar is the highest level registrar gateway. Alternate Registrar 1 is second; Alternate Registrar 2 is the lowest order SPP registrar gateway.
Registrar Options		Registrar Option fields are active only in registrar/client mode and only for registrar VOIP units.
Keep Alive	30 – 300 (seconds)	Time-out duration before a registrar wills un-register a client that does not send its "I'm here" signal. Client normally sends its "I'm here" signal every 20 seconds. Timeout default = 60 seconds.
Proxy/NAT Device Parameters		
Behind Proxy/NAT device	Y/N	Enables MultiVOIP (running in SPP Registrar mode) to operate 'behind' a proxy/NAT device (NAT = Network Address Translation).
Proxy/NAT Device Parameters – Public IP Address	*n.n.n.n*	The public IP address of the proxy/NAT device which the MultiVOIP is behind.

SNMP

If you intend to manage your MultiVOIP remotely using the MultiVOIP Manager software, you will need to set the Simple Network Management Protocol parameters. To make the MultiVOIP controllable by a remote PC running the MultiVOIP Manager software, check the "Enable SNMP Agent" box on the **SNMP Parameters** screen.

The MVP130 MultiVOIPs only have limited SNMP functionality available. Contact Multi-Tech Support for help if you want to use this.

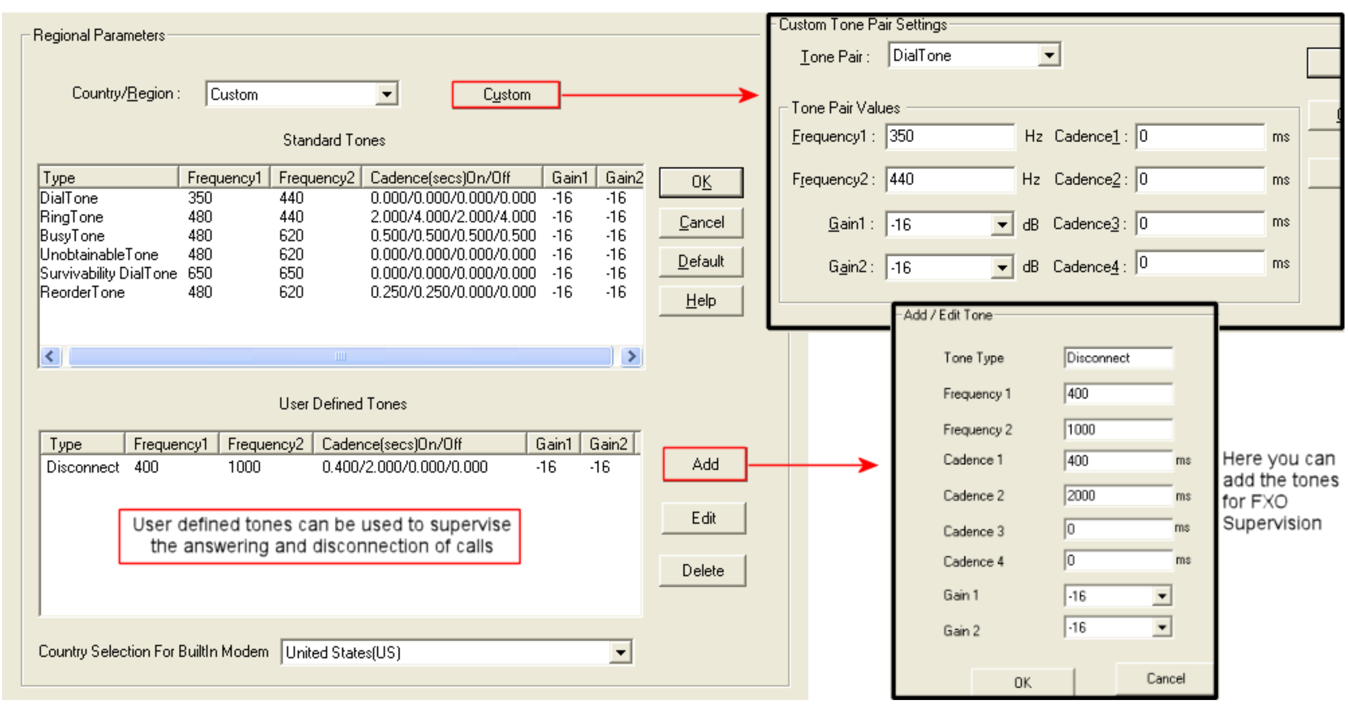


SNMP parameters screen

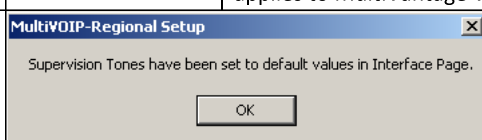
Field Name	Values	Description
Enable SNMP Agent	Y/N	Enables the SNMP code in the firmware of the MultiVOIP. This must be enabled for the MultiVOIP to communicate with and be controllable by the MultiVOIP Manager . Default: disabled
Trap Manager Parameters		
Address	*n.n.n.n*	MultiVOIP Manager computer IP address.
Community Name	--	A community is a group of VOIP endpoints that can communicate with each other. Public is used to designate a grouping where all end users have access to entire VOIP network. Calling permissions can be configured to restrict access as needed.
Port Number	162	The default port number of the SNMP manager receiving the traps is the standard port 162.
Community Name 1	Length = 19 characters (max.) Case sensitive.	First community grouping.
Permissions	Read-Only, Read/Write	If this community needs to change MultiVOIP settings, select Read/Write. Otherwise, select Read-Only to view settings.
Community Name 2	Length = 19 characters (max.) Case sensitive.	Second community grouping
Permissions	Read-Only, Read/Write	If this community needs to change MultiVOIP settings, select Read/Write. Otherwise, select Read-Only to view settings.

Regional

The Regional Parameters are used to set the phone signaling tones and cadences. 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), ring tone, and other, more specialized tones. If you need settings that are not available, the Custom selection will let you set the tones to what is necessary.



Regional parameters

Field Name	Values	Description
Country/Region	USA, Japan, UK, Custom	Name of a country or region that uses a certain set of tone pairs for **dial tone, ring tone, busy tone**, **unobtainable tone** (fast busy tone), **survivability tone** (tone heard briefly, 2 seconds, after going off hook denoting survivable mode of VOIP unit), **re-order tone** (a tone pattern indicating the need for the user to hang up the phone), and **intercept tone** (a tone that warns an a party that has gone off hook but has not begun dialing, within a prescribed time, that an automatic emergency or attendant number will be called; the automatic call can be used to direct an attendant's attention to a disabled or distressed caller, allowing an appropriate response to be made). 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. **Note 1:** **Intercept tone** is applicable only when the FXS telephony interface has been chosen in the **Interface** screen and when the AutoCall / OffHook Alert field is set to OffHook Alert in the **Voice/Fax Parameters** screen. The time allowed for dialing before the automatic calling process begins is set in the OffHook Alert Timer field of the **Voice/Fax Parameters** screen. **Note 2:** "Survivability" tone indicates a special type of call-routing redundancy & applies to MultiVantage VOIP units only
Advisory screen		This message screen appears whenever the Country field is changed. It informs the operator that, upon change of the Country field value, all User Defined Tones will be deleted.
Standard Tones fields		
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 or FXS port. Default: -16dB
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 or FXO port. Default: -16dB
Cadence (ms) On/Off	n/n/n/n four integer time values in milliseconds; 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 Custom to bring up **Custom Tone Pair Settings**. This screen allows the user to specify tone pair attributes that are not found in any of the standard national/regional telephony toning schemes.

Field Name	Values	Description
Country Selection for Built-In Modem (*not applicable to MVP130 or 130-FXS)*	country name	MultiVOIP units operating with the X.06 software release (and above) include a built-in modem. The administrator can dial into this modem to configure the MultiVOIP unit remotely. The country name values in this field set telephony parameters that allow the modem to work in the listed country. This value may be different than the Country/Region value. For example, a user may need to choose "Europe" as the Country/Region value but "Denmark" as the Country-Selection-for-Built-In-Modem value.
User Defined Tones fields		
Type column	*alphanumeric name*	Name of supervisory tone pair. Cannot be same as name of any standard tone pair.
Frequency 1	Freq. in Hertz	Lower frequency of pair.
Frequency 2	Freq. in Hertz	Higher frequency of pair.
Gain 1	+3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This applies to any supervisory tones that the MultiVOIP outputs as audio to the FXS or FXS port. **Default: "Mute"**
Gain 2	+3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This applies to any supervisory tones that the MultiVOIP outputs as audio to the FXS or FXO port. **Default: "Mute"**
Cadence (ms) On/Off	n/n/n/n four integer time values in milliseconds; (zero value indicates continuous tone)	On/off pattern of tone durations used to denote supervisory tones specified by user. Supervisory tones relate to answering and disconnection of calls. 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.

Setting 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 or "re-order" tones or "survivability" tones for your system. 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.

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.
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	+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	+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 milliseconds; zero value	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

Field Name	Values	Description
	for dial-tone indicates continuous tone	duration of first period of tone being "on" in the cadence of the telephony signal.
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.

SMTP

Setting the SMTP Parameters (Log Reports by Email). The SMTP Parameters screen is applicable when the VOIP administrator has chosen to receive log reports by email (this is done by selecting the "SMTP" checkbox in the Others screen and selecting "Enable SMTP" in the SMTP Parameters screen.)

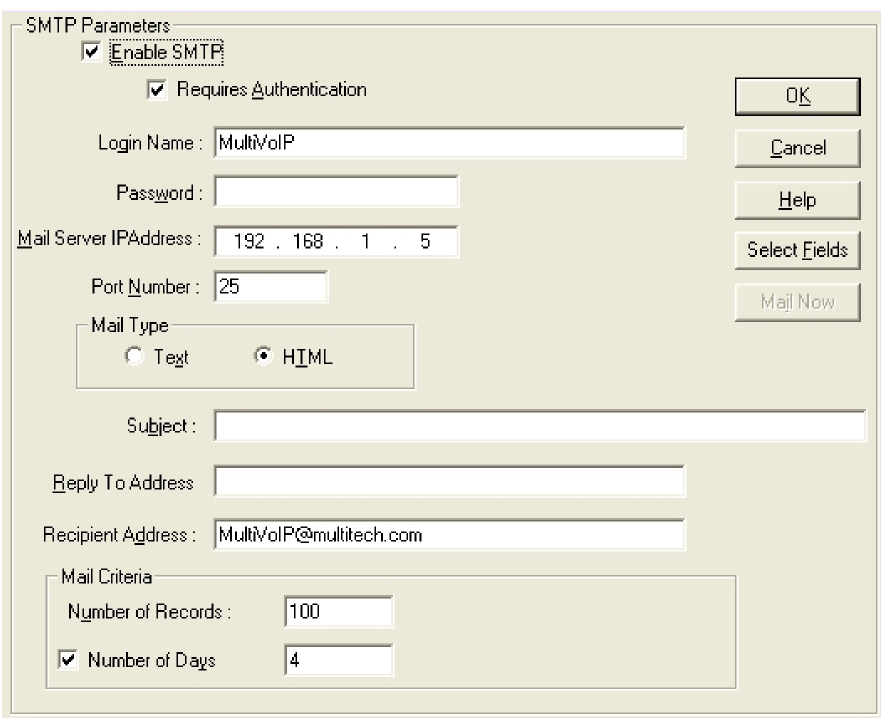
Email Address for VOIP (for email call log reporting)

This is needed only if log reports of VOIP call traffic are to be sent by email.

Ask Mail Server administrator to set up email account (with password) for the MultiVOIP unit itself. Be sure to give a unique identifier to each individual MultiVOIP unit. Get the IP address of the mail server computer, as well.

MultiVOIP as Email Sender. When SMTP is used, the MultiVOIP has its own email account (with Login Name and Password) on some mail server connected to the IP network. Using this account, the MultiVOIP sends email messages containing log report information. The "Recipient" of the log report email is ordinarily the VOIP administrator. Because the MultiVOIP cannot receive email, you must set up a "Reply-To" address. Ordinarily, the "Reply-To" address belongs to a technician who has access to the mail server or MultiVOIP or both, and the VOIP administrator might also be designated as the "Reply-To" party. The Reply-To address receives error or failure messages regarding the emailed reports.

The **SMTP Parameters** screen is shown below:



SMTP Parameters

Field Name	Values	Description
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select “SMTP” in the **Logs** screen.
Requires Authentication	Y/N	If this checkbox is checked, the MultiVOIP will send Authentication information to the SMTP server. The authentication information indicates whether or not the email sender has permission to use the SMTP server.
Login Name	*alpha-numeric*	This is the User Name for the MultiVOIP unit’s email account.
Password	*alpha-numeric*	Login password for MultiVOIP unit’s email account.
Mail Server IP Address	*n.n.n.n*	This is the mail server’s IP address. This mail server must be accessible on the IP network to which the MultiVOIP is connected.
Port Number	25	25 is a standard port number for SMTP.
Mail Type	text or html	Mail type in which log reports will be sent.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP unit.
Reply-To Address	*email address*	User specified. This email address functions as a source email identifier for the MultiVOIP, which, of course, cannot usefully receive email messages. The Reply-To address provides a destination for returned messages indicating the status of messages sent by the MultiVOIP (esp. to indicate when log report email was undeliverable or when an error has occurred).
Recipient Address	*email address*	Email address where VOIP administrator receives log reports.
Mail Criteria		Criteria for sending log summary by email. The log summary email is sent when the user-specified number of log messages has accumulated, or once every day or multiple days, whichever comes first.
Number of Records	integer	The number of log records that must accumulate to trigger the sending of a log-summary email.
Number of Days	integer	The number of days that must pass before triggering the sending of a log-summary email.

The **SMTP Parameters** dialog box has a secondary dialog box, accessed by the *Select Fields* button, which allows you to customize email logging. The MultiVOIP software logs data about many aspects of the call traffic going through the MultiVOIP. The Custom Fields screen lets you pick which aspects will be included in the email log reports.

Custom Field Definitions

Field	Description
Select All	Log report to include all fields shown.
Channel Number	Data channel carrying call.
Duration	Length of call.
Packets Sent	Total packets sent in call.
Bytes Sent	Total bytes sent in call.
Packets Lost	Packets lost in call.
Outbound Digits	The DTMF dialing digits received by this

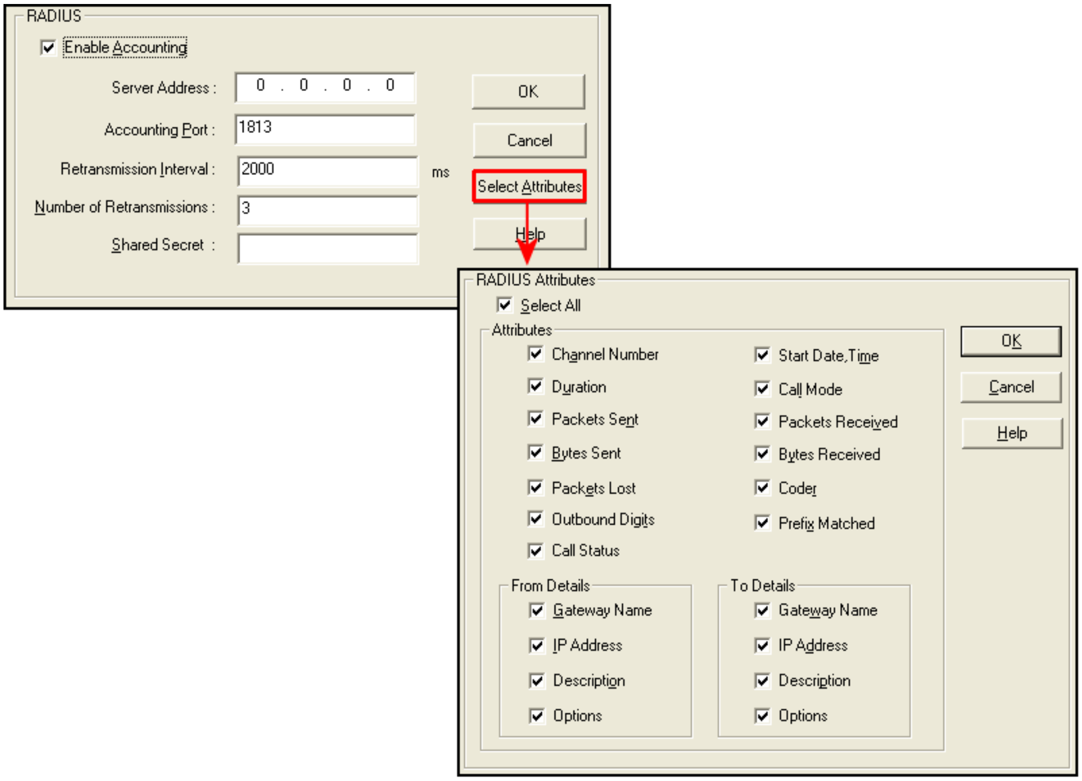
Field	Description
Start Date, Time	Date and time the phone call began.
Call Mode	Voice or fax.
Packets Received	Total packets received in call.
Bytes Received	Total bytes received in call.
Coder	Voice Coder /Compression Rate used for call will be listed in log.
Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Type	Indicates the Call Signaling protocol used

Received	gateway from the remote gateway presuming that DTMF is set to "Out of Band."
Call Status	Successful or unsuccessful.
Call Direction	Indicates call’s originating party.
Server Details	The IP address of the traffic control server (if any) being used (whether an H.323 gatekeeper, a SIP proxy, or an SPP registrar gateway) will be displayed here if the call is handled through that server.
Disconnect Reason	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause occasioned disconnection (for example, a technical error or failure). Values are "Normal" and "Local" disconnection.
From Details	
Gateway Number	Originating gateway
IP Address	IP address where call originated.
Descript	Identifier of site where call originated.
Options	When selected, log will not Silence Compression and Forward Error Correction by call originator.

	for the call (H.323, SIP, or SPP).
DTMF Capability	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different VOIP protocols. For H.323, this field can display "Out of Band" or "Inband". For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition. For SPP it can display "Out of Band RFC2833" or "Inband".
Outbound Digits Sent	The dialing digits sent by this gateway to the remote gateway presuming that DTMF is set to "Out of Band.”
To Details	
Gateway Name	Completing or answering gateway
IP Address	IP address where call was completed or answered.
Descript	Identifier of site where call was completed or answered.
Options	When selected, log will not use Silence Compression and Forward Error Correction by party answering call.

RADIUS

In general, RADIUS is concerned with authentication, authorization, and accounting. The MultiVOIP supports the accounting and authentication functions. The accounting function is well suited for billing of VOIP telephony services. In the *Select Attributes* secondary screen (accessed by clicking on Select Attributes button), the VOIP administrator can select the parameters to be tallied by the RADIUS server.



RADIUS settings

The fields of the RADIUS screen are described in the table below.

Field Name	Values	Description
Enable Accounting	Y/N	When checked, the MultiVOIP will access the accounting functionality of the RADIUS server.
Server Address	*n.n.n.n*	IP address of the RADIUS server that handles accounting (billing) for the current MultiVOIP unit.
Accounting Port	1 - 65535	TDM time slot at which RADIUS accounting information will be transmitted and received.
Retransmission Interval		If the MultiVOIP sends out a packet to the RADIUS server and doesn't receive a response in the retransmit interval, it will retransmit that packet again and wait the retransmit interval again for a response. How many times it does this is determined by the setting in the **Number of Retransmissions** field.
Number of Retransmissions	0 - 255	
Shared Secret	alpha-numeric	Client encryption key for the current VOIP unit.
Select Attributes (button)	--	Gives access to RADIUS Attributes screen. On Attributes screen, one can specify the parameters to be tallied by the RADIUS server for accounting (usually billing) purposes.

The RADIUS dialog box has a secondary dialog box, **RADIUS Attributes**, which allows you to customize accounting information sent to the RADIUS server by the MultiVOIP. The MultiVOIP software logs data about many aspects of the call traffic going through the MultiVOIP. The RADIUS Attributes screen lets you pick which aspects will be included in the accounting reports sent to the RADIUS server.

Field	Description	Field	Description
Select All	Log report to include all fields shown.	Start Date, Time	Date and time the phone call began.
Channel Number	Data channel carrying call.	Call Mode	Voice or fax.
Duration	Length of call.	Packets Received	Total packets received in call.
Packets Sent	Total packets sent in call.	Bytes Received	Total bytes received in call.
Bytes Sent	Total bytes sent in call.	Coder	Voice Coder /Compression Rate used for call will be listed in log.
Packets Lost	Packets lost in call.	Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Outbound Digits Sent	DTMF digits received by this gateway from remote gateway (if that DTMF set to "Out of Band").	Call Status	Successful or unsuccessful.
Server Details	The IP address of the traffic control server being used will be displayed here if the call is handled through that server. The Options field refers to non-mandatory server features that might be activated. For example, with H.323, various H.323 Version 4 options might be listed.		
From Details		**To Details**	
Gateway Number	Originating gateway	Gateway Name	Completing or answering gateway
IP Address	IP address where call originated.	IP Address	IP address where call was completed/answered.
Descript	Identifier of where call originated.	Descript	Identifier of where call was completed/answered.
Options	When selected, log will not use Silence Compression and Forward Error Correction by call originator.	Options	When selected, log will not use Silence Compression and Forward Error Correction by party answering call.

Logs/Traces

The Logs/Traces screen lets you choose how the VOIP administrator will receive log reports about the MultiVOIP's performance and the phone call traffic that is passing through it. Log reports can be received in one of three ways:

- in the MultiVOIP program (interface),
- through email (SMTP)
- at the MultiVOIP Manager remote VOIP system management program (SNMP).

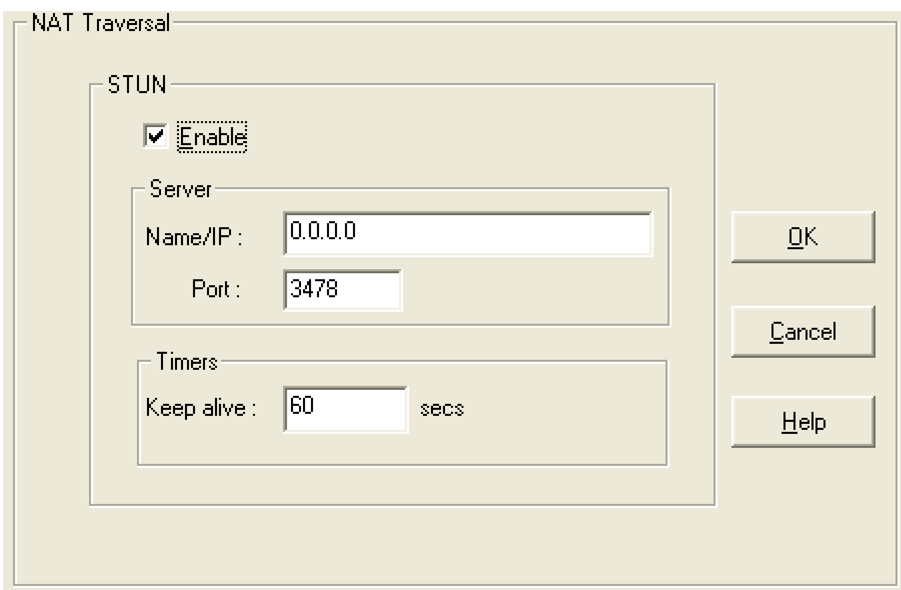
Logs and Filters screens

If you enable console messages, you can customize the types of messages to be included/excluded in log reports by clicking on the *Filters* button and using the **Console Messages Filter Settings** screen. If you use the logging function, select the logging option that applies to your VOIP system design. If you intend to use a SysLog Server program for logging, click in that Enable check box. The common SysLog logical port number is 514. If you intend to use the MultiVOIP web browser interface for configuration and control of MultiVOIP units, be aware that the web browser interface does not support logs directly. However, when the web browser interface is used, log files can still be sent to the VOIP administrator via email (which requires using the SMTP logging option).

Field Name	Values	Description
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic terminal program like HyperTerminal ™ or equivalent. Normally, this should be disabled because it uses MultiVOIP processing resources. Console messages are meant for IT support personnel.
Filters (button)		Click to access secondary screen on where console messages can be included/excluded by category and on a per-channel basis.
Turn Off Logs	Y/N	Check to disable log-reporting function.
Logs Buttons		Only one of these three log reporting methods, GUI, SMTP, or SNMP, may be chosen.
GUI	•	User must view logs at the MultiVOIP configuration program.
SNMP	•	Log messages will be delivered to the MultiVOIP Manager application program.
SMTP	•	Log messages will be sent to user-specified email address.
SysLog Server Enable	Y/N	This box must be checked if logging is to be done in conjunction with a SysLog Server program.
IP Address	*n.n.n.n*	IP address of computer, in VOIP network, on which SysLog Server program is running.
Port	514	Logical port for SysLog Server. 514 is commonly used.
Online Statistics Updation Interval	integer	Set the interval (in seconds) at which logging information will be updated.

NAT Traversal

Setting the NAT Traversal parameters. NAT (Network Address Translation) parameters are applicable only when the MultiVOIP is operating in SIP mode. STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.



Field Name	Values	Description
Enable (STUN)	Y/N	Enables STUN client functionality in the MultiVOIP. STUN (Simple Traversal of UDP through NATs (Network Address Translation)) is a protocol that allows a server to assist client gateways behind a NAT firewall or router with their packet routing.
Name/IP (Server)	*n.n.n.n*	IP address of the STUN server.
Port (Server; NAT/STUN)	*port*; default= 3478	The data port (TDM time slot) at which STUN info will be transmitted and received.
Keep Alive (Timers; NAT/STUN)	60 – 3600 (seconds)	The interval at which the STUN client sends indicator ("Keep Alive") packets to the STUN server to determine whether or not the STUN server is available.

Supplementary Services

Supplementary Services features derive from the H.450 standard, which brings to the VOIP telephony functionality once only available with PSTN or PBX telephony. Even though the H.450 standard refers only to H.323, Supplementary Services are still applicable to the SIP and SPP VOIP protocols.

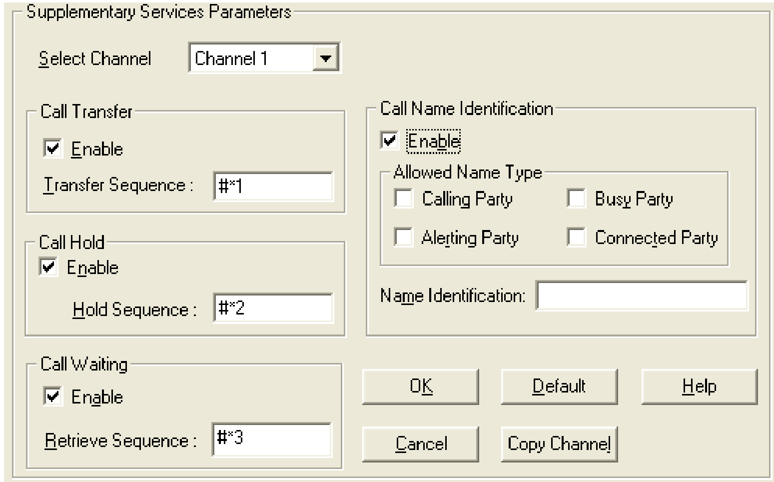
Of the features implemented under Supplementary Services, three are very closely related: Call Transfer, Call Hold, and Call Waiting. Call Name Identification is similar but not identical to the premium PSTN feature commonly known as **Caller ID**.

Call Transfer. Call Transfer allows one party to re-connect the party with whom they have been speaking to a third party. The first party is disconnected when the third party becomes connected. Feature is used by a programmable phone keypad sequence (for example, #7).

Call Hold. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function. Feature is used by a programmable phone keypad sequence (for example, #7).

Call Waiting. Call Waiting notifies an engaged caller of an incoming call and allows them to receive a call from a third party while the party with whom they have been speaking is put on hold. Feature is used by a programmable phone keypad sequence (for example, #7).

Call Name Identification. When enabled for a given VOIP unit (the 'home' VOIP), this feature gives notice to remote VOIPs involved in calls. Notification goes to the remote VOIP administrator, not to individual phone stations. When the home VOIP is the caller, a plain English descriptor will be sent to the remote VOIP identifying the channel over which the call is being originated (for example, "Calling Party - Omaha Sales Office Line 2"). If that VOIP channel is dedicated to a certain individual, the descriptor could say that, as well (for example "Calling Party - Harold Smith in Omaha"). When the home VOIP receives a call from any remote VOIP, the home VOIP sends a status message back to that caller. This message confirms that the home VOIP's phone channel is either busy or ringing or that a connection has been made (for example, "Busy Party - Omaha Sales Office Line 2"). These messages appear in the **Statistics – Call Progress** screen of the remote VOIP.



Field Name	Values	Description
Select Channel	Channel 1	The channel to be configured is selected here.
Call Transfer Enable	Y/N	Select to enable the Call Transfer function in the VOIP unit. This is a "blind" transfer and the sequence of events is as follows: Callers A and B are having a conversation. Caller A wants to put B into contact with C. Caller A dials call transfer sequence. Caller A hears dial tone and dials number for caller C. Caller A gets disconnected while Caller B gets connected to caller C. A brief musical jingle is played for the caller on hold.
Transfer Sequence	Any phone keypad character	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call transfer. The call-transfer sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). The sequences for call transfer, call hold, and call waiting can be from 1 to 4 digits in length consisting of any combination of digits 1234567890*#.
Call Hold Enable	Y/N	Select to enable Call Hold function in VOIP unit. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function.
Hold Sequence	phone keypad characters	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call hold. The call-hold sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #).
Call Waiting Enable	Y/N	Select to enable Call Waiting function in VOIP unit.
Retrieve Sequence	Phone keypad characters, two characters in length	The numbers and/or symbols that the caller must press on the phone keypad to initiate retrieval of a waiting call. The call-waiting retrieval sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). This is the phone keypad sequence that a user must press to retrieve a waiting call. Customize-able. Sequence should be distinct from sequence that might be used to retrieve a waiting call via the PBX or PSTN.
Call Name Identification Enable	Enables CNI function. Call Name Identification is not the same as Caller ID. When enabled on a given VOIP unit currently being controlled by the MultiVOIP interface (the 'home VOIP'), Call Name	

Field Name	Values	Description
	Identification sends an identifier and status information to the administrator of the remote VOIP involved in the call. The feature operates on a channel-by-channel basis (each channel can have a separate identifier). If the home VOIP is originating the call, only the Calling **Party** field is applicable. If the home VOIP is receiving the call, then the **Alerting Party**, **Busy Party**, and **Connected Party** fields are the only applicable fields (and any or all of these could be enabled for a given VOIP channel). The status information confirms back to the originator that the home VOIP, is either busy, or ringing, or that the intended call has been completed and is currently connected. The identifier and status information are made available to the remote VOIP unit and appear in the **Caller ID** field of its **Statistics – Call Progress** screen. (This is how MultiVOIP units handle CNI messages; in other VOIP brands, H.450 may be implemented differently and then the message presentation may vary.)	
Calling Party, Allowed Name Type (CNI)	If the 'home' VOIP unit is originating the call and **Calling Party** is selected, then the identifier (from the **Caller Id** field) will be sent to the remote VOIP unit being called. The **Caller Id** field gives the remote VOIP administrator a plain-language identifier of the party that is originating the call occurring on a specific channel. This field is applicable only when the 'home' VOIP unit is originating the call. **Example**. Suppose a VOIP system has offices in both Denver and Omaha. In the Omaha VOIP unit (the 'home' VOIP in this example), **Call Name Identification** has been enabled, **Calling Party** has been enabled as an **Allowed Name Type,** and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field. When channel 2 of the Omaha VOIP is used to make a call to any other VOIP phone station (for example, the Denver office), the message "Calling Party - Omaha Sales Office Voipchannel 2" will appear in the "Caller Id" field of the **Statistics - Call Progress** screen of the Denver VOIP.	
Alerting Party, Allowed Name Type (CNI)	If the 'home' VOIP unit is receiving the call and **Alerting Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote VOIP unit that the call is ringing. This field is applicable only when the 'home' VOIP unit is receiving the call. **Example**. Suppose a VOIP system has offices in both Denver and Omaha. In the Omaha VOIP unit (the 'home' VOIP unit in this example), **Call Name Identification** has been enabled, **Alerting Party** has been enabled as an **Allowed Name Type**, and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field of the **Supplementary Services** screen. When channel 2 of the Omaha VOIP receives a call from any other VOIP phone station (for example, the Denver office), the message "Alerting Party - Omaha Sales Office Voipchannel 2" will be sent back and will appear in the **Caller Id** field of the **Statistics – Call Progress** screen of the Denver VOIP. This confirms to the Denver VOIP that the phone is ringing in Omaha.	
Busy Party, Allowed Name Type (CNI)	If the 'home' VOIP unit is receiving a call directed toward an already engaged channel or phone station and **Busy Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote VOIP unit that the channel or called party is busy. This field is applicable only when the 'home' VOIP unit is receiving the call. **Example**. Suppose a VOIP system has offices in both Denver and Omaha. In the Omaha VOIP unit (the 'home' VOIP unit in this example), **Call Name Identification** has been enabled, **Busy Party** has been enabled as an **Allowed Name Type**, and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field of the **Supplementary Services** screen. When channel 2 of the Omaha VOIP is busy but still receives a call attempt from any other VOIP phone station (for example, the Denver office), the message "Busy Party - Omaha Sales Office Voipchannel 2" will be sent back and will appear in the **Caller Id** field of the **Statistics – Call Progress** screen of the Denver VOIP. This confirms to the Denver VOIP that the channel or phone station is busy in Omaha.	
Connected Party, Allowed Name Type (CNI)	If the 'home' VOIP unit is receiving a call and **Connected Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote VOIP unit that the attempted call has been completed and the connection is made. This field is applicable only when the 'home' VOIP unit is receiving the call. **Example**. Suppose a VOIP system has offices in both Denver and Omaha. In the Omaha VOIP unit (the 'home' VOIP unit in this example), **Call Name Identification** has been enabled, **Connected Party** has been enabled as an **Allowed Name Type**, and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field of the **Supplementary Services** screen.	

Field Name	Values	Description
	When channel 2 of the Omaha VOIP completes an attempted call from any other VOIP phone station (for example, the Denver office), the message "Connect Party - Omaha Sales Office Voipchannel 2" will be sent back and will appear in the **Caller Id** field of the **Statistics – Call Progress** screen of the Denver VOIP. This confirms to the Denver VOIP that the call has been completed to Omaha.	
Caller ID	This is the identifier of a specific channel of the 'home' VOIP unit. The Caller Id field typically describes a person, office, or location, for example, "Harry Smith," or "Bursar's Office," or "Barnesville Factory."	
Default	When this button is clicked, all Supplementary Service parameters are set to their default values.	
Copy Channel	Copies the Supplementary Service attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.	

Save Settings

Save & Reboot

Saving the MultiVOIP Configuration. When values have been set for all of the MultiVOIP's various operating parameters, click on **Save Setup** in the sidebar, then *Save & Reboot*.

Creating a User Default Configuration. When a "Setup" (complete grouping of parameters) is being saved, you will be prompted about designating that setup as a "User Default" setup. A User Default setup may be useful as a baseline of site-specific values to which you can easily revert. Establishing a User Default Setup is optional.

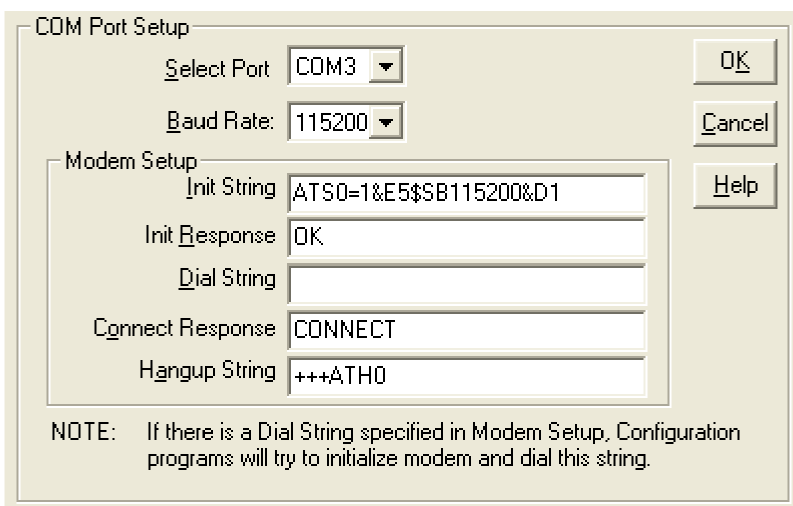
Connection

Settings

This is also accessible from the Start menu in the MultiVOIP software folder.

Set Baud Rate. The **Connection** option in the sidebar menu has a "Settings" item that includes the baud-rate setting for the COM port of the computer running the MultiVOIP software.

First, it is important to note that the default COM port established by the MultiVOIP program is COM1. *Do not accept the default value until you have checked the COM port allocation on your PC*. To do this, check for COM port assignments in the system resource manager of your Windows operating system. If COM1 is not available, you must change the COM port setting to a COM port that you have confirmed as being available on your PC.



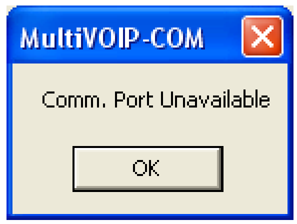
COM port setup

Troubleshooting Software Issues

In the lower left corner of the screen, the connection status of the MultiVOIP will be displayed. The messages in the lower left corner will change as detection occurs. The message “MultiVOIP Found” confirms that the MultiVOIP is in contact with the MultiVOIP configuration program. If the message displayed is “MultiVOIP Not Found!” please try the resolutions below.

Fixing a COM Port Problem

If the MultiVOIP main screen appears but is grayed out and seems inaccessible, the COM port that was specified for its communication with the PC is unavailable and must be changed. An error message will appear.

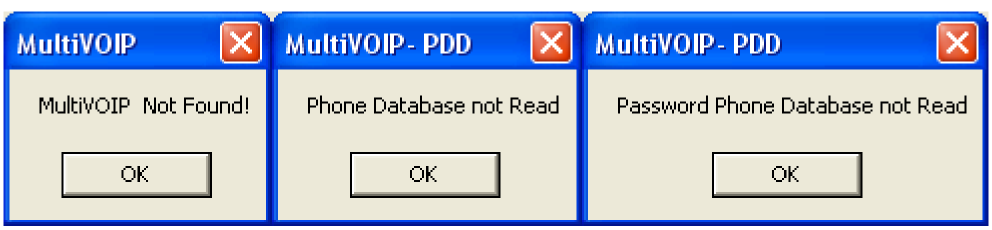


Error pop-up

To change the COM port setting, use the **COM Port Setup** dialog box, by going to the **Connection** pull-down menu and choosing “Settings” or use the left side control panel. In the “Select Port” field, select a COM port that is available on the PC (if no COM ports are currently available, re-allocate COM port resources in the computer’s MS Windows operating system to make one available).

Fixing a Cabling Problem

If the MultiVOIP cannot be located by the computer, three error messages will appear (saying “Multi-VOIP Not Found”, “Phone Database Not Read” and “Password Phone Database Not Read).



Cabling errors

In this case, the MultiVOIP is simply disconnected from the network. For instructions on MultiVOIP cable connections, see the Cabling section of Chapter 3.

Chapter 5 – Phone Book Configuration

Introduction

When a VOIP serves a PBX system, it's important that the operation of the VOIP be transparent to the telephone end user. That is, the VOIP should not entail the dialing of extra digits to reach users elsewhere on the network that the VOIP serves. On the contrary, VOIP service more commonly reduces dialed digits by allowing users (served by PBXs in facilities in distant cities) to dial their co-workers with 3-, 4-, or 5-digit extensions as if they were in the same facility.

Furthermore, the setup of the VOIP generally should allow users to make calls on a non-toll basis to any numbers accessible without toll by users at all other locations on the VOIP system. Consider, for example, a company with VOIP-equipped offices in New York, Miami, and Los Angeles, each served by its own PBX. When the VOIP phone books are set correctly, personnel in the Miami office should be able to make calls without toll not only to the company's offices in New York and Los Angeles, but also to any number that's local in those two cities.

To achieve transparency of the VOIP telephony system and to give full access to all types of non-toll calls made possible by the VOIP system, the VOIP administrator must properly configure the "Outbound" and "Inbound" phone-books of each VOIP in the system.

The "Outbound" phonebook for a particular VOIP unit describes the dialing sequences required for a call to originate locally (typically in a PBX in a particular facility) and reach any of its possible destinations at remote VOIP sites, including non-toll calls completed in the PSTN at the remote site.

The "Inbound" phonebook for a particular VOIP unit describes the dialing sequences required for a call to originate remotely from any other VOIP sites in the system, and to terminate on that particular VOIP.

Briefly stated, *the MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook describes the dialing sequences that can be used to call that MultiVOIP and how those calls will be directed.* The phone numbers are not literally "listed" individually, but are, instead, described by rule.

Identify Remote VOIP Site to Call

When you're done installing the MultiVOIP, you'll want to confirm that it is configured and operating properly. To do so, it's good to have another VOIP that you can call for testing purposes. You'll want to confirm end-to-end connectivity. You'll need IP and telephone information about that remote site.

If this is the very first VOIP in the system, you'll want to coordinate the installation of this MultiVOIP with an installation of another unit at a remote site.

Identify VOIP Protocol to be Used

Will you use H.323, SIP, or SPP? Each has advantages and disadvantages. Although it is possible to mix protocols, it is highly desirable to use the same VOIP protocol for all VOIP units in the system. SPP is a non-standard protocol developed by Multi-Tech. SPP is not compatible with the "Proprietary" protocol used in Multi-Tech's earlier generation of VOIP gateways.

Phonebook Starter Configuration

This section will walk you through the phone book setup with examples that will aid in entering the correct numbers needed to have the MultiVOIP working correctly. To do this part of the setup, you need access to another VOIP that you can call to conduct a test. It should be at a remote location, typically somewhere outside of your building. You must know the phone number and IP address for that site. We are assuming here that the MultiVOIP will operate in conjunction with a PBX.

You must configure both the Outbound Phonebook and the Inbound Phonebook. A starter configuration only means that two VOIP locations will be set up to begin the system and establish VOIP communication. Once this is accomplished, you can easily add other VOIP sites to the network.

Outbound Phonebook

1. Open the MultiVOIP program. (**Start** | **MultiVOIP *n.nn*** | **Configuration**)
2. Go to **Phone Book** | **Outbound Phonebook** | **Add Entry**.
3. On a sheet of paper, write down the calling code of the remote VOIP (area code, country code, city code, and so on) that you'll be calling.

   Follow the example that best fits your situation:

North America, Long-Distance Example	Euro, National Call Example	Euro, International Call Example
Technician in Seattle (area 206) must set up one VOIP there, another in Chicago (area 312, downtown). **Answer**: write down **312**.	Technician in central London (area 0207) to set up VOIP there, another in Birmingham (area 0121). **Answer**: write down **0121**.	Technician in Rotterdam (country 31; city 010) to set up one VOIP there, another in Bordeaux (country 33; area 05). **Answer**: write down **3305**.

Suppose you want to call a phone number outside of your building using a phone station that is an extension from your PBX system (if present). What digits must you dial? Often a "9" or "8" must be dialed to "get an outside line" through the PBX (that is, to connect to the PSTN). Generally, "1 "or "11" or "0" must be dialed as a prefix for calls outside of the calling code area (long-distance calls, national calls, or international calls).

On a sheet of paper, write down the digits you must dial before you can dial a remote area code.

North America, Long-Distance Example	Euro, National Call Example	Euro, International Call Example
Seattle/Chicago system. Seattle VOIP works with PBX that uses "8" for all VOIP calls. "1" must immediately precede area code of dialed number. **Answer**: write down **81**.	London/Birmingham system. London VOIP works with PBX that uses "9" for all out-of-building calls whether by VOIP or by PSTN. "0" must immediately precede area code of dialed number. **Answer**: write down **90**.	Rotterdam/Bordeaux system. Rotterdam VOIP works with PBX where "9" is used for all out-of-building calls. "0" must precede all international calls. **Answer**: write down **90**.

4. In the "Destination Pattern" field of the **Add/Edit Outbound Phonebook** screen, enter the digits from the previous examples.

North America, Long-Distance Example	Euro, National Call Example	Euro, International Call Example
Seattle/Chicago system. **Answer**: enter **81312** as Destination Pat-tern in Outbound Phone-book of Seattle VOIP.	London/Birmingham system. Leading zero of Birmingham area code is dropped when combined with national-dialing access code. (Such practices vary by country.) **Answer**: enter **90121** as Destination Pattern in Outbound Phonebook of London VOIP. *Not 900121.*	Rotterdam/Bordeaux system. **Answer**: enter **903305** as Destination Pattern in Outbound Phonebook of Rotterdam VOIP.

5. In the "Remove Prefix" field, enter the initial PBX access digit ("8" or "9").

North America, Long-Distance Example	Euro, National Call Example	Euro, International Call Example
Seattle/Chicago system. **Answer**: enter **8** in "Remove Prefix" field of Seattle Outbound Phonebook.	London/Birmingham system. **Answer**: enter **9** in "Remove Prefix" field of London Outbound Phonebook.	Rotterdam/Bordeaux system. **Answer**: enter **9** in "Remove Prefix" field of Outbound Phonebook for Rotterdam VOIP.

Note: Some PBXs will not 'hand off' the "8" or "9" to the VOIP. But for those PBX units that do, it's important to enter the "8" or "9" in the "Remove Prefix" field in the Outbound Phonebook. This precludes the problem of having to make two inbound phonebook entries at remote VOIPs, one to account for situations where "8" is used as the PBX access digit and another for when "9" is used.

6. In the "Protocol Type" field group, select the VOIP protocol that you will use (H.323, SIP, or SPP). Use the appropriate screen under **Configuration | Call Signaling** to configure the VOIP protocol in detail.
7. Click **OK** to exit from the **Add/Edit Outbound Phonebook** screen.

Inbound Phonebook

1. Open the MultiVOIP program. (**Start** | **MultiVOIP *n.nn*** | **Configuration**)
2. Go to **Phone Book** | **Inbound Phonebook** | **Add Entry**.
3. In the "Remove Prefix" field, enter your local calling code (area code, country code, city code, and so on) preceded by any other "access digits" that are required to reach your local site from the remote VOIP location (think of it as though the call were being made through the PSTN – even though it will not be).

North America, Long-Distance Example	Euro, National Call Example	Euro, International Call Example
Seattle/Chicago system. Seattle is area 206. Chicago employees must dial 81 before dialing any Seattle number on the VOIP system. Answer: **1206** is prefix to be removed by local (Seattle) VOIP.	London/Birmingham system. Inner London is 0207 area. Birmingham employees must dial 9 before dialing any London number on the VOIP system. Answer: **0207** is prefix to be removed by local (London) VOIP.	Rotterdam/Bordeaux system. Rotterdam is country code 31, city code 010. Bordeaux employees must dial 903110 before dialing any Rotterdam number on the VOIP system. Answer: **03110** is prefix to be removed by local (Rotterdam) VOIP.

4. In the “Add Prefix” field, enter any digits that must be dialed from your local VOIP to gain access to the PSTN.

North America, Long-Distance Example	Euro, National Call Example	Euro, International Call Example
Seattle/Chicago system. On Seattle PBX, “9” is used to get an outside line. Answer: **9** is prefix to be added by local (Seattle) VOIP.	London/Birmingham system. On London PBX, “9” is used to get an outside line. Answer: **9** is prefix to be added by local (London) VOIP.	Rotterdam/Bordeaux system. On Rotterdam PBX, “9” is used to get an outside line. Answer: **9** is prefix to be added by local (Rotterdam) VOIP.

5. In the “Channel Number” field, enter “Hunting.” A “hunting” value means the VOIP unit will assign the call to the first available channel. If desired, specific channels can be assigned to specific incoming calls (that is, to any set of calls received with a particular incoming dialing pattern).
6. In the “Description” field, it is useful to describe the ultimate destination of the calls. For example, in a New York City VOIP system, “incoming calls to Manhattan office,” might describe a phonebook entry, as might the descriptor “incoming calls to NYC local calling area.” The description should make the routing of calls easy to understand. For this, 40 characters are the maximum.

North America, Long-Distance Example	Euro, National Call Example	Euro, International Call Example
Seattle/Chicago system. Possible Description: Free Seattle access, all employees	London/Birmingham system. Possible Description: Local-rate London access, all employees	Rotterdam/Bordeaux system. Possible Description: Local-rate Rotterdam access, all employees

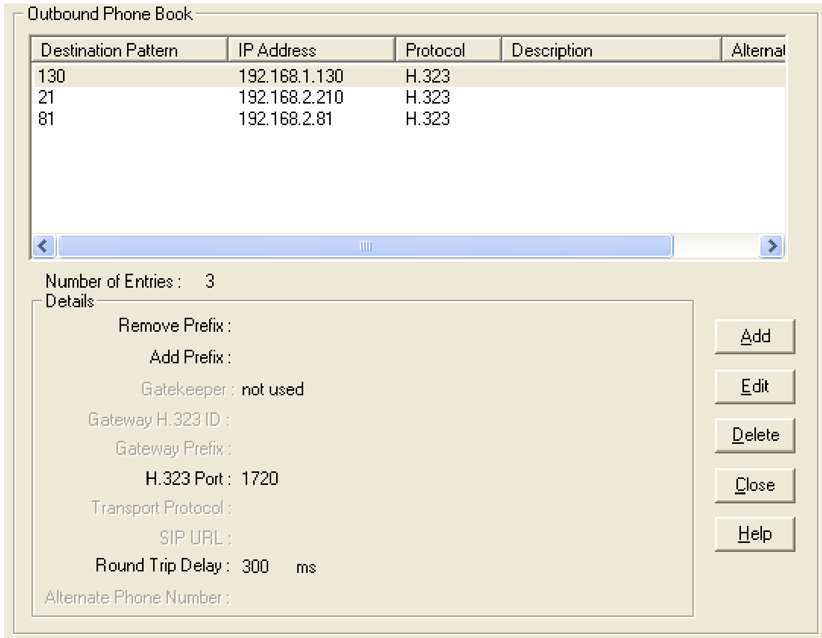
7. Repeat steps 2-6 for each inbound phonebook entry. When all entries are complete, go to step 8.
8. Click **OK** to exit the inbound phonebook screen.
9. Click on **Save Setup**. Highlight **Save and Reboot**. Click **OK**.

Your starter inbound phonebook configuration is complete.

Phone Book Descriptions

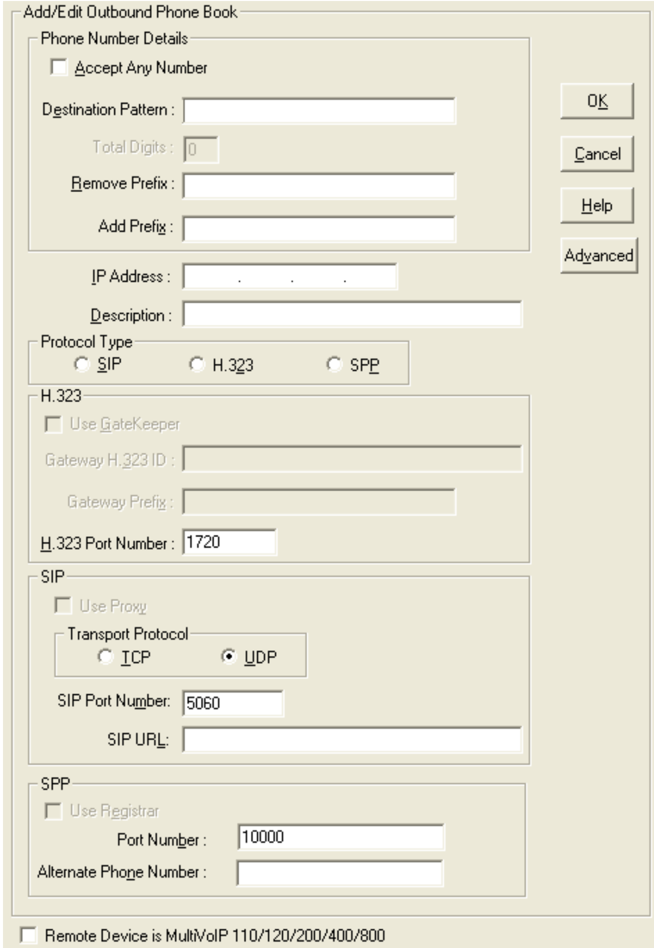
Outbound Phone Book/List Entries

Fields in the "Details" section will differ depending on the protocol (H.323, SIP, or SPP) of the selected list entry to which the details pertain.



Outbound Phone Book

Add/Edit Outbound Phone Book



Add/Edit screen

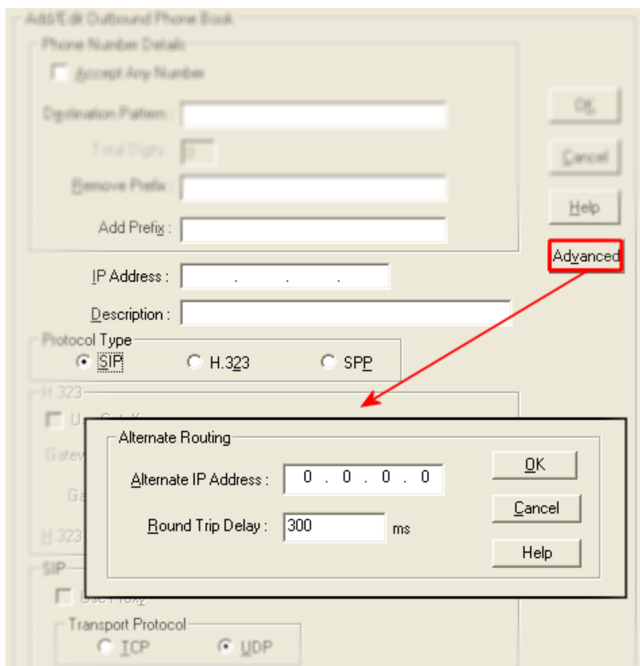
Enter Outbound Phone Book data for your MultiVOIP unit. Note that the Advanced button gives access to the Alternate IP Routing feature, if needed. Alternate IP Routing can be implemented in a secondary screen (as described after the primary screen field definitions below).

Field Name	Values	Description
Accept Any Number	Y/N	When checked, “Any Number” appears as the value in the Destination Pattern field. How **Any Number** works depends on whether or not an external routing device is used. **When no external routing device is used**. If **Any Number** is selected, calls to phone numbers that don’t match a listed Destination Pattern are directed to the **IP Address** in the Add/Edit Outbound Phone Book screen. **Any Number** can be used in addition to one or more Destination Patterns. **When external routing device is used**. If Any Number is selected, calls to phone numbers that don’t match a listed Destination Pattern are directed to the external routing device used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). The IP Address of the external routing device must be set in the Phone Book Configuration screen.
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PSTN and carried on Internet or other IP network.

Field Name	Values	Description
Total Digits	as needed	Number of digits the phone user must dial to reach specified destination. *This field not used in North America*
Remove Prefix	dialed digits	Portion of dialed number to be removed before completing call to destination.
Add Prefix	dialed digits	Digits to be added before completing call to destination.
IP Address	*n.n.n.n*	The IP address to which the call will be directed if it begins with the destination pattern given.
Description	alpha-numeric	Describes the facility or geographical location at which the call will be completed.
Protocol Type	SIP or H.323 or SPP	Indicates protocol to be used in outbound transmission. Single Port Protocol (SPP) is a non-standard protocol designed by Multi-Tech.
H.323 fields		
Use Gatekeeper	Y/N	Indicates whether or not gatekeeper is used.
Gateway H.323 ID	alpha-numeric	The H.323 ID assigned to the destination MultiVOIP. Only valid if “Use Gatekeeper” is enabled for this entry.
Gateway Prefix	numeric	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.
H.323 Port Number	1720	This parameter pertains to Q.931, which is the H.323 call signaling protocol for setup and termination of calls (aka ITU-T Recommendation I.451). H.323 employs only one “well-known” port (1720) for Q.931 signaling. If Q.931 message-oriented signaling protocol is used, 1720 must be chosen as the H.323 Port Number.
SIP Fields		
Use Proxy	Y/N	Select if proxy server is used.
Transport Protocol	TCP or UDP	VOIP administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other *See RFC 3087 (“Control of Service Context using SIP Request-URI,” by the Network Working Group).	The SIP Port Number is a UDP logical port number. The VOIP will “listen” for SIP messages at this logical port. If SIP is used, 5060 is the default, standard or “well known” port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	*sip.userphone@hostserver*, where “userphone” is the telephone number and “hostserver” is the domain name or an address on the network	Looking similar to an email address, a **SIP URL** identifies a user's address. In SIP communications, each caller or recipient is identified by a SIP URL: sip:user_name@host_name. The format of a sip URL is very similar to an email address, except that the “sip:“ prefix is used.
SPP Fields		
Use Registrar	Y/N	Select this checkbox to use registrar when VOIP system is operating in the “Registrar/Client” SPP mode. In this mode, one VOIP (the registrar, as set in Phonebook Configuration screen) has a static IP address and all other VOIPs (clients) point to the registrar’s IP address as functionally their own. However, if your VOIP system overall is operating in “Registrar/Client” mode but you want to make an exception and use Direct mode for the destination pattern of this particular Add/Edit Phonebook entry, leave this checkbox unselected. Also do not select this if your overall VOIP system is operating in the Direct SPP mode – in this mode all VOIPs are peers with unique static IP addresses.

Field Name	Values	Description
Port Number	numeric	When operating in “Registrar/Client” mode, this is the port by which the gateway receives all SPP data and control messages from the registrar gateway. (This ability to receive all data and messages via one port allows the VOIP to operate behind a firewall with only one port open.) When operating in “Direct” mode, this is the Port by which peer VOIPs receive data and messages.
Alternate Phone Number	numeric	Phone number associated with alternate IP routing.
Remote Device is [legacy VOIP]	Y/N	When checked, this MultiVOIP can operate with ‘first-generation’ MultiVOIP units in the same IP network. These include MVP-110/120/200/400/800.
Advanced button	Gives access to secondary screen where an Alternate IP Route can be specified for backup or redundancy of signal paths. For SIP & H.323 operation only.	

Click on **Advanced** to bring up the **Alternate Routing** screen. This provides an alternate path for calls if the primary IP network cannot carry the traffic. Often in cases of failure, call traffic is temporarily diverted into the PSTN. However, this feature could also be used to divert traffic to a redundant (backup) unit in case one VOIP unit fails. The user must specify the IP address of the alternate route for each destination pattern entry in the Outbound Phonebook.



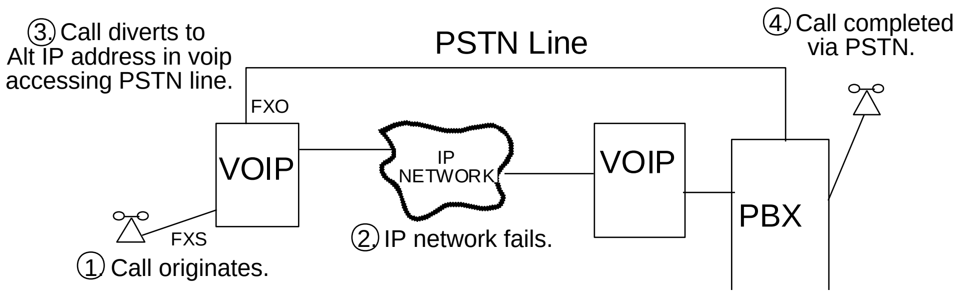
Advanced button

Alternate Routing Field Definitions

Field Name	Values	Description
Alternate IP Address	*n.n.n.n*	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	Default is 300 milliseconds	The Round Trip Delay is the criterion for judging when a data pathway is considered blocked. When the delay exceeds the threshold specified here, the data stream will be diverted to the alternate destination specified as the Alternate IP Address.

The Alternate Routing function facilitates PSTN Failover protection, that is, it allows you to re-route VOIP calls automatically over the PSTN if the VOIP system fails. The MultiVOIP can be programmed to respond to excessive delays in the transmission of voice packets, which the MultiVOIP interprets as a failure of the IP network. Upon detecting an excessive delay in transmission of voice packets (overly high “latency” in the network) the MultiVOIP diverts the call to another IP address, which itself is connected to the PSTN (for example, via an FXO port on the self-same MultiVOIP could be connected to the PSTN).

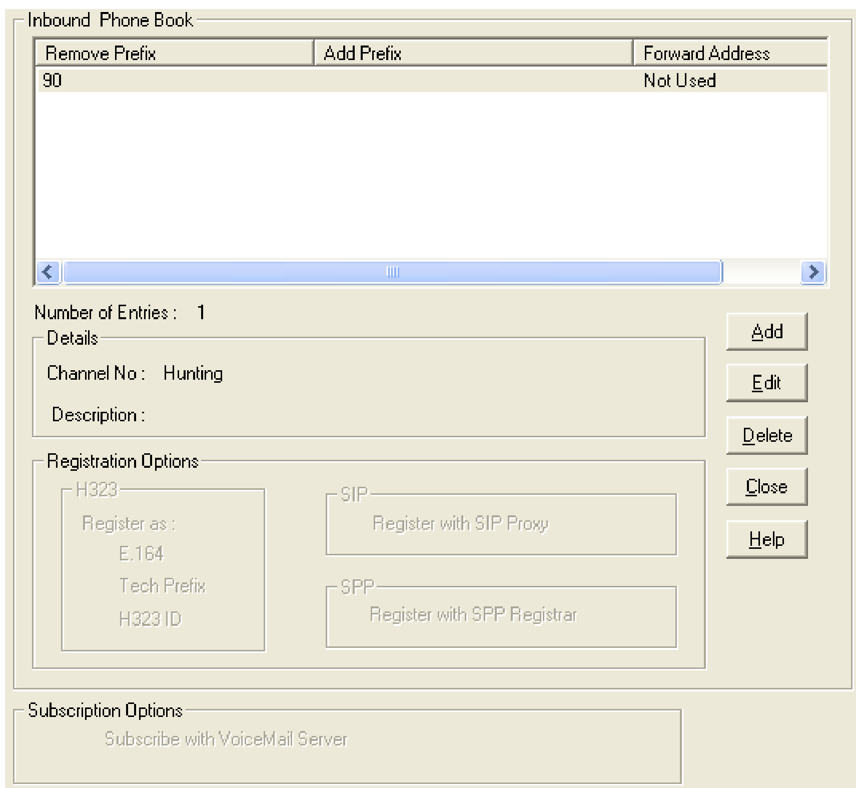
PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails. See Figure 5-4 below for example.



PSTN failover

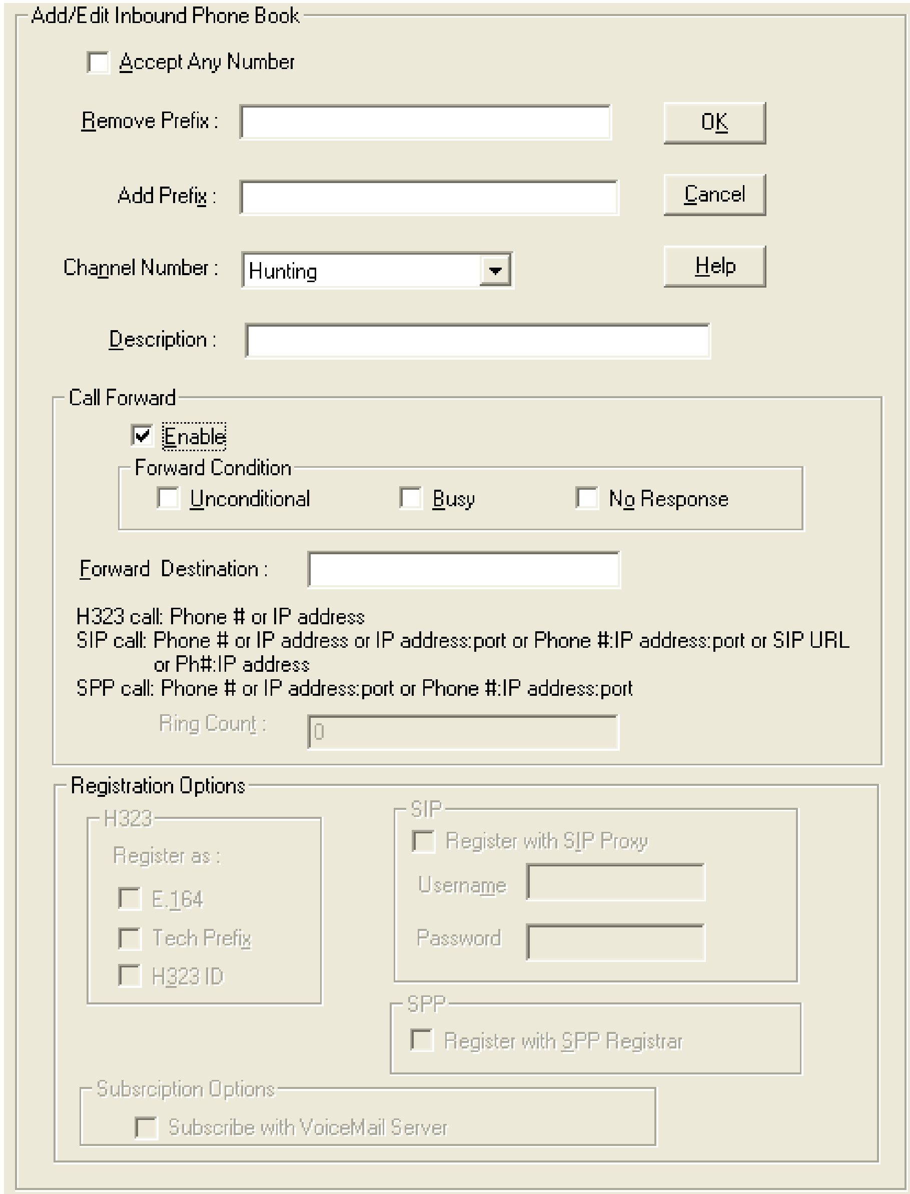
Inbound Phone Book/List Entries

The “Details” and “Registration Options” sections will display information based on the setup and protocols chosen. The “Subscription Options” area is used in conjunction with a Voice Mail Server.



Inbound phonebook entries

Add/Edit Inbound Phone Book



Add/Edit Inbound Phone Book

Field Name	Values	Description
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Remove Prefix field. The **Any Number** feature of the Inbound Phone Book does not work when an external routing device is used (Gatekeeper for H.323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the VOIP on the channel listed in the **Channel Number** field. "Any Number" can be used in addition to one or more Prefixes.
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)

Field Name	Values	Description
Channel Number	channel, or "Hunting"	Channel number to which the call will be assigned as it enters the local telephony equipment (often a local PBX). "Hunting" directs the call to any available channel.
Description	--	Describes the facility or geographical location at which the call originated.
Call Forward Parameters		
Enable	Y/N	Click the check-box to enable the call-forwarding feature.
Forward Condition	Unconditional, Busy, No Response	**Unconditional**. When selected, all calls received will be forwarded. **Busy**. When selected, calls will be forwarded when station is busy. **No Response**. When selected, calls will be forwarded if called party does not answer after a specified number of rings, as specified in **Ring Count** field. Forwarding can be conditioned on both "Busy" and "No Response
Forward Destination	IP address, phone number, port number, etc	Phone number or IP address to which calls will be directed. For **H.323** calls, the Forward Destination can be either a Phone Number or an IP Address. For **SIP** calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address, (c) IP address: port number, (d) phone number: IP address: port number, (e) SIP URL, or (f) phone #: IP address. For **SPP** calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address: port, or (c) phone number: IP address: port.
Ring Count	integer	When "No Response" is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.
Registration Option Parameters	In an **H.323** VOIP system, gateways can register with the system using one of these identifiers: an ***E.164 identifier***, a ***Tech Prefix identifier***, or an ***H.323 ID identifier***. In a **SIP** VOIP system, gateways can register with the SIP Proxy. In an **SPP** VOIP system, gateways can register with the SPP Registrar VOIP unit.	

Authorized User Name and Password for SIP

To enable the Registration Options on the Add/Edit Inbound Phone Book, activate Use SIP Proxy Option on the Call Signaling, SIP Parameters Screen. Then add the IP address for the Primary Proxy in the SIP Proxy Parameters. This allows you to add a Username and Password to the Inbound Phone Book entry.

This feature is used when the MultiVOIP registers with the proxies that support authorization and need the username, password and the endpoint name to be unique.

The VOIP sends Register request to Registrar for each entry with its configured Username and Password. When Authentication is enabled for the endpoint, then the registrar/proxy sends "401 Unauthorized/407 Proxy Authentication Required" response when it receives a REGISTER/INVITE request. Now, the endpoint has to send the authentication details in the Authorization header. In this header one of the fields is "username".

Generally proxies accept requests even if both Endpoint Name and Username are same. But some proxies expect that the Endpoint Name and Username should be different.

To support these proxies, we have the username and password configuration for every inbound phone book entry which gets registered with a proxy.

If the username and password are not configured in the inbound phone book, then the registration will happen with the default username and password that are configured in the SIP Call Signaling Page.

Phone Book Save and Reboot

When Outbound and Inbound Phonebook entries are completed, click on **Save Setup** in the sidebar menu to save your configuration. You can change your configuration at any time as needed for your system.

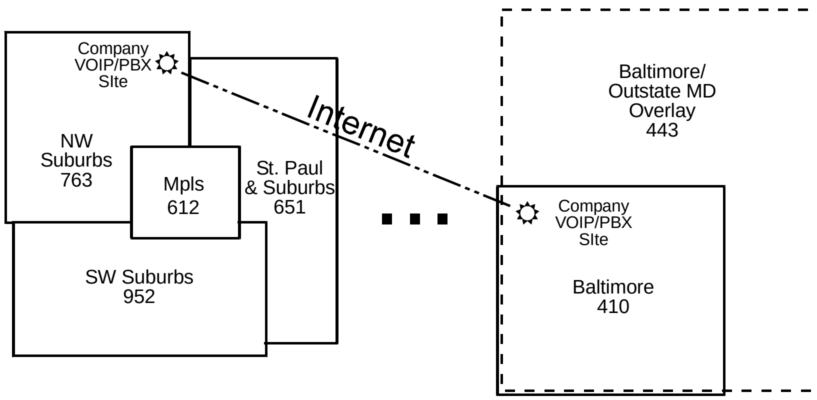
Remember that the initial MultiVOIP setup must be done locally or via the built-in Remote Configuration/Command Modem using the MultiVOIP program. After the initial configuration is complete, all of the MultiVOIP units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVOIP web interface software program or the MultiVOIP program (in conjunction with the built-in modem).

Phonebook Examples

North America

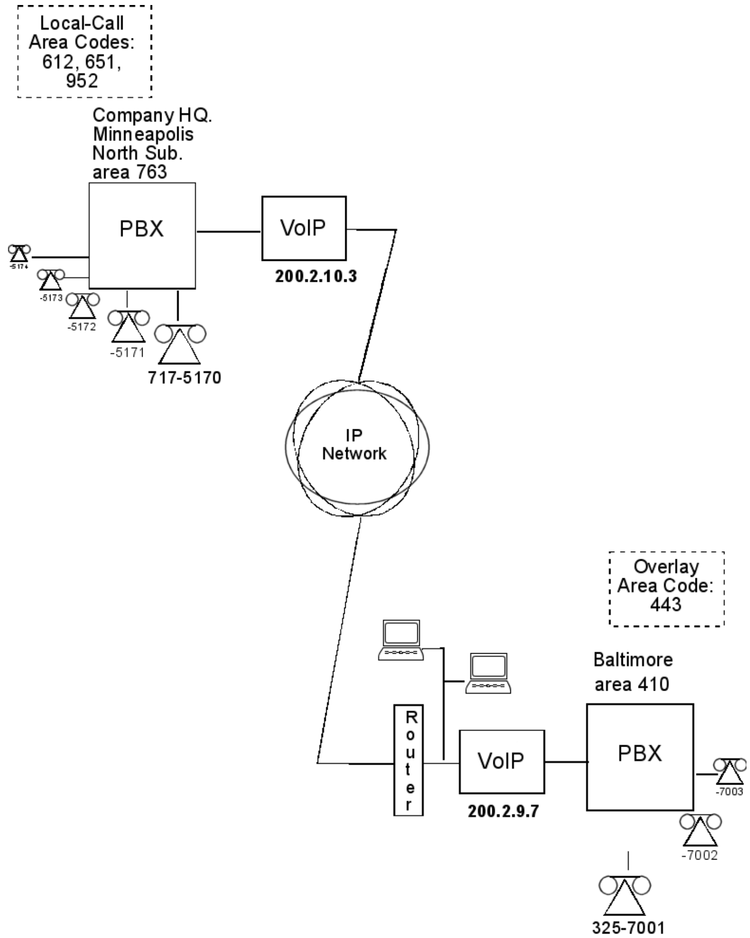
The following example demonstrates how Outbound and Inbound Phonebook entries work in a situation of multiple area codes. Consider a company with offices in Minneapolis and Baltimore.

Notice first the area code situation in those two cities: Minneapolis's local calling area consists of multiple adjacent area codes; Baltimore's local calling area consists of a base area code plus an overlay area code.



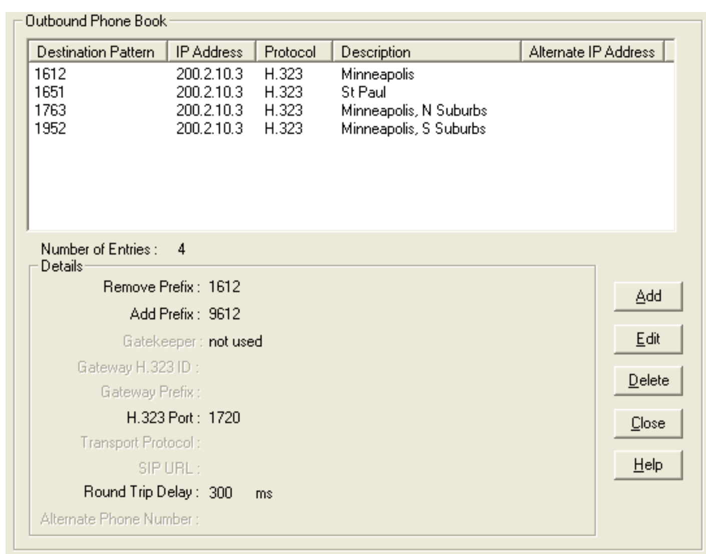
North America example

An outline of the equipment setup in both offices is shown below.



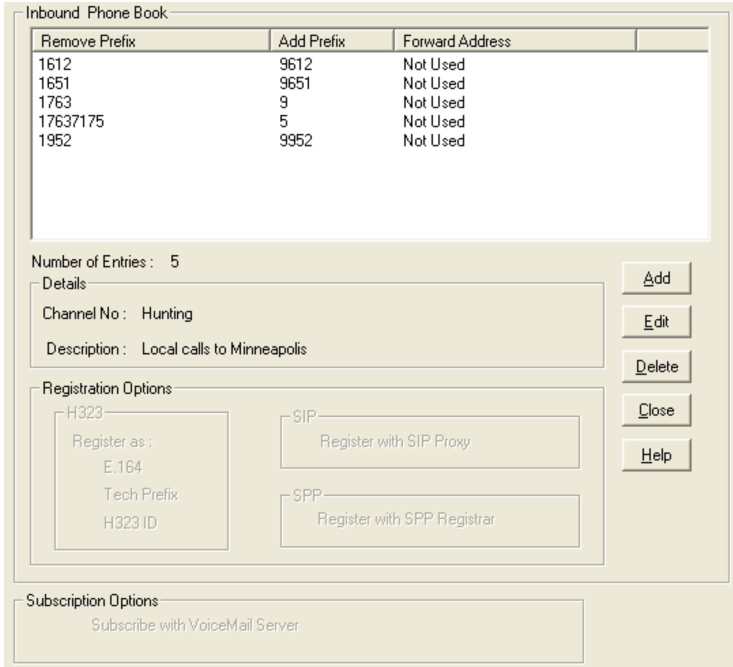
Equipment setup example

The screen below shows Outbound Phonebook entries for the VOIP located in the company's Baltimore facility.



Baltimore example

The entries in the Minneapolis VOIP’s Inbound Phonebook match the Outbound Phonebook entries of the Baltimore VOIP, as shown below.



Minneapolis example

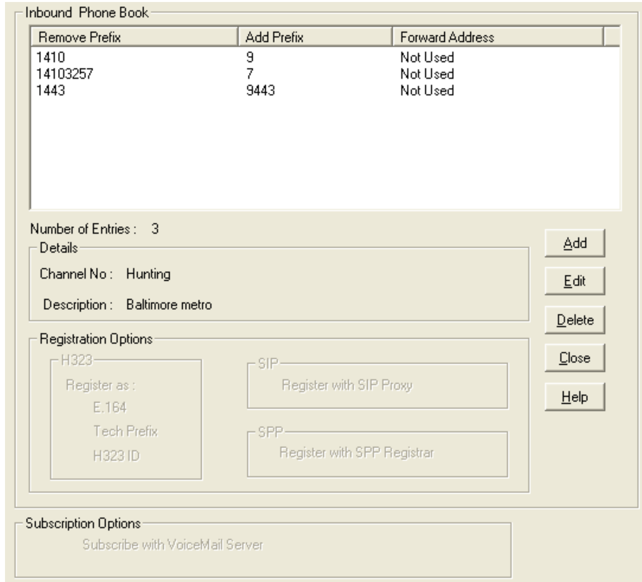
To call the Minneapolis/St. Paul area, a Baltimore employee must dial eleven digits. (In this case, we are assuming that the Baltimore PBX does not require an “8” or “9” to seize an outside phone line.)

If a Baltimore employee dials any phone number in the 612 area code, the call will automatically be handled by the company’s VOIP system. Upon receiving such a call, the Minneapolis VOIP will remove the digits “1612”. But before the suburban-Minneapolis VOIP can complete the call to the PSTN of the Minneapolis local calling area, it must dial “9” (to get an outside line from the PBX) and then a comma (which denotes a pause to get a PSTN dial tone) and then the 10-digit phone number which includes the area code (612 for the city of Minneapolis; which is different than the area code of the suburb where the PBX is actually located -- 763).

A similar sequence of events occurs when the Baltimore employee calls number in the 651 and 952 area codes because number in both of these area codes are local calls in the Minneapolis/St. Paul area.

The simplest case is a call from Baltimore to a phone within the Minneapolis/St. Paul area code where the company’s VOIP and PBX are located, namely 763. In that case, that local VOIP removes 1763 and dials 9 to direct the call to its local 7-digit PSTN.

Finally, consider the longest entry in the Minneapolis Inbound Phonebook, “17637175. Note that the main phone number of the Minneapolis PBX is 763-717-5170. The destination pattern 17637175 means that all calls to Minneapolis employees will stay within the suburban Minneapolis PBX and will not reach or be carried on the local PSTN. Similarly, the Inbound Phone Book for the Baltimore VOIP generally matches the Outbound Phone Book of the Minneapolis VOIP.

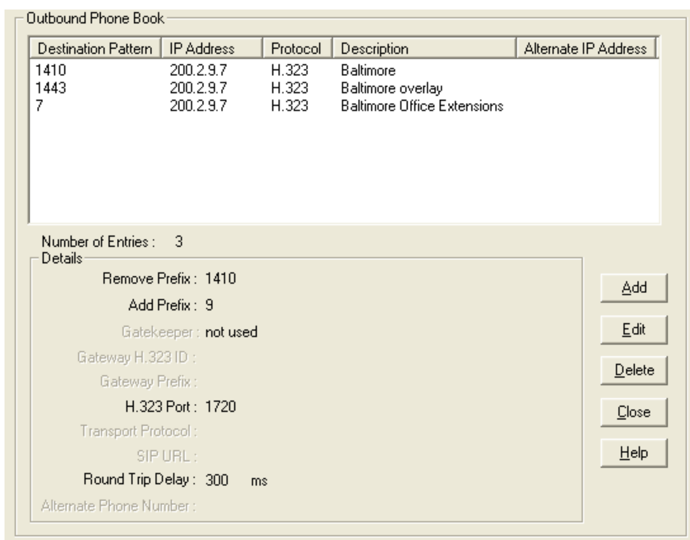


Inbound Baltimore example

Notice the extended prefix to be removed: 14103257. This entry allows Minneapolis users to contact Baltimore co-workers as though they were in the Minneapolis facility, using numbers in the range 7000 to 7999.

Note also that a comma (as in the entry 9,443) denotes a delay in dialing. A one-second delay is commonly used to allow a second dial tone to be generated for calls going outside of the facility's PBX system.

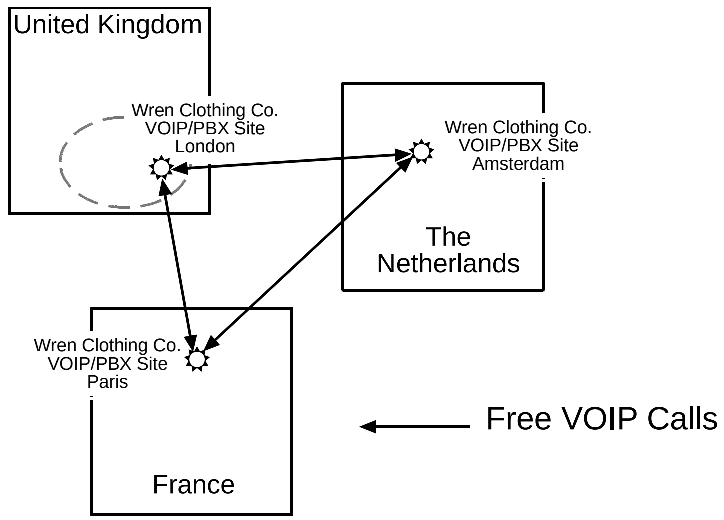
The Outbound Phone Book for the Minneapolis VOIP is shown below. The third destination pattern, "7" facilitates reception of co-worker calls using local-appearing-extensions only. In this case, the "Add Prefix" field value for this phonebook entry would be "1410325".



Outbound Minneapolis example

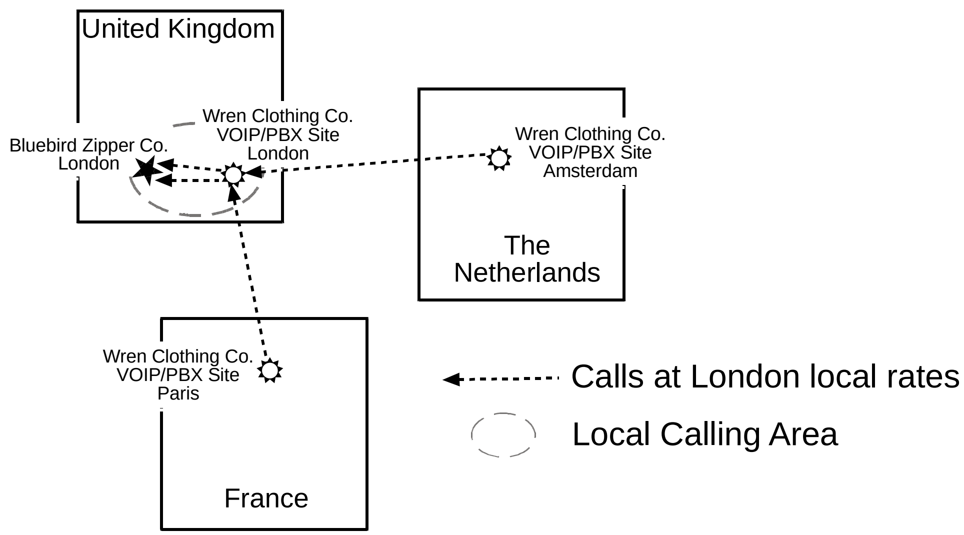
Europe

The most direct use of the VOIP system is making calls between the offices where the VOIPs are located. Consider, for example, the Wren Clothing Company. This company has VOIP-equipped offices in London, Paris, and Amsterdam, each served by its own PBX. VOIP calls between the three offices completely avoid international long-distance charges. These calls are free. The phonebooks can be set up to allow all Wren Clothing employees to contact each other using 3-, 4-, or 5-digit numbers, as though they were all in the same building.



Free VOIP calls

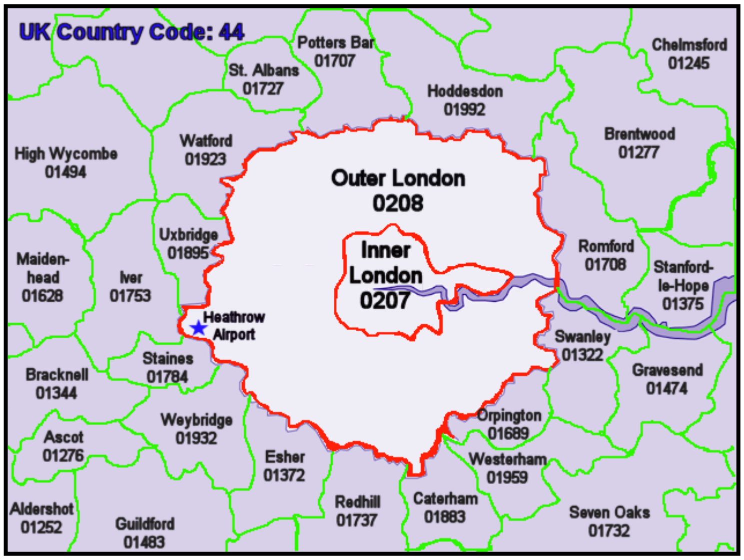
In another use of the VOIP system, the local calling area of each VOIP location becomes accessible to all of the VOIP system's users. As a result, international calls can be made at local calling rates. For example, suppose that Wren Clothing buys its zippers from The Bluebird Zipper Company in the western part of metropolitan London. In that case, Wren Clothing personnel in both Paris and Amsterdam could call the Bluebird Zipper Company without paying international long-distance rates. Only London local phone rates would be charged. This applies to calls completed anywhere in London's local calling area. Generally, local calling rates apply only within a single area code, and, for all calls outside that area code, national rates apply. There are, however, some European cases where local calling rates extend beyond a single area code. Local rates between Inner and Outer London are one example of this. It is also possible, in some locations, that calls within an area code may be national calls - but this is rare.

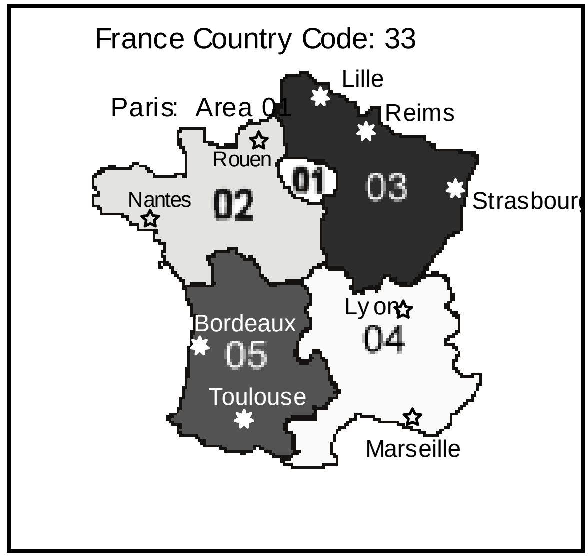


Local calling area

This next example will have the following features:

- Employees in all cities will be able to call each other over the VOIP system using 4-digit extensions.
- Calls to Outer London and Inner London, greater Amsterdam, and greater Paris will be accessible to all company offices as local calls.
- Vendors in Guildford, Lyon, and Rotterdam can be contacted as national calls by all company offices.

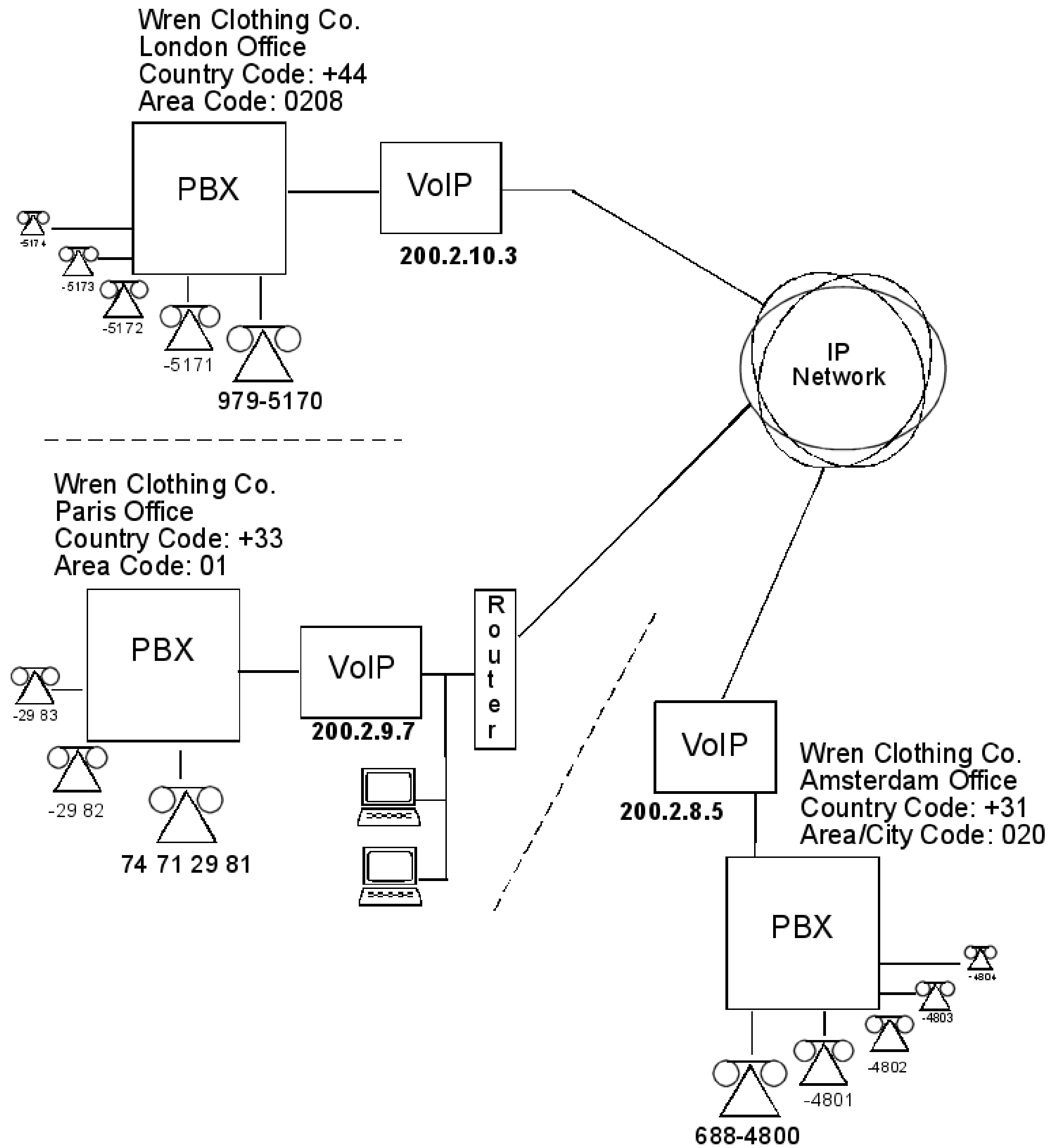




UK & France codes

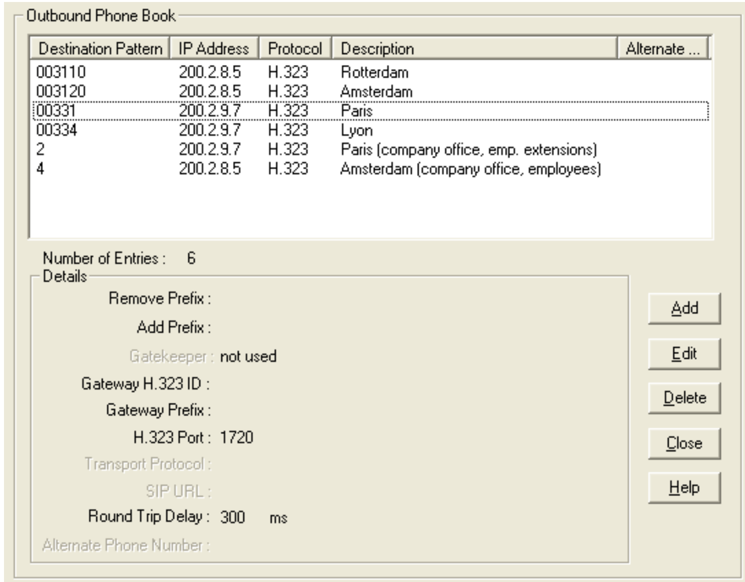
Netherlands codes

An outline of the equipment setup in these three offices is shown below.



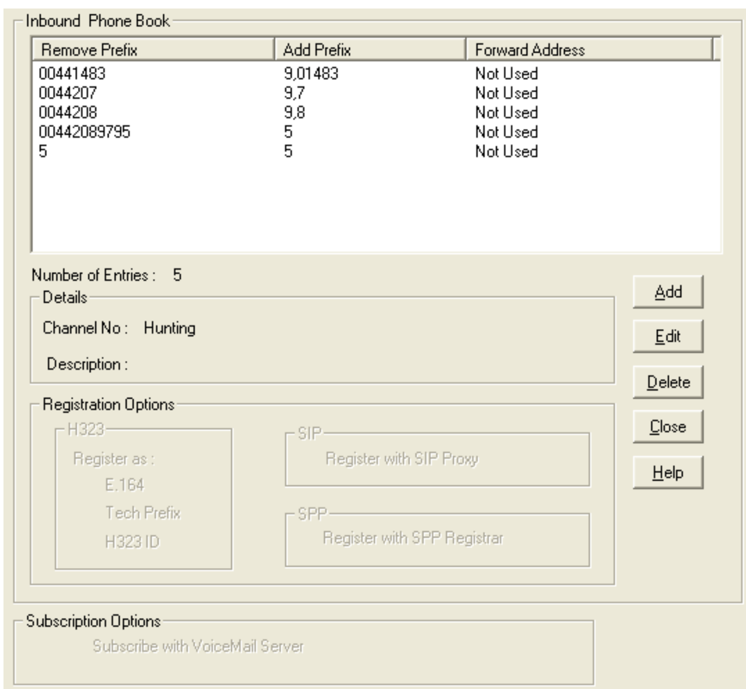
Setup example

The screen below shows Outbound Phone Book entries for the VOIP located in the company's London facility.



London example outbound

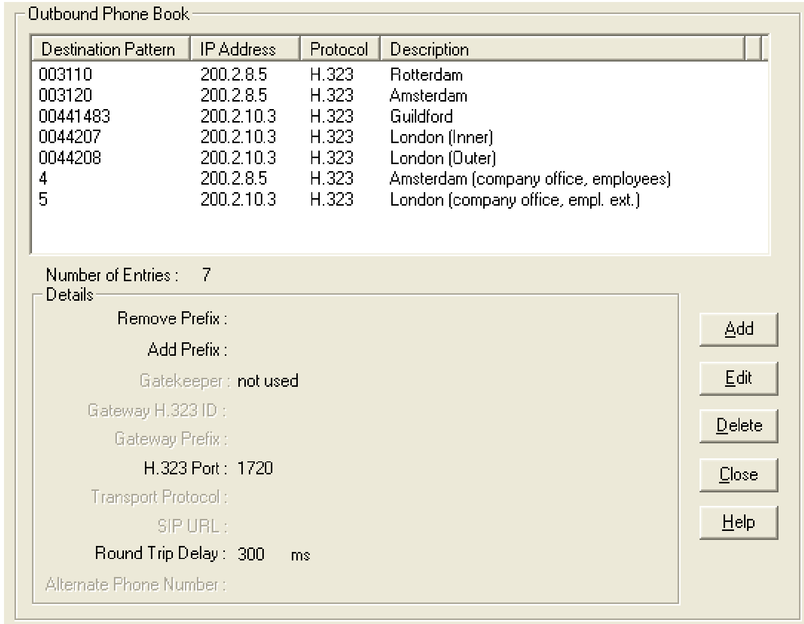
The Inbound Phone Book for the London VOIP is shown below.



London example inbound

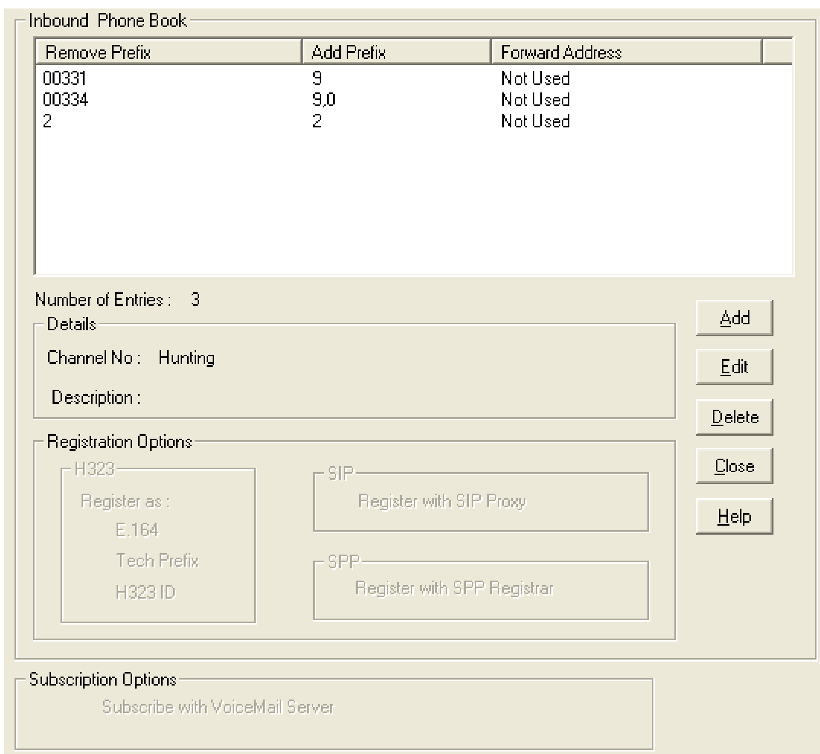
NOTE: Commas are allowed in the Inbound Phonebook, but **not** in the Outbound Phonebook. Commas denote a brief pause for a dial tone, allowing time for the PBX to get an outside line.

The screen below shows Outbound Phone Book entries for the VOIP located in the company's Paris facility.



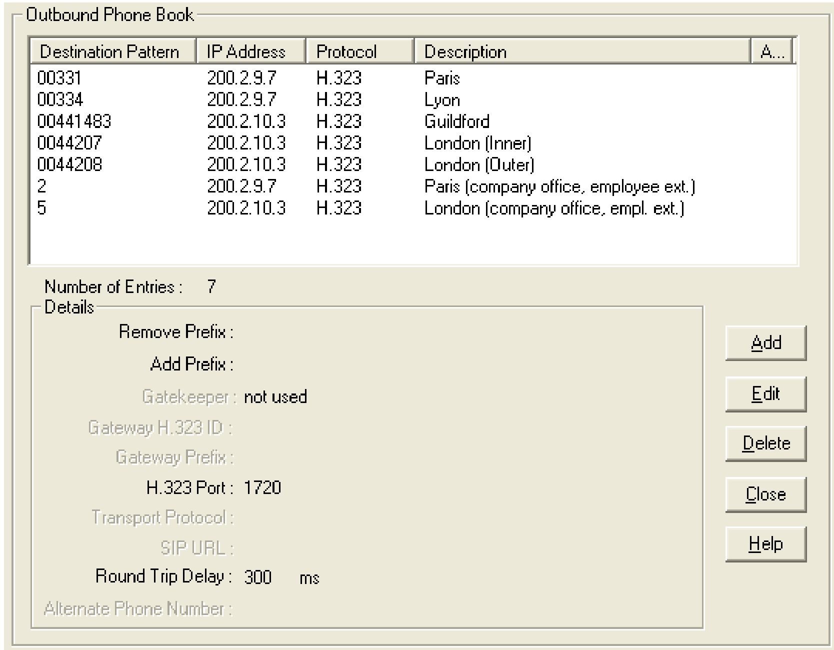
Paris example outbound

The Inbound Phone Book for the Paris VOIP is shown below.



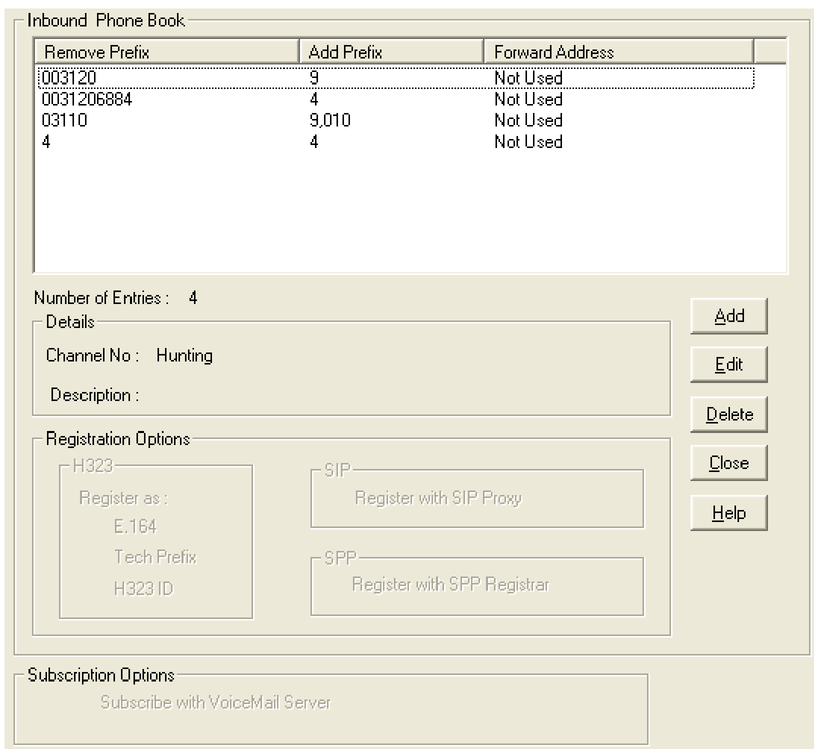
Paris example inbound

The screen below shows Outbound Phone Book entries for the VOIP in the company's Amsterdam facility.



Amsterdam example outbound

The Inbound Phone Book for the Amsterdam VOIP is shown below.

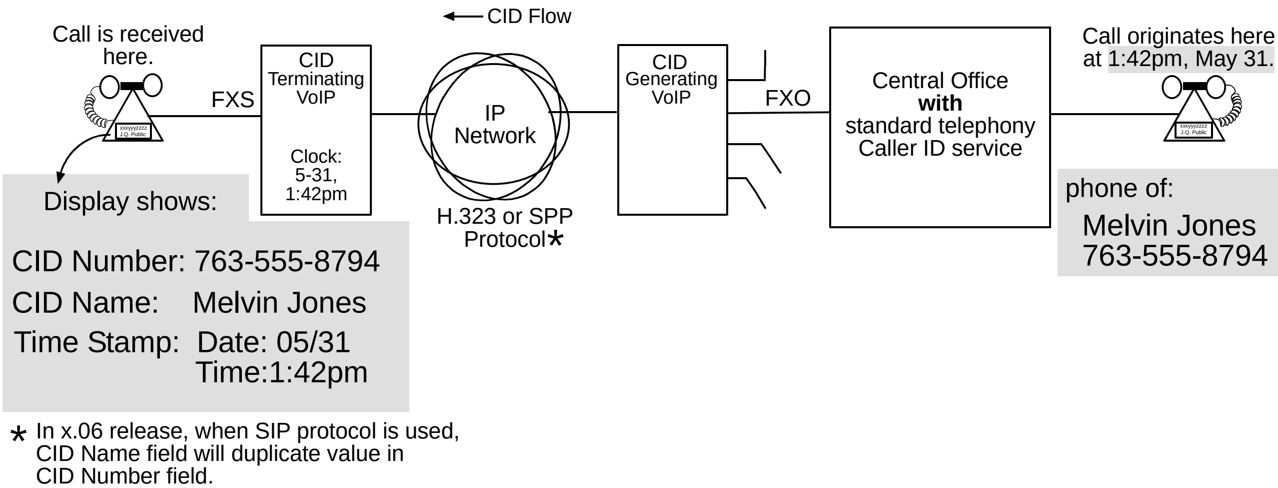


Amsterdam example inbound

Variations of Caller ID

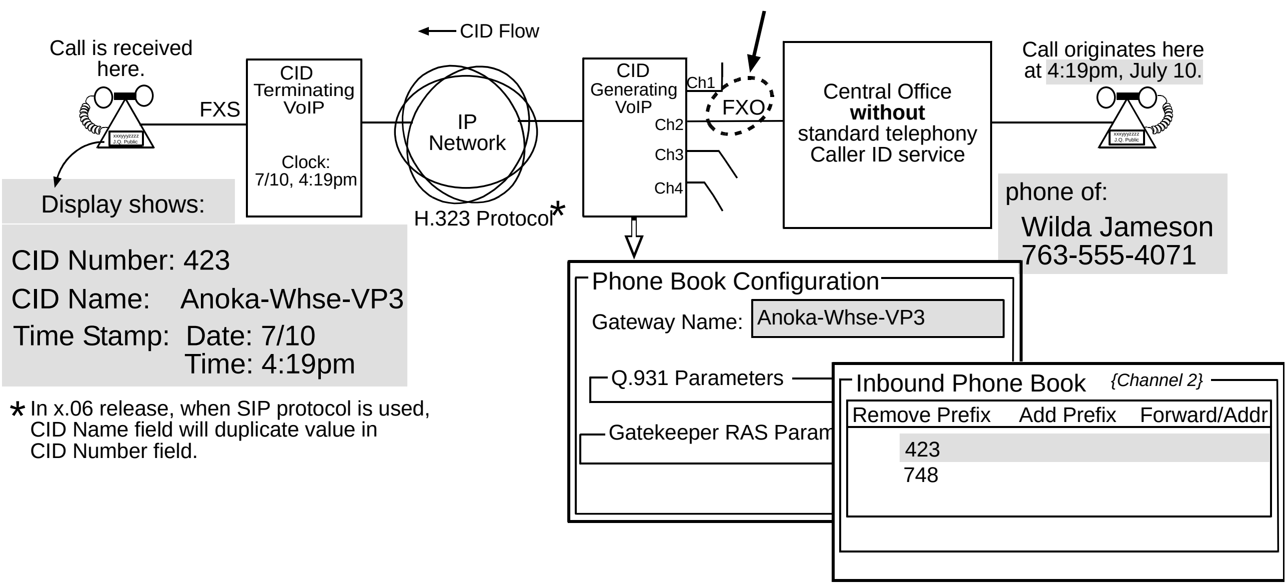
The Caller ID feature has dependencies on both the telco central office and the MultiVOIP phone book. See the diagram series below:

VOIP Caller ID Case #1 – Call, through telco central office *with* standard CID, enters VOIP system.



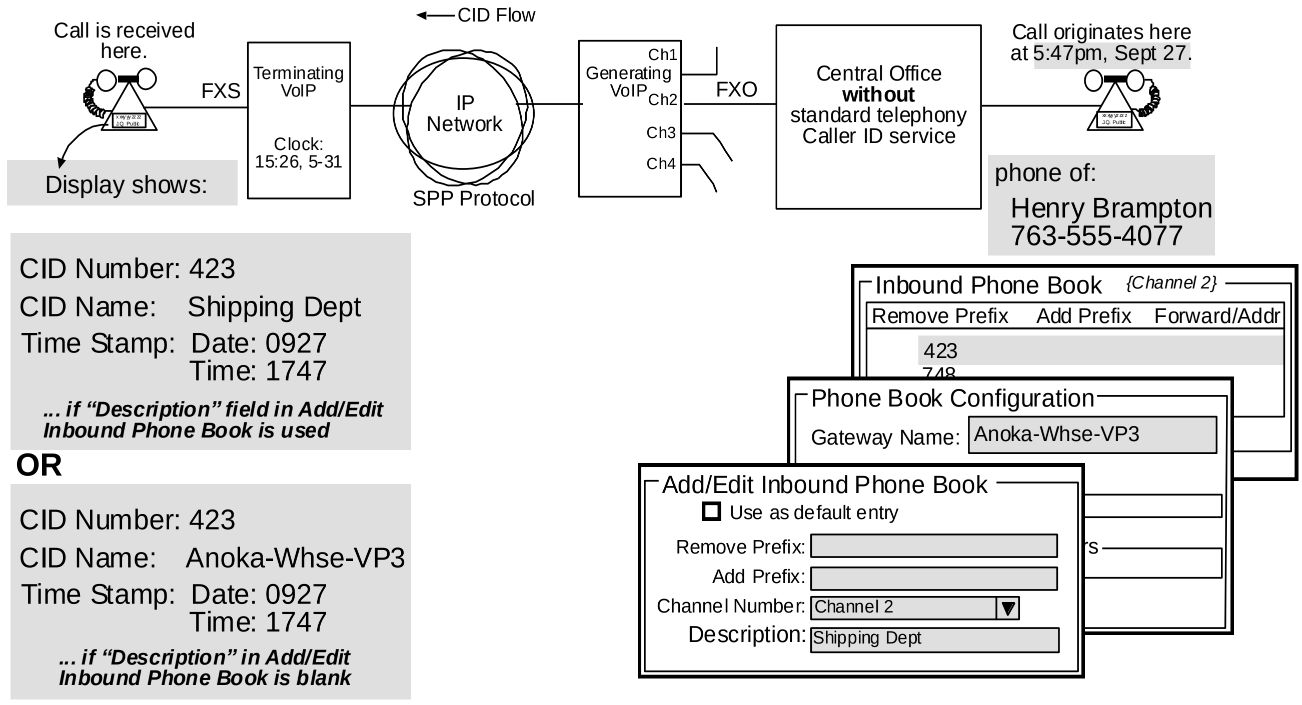
Caller ID example 1

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



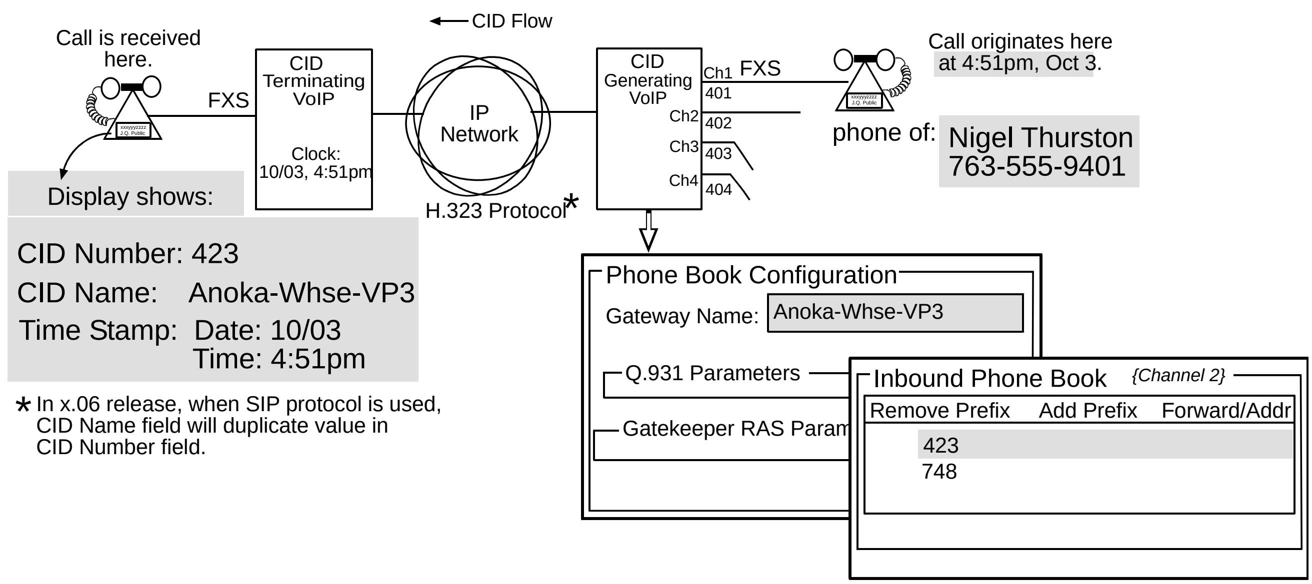
Caller ID example 2

VOIP Caller ID Case #3 – Call, through telco central office *without* standard CID, enters SPP VOIP system.



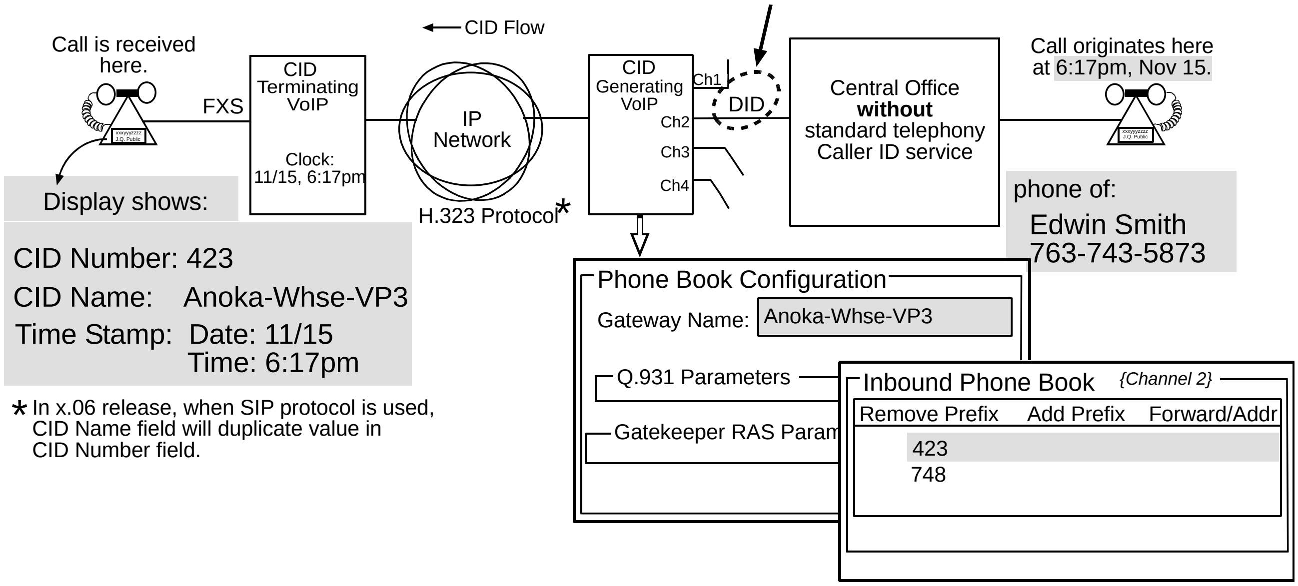
Caller ID example 3

VOIP Caller ID Case #4 – Remote FXS call on H.323 VOIP system.



Caller ID example 4

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



Caller ID example 5

Chapter 6 – Using the Software

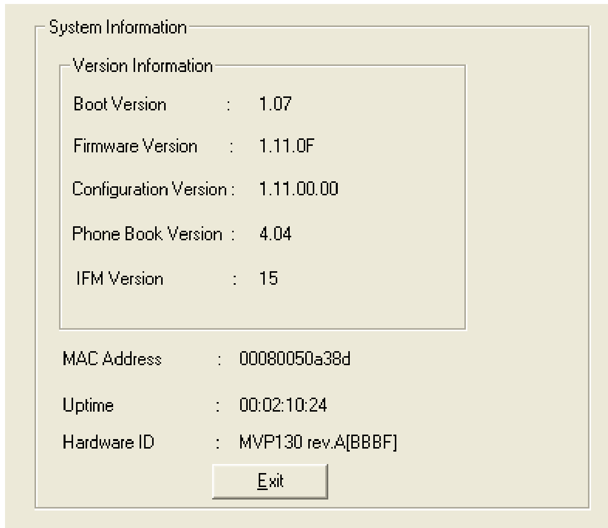
This chapter describes the day to day operation and maintenance sections of the MultiVOIP software. How to update the firmware and software are also covered here should either be needed. This section will mainly focus on the Statistics section of the configuration software, but there are references to a few of the other sections as they are used more in the daily operations than in a setup situation.

Software Categories Covered in This Chapter

- **System Information**
- **Call Progress**
- **Logs**
- **IP Statistics**
- **Link Management**
- **Registered Gateway Details**
- **Servers**
  - **H.323 GateKeepers**
  - **SIP Proxies**
  - **SPP Registrars**
- **Advanced**
  - **Packetization Time**

System Information screen

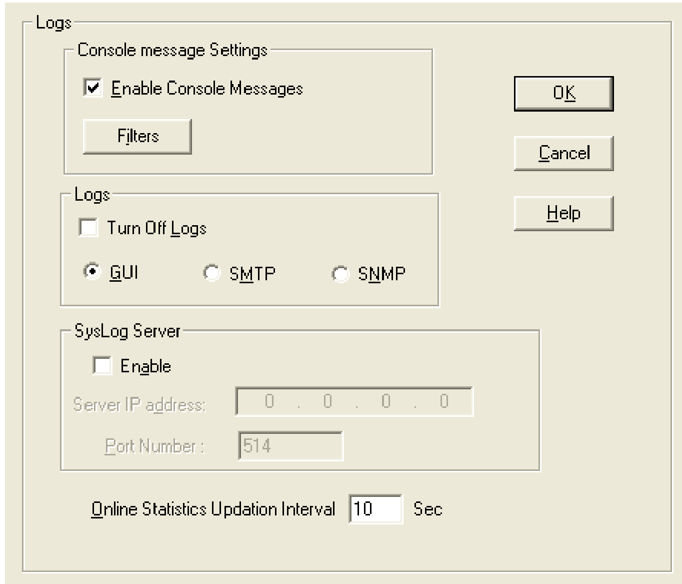
This screen presents system information at a glance. It is found under the Configuration section and its primary use is in troubleshooting. The information presented in figure 6-1 is for reference only and is not meant to be an exact match of your system.



System information

Field Name	Values	Description
Boot Version	*nn.nn* alpha-numeric	Indicates the version of the code that is used at the startup (booting) of the VOIP. The boot code version is independent of the software version.
Firmware Version	*nn.nn.nn* alpha-numeric	Indicates the version of the MultiVOIP firmware.
Configuration Version	*nn.nn.nn.nn* alpha-numeric	Indicates the version of the MultiVOIP configuration software.
Phone Book Version	*nn.nn* alpha-numeric	Indicates the version of the MultiVOIP phone book being used.
IFM Version	*nn* alpha-numeric	Indicates version of the IFM module, the device that performs the transformation between telephony signals and IP signals.
Mac Address	numeric	Denotes the number assigned as the VOIP unit’s unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the VOIP has been running since its last booting.
Hardware ID	alpha-numeric	Indicates version of the MultiVOIP circuit board assembly being used.

The frequency with which the System Information screen is updated is determined by a setting in the Logs/Traces screen (which is under the Configuration section).

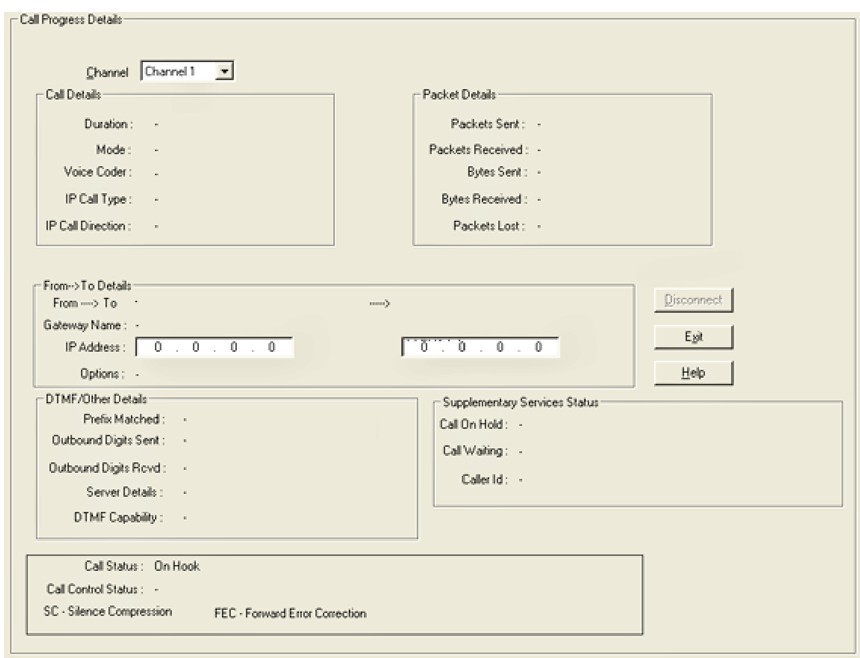


Logs/Traces screen

Statistics Section

Ongoing operation of the MultiVOIP, whether it is in a MultiVOIP/PBX setting or MultiVOIP/telco-office setting, can be monitored for performance using the Statistics functions of the MultiVOIP software. The following screens are examples of what can be shown and are followed by detailed descriptions of the categories involved. The model and signaling used will affect what is available for display.

Call Progress

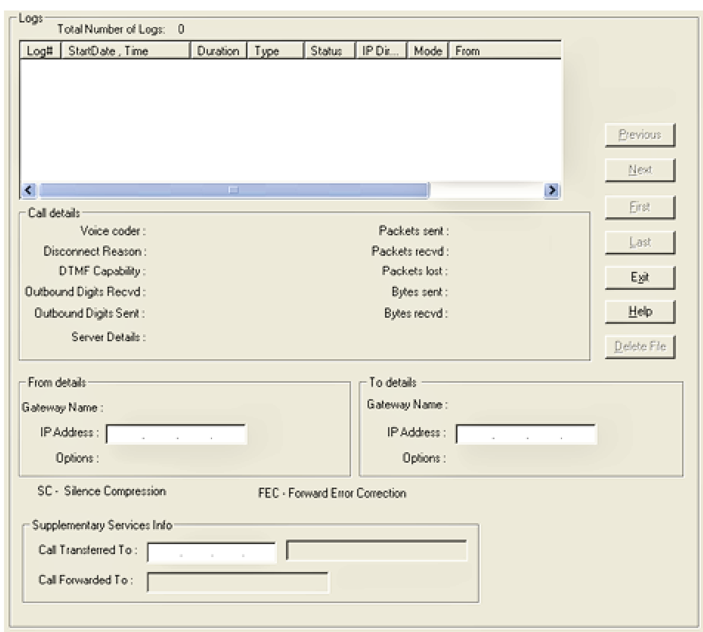


Call progress screen

Field Name	Values	Description
Channel	1-n	Number of data channel or time slot on which the call is carried. This is the channel for which call-progress details are being viewed.
Call Details		
Duration	H/M/S	The length of the call in hours, minutes, and seconds (hh:mm:ss).
Mode	Voice or FAX	Indicates whether the call being described was a voice call or a FAX call.
Voice Coder	G.723, G.729, G.711, and so on	The voice coder being used on this call.
IP Call Type	H.323, SIP, or SPP	Indicates the Call Signaling protocol used for the call (H.323, SIP, or SPP).
IP Call Direction	incoming, outgoing	Indicates whether the call in question is an incoming call or an outgoing call.
Packet Details		
Packets Sent	integer value	The number of data packets sent over the IP network during the call.
Packets Rcvd	integer value	The number of data packets received over the IP network during the call.
Bytes Sent	integer value	The number of bytes of data sent over the IP network during the call.
Bytes Rcvd	integer value	The number of bytes of data received over the IP network during the call.
Packets Lost	integer value	The number of voice packets from this call that were lost after being received from the IP network.
From – To Details		
Gateway Name (from)	alphanumeric string	Identifier for the VOIP gateway that handled the origination of this call.
IP Address (from)	*n.n.n.n*	IP address from which the call was received.
Options	SC, FEC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.
Gateway Name (to)	alphanumeric string	Identifier for the VOIP gateway that handled the completion of this call.
IP Address (to)	*n.n.n.n*	IP address to which the call was sent.
Options	SC, FEC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.
DTMF/Other Details		
Prefix Matched	specified dialing digits	Displays the dialed digits that were matched to a phonebook entry.
Outbound Digits Sent	0-9, #, *	The digits transmitted by the MultiVOIP to the PBX/telco for this call.
Outbound Digits Received	0-9, #, *	Of the digits transmitted by the MultiVOIP to the PBX/telco for this call, these are the digits that were confirmed as being received.
Server Details	*n.n.n.n* and/or other related descriptions	The IP address (and so on) of the traffic control server (if any) being used (whether an H.323 gatekeeper, a SIP proxy, or an SPP registrar gateway) will be displayed here if the call is handled through that server.
DTMF Capability	inband, out of band Expressions differ slightly for different Call Signaling protocols (H.323, SIP, or SPP).	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different VOIP protocols. For H.323, this field can display "Out of Band" or "Inband". For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition. For SPP it can display "Out of Band RFC2833" or "Inband".

Field Name	Values	Description
Supplementary Services Status		
Call on Hold	alphanumeric	Describes held call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers come from **Gateway Name** field in **Phone Book Configuration** screen of remote VOIP.
Call Waiting	alphanumeric	Describes waiting call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers come from **Gateway Name** field in **Phone Book Configuration** screen of remote VOIP.
Caller ID	"Calling Party + *identifier*"; "Alerting Party + *identifier";* "Busy Party + *identifier";* "Connected Party + *identifier*"	This field shows the identifier and status of a remote VOIP (which has Call Name Identification enabled) with which this VOIP unit is currently engaged in some VOIP transmission. The status of the engagement (Connected, Alerting, Busy, or Calling) is followed by the identifier of a specific channel of a remote VOIP unit. This identifier comes from the "Caller Id" field in the **Supplementary Services screen** of the remote VOIP unit.
Call Status fields		
Call Status	hangup, active	Shows condition of current call.
Call Control Status	Tun, FS + Tun, AE, Mux	Displays the H.323 version 4 features in use for the selected call. These include tunneling (Tun), Fast Start with tunneling (FS + Tun), Annex E multiplexed UDP call signaling transport (AE), and Q.931 Multiplexing (Mux).
Silence Compression	SC	"SC" stands for Silence Compression. 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.
Forward Error Correction	FEC	"FEC" stands for Forward Error Correction. **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

Logs

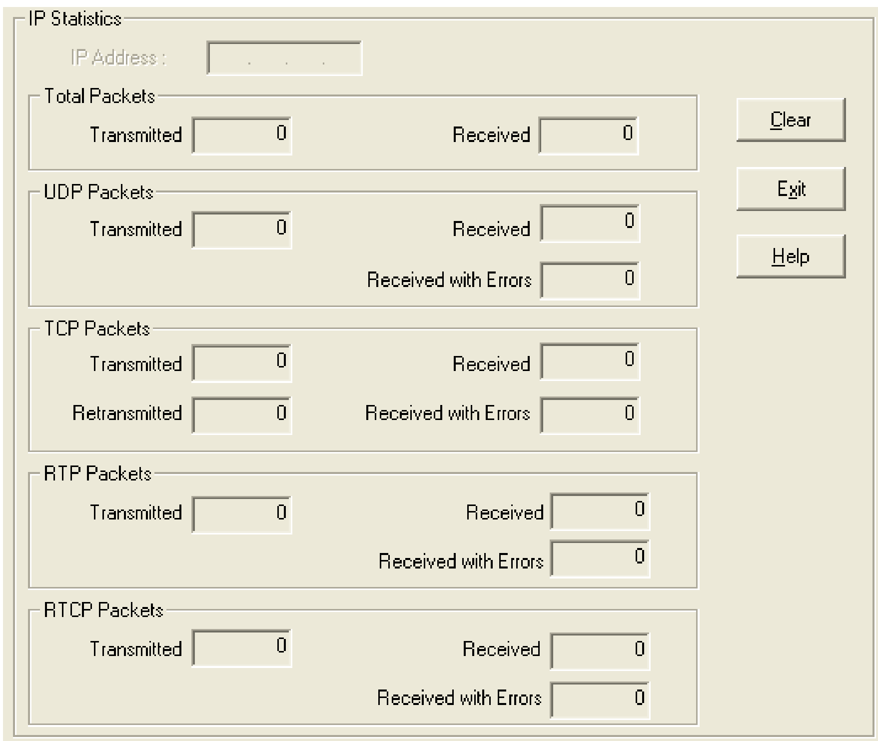


Log statistics screen

Field Name	Values	Description
Log # column	1 or higher	All calls are assigned an event number in chronological order, with the most recent call having the highest event number.
Start Date, Time column	dd:mm:yyyy hh:mm:ss	The starting time of the call. The date is presented as a day and a month of one or two digits, and a four-digit year. This is followed by a time-of-day in a two-digit hour, a two-digit minute, and a two-digit seconds value.
Duration column	hh:mm:ss	This describes how long the call lasted in hours, minutes, and seconds.
Type	H.323, SIP, SPP	Indicates the Call Signaling protocol used for the call (H.323, SIP, or SPP).
Status column	success or failure	Displays the status of the call (whether the call was completed or not).
IP Direction	incoming, outgoing	Indicates whether the call is "incoming" or "outgoing" with respect to the gateway.
Mode column	voice or FAX	Indicates whether the event being described was a voice call or a FAX call.
From column	gateway name	Displays the name of the voice gateway that originates the call.
To column	gateway name	Displays the name of the voice gateway that completes the call.
Special Buttons		
Previous	--	Displays log entry before currently selected one.
Next	--	Displays log entry after currently selected one.
First	--	Displays first log entry
Last	--	Displays last log entry.
Delete File	--	Deletes selected log file.
Call Details		
Voice coder	Coder protocol	The voice coder being used on this call.
Disconnect Reason	"Normal" or "Local" disconnection.	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause occasioned disconnection (for example, a technical error or failure).
DTMF Capability	inband, out of band Expressions differ slightly for different Call Signaling protocols.	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different VOIP protocols. For H.323, this field can display "Out of Band" or "Inband". For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition. For SPP it can display "Out of Band RFC2833" or "Inband".
Outbound Digits Received	0-9, #, *	The digits, sent by MultiVOIP to PBX/telco, that were acknowledged as having been received by the remote VOIP gateway.
Outbound Digits Sent	0-9, #, *	The digits transmitted by the MultiVOIP to the PBX/telco for this call.
Server Details	*n.n.n.n*	When the MultiVOIP is operating in the non-direct mode (with Gatekeeper in H.323 mode; with proxy in SIP mode; or in the client/server configuration of SPP mode), this field shows the IP address of the server that is directing IP phone traffic.
Packets sent	integer value	Number of data packets sent over the IP network in the course of this call.
Packets received	integer value	Number of data packets received over the IP network in the course of this call.
Packets lost	integer value	Number of voice packets from this call that were lost after being received from the IP network.
Bytes sent	integer value	Number of bytes of data sent over the IP network in the course of this call.
Bytes received	integer value	Number of bytes of data received over the IP network in the course of this call.

FROM Details		
Gateway Name	alphanumeric	Identifier for the VOIP gateway that originated this call.
IP Address	*n.n.n.n*	IP address of the VOIP gateway from which the call was received.
Options	FEC, SC	Displays VOIP transmission options used by the VOIP gateway originating the call. These may include Forward Error Correction or Silence Compression.
TO Details		
Gateway Name	alphanumeric	Identifier for the VOIP gateway that completed (terminated) this call.
IP Address	*n.n.n.n*	IP address of the VOIP gateway at which the call was completed.
Options		Displays transmission options used by VOIP gateway terminating the call.
Supplementary Services Info		
Call Transferred To	phone number	Number of party called in transfer.
Call Forwarded To	phone number	Number of party called in forwarding.

IP Statistics



IP statistics screen

UDP versus TCP. (User Datagram Protocol versus Transmission Control Protocol). UDP provides unguaranteed, connectionless transmission of data across an IP network. By contrast, TCP provides reliable, connection-oriented transmission of data.

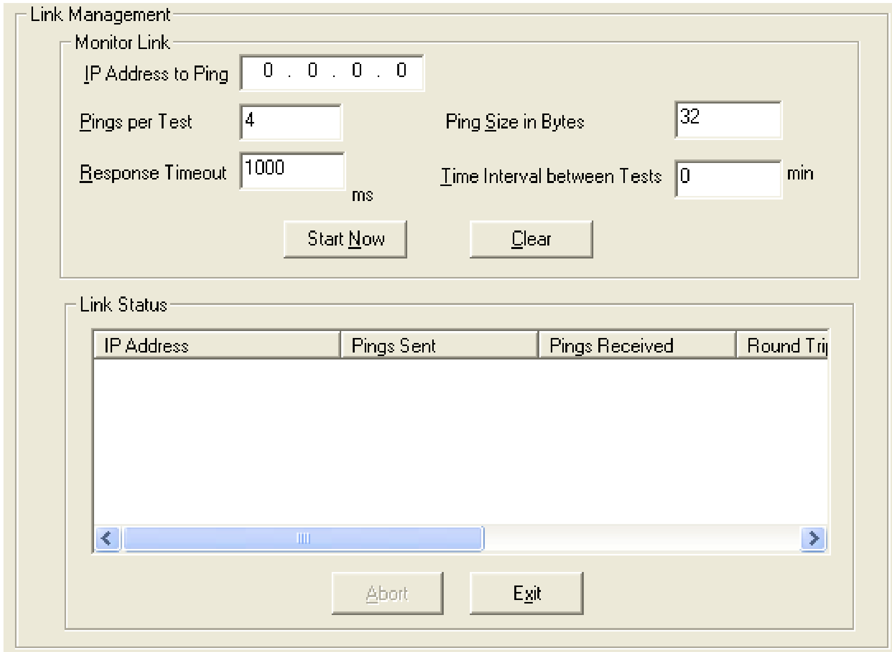
Both TCP and UDP split data into packets called “datagrams.” However, TCP includes extra headers in the datagram to enable retransmission of lost packets and reassembly of packets into their correct order if they arrive out of order. UDP does not provide this. Lost UDP packets are irretrievable; that is, out-of-order UDP packets cannot be reconstituted in their proper order.

Despite these obvious disadvantages, UDP packets can be transmitted much faster than TCP packets -- as much as three times faster. In certain applications, like audio and video data transmission, the need for high speed outweighs the need for verified data integrity. Sound or pictures often remain intelligible despite a certain amount of lost or disordered data packets (which comes through as static).

Field Name	Values	Description
IP Address	*n.n.n.n*	IP address of the MultiVOIP. For an IP address to be displayed here, the MultiVOIP must have DHCP enabled. Its IP address, in such a case, is assigned by the DHCP server.
"Clear" button	--	Clears packet tallies from memory.
Total Packets		Sum of data packets of all types.
Transmitted	integer value	Total number of packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Total number of packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Total number of error-laden packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
UDP Packets		User Datagram Protocol packets.
Transmitted	integer value	Number of UDP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
TCP Packets		Transmission Control Protocol packets.
Transmitted	integer value	Number of TCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
RTP Packets		Voice signals are transmitted in Realtime Transport Protocol packets. RTP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
RTCP Packets		Realtime Transport Control Protocol packets convey control information to assist in the transmission of RTP (voice) packets. RTCP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

Link Management

The Link Management screen is essentially an automated utility for pinging endpoints on your VOIP network. This utility generates pings of variable sizes at variable intervals and records the response to the pings.

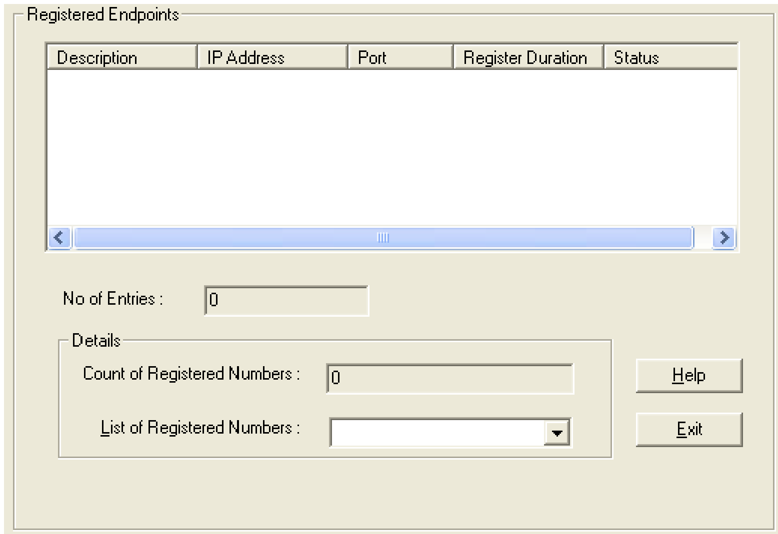


Link management

Field Name	Values	Description
Monitor Link fields		
IP Address to Ping	*n.n.n.n*	This is the IP address of the target endpoint to be pinged.
Pings per Test	1-999	This field determines how many pings will be generated by the Start Now command.
Response Timeout	500 – 5000 milliseconds	The duration after which a ping will be considered to have failed.
Ping Size in Bytes	32 – 128 bytes	This field determines how long or large the ping will be.
Timer Interval between Pings	0 or 30 – 6000 minutes	This field determines how long of a wait there is between one ping and the next.
Start Now command button	--	Initiates pinging.
Clear command button	--	Erases ping parameters in Monitor Link field group and restores default values.
Link Status Parameters		These fields summarize the results of pinging.
IP Address column	*n.n.n.n*	Target of ping.
No. of Pings Sent	as listed	Number of pings sent to target endpoint.
No. of Pings Received	as listed	Number of pings received by target endpoint.
Round Trip Delay (Min/Max/Avg)	as listed, in milliseconds	Displays how long it took from time ping was sent to time ping response was received.
Last Error	as listed	Indicates when last data error occurred.

Registered Gateway Details

The Registered Gateway Details screen presents a real-time display of the special operating parameters of the Single Port Protocol (SPP). These are configured in the **Call Signaling** screen and in the **Add/Edit Outbound Phone Book** screen.

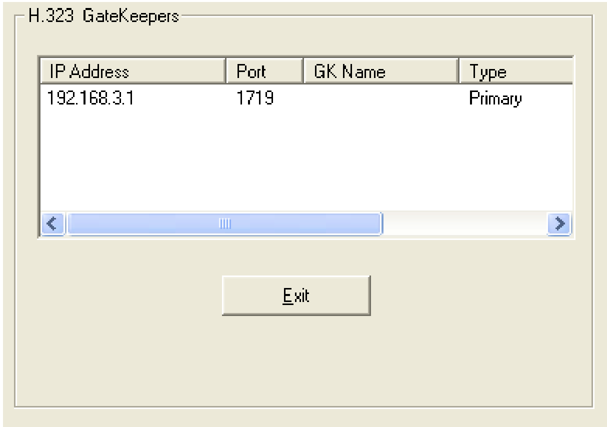


Registered endpoints

Field Name	Values	Description
Column Headings		
Description	alphanumeric	This is a descriptor for a particular VOIP gateway unit. This descriptor should generally identify the physical location of the unit (for example, city, building, and so on) and perhaps even its location in an equipment rack.
IP Address	*n.n.n.n*	The RAS address for the gateway.
Port	*n*	Port by which the gateway exchanges H.225 RAS messages with the gatekeeper.
Register Duration		The time remaining in seconds before the TimeToLive timer expires. If the gateway fails to reregister within this time, the endpoint is unregistered.
Status	Registered/ unregistered	The current status of the gateway either registered or unregistered.
No. of Entries		The number of gateways currently registered to the Registrar. This includes all SPP clients registered and the Registrar itself.
Details		
Count of Registered Numbers		If a registered gateway is selected (by clicking on it in the screen), The "Count of Registered Numbers" will indicate the number of registered phone numbers for the selected gateway. When a client registers, all of its inbound phonebook's phone numbers become registered.
List of Registered Numbers		Lists all of the registered phone numbers for the selected gateway.

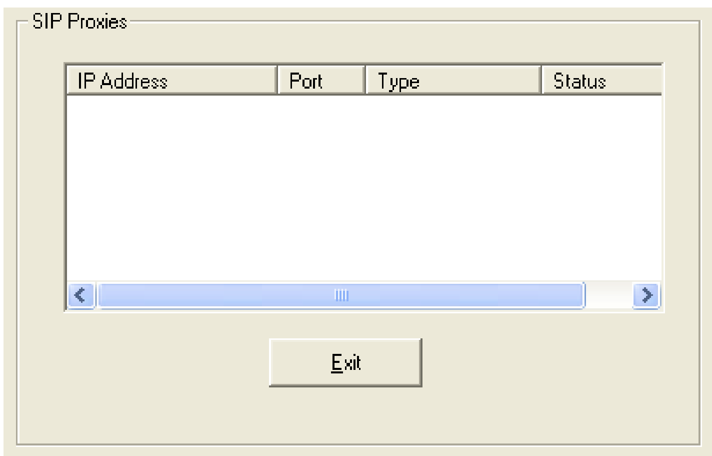
Servers

H.323 GateKeepers



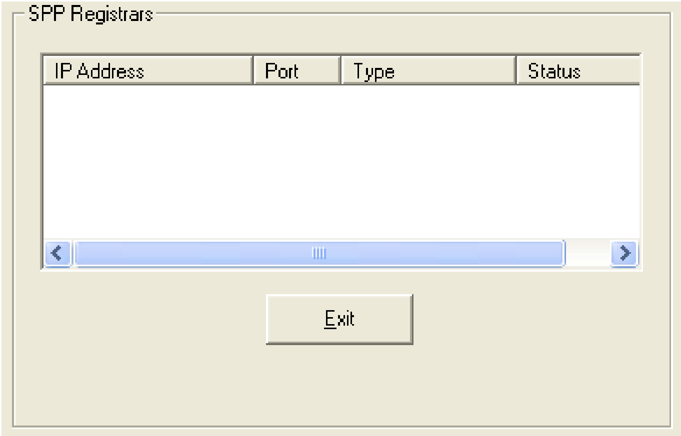
Field Name	Values	Description
Column Headings		
IP Address	*n.n.n.n*	The IP address of the gatekeeper.
Port	*n*	TDMA time slot used for communication between MultiVOIP unit and the gatekeeper that serves it.
GK Name	alpha-numeric string	Identifier for gatekeeper
Type	Primary, Predefined	This field describes the type of gateway as which the MultiVOIP is defined with respect to the gatekeeper
Priority	*n*	Priority level given.
Status	registered, not registered	The current status of the gateway either registered or unregistered.

SIP Proxies



Field Name	Values	Description
Column Headings		
IP Address	*n.n.n.n*	The IP address of the SIP proxy by which the MultiVOIP is governed.
Port	*port*	TDMA time slot used for communication between MultiVOIP unit and the SIP Proxy that governs it.
Type	Primary, Alternate	This field describes the type of gateway as which the MultiVOIP is defined with respect to the gatekeeper.
Status	registered, not registered	The current status of the MultiVOIP gateway with respect to the SIP proxy either registered or unregistered.

SPP Registrars

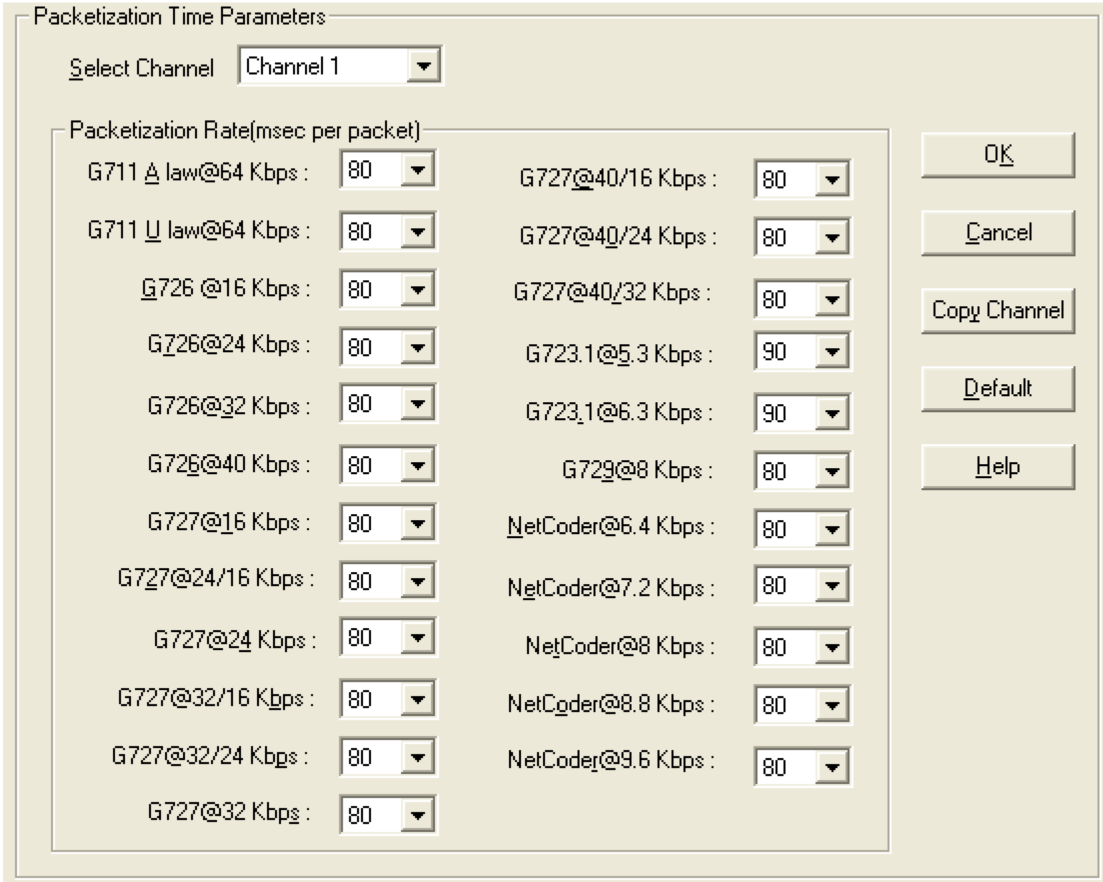


Field Name	Values	Description
Column Headings		
IP Address	*n.n.n.n*	The IP address of the gatekeeper.
Port	*port*	TDMA time slot used for communication between MultiVOIP unit and the gatekeeper that serves it.
Type	Primary, Predefined	This field describes the type of gateway as which the MultiVOIP is defined with respect to the gatekeeper.
Status	registered, not registered	The current status of the gateway either registered or unregistered.

Advanced

Packetization Time

You can use the **Packetization Time** screen to specify definite packetization rates for coders selected in the Voice/FAX Parameters screen (in the “Coder Options” group of fields). The Packetization Time screen is accessible under the “Advanced” options entry in the sidebar list of the main VOIP software screen. In dealing with RTP parameters, the Packetization Time screen is closely related to both Voice/FAX Parameters and to IP Statistics. It is located in the “Advanced” group for ease of use.



Packetization time

Packetization rates can be set separately for each channel.

The table below presents the ranges and increments for packetization rates. The final column represents recommended settings (based on the most common found) when operating with third party devices.

Packetization Ranges and Increments			Recommendations
Coder Types	Range (in Kbps); {default}	Increments (in Kbps)	Setting (in ms)
G711, G726, G727	5-120 {5}	5	20
G723	30-120 {30}	30	30
G729	10-120 {10}	10	20
NetCoder	20-120 {20}	20	20

After you set the packetization rate for one channel, you can copy it into other channels by using the Copy Channel button on the Packetization Time screen. To do so, select the channels you want to copy the settings for.

MultiVOIP Program Menu Items

After the MultiVOIP program is installed on the PC, it can be launched from the Programs group of the Windows **Start** menu (Start | **Programs** | **MultiVOIP *n.nn*** | …). This section describes the software functions available on this menu.

Menu Selection	Description
Configuration	Select this to enter the Configuration program where values for IP, telephony, and other parameters are set.
Configuration Port Setup	Select this to access the **COM Port Setup** screen of the MultiVOIP Configuration program.
Date and Time Setup	Select this for access to set calendar/clock used for data logging.
Download Factory Defaults	Select this to return the configuration parameters to the original factory values.
Download Firmware	Select this to download new versions of firmware as enhancements become available.
Download IFM Firmware	Select this to download new versions of IFM firmware as enhancements become available. The Interface Module (IFM) is the telephony interface for analog MultiVOIP units. There is one IFM for each channel of the MultiVOIP unit. For each channel, the IFM handles the analog signals to and from the attached telephone, PBX or CO line.
Download User Defaults	To be used after a full set of parameter values, values specified by the user, have been saved (using Save Setup). This command loads the saved user defaults into the MultiVOIP.
Set Password	Select this to create a password for access to the MultiVOIP software programs (**Program** group commands, Windows interface, web browser interface, & FTP server). Only the FTP Server function *requires* a password for access. The FTP Server function also requires that a username be set along with the password.
Uninstall	Select this to uninstall the MultiVOIP software (most, but not all components are removed from computer when this command is used).
Upgrade Software	Loads firmware (including H.323 stack) and settings from the controller PC to the MultiVOIP unit. User can choose whether to load Factory Default Settings or Current Configuration settings.

“Downloading” refers to transferring program files from the PC to the nonvolatile “flash” memory of the MultiVOIP. Such transfers are made using the PC’s serial port. This is a “download” from the perspective of the MultiVOIP unit.

When new versions of the MultiVOIP software become available, they are posted on Multi-Tech’s website. Although transferring updated program files from the Multi-Tech website to the user’s PC can generally be considered a download (from the perspective of the PC), this type of download cannot be initiated from the MultiVOIP software’s Program menu command set.

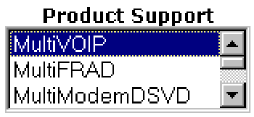
Generally, you must download updated firmware from the Multi-Tech website to the PC before it can be loaded from the PC to the MultiVOIP.

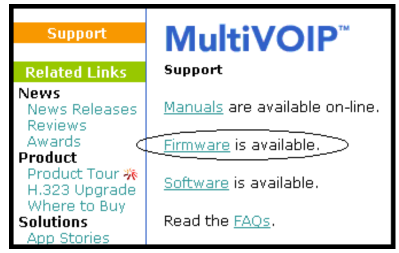
Updating Firmware

Generally, you must download updated firmware from the Multi-Tech website to the PC before it can be loaded from the PC to the MultiVOIP.

Note that the structure of the Multi-Tech website may change without notice. However, you can find most firmware updates by using standard web techniques. For example, you can access updated firmware by doing a search or by clicking on **Support**.

If you choose **Support**, you can select “MultiVOIP” in the **Product Support** menu and then click on **Firmware** to find MultiVOIP resources.

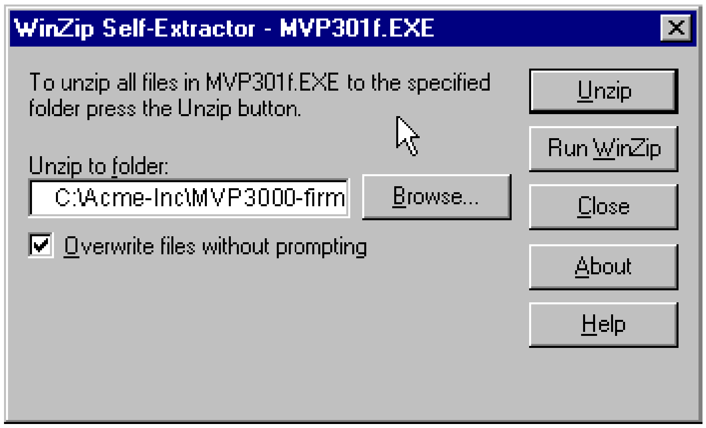




Web locations

After you locate the updated firmware, you can download it from the website.

The firmware file is a self-extracting compressed file (with .zip extension), which you must expand (decompressed, or "unzipped") on the user's PC in a user-specified directory. It is usually best to click the Browse button and select a folder that is easy to get to and remember.



Extract files

Implementing a Software Upgrade

You can upgrade MultiVOIP software locally using a single command at the MultiVOIP Windows interface, namely **Upgrade Software**. This command downloads firmware (including the H.323 stack), and factory default settings from the controller PC to the MultiVOIP unit.

When using the MultiVOIP Windows interface, you can transfer firmware and factory default settings from controller PC to MultiVOIP piecemeal using separate commands.

When using the MultiVOIP web browser interface to control/configure the VOIP remotely, upgrading of software must be done on a piecemeal basis using the FTP Server function of the MultiVOIP unit.

When performing a software upgrade (whether from the Windows interface or web browser interface), follow these steps in order:

1. Identify Current Firmware Version
2. Download Firmware
3. Download Factory Defaults

When upgrading firmware, implement the software commands "Download Firmware," and "Download Factory Defaults" in order. If not done in order, the upgrade is incomplete.

Identifying Current Firmware Version

Before implementing a MultiVOIP firmware upgrade, be sure to verify the firmware version currently loaded on it. The firmware version appears in the MultiVOIP Program menu. Go to **Start** | **Programs** | **MultiVOIP *n.nn***. The final expression, *n.nn*, is the firmware version number.

When a new firmware version is installed, the MultiVOIP software can be upgraded in one step using the **Upgrade Software** command, or piecemeal using the **Download Firmware** command and the **Download Factory Defaults** command.

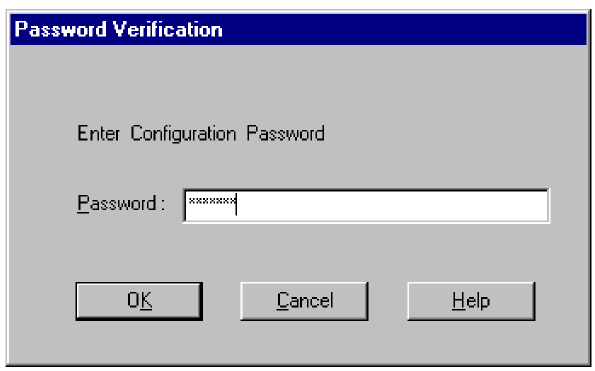
Download Firmware transfers the firmware (including the H.323 protocol stack) in the PC's MultiVOIP directory into the nonvolatile flash memory of the MultiVOIP.

Download Factory Defaults sets all configuration parameters to the standard default values that are loaded at the Multi-Tech factory.

Upgrade Software implements both the **Download Firmware** command and the **Download Factory Defaults** command.

Downloading Firmware

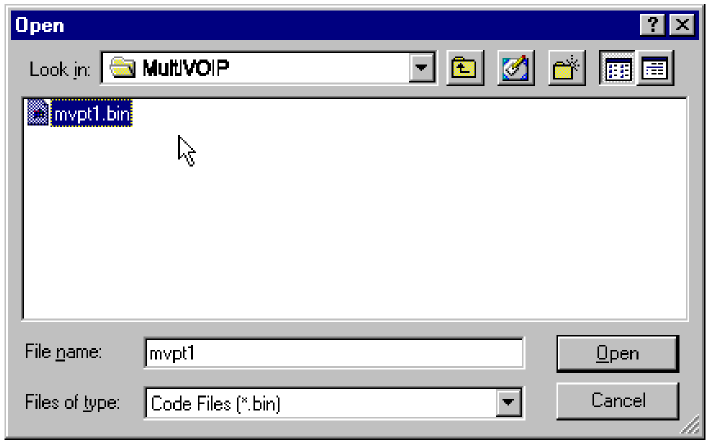
1. The MultiVOIP Configuration program must be off when invoking the Download **Firmware** command. If it is on, the command will not work.
2. To use the Download Factory Defaults command, go to **Start** | **Programs** | **MultiVOIP *n.nn*** | **Download Firmware**.
3. If a password has been established, the **Password Verification** screen will appear.

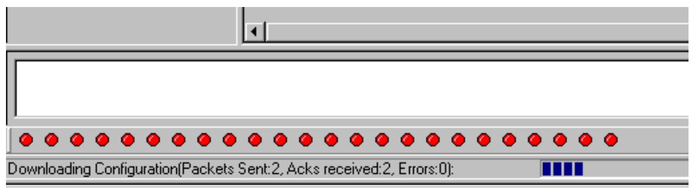
   Type in the password and click **OK**.
4. The **MultiVOIP *n.nn* Firmware** screen appears saying
   "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"

   Click **OK** to download the firmware.

   The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.
5. The program will locate the firmware ".bin" file in the MultiVOIP directory. Highlight the correct (newest) ".bin" file and click **Open**.



6. Progress bars will appear at the bottom of the screen during the file transfer.

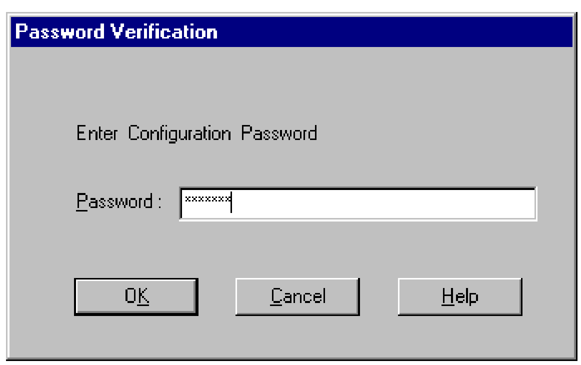


   The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

7. 7. The Download **Firmware** procedure is complete.

Downloading Factory Defaults

1. The MultiVOIP Configuration program must be off when invoking the Download **Factory Defaults** command. If it is on, the command will not work.
2. To use the Download Factory Defaults command, go to Start | Programs | MultiVOIP *n.nn*. | Download Factory Defaults.
3. If a password has been established, the **Password Verification** screen will appear.



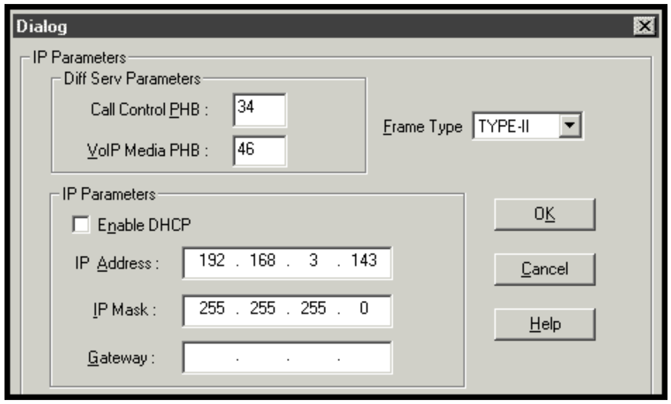
   Type in the password and click **OK**.

4. The MVP ***n.nn*** **- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"

   Click **OK** to download the factory defaults.

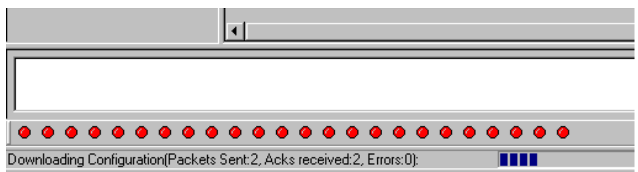
   The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

5. After the PC gets a response from the MultiVOIP, the **Dialog – IP Parameters** screen will appear.



The user should verify that the correct IP parameter values are listed on the screen and revise them if necessary. Then click **OK**.

6. Progress bars will appear at the bottom of the screen during the data transfer.



The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

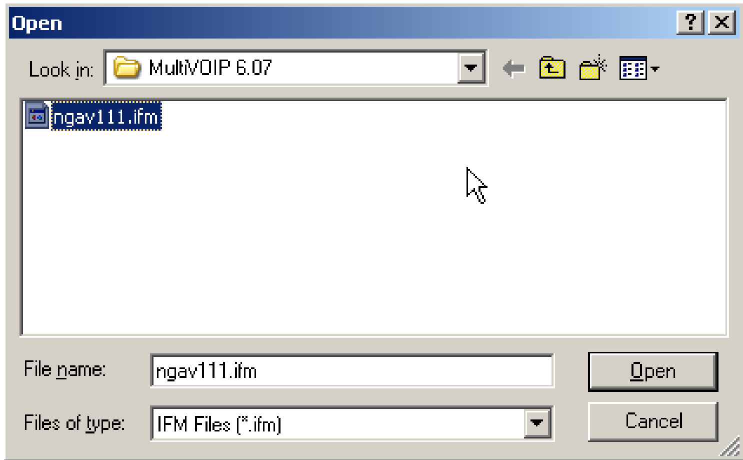
7. The **Download Factory Defaults** procedure is complete.

Downloading IFM Firmware

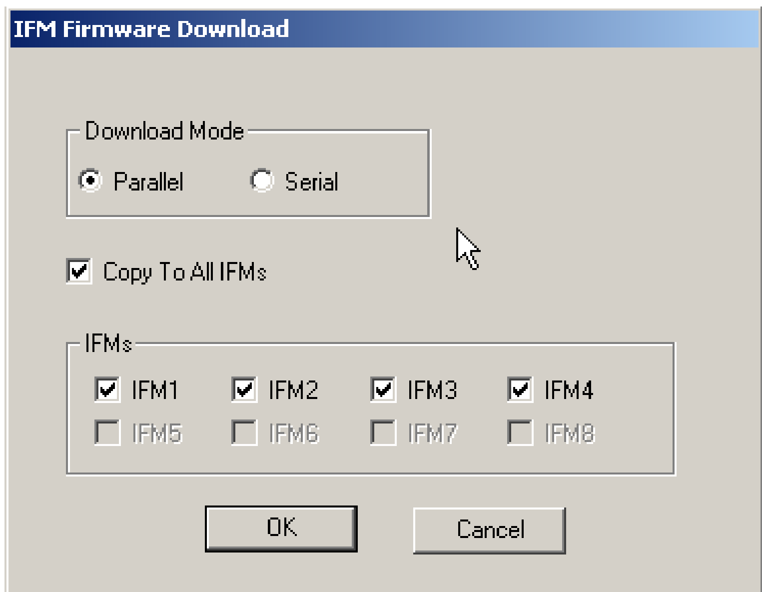
The Interface Module (IFM) is the telephony interface for analog MultiVOIP units. There is one IFM for each channel of the MultiVOIP unit. For each channel, the IFM handles the analog signals to and from the attached telephone, PBX or CO line. The IFM communicates with the main processor indicating the status of the telephone line. For example, it might indicate that a phone is off hook (FXS) or that an incoming ring is present (FXO). The IFM receives operating instructions from the VOIP's main processor. For example, the IFM might be instructed to ring the phone (FXS) or seize the line (FXO). The IFM contains a codec (coder/decoder) to convert the incoming audio to a PCM stream (pulse code modulation) which it sends to the DSP (digital signal processor). The IFM's codec also converts outgoing PCM to audio.

The firmware of the IFMs will change from time to time and you may need to upgrade the firmware on your MultiVOIP unit. To do so, follow these instructions.

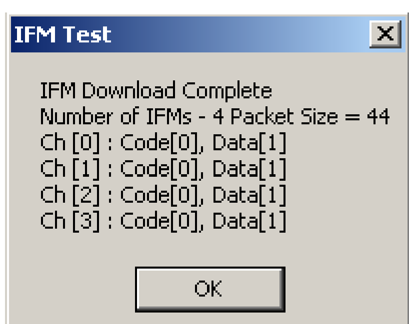
1. In the **System Information** screen of the MultiVOIP Configuration software, check the version number of the IFM firmware already installed on the MultiVOIP unit. Write down the version number.
2. Exit the Configuration software program. The MultiVOIP Configuration program must be off when invoking the **Download IFM Firmware** command. If it is on, the command will not work.
3. To use the **Download IFM Firmware** command, go to **Start** | **Programs** | **MultiVOIP *n.nn*** | **Download IFM Firmware**.
4. A warning window will appear: "Downloading IFM Firmware will reboot the MultiVOIP. Do you want to continue?" Click **OK**.
5. The "Boot" LED on the front panel of the MultiVOIP will come on.
6. The software will search for an IFM firmware file to use to upgrade the system; if the file found represents firmware newer than that already installed on the MultiVOIP (or if you want to overwrite the same version of firmware) click **Open**.



7. The **IFM Firmware Download** screen will appear. Select “Copy to All IFMs” and click **OK**. (Only in very special circumstances would different IFMs in the same VOIP be loaded with different IFM firmware.)



The main MultiVOIP Configuration screen appears. Progress bars can be seen at the bottom of the screen while files are being copied. Then a completion screen entitled **IFM Test** will appear.



8. Click **OK** to close. The MultiVOIP reboots. When the reboot is complete, the MultiVOIP Configuration screen closes. The IFM firmware downloading process is complete.

Setting and Downloading User Defaults

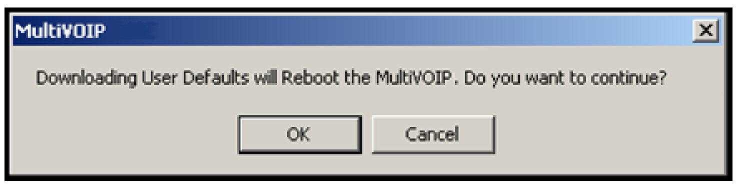
The **Download User Defaults** command allows you to maintain a known working configuration that is specific to your VOIP system. You can then experiment with alterations or improvements to the configurations confident that a working configuration can be restored if necessary.

1. Before you can use the Download User Defaults command, you must first save a set of configuration parameters by using the **Save Setup** command in the sidebar menu of the MultiVOIP software.

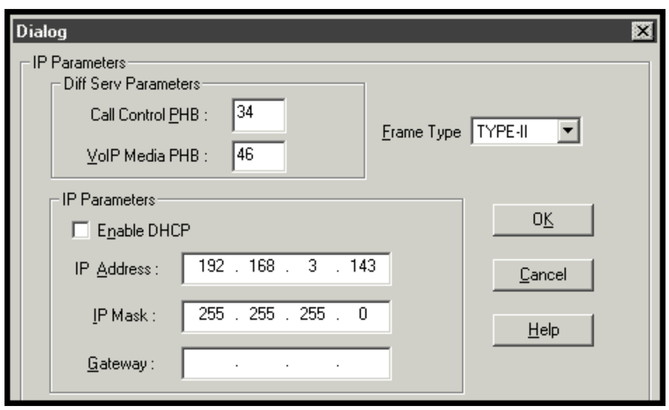
2. Before the setup configuration is saved, you will be prompted to save the setup as the User Default Configuration. Select the checkbox and click **OK**.

   A user default file will be created. The MultiVOIP unit will reboot itself.

3. To download the user defaults, go to **Start | Programs | MultiVOIP *n.nn* | Download User Defaults.**
4. A confirmation screen will appear indicating that this action will entail rebooting the MultiVOIP. Click **OK.**



5. When the file transfer process is complete, the **Dialog / IP Parameters** screen will appear.



6. Set the IP values per your particular VOIP system. Click **OK**. Progress bars will appear as the MultiVOIP reboots itself.

Setting a Password

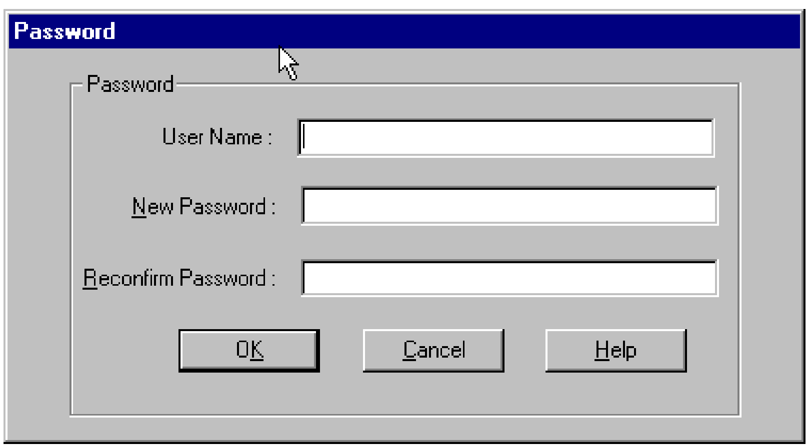
Windows Interface

After a user name has been designated and a password has been set, that password is required to gain access to any functionality of the MultiVOIP software. Only one user name and password can be assigned to a VOIP unit. The user name will be required when communicating with the MultiVOIP via the web browser interface.

Note: Record your user name and password in a safe place. If the password is lost, forgotten, or irretrievable, the user must contact Multi-Tech Tech Support in order to resume use of the MultiVOIP unit.

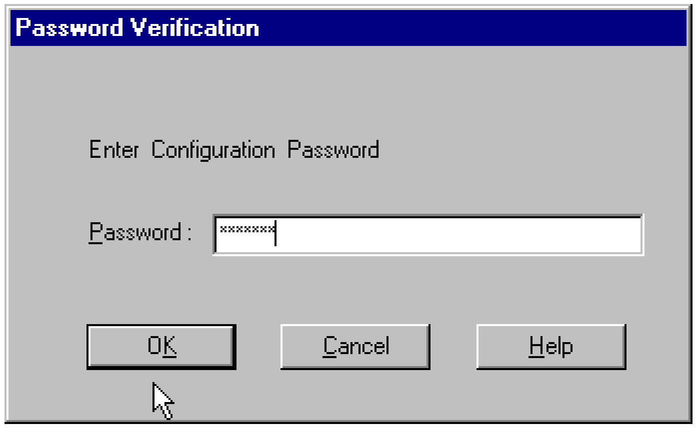
1. The MultiVOIP configuration program must be off when invoking the **Set Password** command. If it is on, the command will not work.
2. To use the Set Password command, go to Start | Programs | MultiVOIP *n.nn* | Set Password.
3. You will be prompted to confirm that you want to establish a password, which will entail rebooting the MultiVOIP (which is done automatically).

   Click **OK** to proceed with establishing a password.
4. The **Password** screen appears. If you intend to use the FTP Server function that is built into the MultiVOIP, enter a user name. (A User Name is not needed to access the local Windows interface, the web browser interface, or the commands in the **Program** group.) Type your password in the **Password** field of the **Password** screen. Type this same password again in the **Confirm Password** field to verify the password you have chosen.

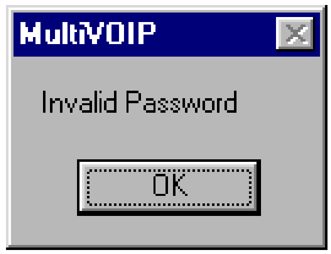
   Click **OK**.

   A message will appear indicating that a password has been set successfully.
   After the password has been set successfully, the MultiVOIP will re-boot itself and, in so doing, its **BOOT** LED will light up.
5. After the password has been set, the user will be required to enter the password to gain access to the web browser interface and any part of the MultiVOIP software listed in the **Program** group menu. User Name and Password are both needed for access to the FTP Server residing in the MultiVOIP.



When MultiVOIP program asks for password, the program shuts down if you select **CANCEL** .

An error message appears if you enter invalid password is entered.



Web Browser Interface

A passwords is optional for the MultiVOIP web browser interface. Only one password can be assigned and it works for all MultiVOIP software function. If a password has been set, it is required to access the MultiVOIP web browser interface.

NOTE: If you lose the password, the contact Multi-Tech Tech Support to resume use of the MultiVOIP web browser interface.

Upgrading Software

As noted earlier the Upgrade Software command transfers, from the controller PC to the MultiVOIP unit, firmware (including the H.323 stack) and settings. The settings can be either Factory Default Settings or Current Configuration Settings.

NOTE: To upgrade a MultiVOIP from software version *x*.04 or earlier, an ftp primer file must first be sent to the VOIP. Contact Multi-Tech Technical Support if you need this the FTP_Primer.bin file.

CAUTION: You cannot go back to x.04 or earlier versions using FTP. You must use 'upgrade software' via the serial port.

NOTE: These ftp upgrade instructions do not apply to software release x.05 and above.

FTP Server File Downloads

Multi-Tech has built an FTP server into the MultiVOIP unit. Therefore, you can use an FTP client program or even a browser to transfer files from the controller PC to the VOIP unit.

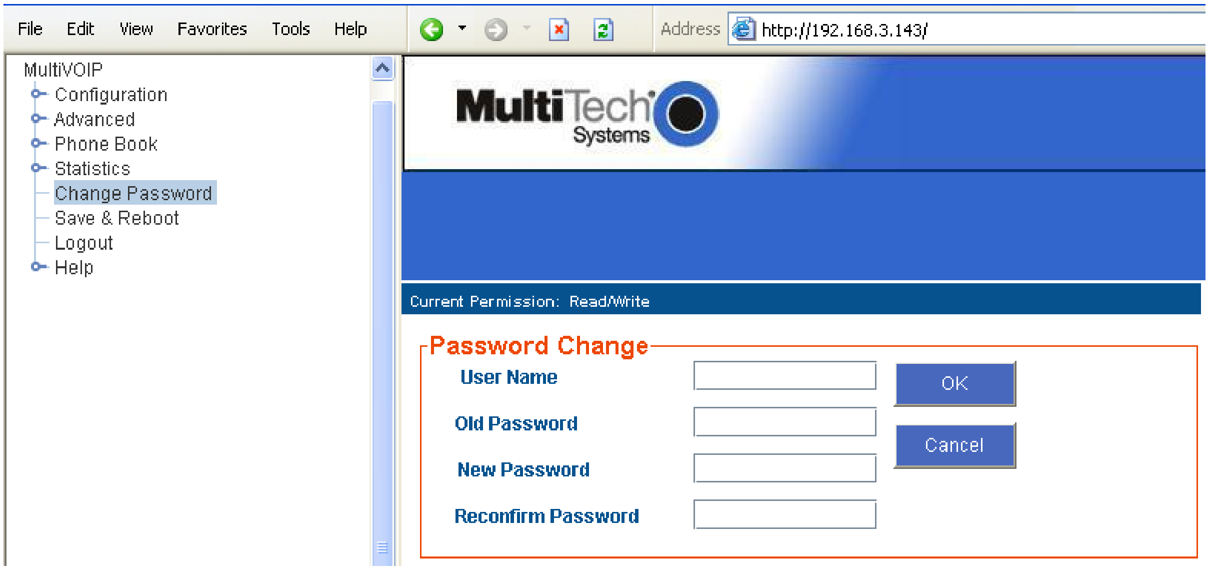
The terminology of "downloads" and "uploads" gets a bit confusing in this context. File transfers from a client to a server are typically considered "uploads." File transfers from a large repository of data to machines with less data capacity are considered "downloads." In this case, these metaphors are contradictory: the FTP server is actually housed in the MultiVOIP unit, and the controller PC, which is actually the repository of the info to be transferred, uses an FTP client program. In this situation, we have chosen to call the transfer of files from the PC to the VOIP "downloads." (Be aware that some FTP client programs may use the opposite terminology, that is, they may refer to the file transfer as an "upload ")

You can download firmware, CAS telephony protocols, default configuration parameters, and phonebook data for the MultiVOIP unit with this FTP functionality. These downloads are done over a network, not by a local serial port connection. Consequently, VOIPs at distant locations can be updated from a central control point.

The phonebook downloading feature greatly reduces the data-entry required to establish inbound and outbound phonebooks for the VOIP units within a system. Although each MultiVOIP unit will require some unique phonebook entries, most will be common to the entire VOIP system. After the phonebooks for the first few VOIP units have been compiled, phonebooks for additional VOIPs become much simpler: you copy the common material by downloading and then do data entry for the few phonebook items that are unique to that particular VOIP unit or VOIP site.

To transfer files using the FTP server function in the MultiVOIP:

1. **Establish Network Connection and IP Addresses**. Both the controller PC and the MultiVOIP unit(s) must be connected to the same IP network. An IP address must be assigned for each.
2. **Establish User Name and Password**. Establish a user name and (optionally) a password for contacting the VOIP over the IP network. (When connection is made through a local serial connection between the PC and the VOIP unit, no user name is needed.)

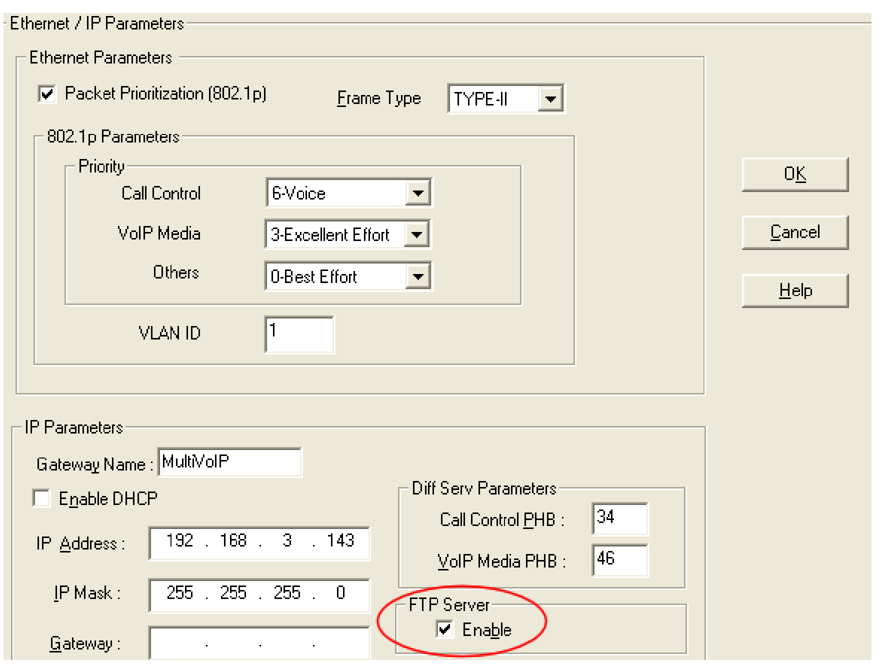


As shown above, the user name and password can be set in the web interface as well as in the Windows interface.

3. **Install FTP Client Program or Use Substitute**. You *should* install an FTP client program on the controller PC. FTP file transfers can be done using a web browser (for example, Netscape or Internet Explorer) in conjunction with a local Windows browser a (for example, Windows Explorer), but this approach is somewhat clumsy (it requires use of two application programs rather than one) and it limits downloading to only one VOIP unit at a time. With an FTP client program, multiple VOIPs can receive FTP file transmissions in response to a single command (the transfers may occur serially however).

   Although Multi-Tech does not provide an FTP client program with the MultiVOIP software or endorse any particular FTP client program, we remind our readers that adequate FTP programs are readily available under retail, shareware and freeware licenses. (Read and observe any End-User License Agreement carefully.) Two examples of this are the "WSFTP" client and the "SmartFTP" client, with the former having an essentially text-based interface and the latter having a more graphically oriented interface, as of this writing. User preferences will vary.

4. **Enable FTP Functionality**. Go to the **IP Parameters** screen and click on the "FTP Server: Enable" box.

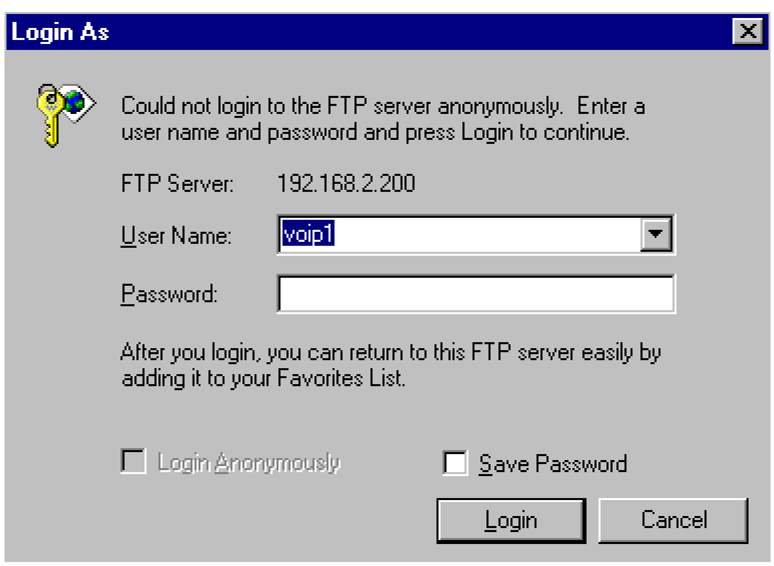


5. **Identify Files to be Updated**. Determine which files you want to update. Six types of files can be updated using the FTP feature. In some cases, the file to be transferred has "Ftp" as the part of its filename just before the suffix (or extension). So, for example, the file "mvpt1Ftp.bin" can be transferred to update the bin file (firmware) residing in the MultiVOIP. Similarly, the file "fxo_loopFtp.cas" could be transferred to enable use of the FXO Loop Start telephony interface in one of the analog VOIP units and the file "r2_brazilFtp.cas" could be transferred to enable a particular telephony protocol used in Brazil. Note, however, that before any CAS file can be used as an update, it must be renamed to CASFILE.CAS so that it overwrites and replaces the default CAS file.

File Type	File Names	Description
firmware "bin" file	mvpt1Ftp.bin	This is the MultiVOIP firmware file. Only one file of this type will be in the directory.
factory defaults	fdefFtp.cnf	This file contains factory default settings for user-changeable configuration parameters. Only one file of this type will be in the directory.
CAS file	fxo_loopFtp.cas, em_winkFtp.cas, r2_brazilFtp.cas r2_chinaFtp.cas	These telephony files are for Channel Associated Signaling. The directory contains many CAS files, some labeled for specific functionality, others for countries or regions where certain attributes are standard. Any CAS file used must first be renamed to "CASFILE.CAS."
inbound phonebook	InPhBk.tmr	This file updates the inbound phonebook in the MultiVOIP unit.
outbound phonebook	OutPhBk.tmr	This file updates the outbound phonebook in the MultiVOIP unit.

6. **Contact MultiVOIP FTP Server**. You must make contact with the FTP Server in the VOIP using either a web browser or FTP client program. Enter the IP address of the MultiVOIP's FTP Server. If you are using a browser, the address must be preceded by "ftp://" (otherwise you'll reach the web interface within the MultiVOIP unit).

7. **Log In**. Use the User Name and password. The login screens differ, depending on whether the FTP file transfer is to be done with a web browser (shown below) or with an FTP client program (varies).

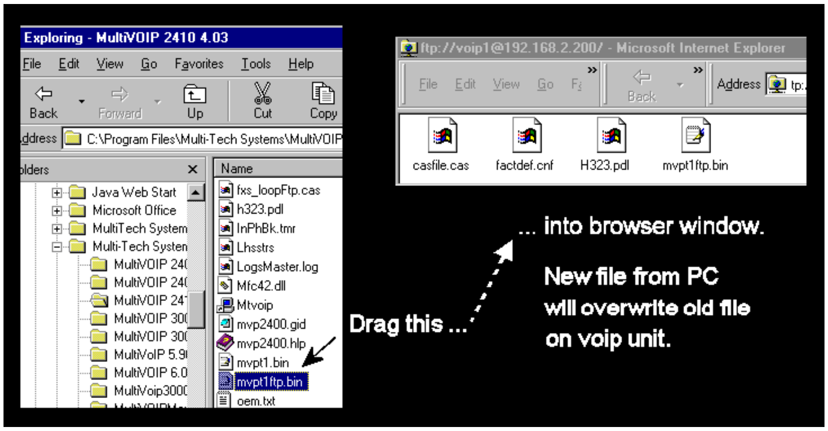


8. **Use Download**. Downloading can be done with a web browser or with an FTP client program.

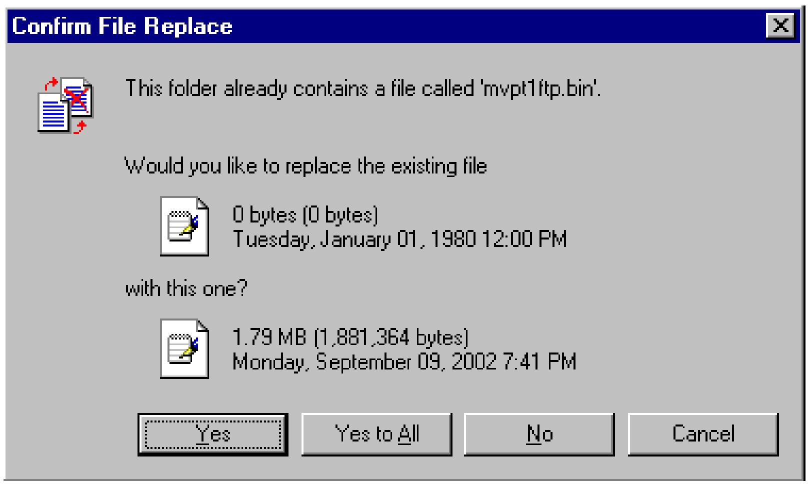
Download with Web Browser:

a. In the local Windows browser, locate the directory holding the MultiVOIP program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP *xxxx* yyyy (where x and y represent MultiVOIP model numbers and software version numbers).

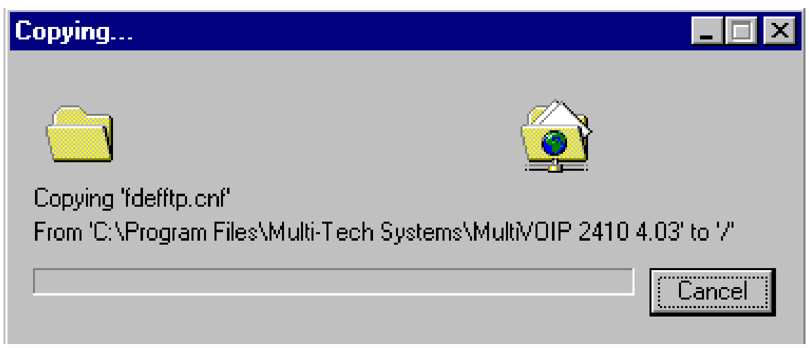
b. Drag-and-drop files from the local Windows browser (for example, Windows Explorer) to the web browser.



c. You may be asked to confirm the overwriting of files on the MultiVOIP. Do so.



d. File transfer between PC and VOIP will look like transfer within VOIP directories.

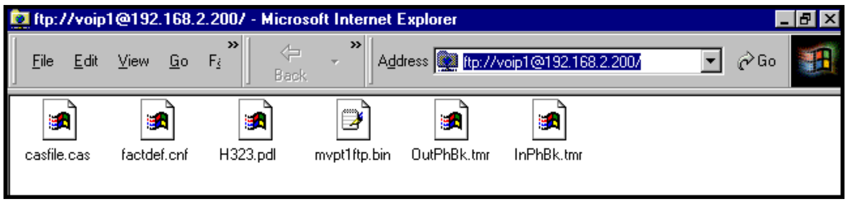


Download with FTP Client Program:

a. In the local directory browser of the FTP client program, locate the directory holding the MultiVOIP program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP xxxx yyyy (where x and y represent MultiVOIP model numbers and software version numbers).

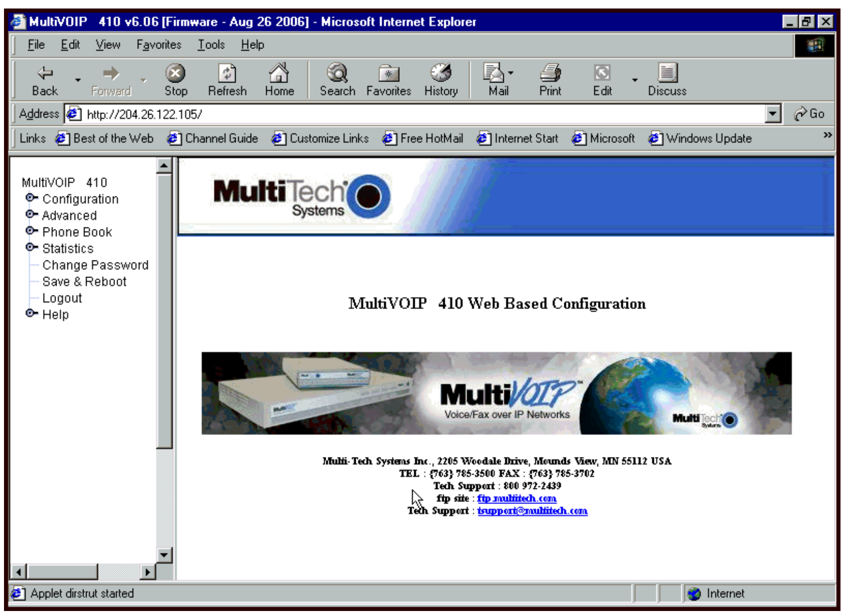
b. In the FTP client program window, drag-and-drop files from the local browser pane to the pane for the MultiVOIP FTP server. FTP client interface operations vary. In some cases, you can choose between immediate and queued transfer. In some cases, there may be automated capabilities to transfer to multiple destinations with a single command.

9. **Verify Transfer**. The files transferred appear in the directory of the MultiVOIP.



10. **Log Out of FTP Session**. Whether the file transfer was done with a web browser or with an FTP client program, you *must* log out of the FTP session before opening the MultiVOIP Windows interface.

Web Browser Interface



You can control the MultiVOIP unit with a graphic user interface (interface) based on the common web browser platform. Qualifying browsers are Internet Explorer 6+, Netscape 6+, and Mozilla Firefox 1.0+.

Function	Remote configuration and control of MultiVOIP units.
Configuration Prerequisite	Local Windows interface must be used to assign IP address to MultiVOIP.
Browser Version Requirement	Internet Explorer 6.0 or higher; or Netscape 6.0 or higher; or Mozilla Firefox 1.0 or higher.
Java Requirement	Java Runtime Environment version 1.4.0_01 or higher (this application program is included with MultiVOIP)

The initial configuration step of assigning the VOIP unit an IP address must still be done locally using the Windows interface. However, all additional configurations can be done via the web interface.

The content and organization of the web interface is directly parallel to the Windows interface. For each screen in the Windows interface, there is a corresponding screen in the web interface. The fields on each screen are the same, as well.

The Windows interface gives access to commands via icons and pull-down menus whereas the web interface does not. The web interface, however, cannot perform logging in the same direct mode done in the Windows interface. However, when the web interface is used, logging can be done by email (SMTP).

The graphic layout of the web interface is also somewhat larger-scale than that of the Windows interface. For that reason, it’s helpful to use as large of a video monitor as possible.

The primary advantage of the web interface is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

To use the web interface, install a Java application program on the controller PC. Java supports drop-down menus and multiple windows in the web interface.

When installation is complete, the Java program runs automatically in the background as a plug-in supporting the MultiVOIP web interface. No user actions are required.

After the Java program has been installed, you can access the MultiVOIP using the web browser interface. Close the MultiVOIP Windows interface. Start the web browser. Enter the IP address of the MultiVOIP unit. Enter a password when prompted. (A password is needed here only if password has been set for the local Windows interface or for the MultiVOIP’s FTP Server function. See “Setting a Password -- Web Browser interface” earlier in this chapter.) The web browser interface offers essentially the same control over the VOIP as can be achieved using the Windows interface. As noted earlier, logging functions cannot be handled via the web interface. And, because network communications will be slower than direct communications over a serial PC cable, command execution will be somewhat slower over the web browser interface than with the Windows interface.

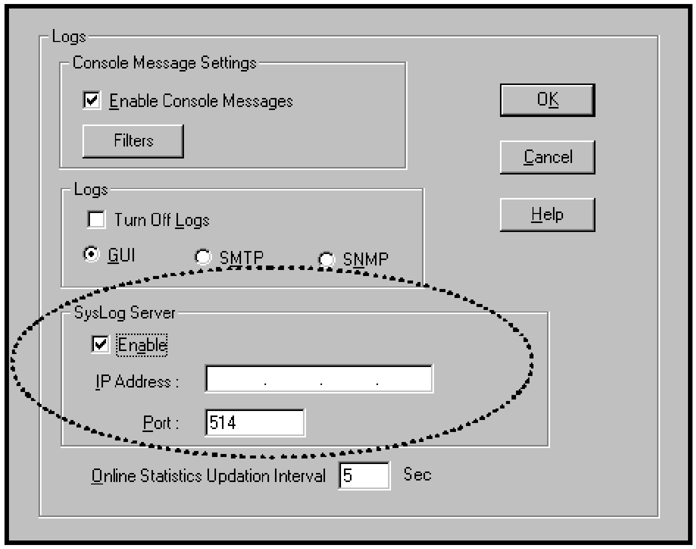
SysLog Server Functions

Multi-Tech has built SysLog server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.

The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a “daemon”). SysLog client programs, both paid and freeware can be obtained from Kiwi Enterprises (search the Internet for kiwi syslog daemon), among other firms. Read the End-User License Agreement carefully and observe license requirements. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

Multi-Tech Systems does not endorse any particular SysLog client program. A SysLog client programs from qualified providers most likely work with MultiVOIP units.

Before using a SysLog client program, enable the SysLog function within the MultiVOIP in the **Logs** menu under **Configuration**.



The IP Address used is that of the MultiVOIP itself.

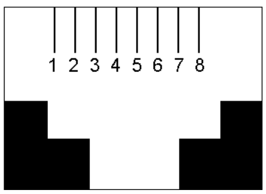
In the **Port** field, entered by default, is the standard (‘well-known’) logical port, 514.

Configuring the SysLog Client Program. Configure the SysLog client program for your own needs. In various SysLog client programs, you can define where log messages are saved and archived, opt for interaction with an SNMP system (like MultiVOIP Manager), set the content and format of log messages, determine disk space allocation limits for log messages, and establish a hierarchy for the seriousness of messages (normal, alert, critical, emergency, and so on.).

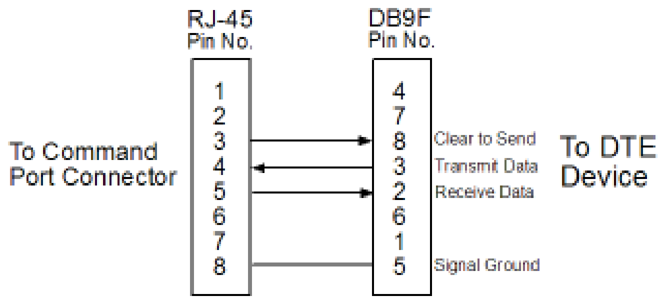
Appendix A – Cable Pin-outs

Command Cable

RJ-45 Connector



End-to-End Pin Info

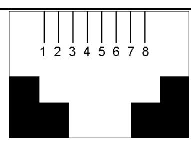


RJ-45 connector plugs into Command Port of MultiVOIP.

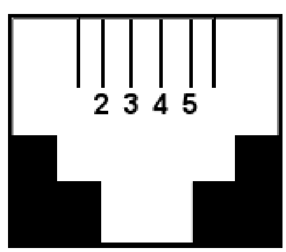
DB-9 connector plugs into serial port of command PC (which runs MultiVOIP configuration software).

Ethernet Connector

The functions of the individual conductors of the MultiVOIP's Ethernet port are shown on a pin-by-pin basis below.

RJ-45 Ethernet Connector	Pin	Circuit Signal Name
	1	TD+ Data Transmit Positive
	2	TD- Data Transmit Negative
	3	RD+ Data Receive Positive
	6	RD- Data Receive Negative

Voice/Fax Channel Connectors



FXS Pin	Description	FXO Pin	Description
2	N/C	2	N/C
3	Ring	3	Tip
4	Tip	4	Ring
5	N/C	5	N/C

Appendix B – TCP/UDP Port Assignments

Well Known Port Numbers

The following description of port number assignments for Internet Protocol (IP) communication is taken from the Internet Assigned Numbers Authority (IANA) web site (www.iana.org).

> “The Well Known Ports are assigned by the IANA and on most systems can only be used by system (or root) processes or by programs executed by privileged users. Ports are used in the TCP [RFC793] to name the ends of logical connections which carry long term conversations. For the purpose of providing services to unknown callers, a service contact port is defined. This list specifies the port used by the server process as its contact port. The contact port is sometimes called the "well-known port". To the extent possible, these same port assignments are used with the UDP [RFC768]. The range for assigned ports managed by the IANA is 0-1023.”

Well-known port numbers especially pertinent to MultiVOIP operation are listed below.

Port Number Assignment List

Function	**Port Number**
telnet	23
tftp	69
snmp	161
snmp tray	162
gatekeeper registration	1719
H.323	1720
SIP	5060
SysLog	514

Appendix C – Regulatory Information

EMC, Safety, and R&TTE Directive Compliance

The CE mark is affixed to this product to confirm compliance with the following European Community Directives:

Council Directive 2004/108/EC of 15 December 2004 on the approximation of the laws of Member States relating to electromagnetic compatibility;

and

Council Directive 2006/95/EC of 12 December 2006 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits;

and

Council Directive 1999/5/EC of 9 March 1999 on radio equipment and telecommunications terminal equipment and the mutual recognition of their conformity.

FCC Part 15 Declaration

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Plug the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This device complies with Part 15 of the 47 CFR rules. Operation of this device is subject to the following conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference that may cause undesired operation.

Warning: Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user’s authority to operate the equipment.

FCC Part 68 Telecom

1. This equipment complies with Part 68 of the 47 CFR rules and the requirements adopted by the ACTA. Located on this equipment is a label that contains, among other information, the registration number and Ringer Equivalence Number (REN) for this equipment or a product identifier in the format:

   For current products: US:AAAEQ##Txxxx.

   For legacy products: AU7USA-xxxxx-xx-x

   If requested, this number must be provided to the telephone company.
2. A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable 47 CFR Part 68 rules and requirements adopted by the ACTA. It's designed to be connected to a compatible modular jack that is also compliant.
3. The Ringer Equivalence Number (REN) is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. For products approved after July 23, 2001, the REN for this product is part of the product identifier that has the format US:AAAEQ##Txxxx. The digits represented by ## are the REN without a decimal point (e.g., 03 is a REN of 0.3). For earlier products, the REN is separately shown on the label.
4. If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.
5. The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.
6. If trouble is experienced with this equipment, please contact Multi-Tech Systems, Inc. at the address shown below for details of how to have the repairs made. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.
7. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.
8. No repairs are to be made by you. Repairs are to be made only by Multi-Tech Systems or its licensees. Unauthorized repairs void registration and warranty.
9. If your home has specially wired alarm equipment connected to the telephone line, ensure the installation of this equipment does not disable your alarm equipment. If you have questions about what will disable alarm equipment, consult your telephone company or a qualified installer.
10. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.
11. This equipment is hearing aid compatible.
12. Manufacturing Information on telecommunications device (modem):

Manufacturer:	Multi-Tech Systems
Trade Name:	MultiVOIP
Model Number:	MTIFM

Registration Number:	US:AU7CN06BMTIFM
Ringer Equivalence:	0.6B
Modular Jack (USOC):	RJ-48C
Service Center in USA:	Multi-Tech Systems, Inc. 2205 Woodale Drive Mounds View, MN 55112 USA (763)785-3500 (763) 785-9874 (Fax)

Industry Canada

This Class A digital apparatus meets all requirements of the Canadian Interference-Causing Equipment Regulations.

Cet appareil numérique de la classe A respecte toutes les exigences du Reglement Canadien sur le matériel brouilleur.

Canadian Limitations Notice

Notice: This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

Notice: The REN assigned to each terminal equipment provides an indication of the maximum number of terminals allowed to be connected to a telephone interface. The termination on an interface may consist of any combination of devices subject only to the requirement that the sum of the Ringer Equivalence Numbers of all the devices does not exceed five.

Restrictions concernant le raccordement de matériel

Avis: Le présent matériel est conforme aux spécifications techniques d'Industrie Canada applicables au matériel terminal. Cette conformité est confirmée par le numéro d'enregistrement. Le sigle IC, placé devant le numéro d'enregistrement, signifie que l'enregistrement s'est effectué conformément à une déclaration de conformité et indique que les spécifications techniques d'Industrie Canada ont été respectées. Il n'implique pas qu'Industrie Canada a approuvé le matériel.

Avis: L'IES assigné à chaque dispositif terminal indique le nombre maximal de terminaux qui peuvent être raccordés à une interface téléphonique. La terminaison d'une interface peut consister en une combinaison quelconque de dispositifs, à la seule condition que la somme d'indices d'équivalence de la sonnerie de tous les dispositifs n'excède pas 5.

Waste Electrical and Electronic Equipment Statement

WEEE Directive

The WEEE Directive places an obligation on EU-based manufacturers, distributors, retailers, and importers to take-back electronics products at the end of their useful life. A sister directive, ROHS (Restriction of Hazardous Substances) complements the WEEE Directive by banning the presence of specific hazardous substances in the products at the design phase. The WEEE Directive covers all Multi-Tech products imported into the EU as of August 13, 2005. EU-based manufacturers, distributors, retailers and importers are obliged to finance the costs of recovery from municipal collection points, reuse, and recycling of specified percentages per the WEEE requirements.

Instructions for Disposal of WEEE by Users in the European Union

The symbol shown below is on the product or on its packaging, which indicates that this product must not be disposed of with other waste. Instead, it is the user's responsibility to dispose of their waste equipment by handing it over to a designated collection point for the recycling of waste electrical and electronic equipment. The separate collection and recycling of your waste equipment at the time of disposal will help to conserve natural resources and ensure that it is recycled in a manner that protects human health and the environment. For more information about where you can drop off your waste equipment for recycling, please contact your local city office, your household waste disposal service or where you purchased the product.

July, 2005

Restriction of the Use of Hazardous Substances (RoHS)

Multi-Tech Systems, Inc.
Certificate of Compliance
2011/65/EU

Multi-Tech Systems confirms that its embedded products comply with the chemical concentration limitations set forth in the directive 2011/65/EU of the European Parliament (Restriction of the use of certain Hazardous Substances in electrical and electronic equipment - RoHS)

These Multi-Tech products do not contain the following banned chemicals[1]:

- Lead, [Pb] < 1000 PPM
- Mercury, [Hg] < 1000 PPM
- Hexavalent Chromium, [Cr+6] < 1000 PPM
- Cadmium, [Cd] < 100 PPM
- Polybrominated Biphenyl, [PBB] < 1000 PPM
- Polybrominated Diphenyl Ether, [PBDE] < 1000 PPM

Environmental considerations:

- Moisture Sensitivity Level (MSL) =1
- Maximum Soldering temperature = 260C (in SMT reflow oven)

[1]Lead usage in some components is exempted by the following RoHS annex, therefore higher lead concentration would be found in some modules (>1000 PPM);

–Resistors containing lead in a glass or ceramic matrix compound.

Information on HS/TS Substances According to Chinese Standards

In accordance with China's Administrative Measures on the Control of Pollution Caused by Electronic Information Products (EIP) # 39, also known as China RoHS, the following information is provided regarding the names and concentration levels of Toxic Substances (TS) or Hazardous Substances (HS) which may be contained in Multi-Tech Systems Inc. products relative to the EIP standards set by China's Ministry of Information Industry (MII).

Component Name	Hazardous/Toxic Substance/Elements					
	Lead (PB)	Mercury (Hg)	Cadmium (CD)	Hexavalent Chromium (CR6+)	Polybrominated Biphenyl (PBB)	Polybrominated Diphenyl Ether (PBDE)
Printed Circuit Boards	O	O	O	O	O	O
Resistors	X	O	O	O	O	O
Capacitors	X	O	O	O	O	O
Ferrite Beads	O	O	O	O	O	O
Relays/Opticals	O	O	O	O	O	O
ICs	O	O	O	O	O	O
Diodes/ Transistors	O	O	O	O	O	O
Oscillators and Crystals	X	O	O	O	O	O
Regulator	O	O	O	O	O	O
Voltage Sensor	O	O	O	O	O	O
Transformer	O	O	O	O	O	O
Speaker	O	O	O	O	O	O
Connectors	O	O	O	O	O	O
LEDs	O	O	O	O	O	O
Screws, Nuts, and other Hardware	X	O	O	O	O	O
AC-DC Power Supplies	O	O	O	O	O	O
Software / Documentation CDs	O	O	O	O	O	O
Booklets and Paperwork	O	O	O	O	O	O
Chassis	O	O	O	O	O	O

X Represents that the concentration of such hazardous/toxic substance in all the units of homogeneous material of such component is higher than the SJ/Txxx-2006 Requirements for Concentration Limits.

O Represents that no such substances are used or that the concentration is within the aforementioned limits.

Information on HS/TS Substances According to Chinese Standards (in Chinese)

依照中国标准的有毒有害物质信息

根据中华人民共和国信息产业部 (MII) 制定的电子信息产品 (EIP) 标准—中华人民共和国《电子信息产品污染控制管理办法》（第 39 号），也称作中国 RoHS，下表列出了 Multi-Tech Systems, Inc. 产品中可能含有的有毒物质 (TS) 或有害物质 (HS) 的名称及含量水平方面的信息。

成分名称	有害/有毒物质/元素					
	铅 (PB)	汞 (Hg)	镉 (CD)	六价铬 (CR6+)	多溴联苯 (PBB)	多溴二苯醚 (PBDE)
印刷电路板	O	O	O	O	O	O
电阻器	X	O	O	O	O	O
电容器	X	O	O	O	O	O
铁氧体磁环	O	O	O	O	O	O
继电器/光学部件	O	O	O	O	O	O
IC	O	O	O	O	O	O
二极管/晶体管	O	O	O	O	O	O
振荡器和晶振	X	O	O	O	O	O
调节器	O	O	O	O	O	O
电压传感器	O	O	O	O	O	O
变压器	O	O	O	O	O	O
扬声器	O	O	O	O	O	O
连接器	O	O	O	O	O	O
LED	O	O	O	O	O	O
螺丝、螺母以及其它五金件	X	O	O	O	O	O
交流-直流电源	O	O	O	O	O	O
软件/文档 CD	O	O	O	O	O	O
手册和纸页	O	O	O	O	O	O
底盘	O	O	O	O	O	O

X 表示所有使用类似材料的设备中有害/有毒物质的含量水平高于 SJ/Txxx-2006 限量要求。

O 表示不含该物质或者该物质的含量水平在上述限量要求之内。