MultiVOIP™

Voice / Fax over IP Networks

User Guide for Voice/IP Gateways

Digital Models (T1, E1, ISDN-PRI):
MVP-2400/2410/3010

Analog/BRI Models: MVP-130/210/410/810
MVP-210G/410G/810G
MVP-410ST/810ST

MultiTech® Systems

User Guide

S000249H

Analog MultiVOIP Units (Models MVP130, MVP210, MVP410, MVP810, MVP210G, MVP410G, and MVP810G)
ISDN-BRI MultiVOIP Units (Models MVP410ST, and MVP810ST)
Digital MultiVOIP Units (Models MVP2400, MVP2410, & MVP3010)
Upgrade Units (MVP24-48 and MVP30-60)

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Record of Revisions

Revision	Description
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B	Index added. (05/24/02)
C	Updated for 4.03/6.03 software. (10/11/02)
D	Updated for 4.04/6.04/8.04/9.04 software. (03/20/03) Add embedded gatekeeper models, ISDN-BRI models, MultiVantage Apx., SPP protocol, & Call State Apx.
E	Remove MultiVantage. (04/18/03)
F	Update ISDN-BRI info in SW version 5.02c. (06/04/03)
G	Add MVP130 information. (06/30/03)
H	Revisions to ISDN-BRI & MVP130 content. (08/15/03)

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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Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, Minnesota 55112
(763) 785-3500 or (800) 328-9717
U.S. Fax: 763-785-9874
Technical Support: (800) 972-2439
http://www.multitech.com

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Chapter 1: Overview

About This Manual

This manual is about Voice-over-IP products made by Multi-Tech Systems, Inc. It describes four product groups.

1. T1 Digital MultiVOIP units, models MVP2400, MVP2410, and the capacity-doubling add-on expansion card, model MVP24-48 (which fits the MVP2410 only).
2. E1 Digital MultiVOIP units, models, MVP3010 and the capacity-doubling add-on expansion card, model MVP30-60.
3. Analog MultiVOIP units,
   models MVP810, MVP410, MVP210, & MVP130 and
   models MVP810G, MVP410G, & MVP210G with embedded gatekeeper function.
4. ISDN-BRI MultiVOIP units, models MVP410ST & MVP810ST.

The table below describes the vital characteristics of these various models.

MultiVOIP Product Family					
Description / Model	MVP 2400	MVP-2410	MVP 24-48	MVP 3010	MVP 30-60
Function	T1 digital VOIP unit	T1 digital VOIP unit	T1 digital VOIP add-on card	E1 digital VOIP unit	E1 digital VOIP add-on card
Capacity	24 channels	24 channels	24 added channels	30 channels	30 added channels
Chassis/ Mounting	Table top	19" 1U rack mount	circuit card only	19" 1U rack mount	circuit card only
Description / Model	MVP 810 (G)	MVP 428 (G)	MVP 410 (G)	MVP 210 (G)	MVP 130
Function	analog voip	add-on card	analog voip	Analog voip	Analog voip
Capacity	8 channels	4 added channels	4 channels	2 channels	1 channel
Chassis/ Mounting	19" 1U rack mount	circuit card only	19" 1U rack mount	Table top	table top
Description / Model	MVP810ST		MVP410ST		
Function	ISDN-BRI voip		ISDN-BRI voip		
Capacity	4 ISDN lines (8 B-channels)		2 ISDN lines (4 B-channels)		
Chassis/ Mounting	19" 1U rack mount		19" 1U rack mount		
	1. "G" models have embedded Gatekeeper. 2. "BRI" means Basic Rate Interface.				

How to Use This Manual. *In short, use the index and the examples*.
When our readers crack open this large manual, they generally need one of two things: information on a very specific software setting or technical parameter (about telephony or IP) *or* they need help when setting up phonebooks for their voip systems. The index gives quick access to voip settings and parameters. It's detailed. Use it. The best way to learn about phonebooks is to wade through examples like those in our chapters on T1 (North American standard) Phonebooks and E1 (Euro standard) Phonebooks. Also, the quick setup info of the printed Quick Start Guide is replicated in this manual for your convenience. Finally, this manual is meant to be comprehensive. If you notice that something important is lacking, please let us know.

Additional Resources. The MultiTech web site (www.multitech.com) offers both a list of Frequently Asked Questions (the MultiVOIP FAQ) and a collection of resolutions of issues that MultiVOIP users have encountered (these are Troubleshooting Resolutions in the searchable Knowledge Base).

Variable Model/Version Icon and Typography. The MultiVOIP product family is a coordinated set of products that can operate with each other in a seamless fashion. For example, both the digital and analog MultiVOIP units use the same graphic user interface (GUI) in the MultiVOIP configuration software and both operate under a single GUI in the MultiVoipManager remote management software. Because this is the case, the various model numbers and version numbers of MultiVOIP family products will each appear in various dialog boxes and commands. But instead of showing these dialog boxes once for each model in this manual, we substitute the following icon.

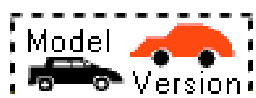


Figure 1-1: Variable Model/Version Icon

It indicates that, whatever MultiVOIP model you are using, all details except the very model and version numbers themselves will be the same regardless of the MultiVOIP model used. Also, in some cases, we will use other typographic devices, like blank underlining ("MultiVOIP ____") to denote information that applies to any and all of the products in this product family.

Introduction to TI MultiVOIPs (MVP2400, MVP2410, & MVP24-48)

We proudly present MultiTech's T1 Digital Multi-VOIP products.
The MVP2400 is a tabletop model; the MVP2410 is a rack-mount model; and the MVP24-48 is an add-on expansion card that doubles the capacity of the MVP2410 without adding another chassis. All of these voice-over-IP products have fax capabilities. All of these models adhere to the North American standard of T1 trunk telephony using digital 24-channel time-division multiplexing, which allows 24 phone conversations to occur on the T1 line simultaneously. All can also accommodate T1 lines of the ISDN Primary Rate Interface type (ISDN-PRI).

Scale-ability. The MVP2400 and MVP2410 are tailored to companies needing more than a few voice-over-IP lines, but not needing carrier-class equipment. When expansion is needed, the MVP2410 can be field-upgraded into a dual T1 unit by installing the MVP24-48 kit, which is essentially a second MultiVOIP motherboard that fits in an open expansion-card slot in the MVP2410. The upgraded dual unit then accommodates two T1 lines.

T1 VOIP Traffic. The MVP-2400/2410 accepts its outbound traffic from a T1 trunk that's connected to either a PBX or to a telco/carrier. The MVP-2400/2410 transforms the telephony signals into IP packets for transmission on LANs, WANs, or the Internet. Inbound IP data traffic is converted to telephony data and signaling.

When connected to PBX. When connected to a PBX, the MVP-2400/2410 creates a network node served by 10/100-Base T connections. Local PBX phone extensions gain toll-free access to all phone stations directly connected to the VOIP network. Phone extensions at any VOIP location also gain toll-free access to the entire local public-switched telephone network (PSTN) at every other VOIP location in the system.

When connected to PSTN. When the T1 line(s) connected to the MVP-2400/2410 are connected directly to the PSTN, the unit becomes a Point-of-Presence server dedicated to local calls off-net.

H.323, SIP & SPP. Being H.323 compatible, the MVP-2400/2410 can place calls to telephone equipment at remote IP network locations that also contain H.323 compatible voice-over-IP gateways. It will interface with H.323 software and H.323 gatekeeper units. H.323 specifications also bring to voip telephony many special features common to conventional telephony. H.323 features of this kind that have been implemented into the MultiVOIP include Call Hold, Call Waiting, Call Name Identification, Call Forwarding (from the H.450 standard), and Call Transfer (H.450.2 from H.323 Version 2). The fourth version of the H.323 standard improves system resource usage (esp. logical port or socket usage) by handling call signaling more compactly and allowing use of the low-overhead UDP protocol instead of the error-correcting TCP protocol where possible.

The MultiVOIP is also SIP-compatible. ("SIP" means Session Initiation Protocol.) However, H.450 Supplementary Services features can be used under H.323 only and not under SIP.

SPP (Single-Port Protocol) 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. SPP offers advantages in certain situations, especially when firewalls are used and when dynamic IP address assignment is needed. However, when SPP is used, certain features of SIP and H.323 will not be available and SPP will not inter-operate with voip systems using H.323 or SIP.

Data Compression & Quality of Service. The MultiVOIP2400/2410 comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

VOIP Functions. The MultiVOIP MVP-2400/2410 gateway performs four basic functions: (a) it converts a dialed number into an IP address, (b) it sends voice over the data network, (c) it establishes a connection with another VOIP gateway at a remote site, and (d) it receives voice over the data network. Voice is handled as IP packets with a variety of compression options. Each T1 connection to the MultiVOIP provides 24 time-slot channels to connect to the telco or to serve phone or fax stations connected to a PBX.

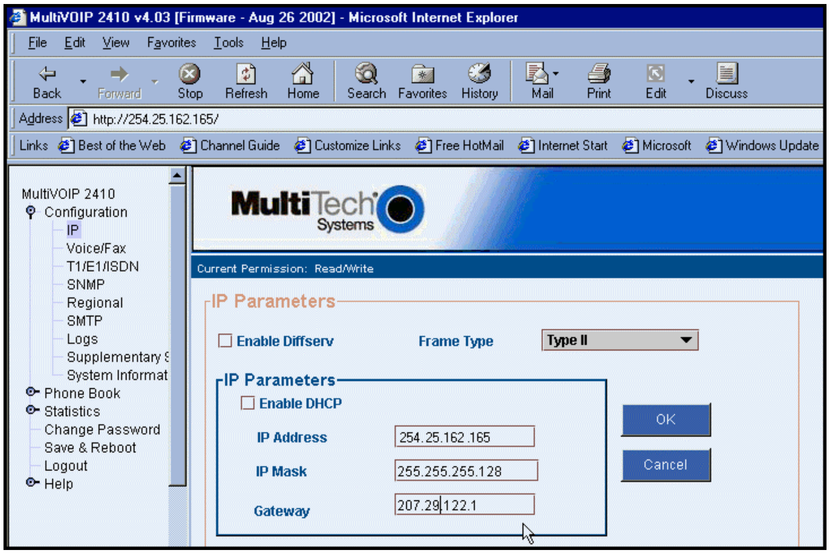
Ports. The MVP2400 and MVP2410 each have one 10/100 Mbps Ethernet LAN interface and one Command port for configuration. An MVP2410 upgraded with the MVP24-48 kit will have two Ethernet LAN interfaces and two Command ports.

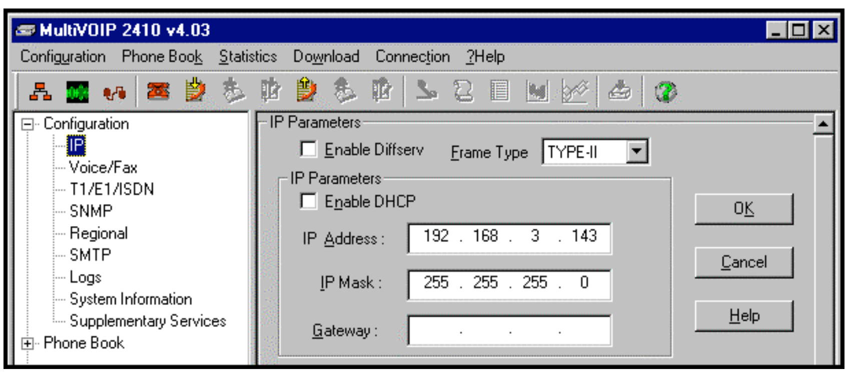
PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

Gatekeeper. T1 voip systems can have gatekeeper functionality either by adding, as an endpoint, either a Multi-Tech standalone gatekeeper (special software residing in separate hardware), or an analog gateway with embedded gatekeeper functionality (MVP210G, MVP410G, or MVP810G). Gatekeepers are optional but useful within voip systems. The gatekeeper acts as the ‘clearinghouse’ for all calls within its zone. MultiTech’s embedded and stand-alone gatekeeper software packages both perform all of the standard gatekeepers functions (address translation, admission control, bandwidth control, and zone management) and also support many valuable optional functions (call control signaling, call authorization, bandwidth management, and call management). The stand-alone gatekeeper is, however, slightly more feature-rich than the embedded gatekeeper. For more details, see the “Embedded Gatekeeper” chapter of this manual and the manual on MultiTech’s stand-alone gatekeeper.

Management. Configuration and system management can be done locally with the MultiVOIP configuration software. After an IP address has been assigned locally, other configuration can be done remotely using the MultiVOIP web browser GUI. Remote system management can be done with the MultiVoipManager SNMP software or via the MultiVOIP web browser GUI. All of these control software packages are included on the Product CD.

While the web GUI's appearance differs slightly, its content and organization are essentially the same as that of the Windows GUI (except for logging).

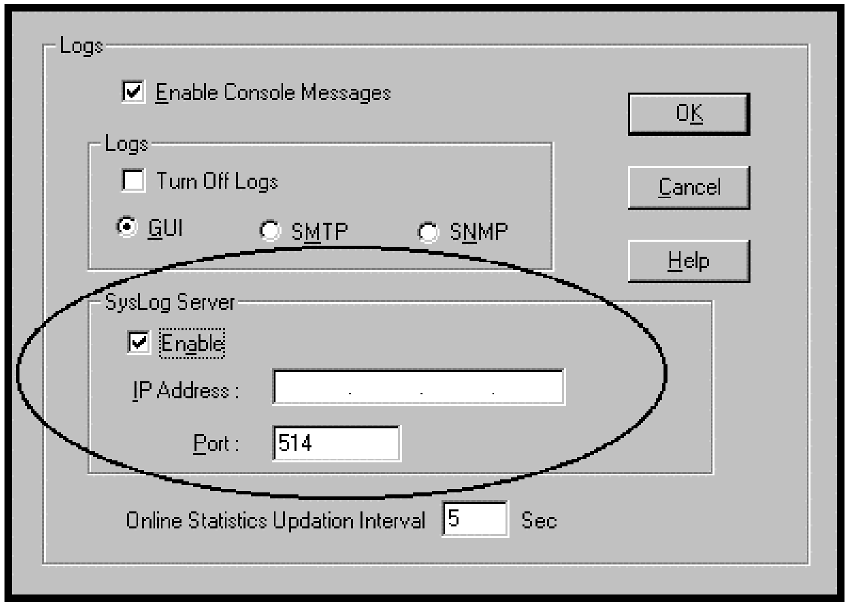




The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

Once you've begun using the web browser GUI, you can go back to the MultiVOIP Windows GUI at any time. However, you must log out of the web browser GUI before using the MultiVOIP Windows GUI.

Logging of System Events. MultiTech 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, among other firms. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by any qualified provider should suffice for use with MultiVOIP units. Kiwi's brief description of their SysLog program indicates the typical scope of such programs. "Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available."

Supplementary Telephony Services. The H.450 standard (an addition to H.323) brings to voip telephony more of the premium features found in PSTN and PBX telephony. MultiVOIP units offer five of these H.450 features: Call Transfer, Call Hold, Call Waiting, Call Name Identification (not the same as Caller ID), and Call Forwarding. (The first four features are found in the “Supplementary Services” window; the fifth, Call Forwarding, appears in the Add/Edit Inbound phonebook screen.) Note that the first three features are closely related. All of these H.450 features are supported for H.323 operation only; they are *not* supported for SIP or SPP.

T1 Front Panel LEDs

The MVP2400, MVP2410, and MVP24-48 all use a common main circuit board or motherboard. Consequently the LED indicators are the same for all.

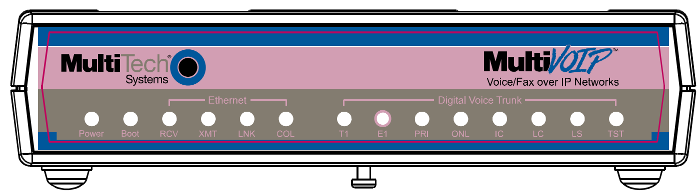


Figure 1-2. MultiVOIP MVP2400 Front Panel

Active LEDs. The MVP2410 front panel has two sets of identical LEDs. In the MVP2410 as shipped (that is, without an expansion card), the left-hand set of LEDs is functional whereas the right-hand set is not.

When the MVP2410 has been upgraded with an MVP24-48 kit, the right-hand set of LEDs will also become active.

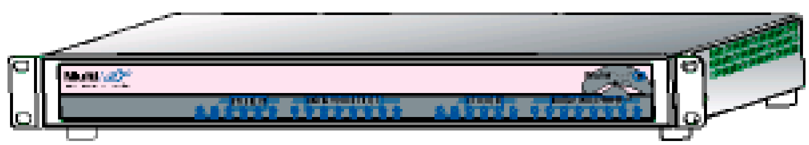


Figure 1-3. MultiVOIP MVP2410x Chassis

T1 LED Descriptions

The descriptions below apply to all digital T1 MultiVOIP units. The MVP2410 has four sets of LEDs plus a lone LED at its far right end. As viewed from the front of the MVP2410, it is the two left groups that are active and present feedback about the operation of the unit. If an MVP24-48 expansion card is added to the MVP2410, the two LED groups on the right become operational with respect to the second T1 connection.

MVP2400/2410 Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on for about 10 seconds while the MVP2400/2410 is booting.
RCV	Receive. Lights when receiving data on Ethernet port.
XMT	Transmit. Lights when transmitting data on Ethernet port.
LNK	Link. When lit, VOIP “sees” the hub or network via the Ethernet connection.
COL	Collision. Lit when data collisions occur.
T1	When lit, indicates presence of T1 connection.
E1	E1. Not supported.
PRI	PRI. On if T1 line is of ISDN-Primary-Rate type.
ONL	Online. This LED is on when frame synchroni-zation has been established on the T1/E1 link.
IC	IC LED is on when Internal Clocking is selected in T1/E1 configuration.
LC	Indicates Loss of Carrier.
LS	Indicates Loss of Signal.
Test	For testing purposes only.

Introduction to El MultiVOIPs (MVP3010 & MVP30-60)

We proudly present MultiTech's E1 Digital Multi-VOIP products. The MVP3010 is a rack-mount model and the MVP30-60 is an add-on expansion card that doubles the capacity of the MVP3010 without adding another chassis. All of these voice-over-IP products have fax capabilities. All adhere to the European standard of E1 trunk telephony using digital 30-channel time-division multiplexing, which allows 30 phone conversations to occur on the E1 line simultaneously. All can also accommodate E1 lines of the ISDN Primary Rate Interface type (ISDN-PRI).

Scale-ability. The MVP3010 is tailored to companies needing more than a few voice-over-IP lines, but not needing carrier-class equipment. When expansion is needed, the MVP3010 can be field-upgraded into a dual E1 unit by installing the MVP30-60 kit, which is essentially a second MultiVOIP motherboard that fits into an open expansion-card slot in the MVP3010. The upgraded dual unit then accommodates two E1 lines.

E1 VOIP Traffic. The MVP3010 accepts its outbound traffic from an E1 trunk that's connected to either a PBX or to a telco/carrier. The MVP3010 transforms the telephony signals into IP packets for transmission on LANs, WANs, or the Internet. Inbound IP data traffic is converted to telephony data and signaling.

When connected to PBX. When connected to a PBX, the MVP3010 creates a network node served by 10/100-Base T connections. Local PBX phone extensions gain toll-free access to all phone stations directly connected to the VOIP network. Phone extensions at any VOIP location also gain local-rate access to the entire local public-switched telephone network (PSTN) at every other VOIP location in the system.

When connected to PSTN. When the E1 line(s) connected to the MVP3010 are connected directly to the PSTN, the unit becomes a Point-of-Presence server dedicated to local calls off-net.

H. 323, SIP, & SPP. Being H.323 compatible, the MVP3010 can place calls to telephone equipment at remote IP network locations that also contain H.323 compatible voice-over-IP gateways. It will interface with H.323 software and H.323 gatekeeper units. H.323 specifications also bring to voip telephony many special features common to conventional telephony. H.323 features of this kind that have been implemented into the MultiVOIP include Call Hold, Call Waiting, Call Identification, Call Forwarding (from the H.450 standard), and Call Transfer (H.450.2 from H.323 Version 2). The fourth version of the H.323 standard improves system resource usage (esp. logical port or socket usage) by handling call signaling more compactly and allowing use of the low-overhead UDP protocol instead of the error-correcting TCP protocol where possible.

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Data Compression & Quality of Service. The MultiVOIP3010 comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

VOIP Functions. The MultiVOIP MVP3010 gateway performs four basic functions: (a) it converts a dialed number into an IP address, (b) it sends voice over the data network, (c) it establishes a connection with another VOIP gateway at a remote site, and (d) it receives voice over the data network. Voice is handled as IP packets with a variety of compression options. Each E1 connection to the MultiVOIP provides 30 time-slot channels to connect to the telco or to serve phone or fax stations connected to a PBX.

Ports. The MVP3010 also has a 10/100 Mbps Ethernet LAN interface, and a Command port for configuration. An MVP3010 upgraded with the MVP30-60 kit will have two Ethernet LAN interfaces and two Command ports.

PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

Gatekeeper. E1 voip systems can have gatekeeper functionality either by adding, as an endpoint, either a Multi-Tech standalone gatekeeper (special software residing in separate hardware) or an analog gateway with embedded gatekeeper functionality (MVP210G, MVP410G, or MVP810G). Gatekeepers are optional but useful within voip systems. The gatekeeper acts as the 'clearinghouse' for all calls within its zone. MultiTech's embedded and stand-alone gatekeeper software packages both perform all of the standard gatekeepers functions (address translation, admission control, bandwidth control, and zone management) and also support many valuable optional functions (call control signaling, call authorization, bandwidth management, and call management). The stand-alone gatekeeper is, however, slightly more feature-rich than the embedded gatekeeper. For more details, see the "Embedded Gatekeeper" chapter of this manual and the manual on MultiTech's stand-alone gatekeeper.

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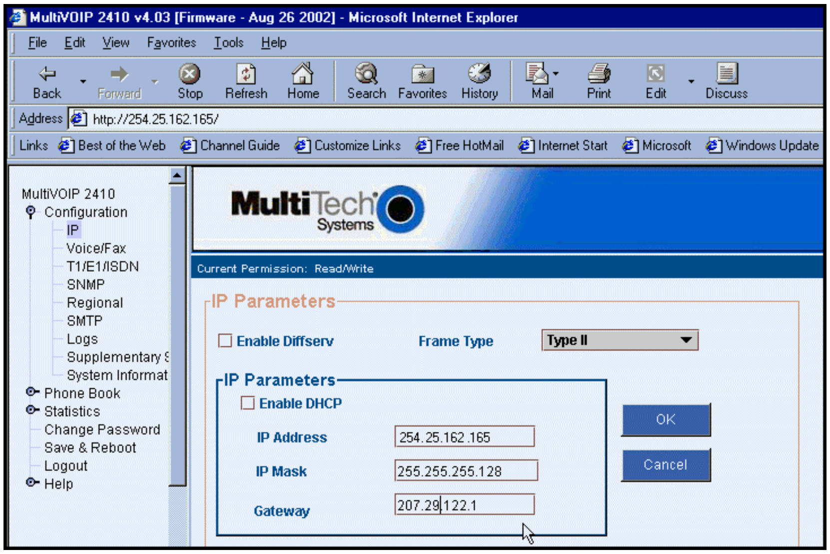

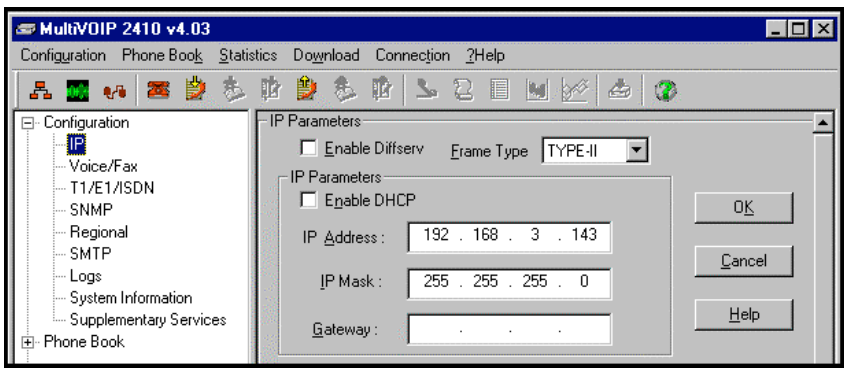

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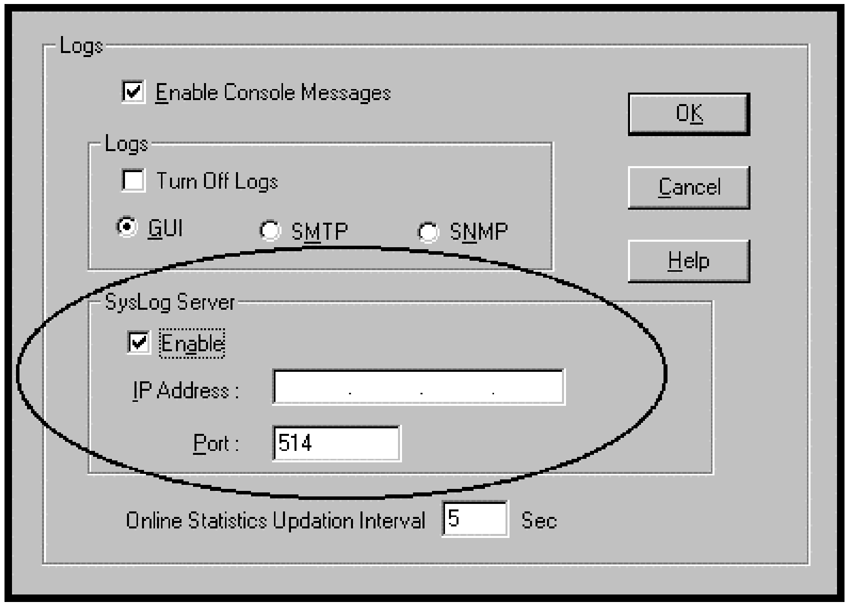

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E1 Front Panel LEDs

Because the MVP3010 and MVP30-60 both use a common main circuit card or motherboard, the LED indicators are the same for both.

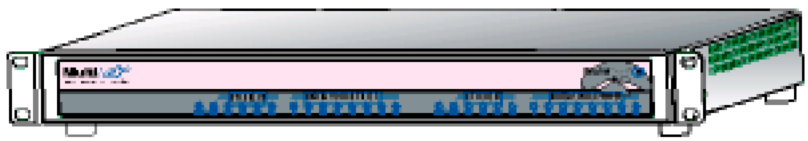


Figure 1-4. MultiVOIP MVP3010 Chassis

Active LEDs. The MVP3010 front panel has two sets of identical LEDs. In the MVP3010 as shipped (that is, without an expansion card), the left-hand set of LEDs is functional whereas the right-hand set is not.

When the MVP3010 has been upgraded with an MVP30-60 kit, the right-hand set of LEDs will also become active.

E1 LED Descriptions

MVP3010 Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on for about 10 seconds while the MVP3010 is booting.
RCV	Receive. Lights when receiving data on Ethernet port.
XMT	Transmit. Lights when transmitting data on Ethernet port.
LNK	Link. When lit, VOIP "sees" the hub or network via the Ethernet connection.
COL	Collision. Lit when data collisions occur.
T1	T1. Not supported.
E1	E1. When lit, indicates presence of E1 connection.
PRI	PRI. On if E1 line is of ISDN-Primary-Rate type.
ONL	Online. This LED is on when frame synchronization has been established on the T1/E1 link.
IC	IC LED is on when Internal Clocking is selected in T1/E1 configuration.
LC	Indicates Loss of Carrier.
LS	Indicates Loss of Signal.
Test	For testing purposes only. For testing purposes only.

Introduction to Analog MultiVOIPs (MVP130, MVP-210/410/810 & MVP428)

VOIP: The Free Ride. We proudly present Multi-Tech's MVP130, MVP-210/410/810 generation of MultiVOIP Voice-over-IP Gateways and models MVP-210G/410G/810G equipped with embedded gatekeeper functionality . All of these models allow voice/fax communication to be transmitted at no additional expense over your existing IP network, which has ordinarily been data only. To access this free voice and fax communication, you simply connect the MultiVOIP to your telephone equipment and your existing Internet connection. These analog MultiVOIPs inter-operate readily with T1 or E1 MultiVOIP units.

Capacity. MultiVOIP models MVP810 and MVP810G are eight-channel units, models MVP410 and MVP410G are four-channel units, and models MVP210 and MVP210G are two-channel units. The MVP130 is a single-channel unit. All of these MultiVOIP units have a 10/100Mbps Ethernet interface and a command port for configuration. The MVP428 is an expansion circuit card for the four-channel MVP410 that turns it into an eight-channel voip.

Mounting. Mechanically, the MVP410 and MVP810 MultiVOIPs are designed for a one-high industry-standard EIA 19-inch rack enclosure. By contrast, MVP130 and the MVP210 are tabletop units. The product 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.

Phone System Transparency. 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 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.

H. 323, SIP, & SPP. Being H.323 compatible, the analog MultiVOIP unit can place calls to telephone equipment at remote IP network locations that also contain H.323 compatible voice-over-IP gateways. It will interface with H.323 software and H.323 gatekeeper units. H.323 specifications also bring to voip telephony many special features common to conventional telephony. H.323 features of this kind that have been implemented into the MultiVOIP include Call Hold, Call Waiting, Call Identification, Call Forwarding (from the H.450

standard), and Call Transfer (H.450.2 from H.323 Version 2). The fourth version of the H.323 standard improves system resource usage (esp. logical port or socket usage) by handling call signaling more compactly and allowing use of the low-overhead UDP protocol instead of the error-correcting TCP protocol where possible.

The MultiVOIP is also SIP-compatible. ("SIP" means Session Initiation Protocol.) However, H.450 Supplementary Services features can be used under H.323 only and not under SIP.

SPP (Single-Port Protocol) 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. SPP offers advantages in certain situations, especially when firewalls are used and when dynamic IP address assignment is needed. However, when SPP is used, certain features of SIP and H.323 will not be available and SPP will not inter-operate with voip systems using H.323 or SIP.

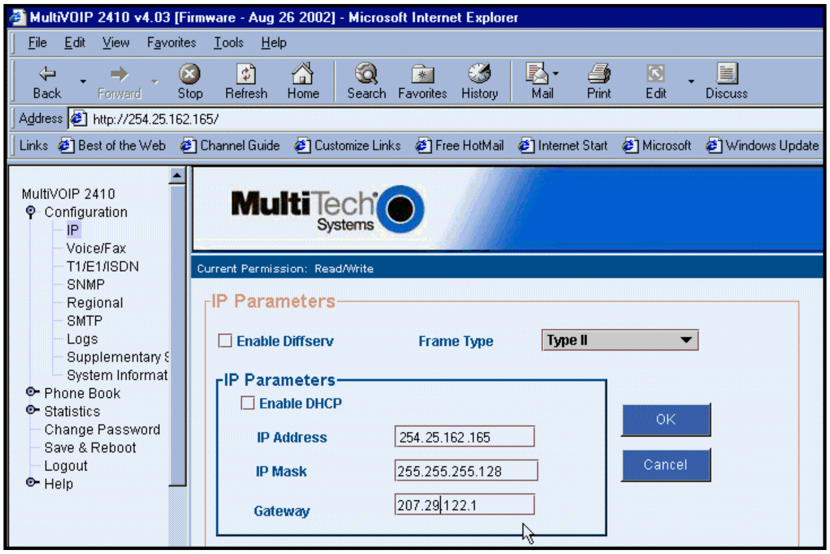
Data Compression & Quality of Service. The analog MultiVOIP unit comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

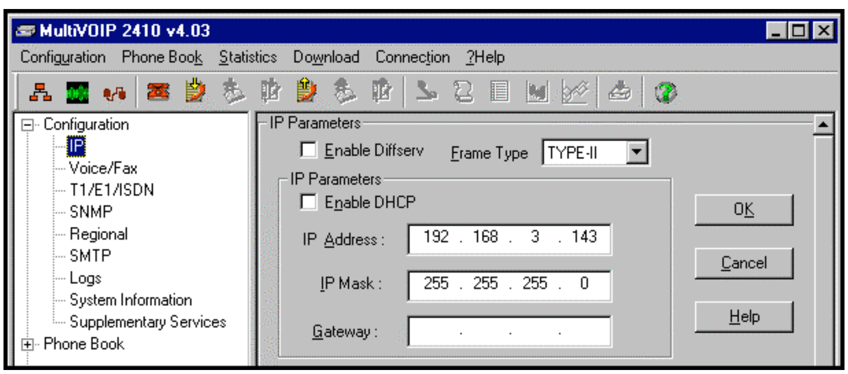
PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

Gatekeepers. For voip systems built with MultiTech's analog gateway units, users can have either an embedded gatekeeper (built into an MVP210G, MVP410G, or MVP810G) or a stand-alone gatekeeper (gatekeeper software residing in separate hardware). Gatekeepers are optional but useful within voip systems. The gatekeeper acts as the 'clearinghouse' for all calls within its zone. MultiTech's embedded and stand-alone gatekeeper software packages both perform all of the standard gatekeepers functions (address translation, admission control, bandwidth control, and zone management) and also support many valuable optional functions (call control signaling, call authorization, bandwidth management, and call management). The stand-alone gatekeeper is, however, slightly more feature-rich than the embedded gatekeeper. For more details, see the "Embedded Gatekeeper" chapter of this manual and the manual on MultiTech's stand-alone gatekeeper.

Management. Configuration and system management can be done locally with the MultiVOIP configuration software. After an IP address has been assigned locally, other configuration can be done remotely using the MultiVOIP web browser GUI. Remote system management can be done with the MultiVoipManager SNMP software or via the MultiVOIP web browser GUI. All of these control software packages are included on the Product CD.

While the web GUI's appearance differs slightly, its content and organization are essentially the same as that of the Windows GUI (except for logging).

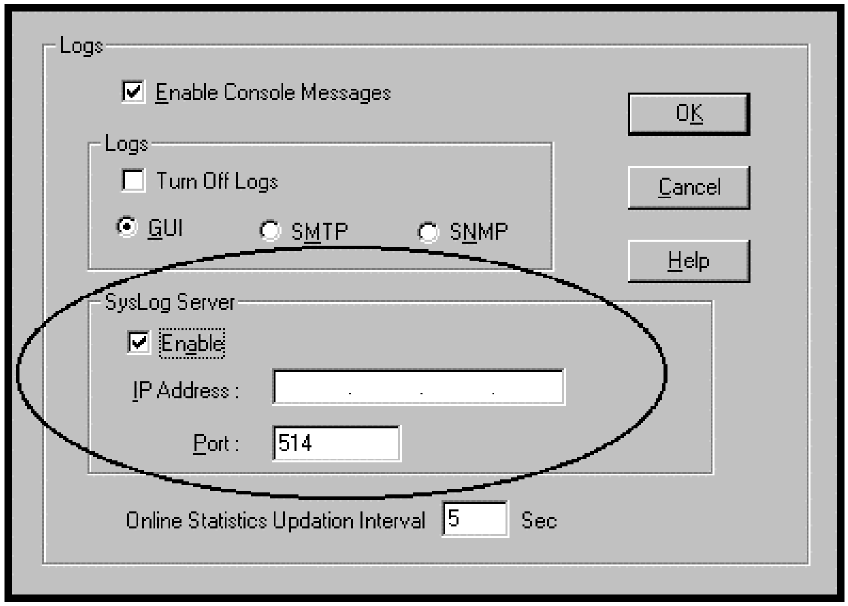




The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

Once you've begun using the web browser GUI, you can go back to the MultiVOIP Windows GUI at any time. However, you must log out of the web browser GUI before using the MultiVOIP Windows GUI.

Logging of System Events. MultiTech 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, among other firms. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by any qualified provider should suffice for use with MultiVOIP units. Kiwi's brief description of their SysLog program indicates the typical scope of such programs. "Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available."

Supplementary Telephony Services. The H.450 standard (an addition to H.323) brings to voip telephony more of the premium features found in PSTN and PBX telephony. MultiVOIP units offer five of these H.450 features: Call Transfer, Call Hold, Call Waiting, Call Name Identification (not the same as Caller ID), and Call Forwarding. (The first four features are found in the "Supplementary Services" window; the fifth, Call Forwarding, appears in the Add/Edit Inbound phonebook screen.) Note that the first three features are closely related. All of these H.450 features are supported for H.323 operation only; they are *not* supported for SIP or SPP.

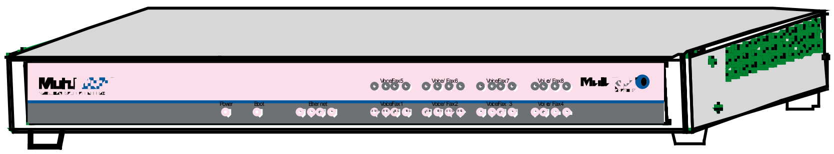


Figure 1-5: MVP-410/810 Chassis

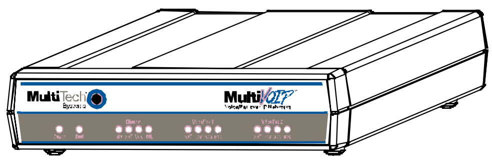


Figure 1-6: MVP-210 Chassis

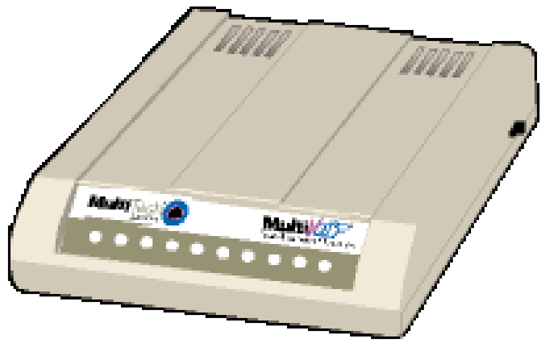


Figure 1-7. MultiVOIP MVP130Chassis

Analog MultiVOIP Front Panel LEDs

LED Types. The MultiVOIPs have two types of LEDs on their front panels:

(1) general operation LED indicators (for power, booting, and ethernet functions), and
(2) channel operation LED indicators that describe the data traffic and performance in each VOIP data channel.

Active LEDs. On both the MVP410 and MVP810, there are eight sets of channel-operation LEDs. However, on the MVP410, only the lower four sets of channel-operation LEDs are functional. On the MVP810, all eight sets are functional.

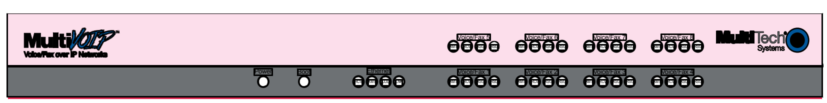


Figure 1-8. MVP410/810 Front Panel

Similarly, the MVP210 has the general-operation indicator LEDs and two sets of channel-operation LEDs, one for each channel.

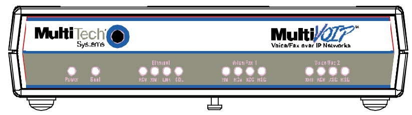


Figure 1-9. MVP210 Front Panel

Finally, the MVP130 has the general-operation indicator LEDs and a set of channel-operation LEDs for its single voip channel.

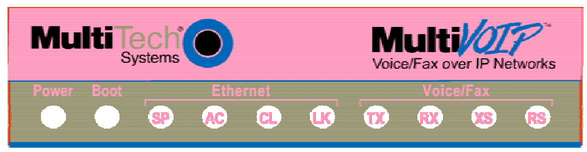


Figure 1-10. MVP130 Front Panel

Analog MultiVOIP LED Descriptions

MVP210/410/810 Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
General Operation LEDs (one set on each MultiVOIP model)	
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	**RCV**. Receive. Lights (blinks) when receiving data on Ethernet port. **XMT**. Transmit. Lights (blinks) when transmitting data on Ethernet port. .. **LNK**. Link. When lit, VOIP "sees" the hub or network via the Ethernet connection. .. **COL**. Collision. Lit when data collisions occur. ..
Channel-Operation LEDs (one set for each channel)	
XMT	**Transmit**. This indicator blinks when voice packets are being transmitted to the local area network.
RCV	**Receive**. This indicator blinks when voice packets are being received from the local area network.
XSG	**Transmit Signal**. This indicator lights when the FXS-configured channel is off-hook, the FXO-configured channel is receiving a ring from the Telco, or the M lead is active on the E&M configured channel. That is, it lights when the MultiVOIP is receiving a ring from the PBX.
RSG	**Receive Signal**. This indicator lights when the FXS-configured channel is ringing, the FXO-configured channel has taken the line off-hook, or the E lead is active on the E&M-configured channel.

MVP130 Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
General Operation LEDs	
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	**SP.** During normal operation, the SP LED lights to indicate 100Mbps is selected. **AC.** During normal operation, the AC LED lights when transmitting or receiving. It will flash at a rate of 50ms high and 50ms low when active. **CL.** During normal operation, the CL LED lights to indicate a collision. It will flash at a rate of 50ms high and 50ms low when active. **LK.** During normal operation, the LK LED lights to indicate a good link is detected.
Channel-Operation LEDs	
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 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 has taken the line off-hook.

Introduction to ISDN-BRI MultiVOIPs (MVP410ST & MVP810ST)

VOIP: The Free Ride. We proudly present Multi-Tech's MVP-410ST/810ST generation of MultiVOIP Voice-over-IP Gateways. All of these models allow voice/fax communication to be transmitted at no additional expense over your existing IP network, which has ordinarily been data only. To access this free voice and fax communication, you simply connect the MultiVOIP to your telephone equipment and your existing Internet connection. These ISDN Basic Rate Interface (ISDN-BRI) MultiVOIPs inter-operate readily with T1 or E1 MultiVOIP units (T1 and E1 MultiVOIP units can operate in ISDN Primary Rate Mode, ISDN-PRI, as well).

Capacity. MultiVOIP model MVP810ST accommodates four ISDN-BRI lines (eight B-channels) and model MVP410ST accommodates two ISDN-BRI channels (four B-channels). Both of these MultiVOIP units have a 10/100Mbps Ethernet interface and a command port for configuration.

Mounting. Mechanically, the MVP410ST and MVP810ST MultiVOIPs are designed for a one-high industry-standard EIA 19-inch rack enclosure. The product 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.

Phone System Transparency. 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 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.

H. 323, SIP, & SPP. Being H.323 compatible, the BRI MultiVOIP unit can place calls to telephone equipment at remote IP network locations that also contain H.323 compatible voice-over-IP gateways. It will interface with H.323 software and H.323 gatekeeper units. H.323 specifications also bring to voip telephony many special features common to conventional telephony. H.323 features of this kind that have been implemented into the MultiVOIP include Call Hold, Call Waiting, Call Identification, Call Forwarding (from the H.450 standard), and Call Transfer (H.450.2 from H.323 Version 2). The fourth version of the H.323 standard improves system resource usage (esp. logical port or socket usage) by handling call signaling more compactly and allowing use of the low-overhead UDP protocol instead of the error-correcting TCP protocol where possible.

The MultiVOIP is also SIP-compatible. ("SIP" means Session Initiation Protocol.) However, H.450 Supplementary Services features can be used under H.323 only and not under SIP.

SPP (Single-Port Protocol) 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. SPP offers advantages in certain situations, especially when firewalls are used and when dynamic IP address assignment is needed. However, when SPP is used, certain features of SIP and H.323 will not be available and SPP will not inter-operate with voip systems using H.323 or SIP.

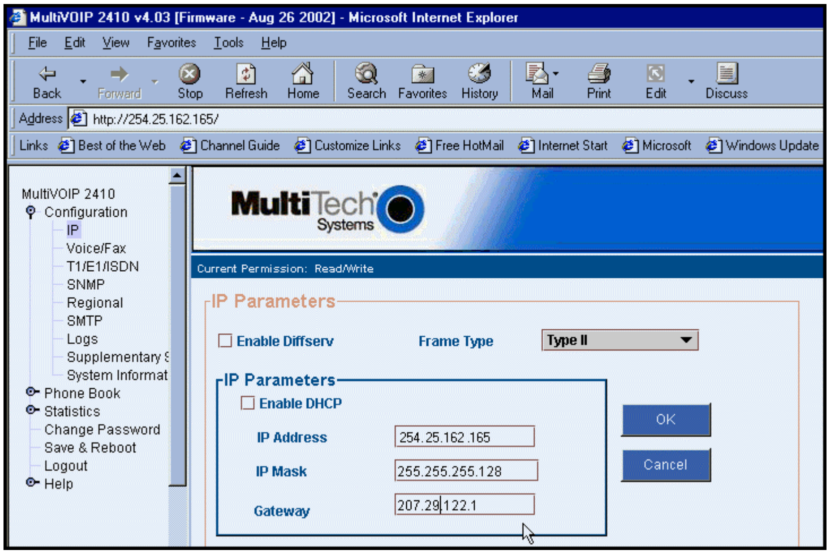
Data Compression & Quality of Service. The BRI MultiVOIP unit comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

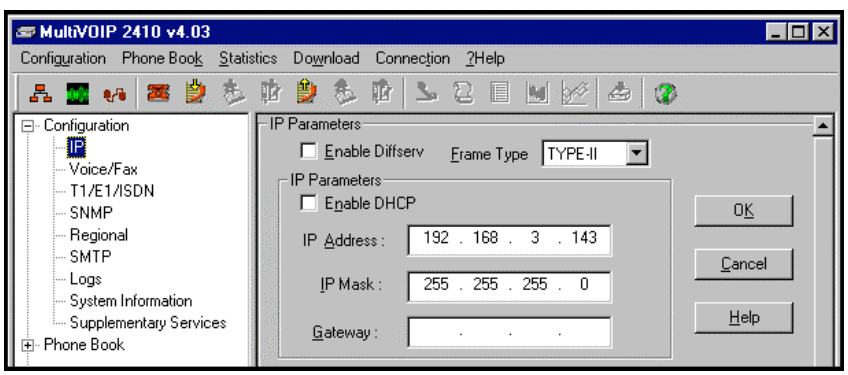
PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

Gatekeeper. At this writing, ISDN-BRI MultiVOIP systems can have gatekeeper functionality only by adding, as an endpoint, a standalone gatekeeper (special software residing in separate hardware). Gatekeepers are optional but useful within voip systems. The gatekeeper acts as the 'clearinghouse' for all calls within its zone. MultiTech's embedded and stand-alone gatekeeper software packages both perform all of the standard gatekeepers functions (address translation, admission control, bandwidth control, and zone management) and also support many valuable optional functions (call control signaling, call authorization, bandwidth management, and call management). The stand-alone gatekeeper is, however, slightly more feature-rich than the embedded gatekeeper. For more details, see the "Embedded Gatekeeper" chapter of this manual and the manual on MultiTech's stand-alone gatekeeper.

Management. Configuration and system management can be done locally with the MultiVOIP configuration software. After an IP address has been assigned locally, other configuration can be done remotely using the MultiVOIP web browser GUI. Remote system management can be done with the MultiVOIP web browser GUI. Neither of these is available yet. The web GUI will be in release 5.04, however. All of these control software packages are included on the Product CD.

While the web GUI's appearance differs slightly, its content and organization are essentially the same as that of the Windows GUI (except for logging).

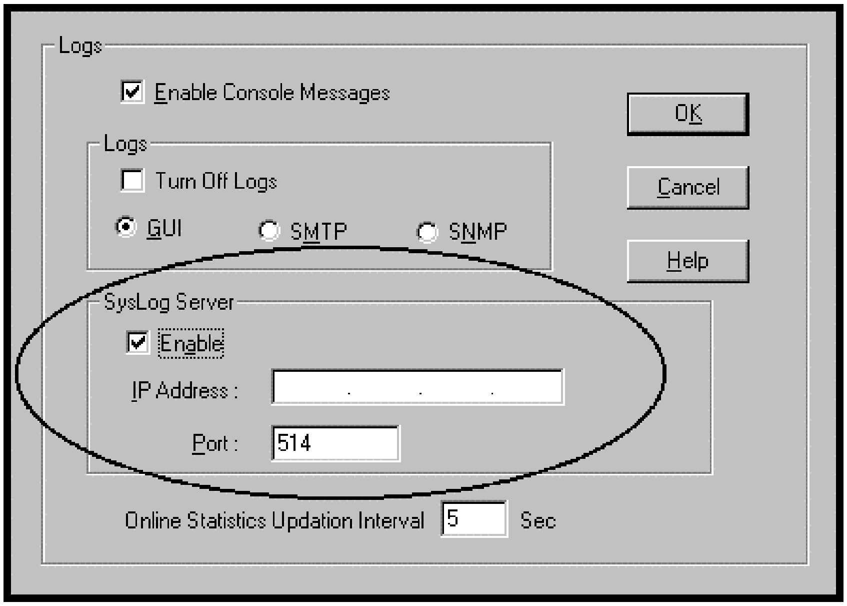




The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

Once you've begun using the web browser GUI, you can go back to the MultiVOIP Windows GUI at any time. However, you must log out of the web browser GUI before using the MultiVOIP Windows GUI.

Logging of System Events. MultiTech 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, among other firms. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by any qualified provider should suffice for use with MultiVOIP units. Kiwi's brief description of their SysLog program indicates the typical scope of such programs. "Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available."

Supplementary Telephony Services. This is available in 5.04 but not 5.02c. The H.450 standard (an addition to H.323) brings to voip telephony more of the premium features found in PSTN and PBX telephony. MultiVOIP units offer five of these H.450 features: Call Transfer, Call Hold, Call Waiting, Call Name Identification (not the same as Caller ID), and Call Forwarding. (The first four features are found in the "Supplementary Services" window; the fifth, Call Forwarding, appears in the Add/Edit Inbound phonebook screen.) Note that the first three features are closely related. All of these H.450 features are supported for H.323 operation only; they are *not* supported for SIP or SPP.

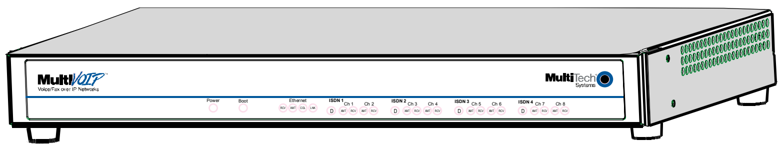


Figure 1-11: MVP-410ST/810ST Chassis

ISDN BRI MultiVOIP Front Panel LEDs

LED Types. The MultiVOIPs have two types of LEDs on their front panels:

(1) general operation LED indicators (for power, booting, and ethernet functions), and
(2) channel operation LED indicators that describe the data traffic and performance in each VOIP data channel.

Active LEDs. On the MVP810ST, there are four sets of ISDN-operation LEDs. On the MVP410ST, there are two sets of ISDN-operation LEDs. Each set contains one "D" LED and two sets of channel operation LEDs (XMT and RCV).

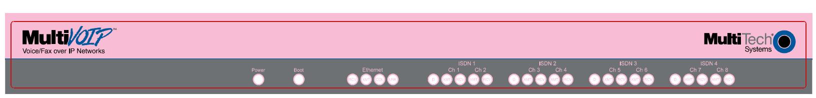


Figure 1-12. MVP-410ST/810ST Front Panel

ISDN-BRI MultiVOIP LED Descriptions

MVP-410ST/810ST Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
General Operation LEDs (one set on each MultiVOIP model)	
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	**RCV**. Receive. Lights (blinks) when receiving data on Ethernet port. **XMT**. Transmit. Lights (blinks) when transmitting data on Ethernet port. .. **LNK**. Link. When lit, VOIP "sees" the hub or network via the Ethernet connection. .. **COL**. Collision. Lit when data collisions occur. ..
D-Channel Operation LEDs (one for each ISDN line)	
D	ISDN D-channel & physical layer indicator. One "D" LED for each ISDN-BRI connection. The "D" LED is off when the BRI physical layer is de-activated.* It flashes when a connection is being established on the physical layer. It is on when the physical layer has been activated. It flickers to indicate D-channel traffic. *If the voip is running in terminal mode and its BRI line is unplugged, the D LED goes off. However, if the voip is running in network mode and its BRI line is unplugged, its LED will flash at regular interval.
B-Channel Operation LEDs (one for each B-channel)	
XMT	**Transmit**. This indicator blinks when voice packets are being transmitted onto the B-channel.
RCV	**Receive**. This indicator blinks when voice packets are being received on the B-channel.

Computer Requirements

The computer on which the MultiVOIP's configuration program is installed must meet these requirements:

- must be IBM-compatible PC with MS Windows operating system;
- must have an available COM port for connection to the MultiVOIP.

However, this PC does not need to be connected to the MultiVOIP permanently. It only needs to be connected when local configuration and monitoring are done. Nearly all configuration and monitoring functions can be done remotely via the IP network.

Specifications

Specs for Digital T1 MultiVOIP Units

Digital T1 MultiVOIP Specifications			
Parameter ....../Model	**MVP-2400**	**MVP-2410 MVP-2410g**	**MVP-2410 w/ MVP24-48 Expansion Card**
Operating Voltage/Current	External transformer: 1.6A@5v	100-240 VAC 1.2 - 0.6 A	100-240 VAC 1.2 - 0.6 A
Mains Frequencies	50/60 Hz	50/60 Hz	50/60 Hz
Power Consumption	13 watts	17 watts	27 watts
Mechanical Dimensions	6.2" W x 9" D x 1.4" H 15.8cm W x 22.9cm D x 3.6cm H	1.75"H x 17.4"W x 8.75"D 4.5cm H x 44.2 cm W x 22.2 cm D	1.75"H x 17.4"W x 8.75"D 4.5cm H x 44.2 cm W x 22.2 cm D
Weight	1.8lbs (.82kg) 2.2lbs (.98kg) with transformer	7.1 lbs. (3.2 kg)	7.5 lbs. (3.4 kg)

Specs for Digital E1 MultiVOIP Units

Digital E1 MultiVOIP Specifications		
Parameter ……/Model	**MVP-3010**	**MVP-3010 w/ MVP30-60 Expansion Card**
Operating Voltage/Current	100-240 VAC 1.2 - 0.6 A	100-240 VAC 1.2 - 0.6 A
Mains Frequencies	50/60 Hz	50/60 Hz
Power Consumption	17 watts	27 watts
Mechanical Dimensions	1.75"H x 17.4"W x 8.75"D 4.5cm H x 44.2 cm W x 22.2 cm D	1.75"H x 17.4"W x 8.75"D 4.5cm H x 44.2 cm W x 22.2 cm D
Weight	7.1 lbs. (3.2 kg)	7.5 lbs. (3.4 kg)

Specs for Analog/BRI MultiVOIP Units

Parameter /Model	MVP210 MVP210G	MVP410 MVP410G	MVP810or MVP410 + 428 MVP810G
Operating Voltage/ Current	External transformer: 3A @5V	100-240 VAC 1.2 - 0.6 A	100-240 VAC 1.2 - 0.6 A
Mains Frequencies	50/60 Hz	50/60 Hz	50/60 Hz
Power Consumption	19 watts	29 watts	46 watts
Mechanical Dimensions	6.2” W x 9” D x 1.4” H 15.8cm W x 22.9cm D x 3.6cm H	1.75” H x 17.4” W x 8.5” D 4.5cm H x 44.2 cm W x 21.6 cm D	1.75” H x 17.4” W x 8.5” D 4.5cm H x 44.2 cm W x 21.6 cm D
Weight	1.8lbs (.82kg) 2.6lbs (1.17kg) with transformer	7.1 lbs. (3.2 kg)	7.7 lbs. (3.5 kg)
Parameter ……/Model	**MVP410ST**	**MVP410 MVP410G MVP410ST**	**MVP130**
Operating Voltage/ Current	100-240VAC 1.2-0.6 A	100-240VAC 1.2-0.6 A	100-240VAC 1.0 A
Mains Frequencies	50/60 Hz	50/60 Hz	50/60 Hz
Power Consumption	12 watts	18 watts	9.7 watts (with phone off hook)
Mechanical Dimensions	Same as MVP410	Same as MVP810	4.3" W x 5.6" D 1.0" H 10.8 cm W X 14.2 cm D X 2.95 cm H
Weight	6.61 lbs. (3.00 kg)	6.75 lbs. (3.06 kg)	8 oz. (23 g)

Installation at a Glance

The basic steps of installing your MultiVOIP network involve unpacking the units, connecting the cables, and configuring the units using management software (MultiVOIP Configuration software) and confirming connectivity with another voip site. This process results in a fully functional Voice-Over-IP network.

Related Documentation

The MultiVOIP User Guide (the document you are now reading) comes in electronic form and is included on your system CD. It presents in-depth information on the features and functionality of Multi-Tech's MultiVOIP Product Family.

The CD media is produced using Adobe Acrobat™ for viewing and printing the user guide. To view or print your copy of a user guide, load Acrobat Reader™ on your system. The Acrobat Reader is included on the MultiVOIP CD and is also a free download from Adobe's Web Site:

www.adobe.com/prodindex/acrobat/readstep.html

This MultiVOIP User Guide is also available on Multi-Tech's Web site at:

http://www.multitech.com

Viewing and printing a user guide from the Web also requires that you have the Acrobat Reader loaded on your system. To select the MultiVOIP User Guide from the Multi-Tech Systems home page, click **Documents** and then click **MultiVOIP Family** in the product list drop-down window. All documents for this MultiVOIP Product Family will be displayed. You can then choose *User Guide (MultiVOIP Product Family)* to view or download the **.pdf** file.

Entries (organized by model number) in the "knowledge base" and 'troubleshooting resolutions' sections of the MultiTech web site (found under "Support") constitute another source of help for problems encountered in the field.

Chapter 2: Quick Start Instructions

Introduction

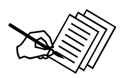
This chapter gets the MultiVOIP up and running quickly. The details we've skipped to make this brief can be found elsewhere in the manual (see Table of Contents and Index).

MultiVOIP Startup Tasks

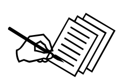
Task	Summary
● **Collecting Phone/IP Details (vital!)**	The MultiVOIP must be configured to interface with your particular phone system and IP network. To do so, certain details must be known about those phone and IP systems.
● **Placement**	Decide where you'll mount the voip.
● **Command/Control Computer Setup: Specs & Settings**	Some modest minimum specifications must be met. A COM port must be set up.
● **Hookup**	Connect power, phone, and data cables per diagram.
● **Software Installation**	This is the configuration program. It's a standard Windows software installation.
● **Phone/IP Starter Configuration**	You will enter phone numbers and IP addresses. You'll use default parameter values where possible to get the system running quickly.
● **Phonebook Starter Configuration**	The phonebook is where you specify how calls will be routed. To get the system running quickly, you'll make phonebooks for just two voip sites.
● **Connectivity Test**	You'll find out if your voip system can carry phone calls between two sites. That means you're up and running!
● **Troubleshooting**	Detect and remedy any problems that might have prevented connectivity.

Phone/IP Details *Absolutely Needed* Before Starting the Installation

Gather IP Information

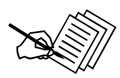
➨	*Ask your computer network administrator.*	*Info needed to operate:* **all MultiVOIP models.**
	 IP Network Parameters: Record for each VOIP Site in System	
	• IP Address	
	• IP Mask	
	• Gateway	
	• Domain Name Server (DNS) Info *(not implemented; for future use)*	

Gather Telephone Information (T1)

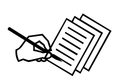
➨	**T1 Phone Parameters** *Ask phone company or PBX maintainer.*	*Info needed to operate:* **MVP2400 MVP2410**
	 T1 Telephony Parameters: Record for this VOIP Site	
	• Which frame format is used? ESF___ or D4___	
	• Which CAS or PRI protocol is used? _______________	
	• Clocking: Does the PBX or telco switch use internal or external clocking? __________________ Note that the setting used in the voip unit will be the opposite of the setting used by the telco/PBX.	
	• Which line coding is used? AMI___ or B8ZS___	
	• Pulse shape level?: (most commonly 0 to 40 meters)	

Phone/IP Details *Absolutely Needed* (cont'd)

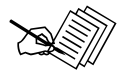
Gather Telephone Information (E1)

➨	**E1 Phone Parameters** *Ask phone company or PBX maintainer.*	*Info needed to operate:* **MVP3010**
	 E1 Telephony Parameters: Record for this VOIP Site	
	• Which frame format is used? Double Frame_____ MultiFrame w/ CRC4_____ MultiFrame w/ CRC4 modified_____	
	• Which CAS or PRI protocol is used? _______________	
	• Clocking: Does the PBX or telco switch use internal or external clocking? _________________ Note that the setting used in the voip unit will be the opposite of the setting used by the telco/PBX.	
	• Which line coding is used? AMI___ or HDB3___	
	• Pulse shape level?: (most commonly 0 to 40 meters)	

Gather Telephone Information (Analog)

➨	**Analog Phone Parameters** *Ask phone company or telecom manager.*	*Needed for:* **MVP810** **MVP410** **MVP210** **MVP130**
	 Analog Telephony Interface Parameters: Record for this VOIP Site	
	• Which interface type (or "signaling") is used? E&M_____ FXS/FXO_____	
	• If FXS, determine whether the line will be used for a phone, fax, or KTS (key telephone system)	
	• If FXO, determine if line will be an analog PBX extension or an analog line from a telco central office	
	• If E&M, determine these aspects of the E&M trunk line from the PBX: • What is its Type (1, 2, 3, 4, or 5)? • Is it 2-wire or 4-wire? • Is it Dial-Tone or Wink?	

Gather Telephone Information (ISDN BRI)

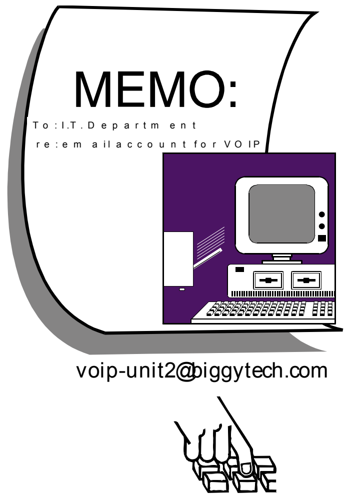
➨	**ISDN-BRI Phone Parameters** *Ask phone company or telecom manager.*	*Needed for:* **MVP810ST** **MVP410ST**
	 ISDN-BRI Telephony Interface Parameters: Record them for this VOIP Site	
	• In which country is this voip installed?	
	• Which operator (switch type) is used?	
	• What type of line coding use required, A-law or u-law?	
	• Determine which BRI ports will be network side and which BRI ports will be terminal side.	
	• If you are connecting the MultiVOIP to network equipment with a "U" interface, an NT1 device must be connected between them.	

Phone/IP Details Often Needed/Wanted

Obtain Email Address for VOIP (for email call log reporting)

<table>
<tr><td>required if log reports of
VOIP call traffic
are to be sent by email</td><td>Optional</td></tr>
<tr><td>SMTP Parameters
Preparation Task:

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.</td><td>
</td></tr>
</table>

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 in a single VOIP system, 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.

Placement

Mount your MultiVOIP in a safe and convenient location where cables for your network and phone system are accessible. Rack-mounting instructions are in *Chapter 3: Mechanical Installation & Cabling.*

The Command/Control Computer (Specs & Settings)

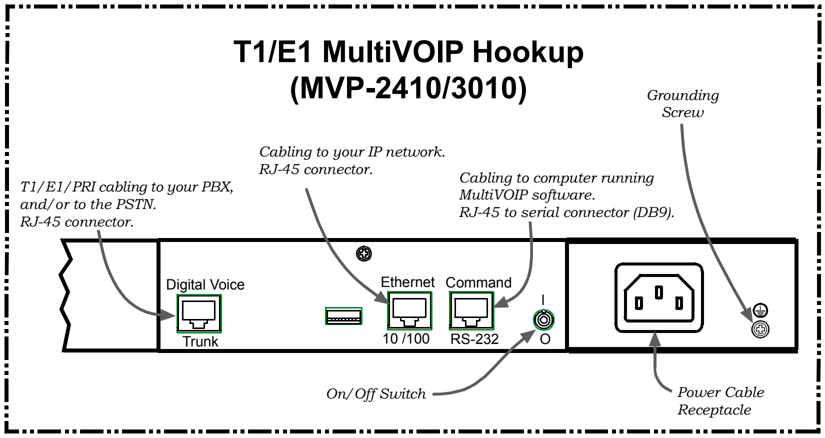
The computer used for command and control of the MultiVOIP
(a) must be an IBM-compatible PC,
(b) must use a Microsoft operating system,
(c) must be connected to your local network (Ethernet) system, and
(d) must have an available serial COM port.

The configuration tasks and control tasks the PC will have to do with the MultiVOIP are not especially demanding. Still, we recommend using a reasonably new computer. The computer that you use to configure your MultiVOIP need not be dedicated to the MultiVOIP after installation is complete.

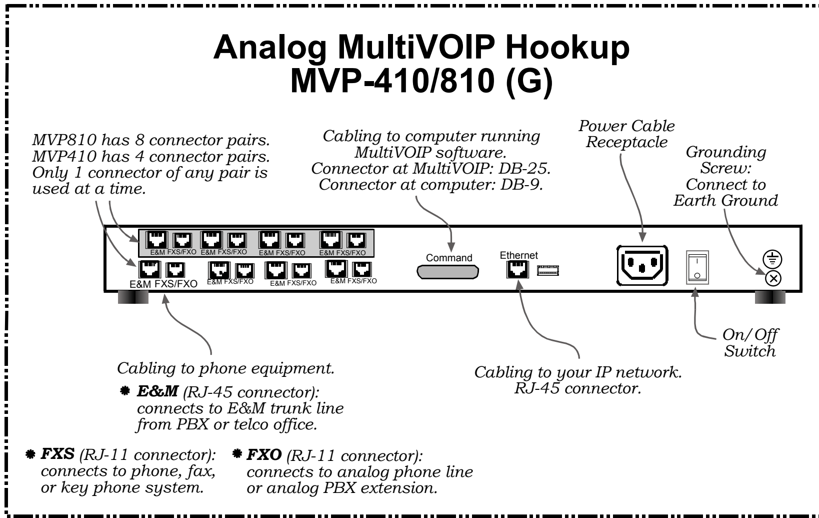
COM port on controller PC. You'll need an available COM port on the controller PC. You'll need to know which COM port is available for use with the MultiVOIP (COM1, COM2, etc.).

Quick Hookups

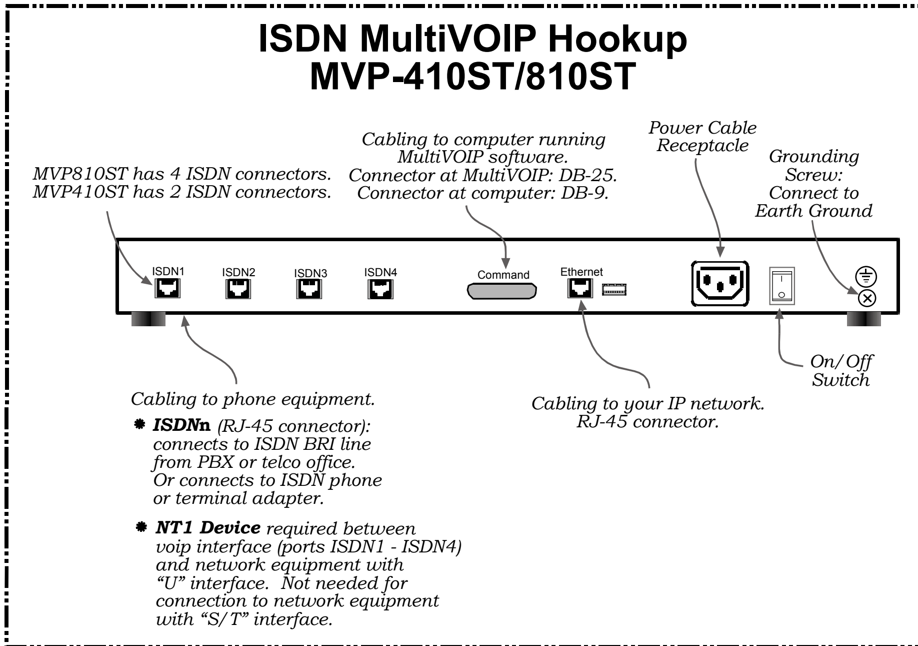
Hookup for MVP2410 & MVP3010



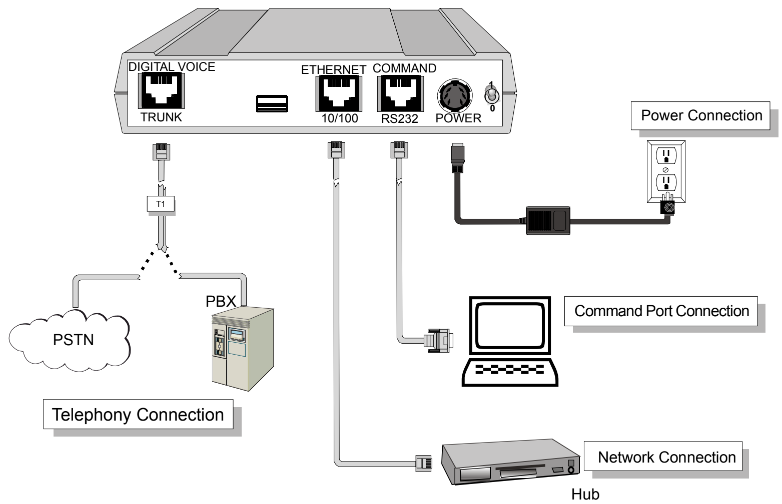
Hookup for MVP-410/410G & MVP-810/810G



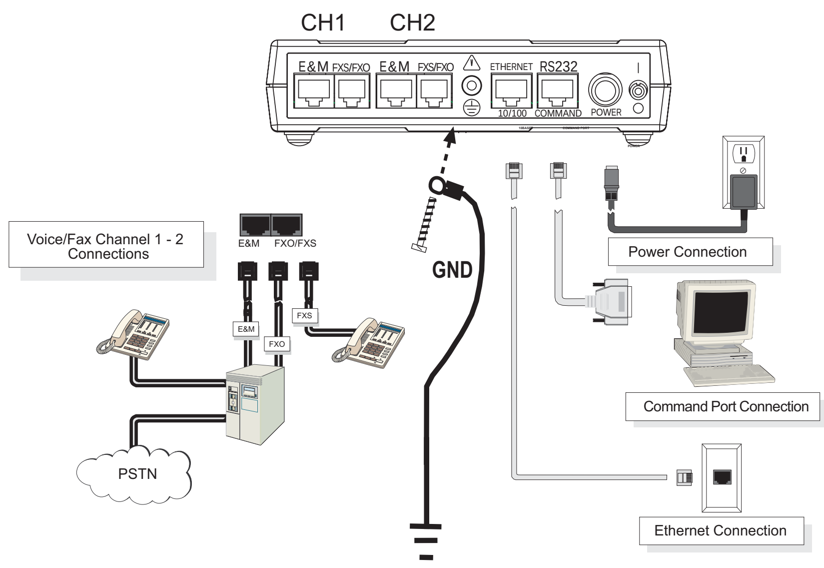
Hookup for MVP410ST & MVP810ST



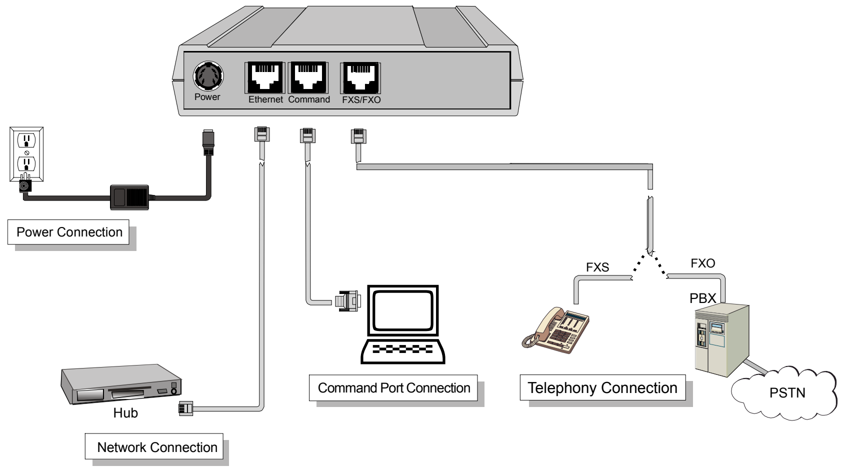
Hookup for MVP2400



Hookup for MVP210x



Hookup for MVP130



Load MultiVOIP Control Software onto PC

For more details, see *Chapter 4: Software Installation*.

1. MultiVOIP must be properly cabled. Power must be turned on.
2. Insert MultiVOIP CD into drive. Allow 10-20 seconds for Autorun to start. If Autorun fails, go to
   My Computer | CD ROM drive | Open. Click **Autorun** icon.
3. At first dialog box, click **Install Software**.
4. At ‘welcome’ screen, click **Next**.
5. Follow on-screen instructions. Accept default program folder location and click **Next.**
6. Accept default icon folder location. Click **Next**. Files will be copied.
7. Select available COM port on command/control computer.
8. At completion screen, click **Finish**.
9. At the prompt “Do you want to run MultiVOIP Configuration?,” click **No**. Software installation is complete.

Phone/IP Starter Configuration

Full details here:

MVP2400 MVP2410x MVP3010	*Chapter 5: Technical Configuration for Digital T1/E1 MultiVOIPs* in User Guide.
MVP130 MVP210x MVP410x MVP810x	*Chapter 6: Technical Configuration for Analog/BRI MultiVOIPs* in User Guide

1. Open MultiVOIP program: **Start | MultiVOIP *xxx* | Configuration**.
2. Go to **Configuration | IP**. Enter the IP parameters for your voip site.
3. Do you want to configure and operate the MultiVOIP unit using the web browser GUI? (It has the same functionality as the local Windows GUI, but offers remote access.)
   If NO, skip to step 5.
   If YES, continue with step 4.
4. **Enable Web Browser GUI (Optional)**. To do configuration and operation procedures using the web browser GUI, you must first enable it. To do so, follow these steps. (The browser used must be Internet Explorer 6.0 or above; or Netscape 6.0 or above.)

A. Be sure an IP address has been assigned to the MultiVOIP unit (this must be done in the MultiVOIP Windows GUI).	**E**. Open web browser. (Note: The PC being used must be connected to and have an IP address on the same IP network that the voip is on.)
B. Save Setup in Windows GUI.	**F**. Browse to IP address of MultiVOIP unit.
C. Close the MultiVOIP Windows GUI.	**G**. If username and password have been established, enter them when prompted by voip.
D. Install Java program from MultiVOIP product CD. (Must be Java Runtime Environment 1.4.0_01 or above.) *NOTE: Required on first use of Web Browser GUI only.*	**H**. Use web browser GUI to configure or operate voip.
Need more info?	See "Web Browser Interface" in *Operation & Maintenance* chapter of **User Guide** (on CD).

Once you've begun using the web browser GUI, you can go back to the MultiVOIP Windows GUI at any time. However, you must log out of the web browser GUI before using the MultiVOIP Windows GUI.

5. Go to **Configuration | Voice/Fax**. Select **Coder** | "Automatic." At the right-hand side of the dialog box, click **Default**. If you know any specific parameter values that will apply to your system, enter them. Click **Copy Channel**. Select **Copy to All**. Click **Copy**. At main Voice/Fax Parameters screen, click **OK** to exit from the dialog box.
6. Enter telephone system information.

Analog MultiVOIPs
MVP130,
MVP-210/410/810
MVP-210G/410G/810G
Go to
Configuration | Interface.
Enter parameters obtained from phone company or PBX administrator.

Digital MultiVOIPs
MVP-2400/2410x/3010
Go to
Configuration | T1/E1/ISDN.
Enter parameters obtained from phone company or PBX administrator.

ISDN-BRI MultiVOIPs
MVP-410ST/810ST
Go to **Configuration | ISDN BRI**.
Enter parameters obtained from phone company or PBX administrator.
If the voip is connected to BRI extensions of a PBX or a phone company, then select "Terminal" in the **ISDN BRI Parameters** screen.
If the voip is connected to ISDN terminal adapters and/or ISDN phones, then select "Network" in the **ISDN BRI Parameters** screen.

7. Go to **Configuration | Regional Parameters**. Select the **Country/Region** that fits your situation. Click **Default** and confirm. Click **OK** to exit from the dialog box.
8. Do you want the phone-call logs produced by the MultiVOIP to be sent out by email (to your Voip Administrator or someone else)?
   If NO, skip to step 10.
   If YES, continue with step 9.

9. Go to **Configuration | SMTP**.
SMTP lets you send phone-call log records to the Voip Administrator by email. Select **Enable SMTP**.
You should have already obtained an email address for the MultiVOIP itself (this serves as the origination email account for email logs that the MultiVOIP can email out automatically).

Enter this email address in the “Login Name” field.
Type the password for this email account.

Enter the IP address of the email server where the MultiVOIP’s email account is located in the “Mail Server IP Address” field.

Typically the email log reports are sent to the Voip Administrator but they can be sent to any email address. Decide where you want the email logs sent and enter that email address in the “Recipient Address” field.

Whenever email log messages are sent out, they must have a standard Subject line. Something like “Phone Logs for Voip N” is useful. If you have more than one MultiVoip unit in the building, you’ll need a unique identifier for each one (select a useful name or number for “N”). In this “Subject” field, enter a useful subject title for the log messages.

In the “Reply-To Address” field, enter the email address of your Voip Administrator.

10. Go to **Configuration | Logs**.
Select “Enable Console Messages.” (*Not applicable if using Web GUI.*)

To allow log reports by email (if desired), click **SMTP**. Click **OK**.

To do logging with a SysLog client program, click on “SysLog Server – Enable” in the **Logs** screen. To implement this function, you must install a SysLog client program. For more info, see the “SysLog Server Functions” section of the *Operation & Maintenance* chapter of the **User Guide**.

Phone/IP Starter Configuration (continued)

11. Enable premium (H.450) telephony features. (Not supported in BRI 502c software.)

    Go to **Supplementary Services**. Select any features to be used.
    For Call Hold, Call Transfer, & Call Waiting, specify the key sequence that the phone user will press to invoke the feature. For Call Name Identification, specify the allowed name types to be used and a caller-id descriptor.

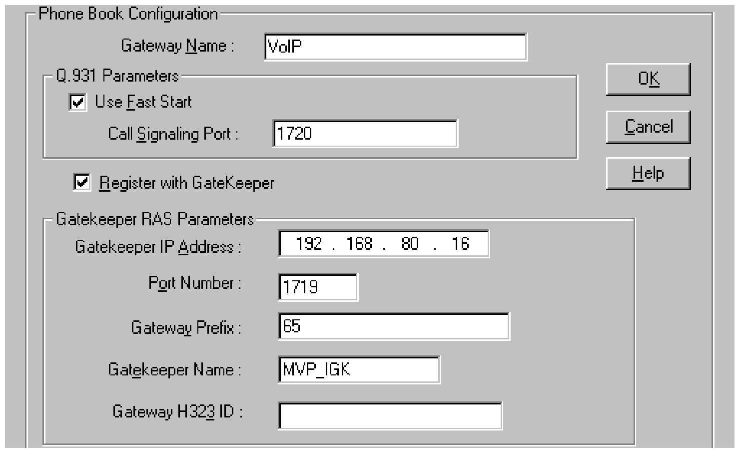
    If Call Forwarding is to be used, enable this feature in the Add/Edit Inbound Phone Book screen.

    After making changes, click on **OK** in the current configuration screen before moving on to the next configuration screen.

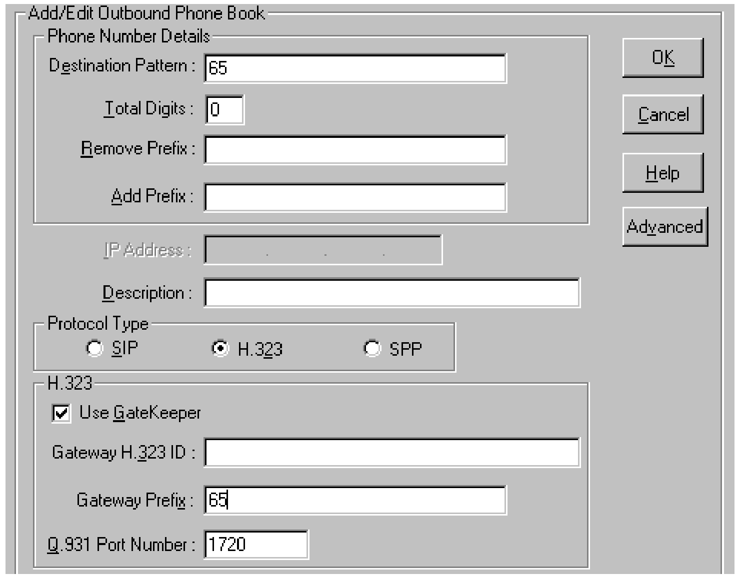
12. (**For analog gatekeeper-equipped models only. These have model numbers with a "G" suffix.**
    For MVP2410G, skip to step 13 and see User Guide for embedded gatekeeper info.
    For units without embedded gatekeeper, skip to step 13.)
    For quick-start purposes, we will arrange for the gatekeeper-equipped voip unit to register itself as a client of its own gatekeeper capability. Then we will set up a gatekeeper-controlled call from one channel to another of that self-same gatekeeper-equipped voip unit to demonstrate that the gatekeeper functionality is active. Thereafter, you can register additional voip units (and other endpoints) with the gatekeeper-equipped voip per instructions in the **User Guide**.

12A. For the "G" voip unit, set the gatekeeper IP address to be the same as the IP address used for its gateway function. To do so, go to the **PhoneBook Configuration** screen. Click on "Register with Gatekeeper."

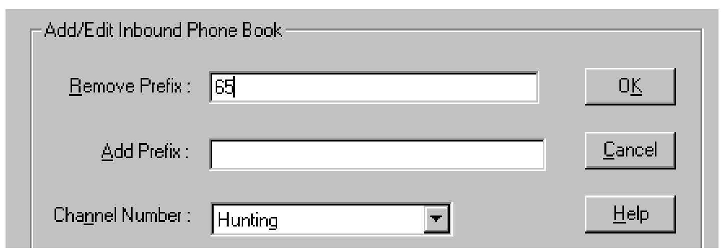
In the "Gatekeeper IP Address" field, enter the same IP address as entered in Step 2 (of this procedure). In the "Gatekeeper Name" field, enter the default name for gatekeeper-equipped units,
which is MVP_IGK. Click **OK**.



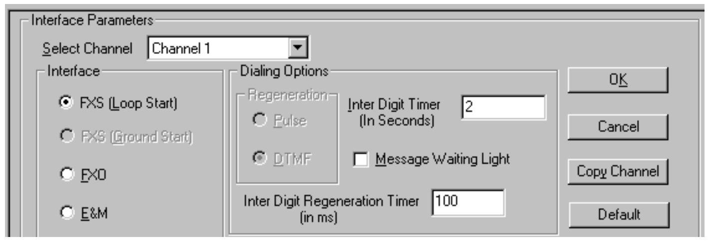
12**B**. In the "Destination Pattern" field of the **Add/Edit Outbound Phonebook** screen, enter 65. Click on "Use Gatekeeper." In the "Gateway Prefix" field, enter 65. Click **OK**.



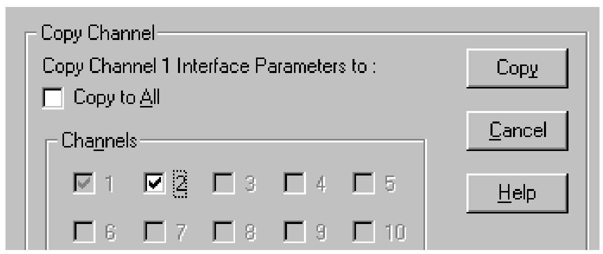
12**C**. In the "Remove Prefix" field of the **Add/Edit Inbound Phonebook** screen, enter 65. Click **OK**.



12**D**. To enable a call between two analog phones on the same voip, we will set up two channels for FXS Loop Start telephony. To do so, go to the **Interface** screen. Click on "FXS Loop Start" for Channel 1.



Click on "Copy Channel" and select Channel 2. Click **Copy**.



Click **OK** to acknowledge the copy. Click **OK** again when the main **Interface** screen returns.

13. Go to **Save Setup | Save and Reboot**. Click OK. This will save the parameter values that you have just entered.

    The MultiVOIP's "BOOT" LED will light up while the configuration file is being saved and loaded into the MultiVOIP. Don't do anything to the MultiVOIP until the "BOOT "LED is off (a loss of power at this point could cause the MultiVOIP unit to lose the configuration settings you have made).

14. (**For analog gatekeeper-equipped models only**. **These have model numbers with a "G" suffix. For non-gatekeeper units and for MVP2410G, skip this step.**) Connect two standard analog telephone sets to the Channel 1 and Channel 2 FXS/FXO ports on the back of the "G" voip unit.

    At either phone, dial 65. The completion of the call to the other phone confirms that the embedded gatekeeper of the "G" voip unit is mediating calls.

    For more information, see the "Embedded Gatekeeper" chapter of the **User Guide**.

END OF PROCEDURE.

Phonebook Starter Configuration (*with remote voip*)

If the topic of voip phone books is new to you, it may be helpful to read the PhoneBook Tips section (page 31) before starting this procedure.

To do this part of the quick setup, you need to know of 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.

Outbound Phonebook

1. Open the MultiVOIP program (**Start | MultiVOIP xxx | Configuration**
2. Go to **Phone Book | PhoneBook Modify | 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, etc.) that you'll be calling.

   Follow the example that best fits your situation.

North America, Long-Distance Example

Technician in Seattle (area 206) must set up one voip there, another in Chicago (area 312, downtown).

Answer: Write down **312**.

Euro, National Call Example

Technician in central London (area 0207) to set up voip there, another in Birmingham (area 0121).

Answer: write down **0121**.

Euro, International Call Example

Technician in Rotterdam (country 31; city 010) to set up one voip there, another in Bordeaux (country 33; area 05).

Answer: write down **3305**.

4. 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 (i.e., 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 that you must dial before you can dial a remote area code.

North America, Long-Distance 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**.

Euro, National Call Example

London/Birming. 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**.

Euro, International Call Example

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**.

5. In the “Destination Pattern” field of the **Add/Edit Outbound Phonebook** screen, enter the digits from step 4 followed by the digits from step 3.

North America, Long-Distance Example

Seattle-Chicago system.

Answer: enter **81312** as Destination Pat-tern in Outbound Phone book of Seattle voip.

Euro, National Call Example

London/Birming. 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 Pat-tern in Outbound Phonebook of London voip. *Not 900121.*

Euro, International Call Example

Rotterdam/Bordeaux system.

Answer: enter **903305** as Destination Pattern in Outbound Phonebook of Rotterdam voip.

6. Tally up the number of digits that must be dialed to reach the remote voip site (including prefix digits of all types). Enter this number in the "Total Digits" field.

North America, Long-Distance Example

Seattle-Chicago system.

To complete Seattle-to-Chicago call, **81312** must be followed by the 7-digit local phone number in Chicago.

Answer: enter **12** as number of Total Digits in Outbound Phone book of Seattle voip.

Euro, National Call Example

London/Birming. system.

To complete London-to-Birmingham call, **90121** must be followed by the 7-digit local phone number in Birmingham.

Answer: enter **12** as number of Total Digits in Outbound Phone book of London voip.

Euro, International Call Example

Rotterdam/Bordeaux system.

To complete Rotterdam-to-Bordeaux call, **903305** must be followed by 8-digit local phone number in Bordeaux.

Answer: enter **14** as number of Total Digits in Outbound Phonebook of Rotterdam voip.

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

North America, Long-Distance Example

Seattle-Chicago system.

Answer: enter **8** in "Remove Prefix" field of Seattle Outbound Phonebook.

Euro, National Call Example

London/Birming. system.

Answer: enter **9** in "Remove Prefix" field of London Outbound Phonebook.

Euro, International Call Example

Rotterdam/Bordeaux system.

Answer: enter **9** in "Remove Prefix" field of Outbound Phonebook for Rotterdam voip.

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.

8. Select the voip protocol that you will use (H.323 or SIP).
9. Click **OK** to exit from the **Add/Edit Outbound Phonebook** screen.

Inbound Phonebook

1. Open the MultiVOIP program.
   (**Start | MultiVOIP xxx | Configuration**
2. Go to **Phone Book | PhoneBook Modify | Inbound Phonebook | Add Entry**.
3. In the "Remove Prefix" field, enter your local calling code (area code, country code, city code, etc.) 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

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.

Euro, National Call Example

London/Birming. 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.

Euro, International Call Example

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

Seattle-Chicago system.

On Seattle PBX, "8" is used to get an outside line.

Answer: **8** is the prefix to be added by local (Seattle) voip.

Euro, National Call Example

London/Birming. system.

On London PBX, "9" is used to get an outside line.

Answer: **9** is the prefix to be added by local (London) voip.

Euro, International Call Example

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 "0." A zero value means the voip unit will assign the call to an available channel. If desired, specific channels can be assigned to specific incoming calls (i.e., 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. (40 characters max.)

North America, Long-Distance Example

Seattle-Chicago system.

Possible Description:.
Free Seattle access, all employees

Euro, National Call Example

London/Birming. system.

Possible Description:.
Local-rate London access, all employees

Euro, International Call Example

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.

Phonebook Tips

Preparing the phonebook for your voip system is a complex task that, at first, seems quite daunting. These tips may make the task easier.

1. **Use Dialing Patterns, Not Complete Phone Numbers**. You will not generally enter complete phone numbers in the voip phonebook. Instead, you'll enter "destination patterns" that involve area codes and other digits. If the destination pattern is a whole area code, you'll be assigning all calls to that area code to go to a particular voip that has a unique IP address. If your destination pattern includes an area code plus a particular local phone exchange number, then the scope of calls sent through your voip system will be narrowed (only calls within that local exchange will be handled by the designated voip, not all calls in that whole area code). In general, when there are fewer digits in your destination pattern, you are asking the voip to handle calls to more destinations.

2. **The Four Types of Phonebook Digits Used. Important!**
"Destination patterns" to be entered in your phonebook will generally consist of:

(a) calling area codes,
(b) access codes,
(c) local exchange numbers, and
(d) specialized codes.

Although voip phonebook entries may look confusing at first, it's useful to remember that all the digits in any phonebook entry must be of one of these four types.

(a) **calling area codes**. There are different names for these around the world: "area codes," "city codes," "country codes," etc. These codes, are used when making non-local calls. They always precede the phone number that would be dialed when making a local call.

(b) **access codes**. There are digits (*PSTN access codes*) that must be dialed to gain access to an operator, to access the publicly switched 'long-distance' calling system(North America), to access the publicly switched 'national' calling system (Europe and elsewhere), or to access the publicly switched 'international' calling system (worldwide).

There are digits (*PBX access codes*) that must be dialed by phones connected to PBX systems or key systems. Often a "9" must be dialed on a PBX phone to gain access to the PSTN ('to get an outside line'). Sometimes "8" must be dialed on a PBX phone to divert calls onto a leased line or to a voip system. However, sometimes PBX systems are 'smart' enough to route calls to a voip system without a special access code (so that "9" might still be used for all calls outside of the building).

There are also digits (*special access codes*) that must be dialed to gain access to a particular discount long-distance carrier or to some other closed or proprietary telephone system.

(c) **local exchange numbers**. Within any calling area there will be many local exchange numbers. A single exchange may be used for an entire small town. In cities, an exchange may be used for a particular neighborhood (although exchanges in cities do not always cover easily discernible areas). Organizations like businesses, governments, schools, and universities are also commonly assigned exchange numbers for their exclusive use. In some cases, these organizational-assigned exchanges can become non-localized because the exchange is assigned to one facility and linked, by the organization's private network, to other sometimes distant locations.

(d) **specialized codes**. Some proprietary voip units assign, to sites and phone stations, numbers that are not compatible with PSTN numbering. This can also occur in PBX or key systems. These specialized numbers must be handled on a case-by-case basis.

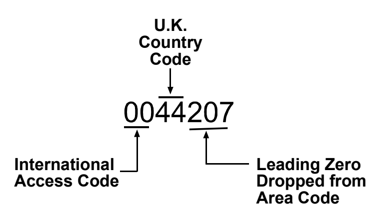
3. **Knowing When to Drop Digits**.

When calling area codes and access codes are used in combination, a leading "1" or "0" must sometimes be dropped.

Example

Area code for Inner London is listed as "0207." However, in international calls the leading "0" is dropped.

Phonebook Entry ➠



4. **Using a Comma**.

Commas are used in telephone dialing strings to indicate a pause to allow a dial tone to appear (common on PBX and key systems). Commas may be used only in the "Add Prefix" field of the Inbound Phonebook.

Detail
, **= 1-second pause**
In many PBX systems (not needed in all)

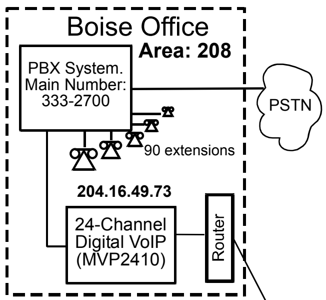
5. **Ease of Use**. The phonebook setup determines how easy the voip system is to use. Generally, you'll want to make it so dialing a voip call is very similar to dialing any other number (on the PSTN or through the PBX).

6. **Avoid Unintentional Calls to Official/Emergency Numbers**. Dialing a voip call will typically be somewhat different than ordinary dialing. Because of this, it's possible to set up situations, quite unwittingly, where phone users may be predisposed to call official numbers without intending to do so. Conversely, a voip/PBX system might also make it difficult to place an official/emergency call when one intends to do so. Study your phonebook setup and do some dialing on the system to avoid these pitfalls.

7. **Inbound/Outbound Pattern Matching**. In general, the Inbound Phonebook entries of the local voip unit will match the Outbound Phonebook entries of the remote voip unit. Similarly, the Outbound Phonebook entries of the local voip unit will match the Inbound Phonebook entries of the remote voip unit. There will often be non-matching entries, but it's nonetheless useful to notice the matching between the phonebooks.

8. **Simulating Network in-lab/on-benchtop**. One common method of configuring a voip network is to set up a local IP network in a lab, connect voip units to it, and perhaps have phones connected on channel banks to make test calls.

Phonebook Example



One Common Situation

Voip Example. This company has offices in three different cities. The PBX units all operate alike. Notably, they all give access to outside lines using “9.” They all are ‘smart’ enough to identify voip calls without using a special access digit (“8” is used in some systems). Finally, the system operates so that employees in any office can dial employees in any other office using only three digits. Here are the phonebooks needed for that system.

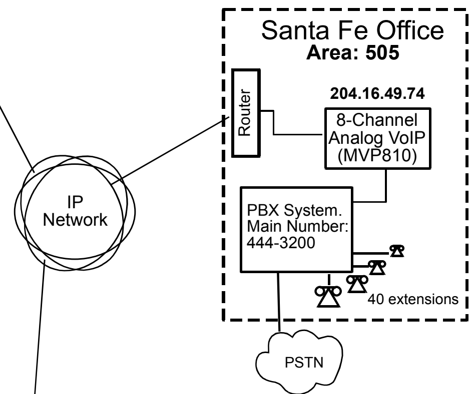
Inbound Phonebook

Each **Inbound Phonebook** contains two entries. The first entry (4 digits) specifies how incoming calls from the other voip sites will be handled if they go out onto the local PSTN. Essentially, all those calls come to the receiving voip with a pattern beginning with **1+area code**. The local voip removes those four digits because they aren’t needed when dialing locally. The local voip attaches a “9” at the beginning of the number to get an outside line. The PBX then completes the call to the PSTN.

The second **Inbound** Phonebook entry (8 digits) is for receiving calls from company employees in the other two cities. The out-of-town employee simply dials 3 digits. The first of the three digits is uniquely used at each site and so acts as a destination pattern (Boise extensions are 7xx, Santa Fe extensions 2xx, Flagstaff extensions 6xx).

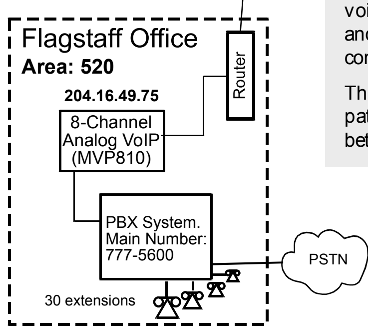
As the remote voip sends out the call, it automatically attaches all of the foregoing digits that would normally have to be dialed using the PSTN.

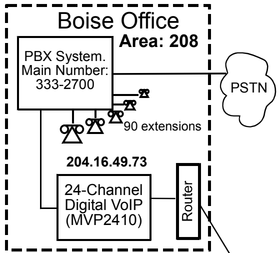
The local (receiving) voip sees the extended pattern in its Inbound Phonebook and so strips off the long telltale pattern of digits needed for 3-digit calling. It must finally add back the last digit before handing the call to the PBX, which completes the call to a specific extension.



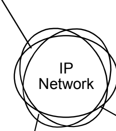
Each **Outbound Phonebook** contains two pairs of entries, two entries for each remote site. Whenever an out-of-town employee dials a 12-digit number beginning with the listed 5-digit destination pattern (9+1+area code) of another company location, the PBX hands the call to the voip system. The local voip strips off the “9” and directs the call to the IP address of the remote voip. The remote voip receives the call and hands it to its PBX. The PBX then completes the call to the PSTN.

The one-digit **Outbound** destination patterns pertain to 3-digit calling between company employees.

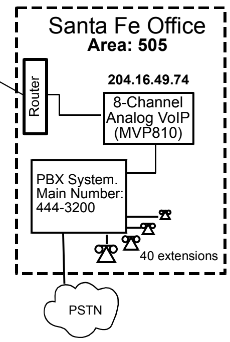


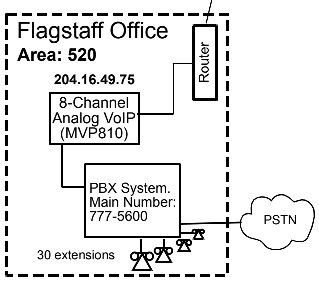


Boise Voip			Boise Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
1208	9	Incoming calls to PSTN, Boise Area	91505	12	9	none	204.16 .49.74	Outgoing calls to Santa Fe area
12083332	2	Incoming calls to extensions of company's PBX system in Boise	2	3	none	1505 444 3	204.16 .49.74	3-digit calls to Santa Fe employees
			91520	12	9	none	204.1 6.49.7 5	Outgoing calls to Flagstaff area
			6	3	none	1520 777 5	204.1 6.49.7 5	3-digit calls to Flagstaff employees



Santa Fe Voip			Santa Fe Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
1505	9,	Incoming calls to PSTN, Santa Fe local calls	91208	12	9	none	204. 16.49. 73	Outgoing calls to Boise area
150544432	2	Incoming calls to extensions of company's PBX system in Santa Fe	7	3	none	1208 333 2	204.1 6.49. 73	Outgoing calls to extensions of company's Boise PBX (3-digit dialing)
			91520	12	9	none	204. 16.49. 75	Outgoing calls to Flagstaff area
			6	3	none	1520 777 5	204. 16.49. 75	3-digit calls to Flagstaff employees





Flagstaff Voip			Flagstaff Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
1520	9	Incoming calls to PSTN, Flagstaff local calls	91505	12	9	none	204.16 .49.74	Outgoing calls to Santa Fe area
15207775	5	Incoming calls to extensions of company's PBX system in Flagstaff	2	3	none	1505 444 3	204.16 .49.74	3-digit calls to Santa Fe employees
			91208	12	9	none	204.16 .49.73	Outgoing calls to Boise area
			7	3	none	1208 333 2	204.16 .49.73	3-digit calls to Boise employees

Sample Phonebooks Enlarged

Boise Voip			Boise Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
1208	9,	Incoming calls to PSTN, Boise Area	91505	12	9	none	204. 16.49. 74	Outgoing calls to Santa Fe area
120833327	7	Incoming calls to extensions of company's PBX system in Boise	2	3	none	1505 444 3	204. 16.49. 74	3-digit calls to Santa Fe employees (extensions 200 to 240)
			91520	12	9	none	204. 16.49. 75	Outgoing calls to Flagstaff area
			6	3	none	1520 777 5	204. 16.49. 75	3-digit calls to Flagstaff employees (extensions 600-630)

Santa Fe Voip			Santa Fe Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
1505	9,	Incoming calls to PSTN, Santa Fe local calls	91208	12	9	none	204. 16.49. 73	Outgoing calls to Boise area
150544432	2	Incoming calls to extensions of company's PBX system in Santa Fe	7	3	none	1208 333 2	204. 16.49. 73	3-digit calls to Boise employees (extensions 700-790)
			91520	12	9	none	204. 16.49. 75	Outgoing calls to Flagstaff area
			6	3	none	1520 777 5	204. 16.49. 75	3-digit calls to Flagstaff employees (extensions 600-630)

Flagstaff Voip			Flagstaff Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
1520	9,	Incoming calls to PSTN, Flagstaff local calls	91505	12	9	none	204.16 .49.74	Outgoing calls to Santa Fe area
152077756	6	Incoming calls to extensions of company's PBX system in Flagstaff	2	3	none	1505 444 3	204.16 .49.74	3-digit calls to Santa Fe employees (extensions 200-240)
			91208	12	9	none	204.16 .49.73	Outgoing calls to Boise area
			7	3	none	1208 333 2	204.16 .49.73	3-digit calls to Boise employees (extensions 700-790)

Phonebook Worksheet

Voip Location/ID:____________________________

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Voip Location/ID:____________________________

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Voip Location/ID:____________________________

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Enlarged Phonebook Worksheet

Voip Location/ID:

Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls

Other Details:

Connectivity Test

The procedures "Phone/IP Starter Configuration" and "Phonebook Starter Configuration" must be completed before you can do this procedure.

1. These connections must be made:

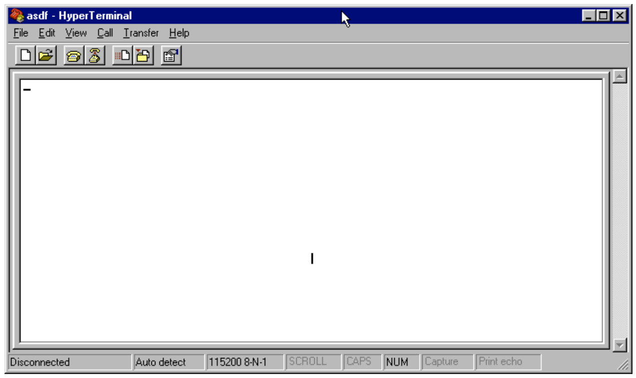
Connections	
for digital MultiVOIPs (MVP-2400/2410/3010	**for analog MultiVOIPs (MVP-130/210/410/810, MVP-210G/410G/810G)**
MultiVOIP to local PBX	MultiVOIP to local phone station –OR-- MultiVOIP to extension of key phone system
MultiVOIP to command PC	MultiVOIP to command PC
MultiVOIP to Internet	MultiVOIP to Internet

2. Inbound Phonebook and Outbound Phonebook must both be set up with at least one entry in each. These entries must allow for connection between two voip units.

3. Console messages must be enabled. (If this has not been done already, go, in the MultiVOIP GUI, to Configuration | Logs and select the "Console Messages" checkbox.

4. You now need to free up the COM port connection (currently being used by the MultiVOIP program) so that the HyperTerminal program can use it. To do this, you can either (a) click on **Connection** in the sidebar and select "Disconnect" from the drop-down box, or (b) close down the MultiVOIP program altogether.

5. Open the **HyperTerminal** program.



6. Use HyperTerminal to receive and record console messages from the MultiVOIP unit. To do so, set up HyperTerminal as follows (setup shown is for Windows NT4; details will differ slightly in other MS operating systems):

In the upper toolbar of the HyperTerminal screen, click on the **Properties** button.

In the "Connect To" tab of the **Connection Properties** dialog box, click on the **Configure** button.

In the next dialog box, on the "General" tab, set "Maximum Speed" to 115200 bps.

On the "Connection" tab, set connection preferences to:

Data bits: 8

Parity: none

Stop bits: 1

Click **OK** twice to exit settings dialog boxes.

7. Make VOIP call.

for digital MultiVOIPs (MVP-2400/2410/3010	**for analog MultiVOIPs (MVP-130/210/410/810)**
Make call from an extension of the local PBX.	Make call on a local phone line accessing PSTN directly or through key system

8. Read console messages recorded on HyperTerminal.

Console Messages from **Originating VOIP**. The voip unit that originates the call will send back messages like that shown below.

```
[00026975]  CAS[0] : RX : ABCD = 1, 1, 1, 1,Pstn State[1]
             TimeStamp : 26975
[00027190]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00027190]  PSTN: cas seizure detected on 0
[00027440]  CAS[0] : TX : ABCD = 0, 0, 0, 0
[00033290]  PSTN:call detected on 0 num=17637175662*
[00033290] H323IF[0]:destAddr =
             TA:200.2.10.5:1720,NAME:Mounds
             View,TEL:17637175662,17637175662
[00033290]  H323IF[0]:srcAddr  = NAME:New
             York,TA:200.2.9.20
[00033440]  H323IF [0]:cmCallStateProceeding
[00033500]  H323[0]: Remote Information (Q931): MultiVOIP
             - T1
[00033565]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00033675]  H323IF [0]: MasterSlaveStatus=Slave
[00033675]  H323IF[0]:FastStart Setup Not Used
[00033690]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00033755]  H323IF[0]: Coder used 'g7231'
[00033810]  PSTN:pstn call connected on 0
```

Console Messages from **Terminating VOIP**. The voip unit connected to the phone where the call is answered will send back messages like that shown below.

```
[00170860]  H323[0]: New incoming call
[00170860]  PSTNIF : Placing call on channel 0  Outbound
             digit 7175662
[00170885]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00171095]  H323IF [0]: MasterSlaveStatus=Master
[00171105]  CAS[0] : RX : ABCD = 1, 1, 1, 1,Pstn State[7]
             TimeStamp : 171105
[00171105]  H323IF[0]: Coder used 'g7231'
[00171110]  H323IF[0]:FastStart Setup Not Used
[00171110]  H323IF[0]: Already opened the outgoing logical
             channel
[00171110]  H323IF[0]: Coder used 'g7231'
[00171315]  CAS[0] : RX : ABCD = 0, 0, 0, 0,Pstn State[9]
             TimeStamp : 171315
[00172275]  PSTN: dialing digit ended on 0
[00172285]  PSTN: pstn proceeding indication on 0
[00172995]  CAS[0] : RX : ABCD = 1, 1, 1, 1,Pstn State[12]
             TimeStamp : 172995
[00173660]  CAS[0] : TX : ABCD = 1, 1, 1, 1
[00173760]  PSTN:pstn call connected on 0
```

9. When you see the following message, end-to-end voip connectivity has been achieved.

   "**PSTN: pstn call connected on X**"

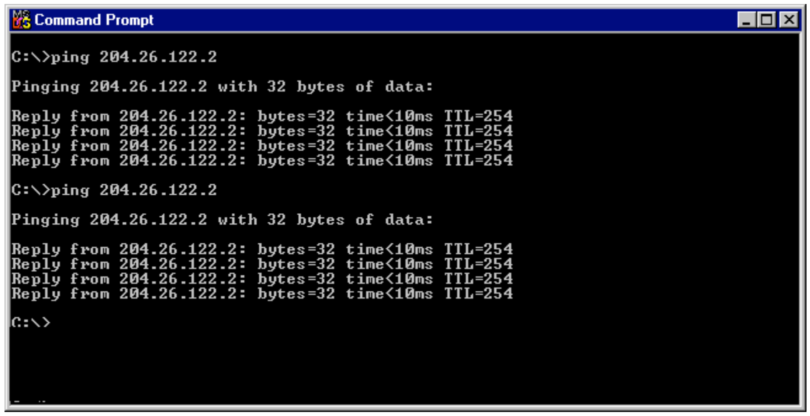
   where x is the number of the voip channel carrying the call

10. If the HyperTerminal messages do not confirm connectivity, go to the ***Troubleshooting*** procedure below.

Troubleshooting

If you cannot establish connectivity between two voips in the system, follow the steps below to determine the problem.

1. Ping both MultiVOIP units to confirm connectivity to the network.



2. Verify the telephone connections.

A. **For MVP2400, MVP2410, or MVP3010**.
Check cabling. Are connections well seated? To correct receptacle?
Is the **ONL** LED on?
(If on, ONL indicates that the MultiVOIP is online on the network.)
Are T1/E1/PRI Parameter settings correct?

B. **For MVP130, MVP210, MVP410, or MVP810**.
Check cabling. Are connections well seated? To correct receptacle?
Are telephone Interface Parameter settings correct?

C. **For MVP410ST** or **MVP810ST**.
Check cabling. Are connections well seated? To correct receptacle?
If terminal equipment is connected to the voip, then "Network" should be selected for that BRI interface in the **ISDN BRI Parameters** screen.
Note: *Each BRI interface is separately configurable.*
If network equipment such as an ISDN BRI PBX or an ISDN BRI line from a phone company is connected to the voip, then "Terminal" should be selected for that BRI interface in the **ISDN BRI Parameters** screen.
Was the proper country and operator chosen?
Was the proper type of line coding (A-law or u-law) chosen?

3. Verify phonebook configuration.
4. Observe console messages while placing a call. Look for error messages indicating phonebook problems, network problems, voice-coder mismatches, etc.

Chapter 3: Mechanical Installation and Cabling

Introduction

The MultiVOIP models MVP130, MVP210, and MVP2400 are tabletop units and can be handled easily by one person. However, the MVP410, MVP810, MVP2410, and MVP3010 are somewhat heavier units. When these units are to be installed into a rack, two able-bodied persons should participate.

Please read the safety notices before beginning installation.

Safety Warnings

Lithium Battery Caution

A lithium battery on the voice/fax channel board provides backup power for the timekeeping capability. The battery has an estimated life expectancy of ten years.

When the 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.

Warning: There is danger of explosion if the battery is incorrectly replaced.

Safety Warnings Telecom

1. Never install telephone wiring during a lightning storm.
2. Never install a telephone jack in wet locations unless the jack is specifically designed for wet locations.
3. This product is to be used with UL and UL listed computers.
4. Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
5. Use caution when installing or modifying telephone lines.
6. Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electrical shock from lightning.
7. Do not use a telephone in the vicinity of a gas leak.
8. To reduce the risk of fire, use only a UL-listed 26 AWG or larger telecommunication line cord.

Unpacking Your MultiVOIP

When unpacking your MultiVOIP, check to see that all of the items shown are included in the box. For the various MultiVOIP models, the contents of the box will be different. Study the particular illustration below that is appropriate to the model you have purchased. If any box contents are missing, contact MultiTech Tech Support at 1-800-972-2439.

Unpacking the MVP2410/3010

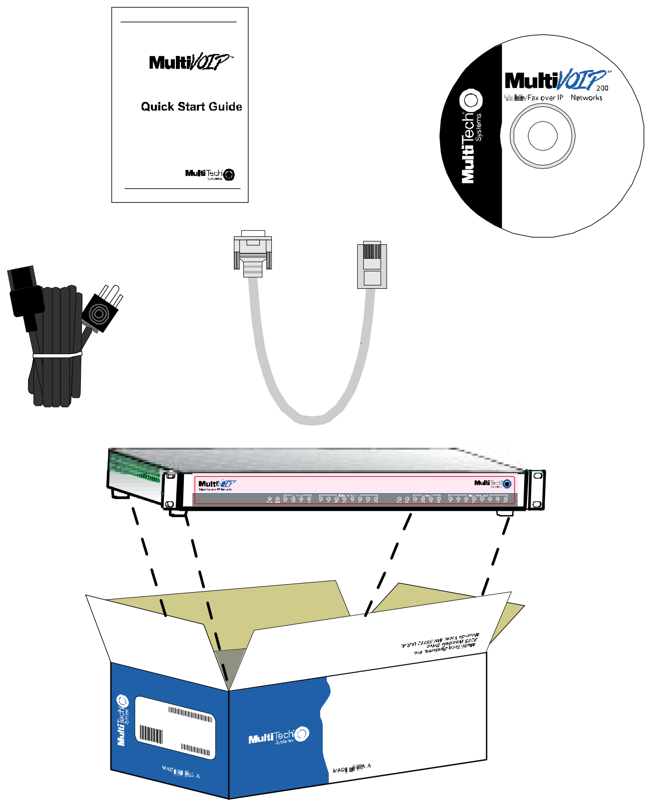


Figure 3-1: Unpacking the MVP2410/3010

Unpacking the MVP2400

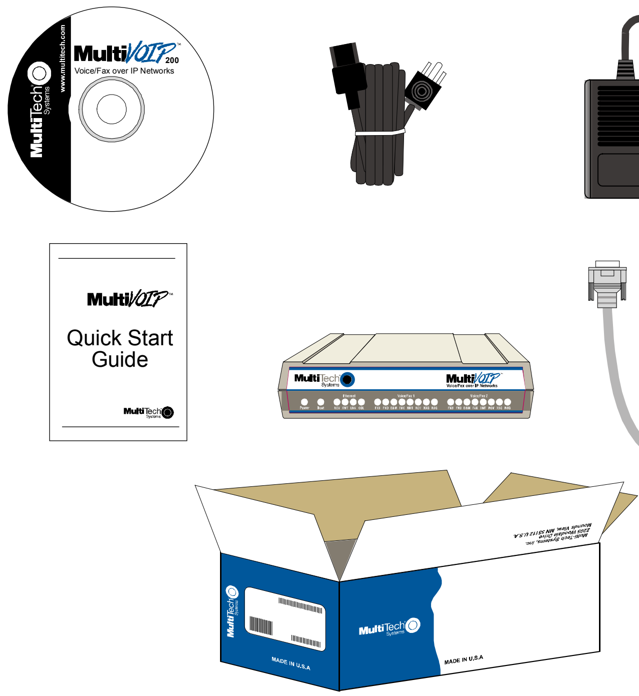


Figure 3-2: Unpacking the MVP2400

Unpacking the MVP-410x/810x

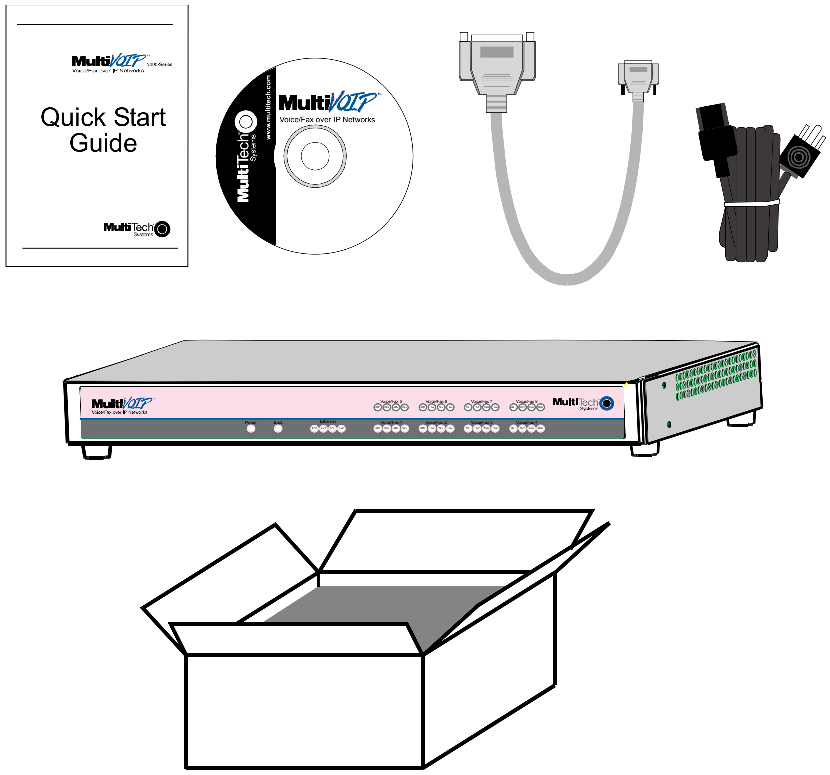


Figure 3-3: Unpacking the MVP-410x/810x

Unpacking the MVP210x

Figure 3-4: Unpacking the MVP210x

Unpacking the MVP130

Figure 3-5: Unpacking the MVP130

Rack Mounting Instructions for MVP-2410/3010 & MVP-410x/810x

The MultiVOIPs can be mounted in an industry-standard EIA 19-inch rack enclosure, as shown in Figure 3-6.

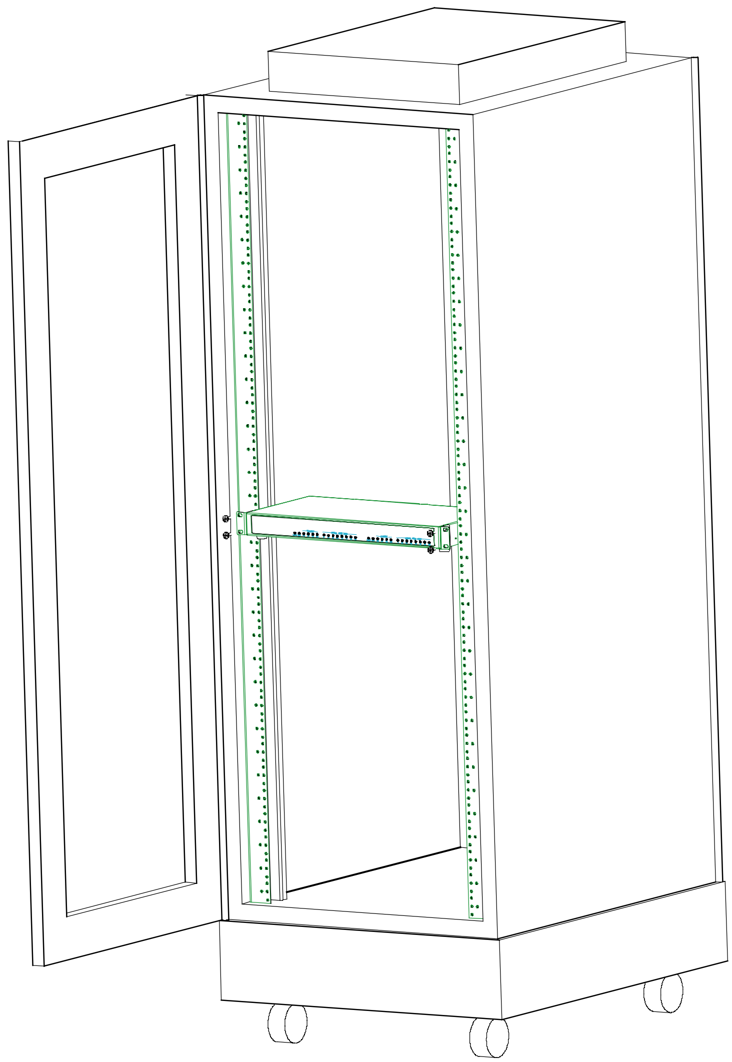


Figure 3-6: Rack-Mounting (MVP2410/3010 or MVP410x/810x)

Safety Recommendations for Rack Installations

Ensure proper installation of the unit in a closed or multi-unit enclosure by following the recommended installation as defined by the enclosure manufacturer. Do not place the unit directly on top of other equipment or place other equipment directly on top of the unit. If installing the unit in a closed or multi-unit enclosure, ensure adequate airflow within the rack so that the maximum recommended ambient temperature is not exceeded. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack. If a power strip is used, ensure that the power strip provides adequate grounding of the attached apparatus.

When mounting the equipment in the rack, make sure mechanical loading is even to avoid a hazardous condition, such as loading heavy equipment in rack unevenly. The rack used should safely support the combined weight of all the equipment it supports.

Ensure that the mains supply circuit is capable of handling the load of the equipment. See the power label on the equipment for load requirements (full specifications for MultiVOIP models are presented in chapter 1 of this manual).

Maximum ambient temperature for the unit is 40 degrees Celsius (104 degrees Fahrenheit). This equipment should only be installed by properly qualified service personnel. Only connect like circuits. In other words, connect SELV (Secondary Extra Low Voltage) circuits to SELV circuits and TN (Telecommunications Network) circuits to TN circuits.

19-Inch Rack Enclosure Mounting Procedure

Attaching the MultiVOIP to a rack-rail of an EIA 19-inch rack enclosure will certainly require two persons. Essentially, the technicians must attach the brackets to the MultiVOIP chassis with the screws provided, as shown in Figure 3-7, and then secure unit to rack rails by the brackets, as shown in Figure 3-8. Because equipment racks vary, screws for rack-rail mounting are not provided. Follow the instructions of the rack manufacturer and use screws that fit.

1. Position the right rack-mounting bracket on the MultiVOIP using the two vertical mounting screw holes.
2. Secure the bracket to the MultiVOIP using the two screws provided.
3. Position the left rack-mounting bracket on the MultiVOIP using the two vertical mounting screw holes.
4. Secure the bracket to the MultiVOIP using the two screws provided.
5. Remove feet (4) from the MultiVOIP unit.
6. Mount the MultiVOIP in the rack enclosure per the rack manufacture's mounting procedure.

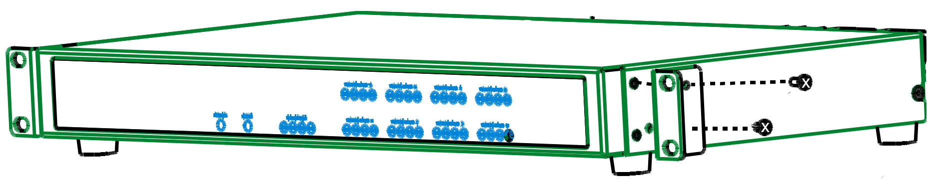


Figure 3-7: Bracket Attachment for Rack Mounting (MVP-2410/3010 & MVP-410x/810x)

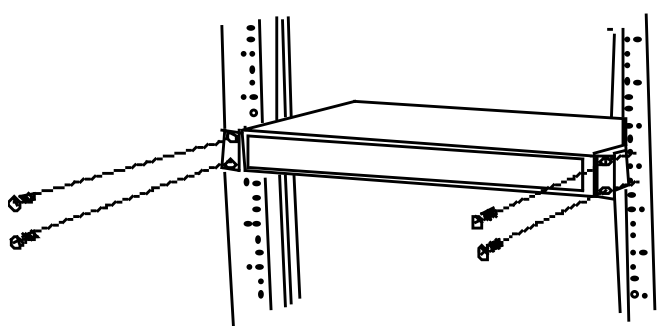


Figure 3-8: Attaching MultiVOIP to Rack Rail (MVP-2410/3010 & MVP-410x/810x)

Cabling

Cabling Procedure for MVP2410/3010

Cabling your MultiVOIP entails making the proper connections for power, command port, phone system (T1/E1 line connected to PBX or telco office), and Ethernet network. Figure 3-9 shows the back panel connectors and the associated cable connections. The following procedure details the steps necessary for cabling your MultiVOIP.

1. Connect the power cord to a live AC outlet, then connect it to the MultiVOIP's power receptacle shown at top right in Figure 3-9.

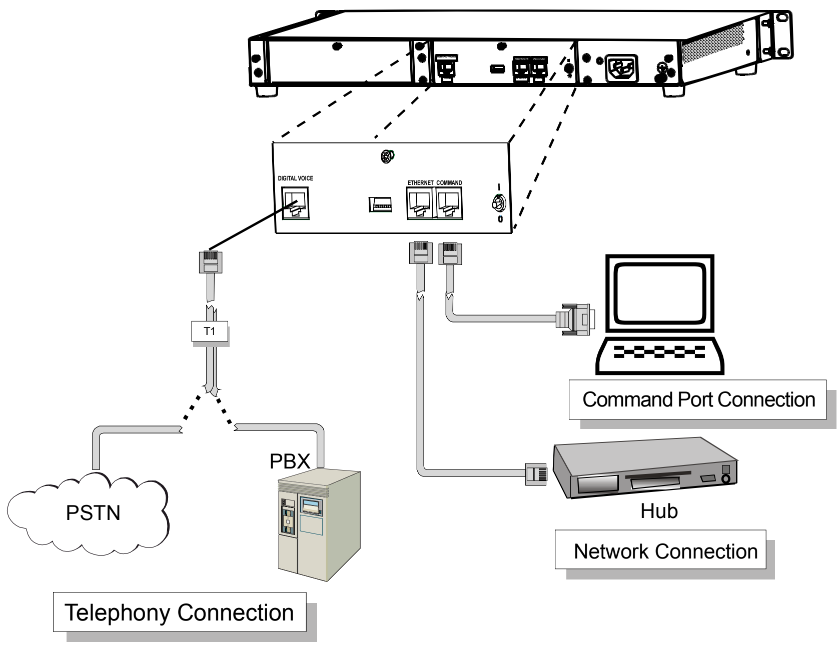


Figure 3-9. Cabling for MVP2410/3010

2. Connect the MultiVOIP to the PC (the computer that will hold the MultiVOIP software) using the RJ-45 to DB9 (female) cable provided with your unit. Plug the RJ-45 end of the cable into the **Command** port of the MultiVOIP and connect the other end (the DB9 connector) to the PC serial port you are using (typically COM1 or COM2). See Figure 3-9.

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.

4. Turn on power to the MultiVOIP by setting the power switch on the right side panel to the **ON** position. Wait for the **Boot** LED on the MultiVOIP to go off before proceeding. This may take a couple of minutes.

Proceed to Chapter 4 "Software Installation."

Cabling Procedure for MVP2400

Cabling your MultiVOIP entails making the proper connections for power, command port, phone system (T1 line connected to PBX or telco office), and Ethernet network. Figure 3-10 shows the back panel connectors and the associated cable connections. The following procedure details the steps necessary for cabling your MultiVOIP.

1. Connect the power supply to a live AC outlet, then connect it to the MultiVOIP as shown in Figure 3-10.

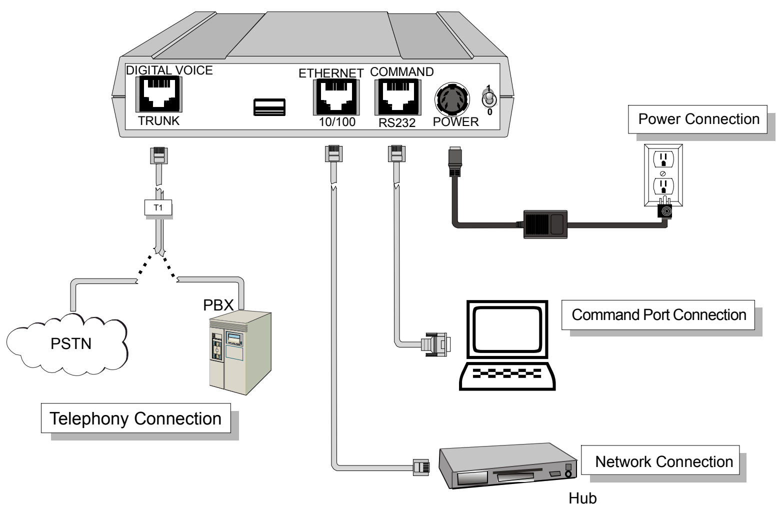


Figure 3-10: Cabling for MVP2400

2. Connect the MultiVOIP to the PC (the computer that will hold the MultiVOIP software) using the RJ-45 to DB9 (female) cable provided with your unit. Plug the RJ-45 end of the cable into the **Command** port of the MultiVOIP and connect the other end (the DB9 connector) to the PC serial port you are using (typically COM1 or COM2). See Figure 3-10.

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.

4. Turn on power to the MultiVOIP by setting the power switch on the right side panel to the **ON** position. Wait for the **Boot** LED on the MultiVOIP to go off before proceeding. This may take a couple of minutes.

Proceed to Chapter 4 “Software Installation.”

Cabling Procedure for MVP-410/410G/810/810G

Cabling involves connecting the MultiVOIP to your LAN and telephone equipment.

1. Connect the power cord supplied with your MultiVOIP to a live AC outlet and to the power connector on the back of the MultiVOIP as shown at top right in Figure 3-11.

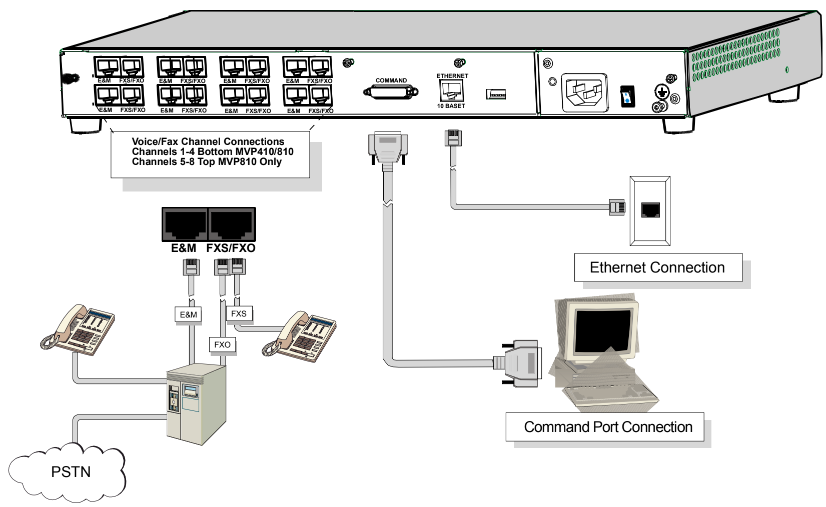


Figure 3-11: Cabling for MVP-410/410G/810/810G

2. Connect the MultiVOIP to a PC by using a DB-25 (male) to DB-9 (female) cable. Plug the DB-25 end of the cable into the Command port of the MultiVOIP and the other end into the PC serial port. See Figure 3-11.
3. Connect a network cable to the **ETHERNET 10BASET** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.
4. If you are connecting a station device such as an analog telephone, a fax machine, or a Key Telephone System (KTS) (FXS interface), or a PBX extension (FXO interface) to your MultiVOIP, connect one end of an RJ-11 phone cord to the Channel **1 FXS/FXO** connector on the back of the MultiVOIP and the other end to the device or phone jack. You will define the interface in the Interface dialog box in the software when you configure the unit.

If you are connecting an E&M trunk from a telephone switch to your MultiVOIP, connect one end of an RJ-45 phone cord to the Channel **1 E&M** connector on the back of the MultiVOIP and the other end to the trunk. Verify that the E&M Type in the E&M Options group of the Interface dialog box is the same as the E&M trunk type support by the telephone switch. See Appendix B for an E&M cabling pinout.

5. Repeat the above step to connect the remaining telephone equipment to each channel on your MultiVOIP.
6. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack.

   This can be accomplished by connecting a grounding wire between the chassis and a metallic object that will provide an electrical ground.
7. Turn on power to the MultiVOIP by placing the ON/OFF switch on the back panel to the ON position. Wait for the **Boot** LED on the MultiVOIP to go off before proceeding. This may take a few minutes.

   Proceed to Chapter 4 to load the MultiVOIP software.

Cabling Procedure for MVP-410ST/810ST

Cabling involves connecting the MultiVOIP to your LAN and telephone equipment.

1. Connect the power cord supplied with your MultiVOIP to a live AC outlet and to the power connector on the back of the MultiVOIP as shown at top right in Figure 3-12.

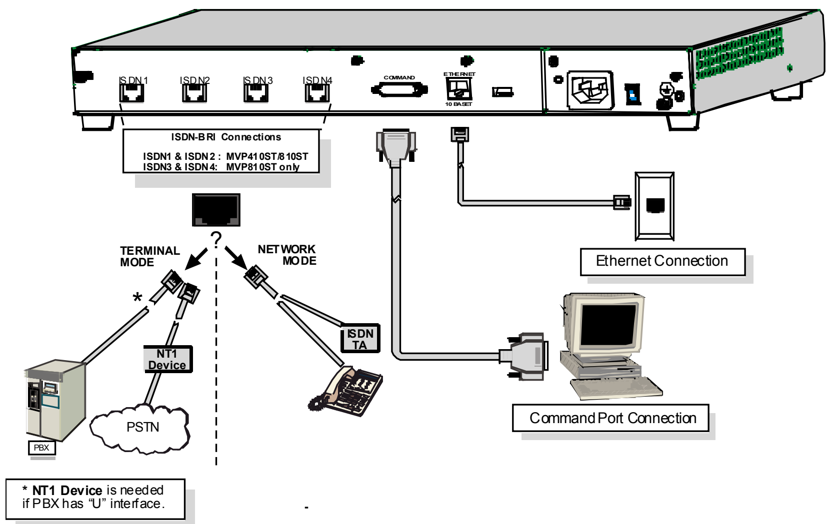


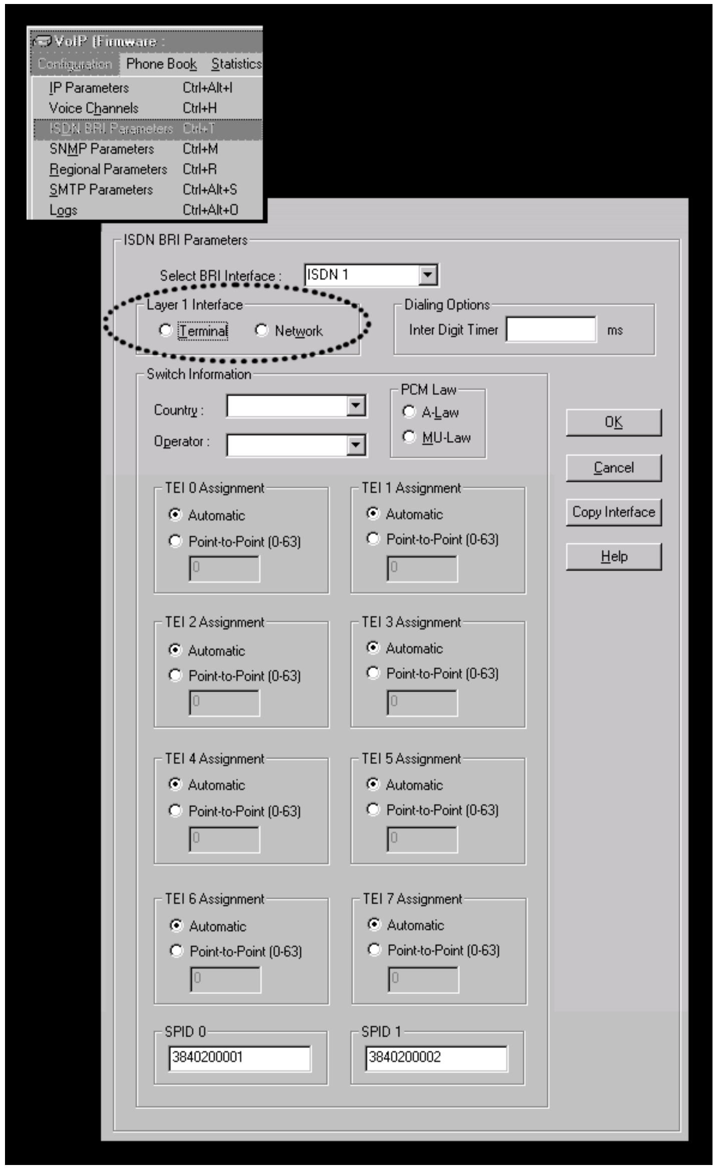
Figure 3-12: Cabling for MVP-410ST/810ST

2. Connect the MultiVOIP to a PC by using a DB-25 (male) to DB-9 (female) cable. Plug the DB-25 end of the cable into the Command port of the MultiVOIP and the other end into the PC serial port. See Figure 3-12.

3. Connect a network cable to the **ETHERNET 10BASET** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.

4. **Terminal Mode**. When a voip ISDN connector is to be connected to a PBX extension line or to a telco line, select "Terminal" as the "Layer 1 Interface" in the **ISDN Parameters** screen. When making cable connections, an NT1 device will be needed between the MultiVOIP and the PSTN or between the MultiVOIP and any PBX with a "U" interface. (For more information, see *Appendix B: Cable Pinouts* in this manual.) Connect cables between voip ISDN connectors and network equipment.

   NOTE: In order to operate in **Terminal** mode, the network equipment to which you will be connecting (e.g., PBX) must support D-channel signaling in its ISDN-S/T interface.



Network Mode. When a voip ISDN connector is to be connected to an ISDN phone station or to an ISDN terminal adapter (TA), select “Network” as the “Layer 1 Interface” in the **ISDN Parameters** screen of the MultiVOIP software. Connect cables between voip ISDN connectors and phone or TA.

NOTE. Any ISDN phone stations connected to the MVP-410ST/810ST must provide their own operating power. That is, the MVP-410ST/810ST does not supply power for ISDN phone stations.

5. Repeat the above step to connect the remaining ISDN telephone equipment to each ISDN connector on your MultiVOIP. Be aware that you can assign each ISDN line separately and independently to either Network mode or Terminal mode. That is, all ISDN lines do not have to be assigned in to the same operating mode.
6. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack.

   This can be accomplished by connecting a grounding wire between the chassis and a metallic object that will provide an electrical ground.
7. Turn on power to the MultiVOIP by placing the ON/OFF switch on the back panel to the ON position. Wait for the **Boot** LED on the MultiVOIP to go off before proceeding. This may take a few minutes.

   Proceed to Chapter 4 to load the MultiVOIP software.

Cabling Procedure for MVP210x

Cabling involves connecting the MultiVOIP to your LAN and telephone equipment.

1. Connect the power cord supplied with your MultiVOIP to the power connector on the back of the MultiVOIP and to a live AC outlet as shown in Figure 3-13.

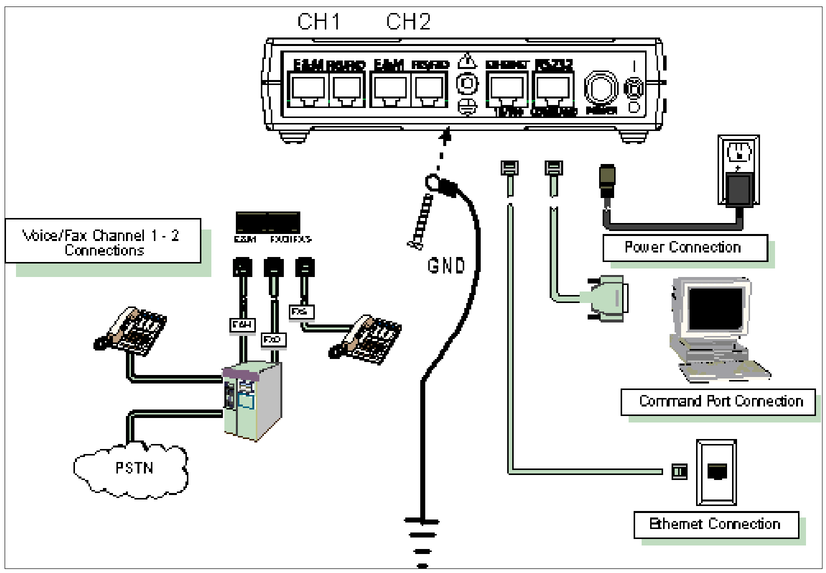


Figure 3-13: Cabling for MVP210x

2. Connect the MultiVOIP to a PC by using a 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. See Figure 3-13.
3. Connect a network cable to the **ETHERNET 10/100** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.
4. If you are connecting a station device such as an analog telephone, a fax machine, or a Key Telephone System (KTS) (FXS interface), or a PBX extension (FXO interface) to your MultiVOIP, connect one end of an RJ-11 phone cord to the Channel **1 FXS/FXO** connector on the back MultiVOIP and the other end to the device or phone jack. You will define the interface in the Interface dialog box in the software when you configure the unit.

If you are connecting an E&M trunk from a telephone switch to your MultiVOIP, connect one end of an RJ-45 phone cord to the Channel **1 E&M** connector on the back of the MultiVOIP and the other end to the trunk. Verify that the E&M Type in the E&M Options group of the Interface dialog box is the same as the E&M trunk type support by the telephone switch. See Appendix B for an E&M cabling pinout.

5. Repeat the above step to connect the remaining telephone equipment to the second channel on your MultiVOIP.
6. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack.

   This can be accomplished by connecting a grounding wire between the chassis and a metallic object that will provide an electrical ground.
7. Turn on power to the MultiVOIP by placing the ON/OFF switch on the back panel to the ON position. Wait for the **BOOT** LED on the MultiVOIP to go off before proceeding. This may take a few minutes.

   Proceed to Chapter 4 to load the MultiVOIP software.

Cabling Procedure for MVP130

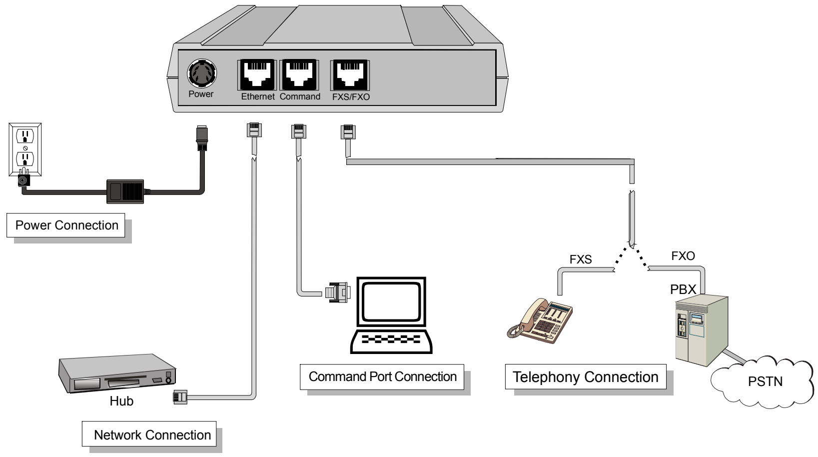


Figure 3-14: Cabling for MVP130

Cabling involves connecting the MultiVOIP to your LAN and telephone equipment.

1. Connect the power cord supplied with your MultiVOIP to the power connector on the back of the MultiVOIP and to a live AC outlet as shown in Figure 3-14.
2. Connect the MultiVOIP to a PC by using a 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. See Figure 3-14.
3. Connect a network cable to the **ETHERNET 10/100** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.
4. If you are connecting a station device such as an analog telephone, a fax machine, or a Key Telephone System (KTS) (FXS interface), or a PBX extension (FXO interface) to your MultiVOIP, connect one end of an RJ-11 phone cord to the Channel **1 FXS/FXO** connector on the back MultiVOIP and the other end to the device or phone jack. You will define the interface in the Interface dialog box in the software when you configure the unit.

Proceed to Chapter 4 to load the MultiVOIP software.

Chapter 4: Software Installation

Introduction

Configuring software for your MultiVOIP entails three tasks:
(1) loading the software onto the PC (this is "Software Installation and is discussed in this chapter),
(2) setting values for telephony and IP parameters that will fit your system (this is "Technical Configuration" and it is discussed in Chapter 5 for T1/E1 MultiVOIP units and in Chapter 6 for analog MultiVOIP units), and
(3) establishing "phonebooks" that contain the various dialing patterns for VOIP calls made to different locations (this is "Phonebook Configuration" and it is discussed in Chapters 7, 8, and 9 for T1, E1, and analog MultiVOIP units respectively).

Loading MultiVOIP Software onto the PC

The software loading procedure does not present every screen or option in the loading process. It is assumed that someone with a thorough knowledge of Windows and the software loading process is performing the installation.

The MultiVOIP software and User Guide are contained on the MultiVOIP product CD. Because the CD is auto-detectable, it will start up automatically when you insert it into your CD-ROM drive. When you have finished loading your MultiVOIP software, you can view and print the User Guide by clicking on the **View Manuals** icon.

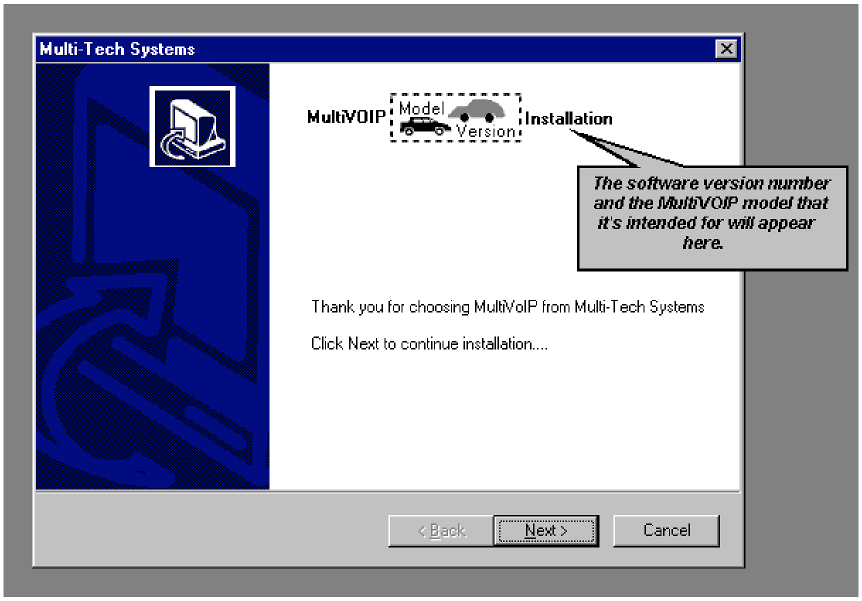
1. Be sure that your MultiVOIP has been properly cabled and that the power is turned on.

2. Insert the MultiVOIP CD into your CD-ROM drive. The CD should start automatically. It may take 10 to 20 seconds for the Multi-Tech CD installation window to display.

If the Multi-Tech Installation CD window does not display automatically, click **My Computer**, then right click the **CD ROM drive** icon, click **Open**, and then click the **Autorun** icon.

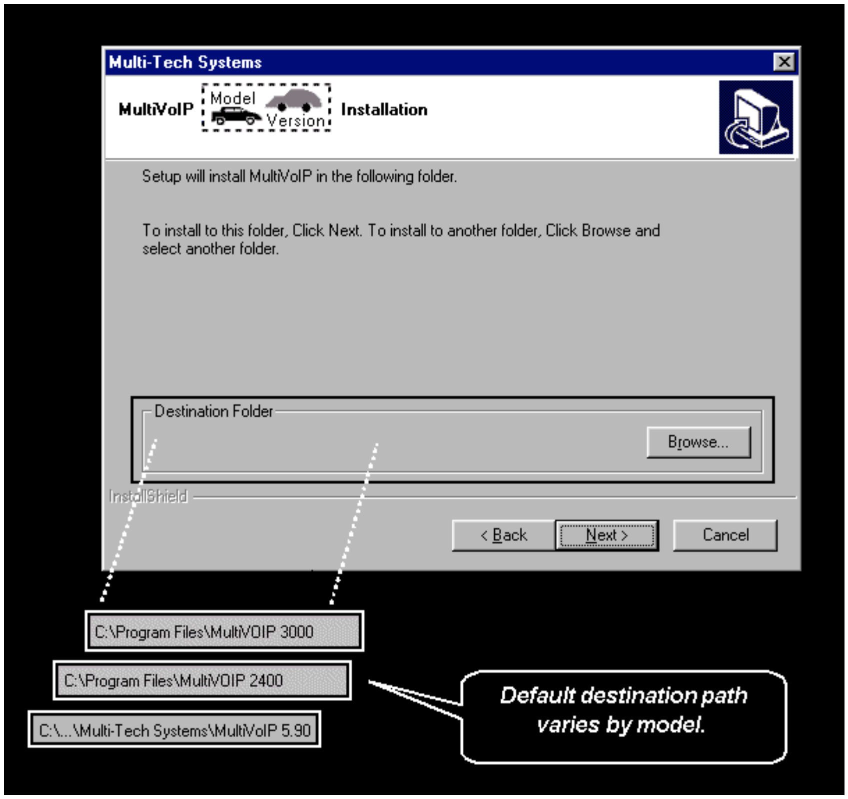
3. When the Multi-Tech Installation CD dialog box appears, click the **Install Software** icon.

4. A ‘welcome’ screen appears.



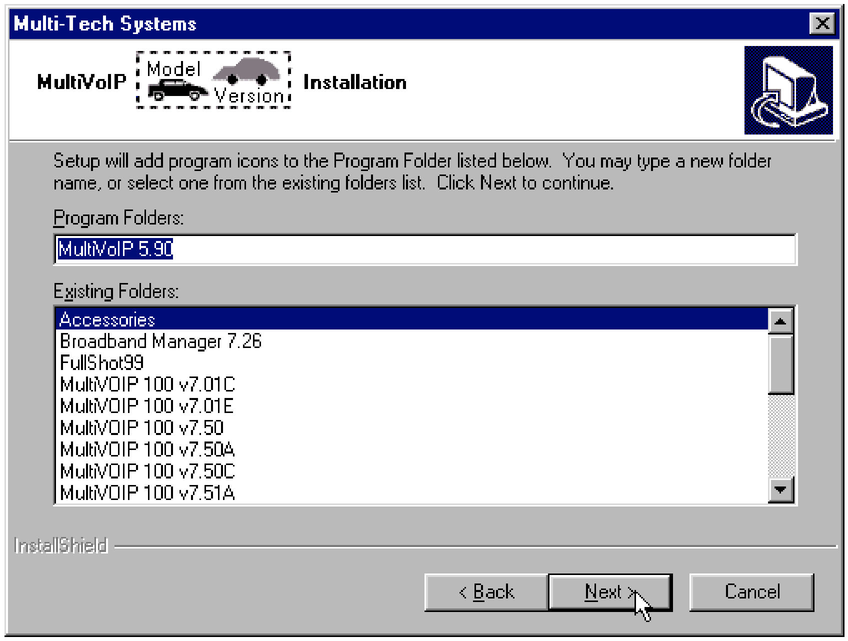
Press **Enter** or click **Next** to continue.

5. Follow the on-screen instructions to install your MultiVOIP software. The first screen asks you to choose the folder location of the files of the MultiVOIP software.



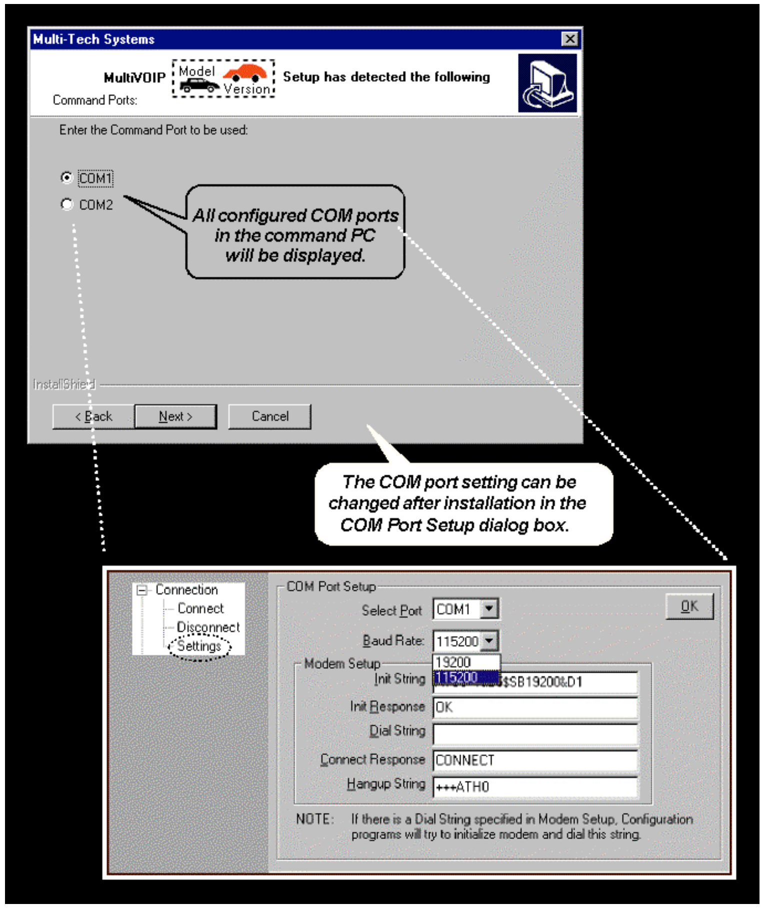
Choose a location and click **Next.**

6. At the next screen, you must select a program folder location for the MultiVOIP software program icon.



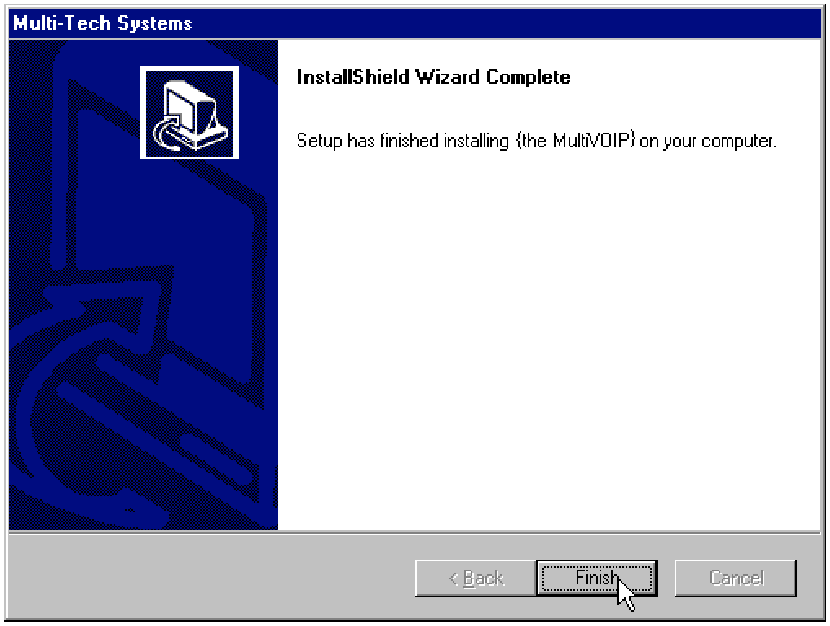
Click **Next**. Transient progress screens will appear while files are being copied.

7. On the next screen you can select the COM port that the command PC will use when communicating with the MultiVoip unit. After software installation, the COM port can be re-set in the MultiVOIP Software (from the sidebar menu, select **Connection | Settings** to access the **COM Port Setup** screen or use the keyboard shortcut Ctrl + G).



NOTE: If the COM port setting made here conflicts with the actual COM port resources available in the command PC, this error message will appear when the MultiVOIP program is launched. If this occurs, you must reset the COM port.	

8. A completion screen will appear.



Click **Finish**.

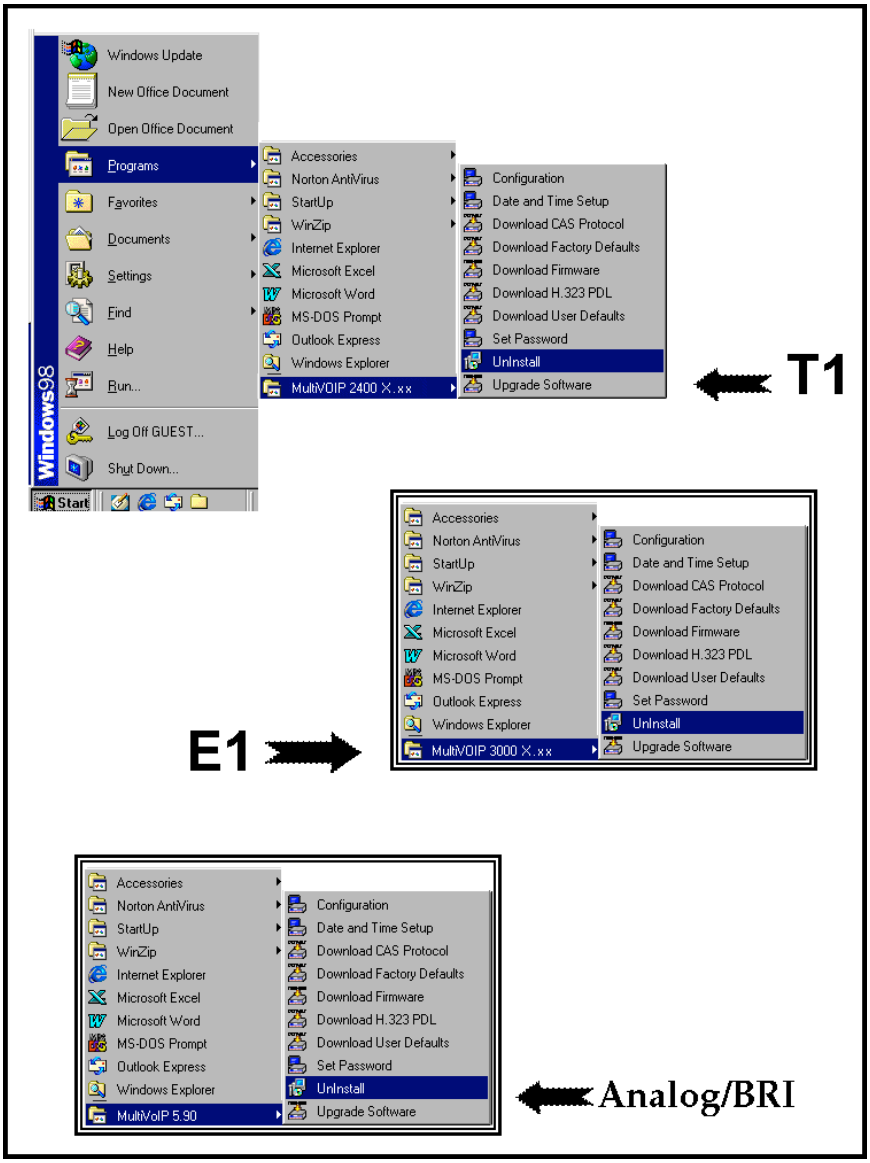
9. When setup of the MultiVOIP software is complete, you will be prompted to run the MultiVOIP software to configure the VOIP.

Software installation is complete at this point. You may proceed with Technical Configuration now or not, at your convenience.

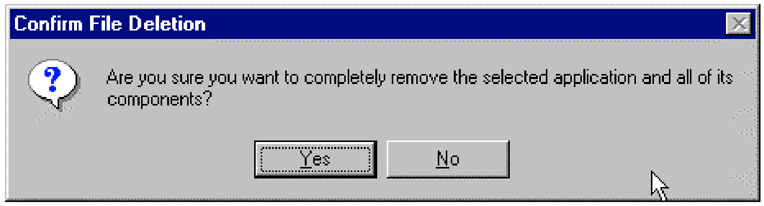
Technical Configuration instructions are in the next two chapters of this manual: Chapter 5 for T1/E1 MultiVOIP units and Chapter 6 for Analog MultiVOIP units.

Un-Installing the MultiVOIP Configuration Software

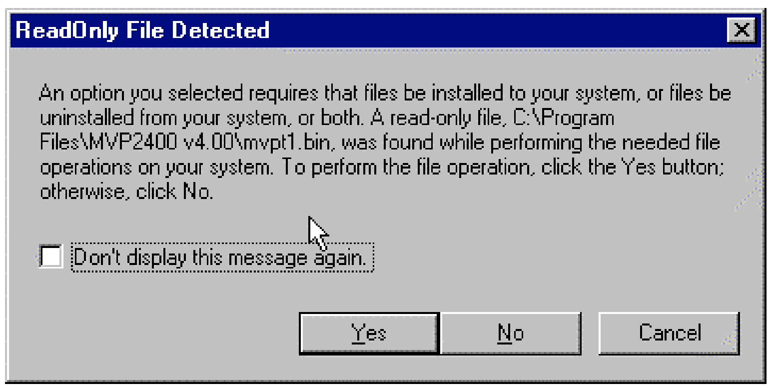
1. To un-install the MultiVOIP configuration software, go to **Start** | **Programs** and locate the entry for the MultiVOIP program. Select **Uninstall**.



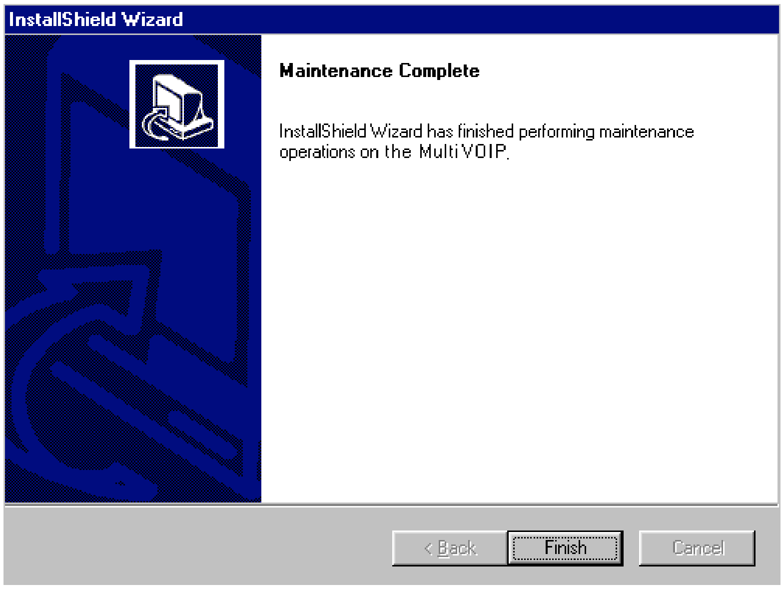
2. Two confirmation screens will appear. Click **Yes** and **OK** when you are certain you want to continue with the uninstallation process.



3. A special warning message similar to that shown below may appear concerning the MultiVOIP software's ".bin" file. Click **Yes**.



4. A completion screen will appear.



Click **Finish**.

Chapter 5: Technical Configuration for Digital T1/E1 MultiVOIPs (MVP2400, MVP2410, MVP3010)

Configuring the Digital T1/E1 MultiVOIP

There are two ways in which the MultiVOIP must be configured before operation: technical configuration and phonebook configuration.

Technical Configuration. First, the MultiVOIP must be configured to operate with technical parameter settings that will match the equipment with which it interfaces. There are seven types of technical parameters that must be set.

These technical parameters pertain to
(1) its operation in an IP network,
(2) its operation with T1/E1 telephony equipment,
(3) its transmission of voice and fax messages,
(4) its interaction with SNMP (Simple Network Management Protocol) network management software (MultiVoipManager),
(5) certain telephony attributes that are common to particular nations or regions,
(6) its operation with a mail server on the same IP network (per SMTP parameters) such that log reports about VoIP telephone call traffic can be sent to the administrator by email,
(7) implementing some common premium telephony features (Call Transfer, Call Hold, Call Waiting, Call ID – "Supplementary Services"), and
(8) selecting the method by which log reports will be made accessible.

The process of specifying values for the various parameters in these seven categories is what we call "technical configuration" and it is described in this chapter.

Phonebook Configuration. The second type of configuration that is required for the MultiVOIP pertains to the phone number dialing sequences that it will receive and transmit when handling calls. Both the PBX/telephony equipment and the other VOIP devices that the MultiVOIP unit interacts with will affect dialing patterns. We call this "Phonebook Configuration," and it is described in *Chapter 7: T1 Phonebook Configuration* and *Chapter 8: E1 Phonebook Configuration* of this manual. Chapter 2, the *Quick Start Instructions*, presents additional examples relevant to the T1/E1 voips.

Local/Remote Configuration. The MultiVOIP must be configured locally at first (to establish an IP address for the MultiVOIP unit). But changes to this initial configuration can be done either locally or remotely.

Local configuration is done through a connection between the "Command" port of the MultiVOIP and the COM port of the computer; the MultiVOIP configuration program is used.

Remote configuration is done through a connection between the MultiVOIP's Ethernet (network) port and a computer connected to the same network. The computer could be miles or continents away from the MultiVOIP itself. There are two ways of doing remote configuration and operation of the MultiVOIP unit: (1) using the MultiVoipManager SNMP program, or (2) using the MultiVOIP web browser interface program.

MultiVoipManager. MultiVoipManager is an SNMP agent program (Simple Network Management Protocol) that extends the capabilities of the MultiVOIP configuration program: MultiVoipManager allows the user to manage any number of VOIPs on a network, whereas the MultiVOIP configuration program can manage only the VOIP to which it is directly/locally connected. The MultiVoipManager can configure multiple VOIPs simultaneously, whereas the MultiVOIP configuration program can configure only one at a time.

MultiVoipManager may (but does not need to) reside on the same PC as the MultiVOIP configuration program. The MultiVoipManager program is on the MultiVOIP Product CD. Updates, when applicable, may be posted at on the MultiTech FTP site. To download, go to ftp://ftp.multitech.com/MultiVoip/.

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

Functional Equivalence of Interfaces. The MultiVOIP configuration program is required to do the initial configuration (that is, setting an IP address for the MultiVOIP unit) so that the VOIP unit can communicate with the MultiVoipManager program or with the web browser GUI. Management of the VOIP after that point can be done from any of these three programs since they all offer essentially the same functionality. Functionally, either the MultiVoipManager program or the web browser GUI can replace the MultiVOIP configuration program after the initial configuration is complete (with minor exceptions, as noted).

WARNING: Do not attempt to interface the MultiVOIP unit with two control programs simultaneously (that is, by accessing the MultiVOIP configuration program via the Command Port and either the MultiVoipManager program or the web browser interface via the Ethernet Port). The results of using two programs to control a single VOIP simultaneously would be unpredictable.

Local Configuration

This manual primarily describes local configuration with the Windows GUI. After IP addresses have been set locally using the Windows GUI, however, most aspects of configuration (logging functions are an exception) can be handled through the web browser GUI, as well (see the *Operation and Maintenance* chapter of this manual). In most aspects of configuration, the Windows GUI and web-browser GUI differ only graphically, not functionally. For information on SNMP remote configuration and management, see the MultiVoipManager documentation.

Pre-Requisites

To complete the configuration of the MultiVOIP unit, you ***must*** know several things about the overall system.

Before configuring your MultiVOIP Gateway unit, you must know the values for several IP and T1/E1 parameters that describe the IP network system and telephony system (PBX or telco central office equipment) with which the digital MultiVOIP will interact. If you plan to receive log reports on phone traffic by email (SMTP), you must arrange to have an email address assigned to the VOIP unit on the email server on your IP network.

IP Parameters

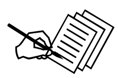
The following parameters must be known about the network (LAN, WAN, Internet, etc.) to which the MultiVOIP will connect:

➨	*Ask your computer network administrator.*	*Info needed to operate:* **all MultiVOIP models.**
	IP Network Parameters: Record for each VOIP Site in System	
	• IP Address	
	• IP Mask	
	• Gateway	
	• Domain Name Server (DNS) Info *(not implemented; for future use)*	

Write down the values for these IP parameters. You will need to enter these values in the "IP Parameters" screen in the Configuration section of the MultiVOIP software. You must have this IP information about *every* VOIP in the system.

T1 Telephony Parameters (for MVP2400 & MVP2410)

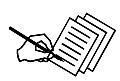
The following parameters must be known about the PBX or telco central office equipment to which the T1 MultiVOIP will connect:

➨	**T1 Phone Parameters** *Ask phone company or PBX maintainer.*	*Info needed to operate:* **MVP2400** **MVP2410**
	 T1 Telephony Parameters: Record for this VOIP Site	
	• Which frame format is used? ESF___ or D4___	
	• Which CAS or PRI protocol is used? _______________	
	• Clocking: Does the PBX or telco switch use internal or external clocking? __________________ Note that the setting used in the voip unit will be the opposite of the setting used by the telco/PBX.	
	• Which line coding is used? AMI___ or B8ZS___	

Write down the values for these T1 parameters. You will need to enter these values in the "T1/E1 Parameters" screen in the Configuration section of the MultiVOIP software.

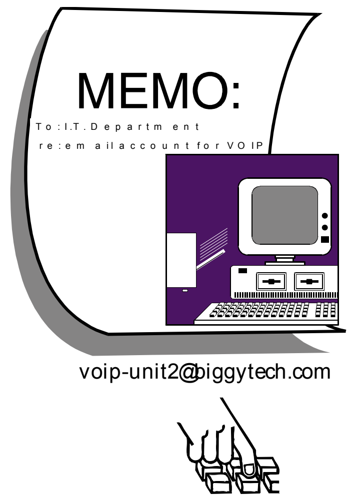
E1 Telephony Parameters (for MVP3010)

The following parameters must be known about the PBX or telco central office equipment to which the E1 MultiVOIP will connect:

➨	**E1 Phone Parameters** *Ask phone company or PBX maintainer.*	*Info needed to operate:* **MVP3010**
	 E1 Telephony Parameters: Record for this VOIP Site	
	• Which frame format is used? Double Frame_____ MultiFrame w/ CRC4_____ MultiFrame w/ CRC4 modified_____	
	• Which CAS or PRI protocol is used? _______________	
	• Clocking: Does the PBX or telco switch use internal or external clocking? _________________ Note that the setting used in the voip unit will be the opposite of the setting used by the telco/PBX.	
	• Which line coding is used? AMI___ or HDB3___	
	• Pulse shape level?: (most commonly 0 to 40 meters)	

Write down the values for these E1 parameters. You will need to enter these values in the “T1/E1 Parameters” screen in the Configuration section of the MultiVOIP software.

SMTP Parameters (for email call log reporting)

required if log reports of VOIP call traffic are to be sent by email	**Optional**
SMTP Parameters Preparation Task: 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.	

Local Configuration Procedure (Summary)

After the MultiVOIP configuration software has been installed in the 'Command' PC (which is connected to the MultiVOIP unit), several steps must be taken to configure the MultiVOIP to function in its specific setting. Although the summary below includes all of these steps, some are optional.

1. Check Power and Cabling.

2. Start MultiVOIP Configuration Program.

3. Confirm Connection.

4. Solve Common Connection Problems.
   A. Fixing a COM Port Problem.
   B. Fixing a Cabling Problem.

5. Familiarize yourself with configuration parameter screens and how to access them.

6. Set IP Parameters.

7. Enable web browser GUI (optional).

8. Set Voice/Fax Parameters.

9. Set T1/E1 Parameters.

10. Set ISDN Parameters (if applicable).

11. Set SNMP Parameters (applicable if MultiVoipManager remote management software is used).

12. Set Regional Parameters (Phone Signaling Tones and Cadences).

13. Set Custom Tones and Cadences (optional).

14. Set SMTP Parameters (applicable if Log Reports are via Email).

15. Set Log Reporting Method (GUI, locally in MultiVOIP Configuration program; SNMP, remotely in MultiVoipManager program; or SMTP, via email).

16. Set Supplementary Services Parameters. The Supplementary Services screen allows voip deployment of features that are normally found in PBX or PSTN systems (e.g., call transfer and call waiting).

17. Set Baud Rate (of COM port connection to 'Command' PC).

18. View System Information and set updating interval (optional).

19. Save the MultiVOIP configuration.

20. Create a User Default Configuration (optional).

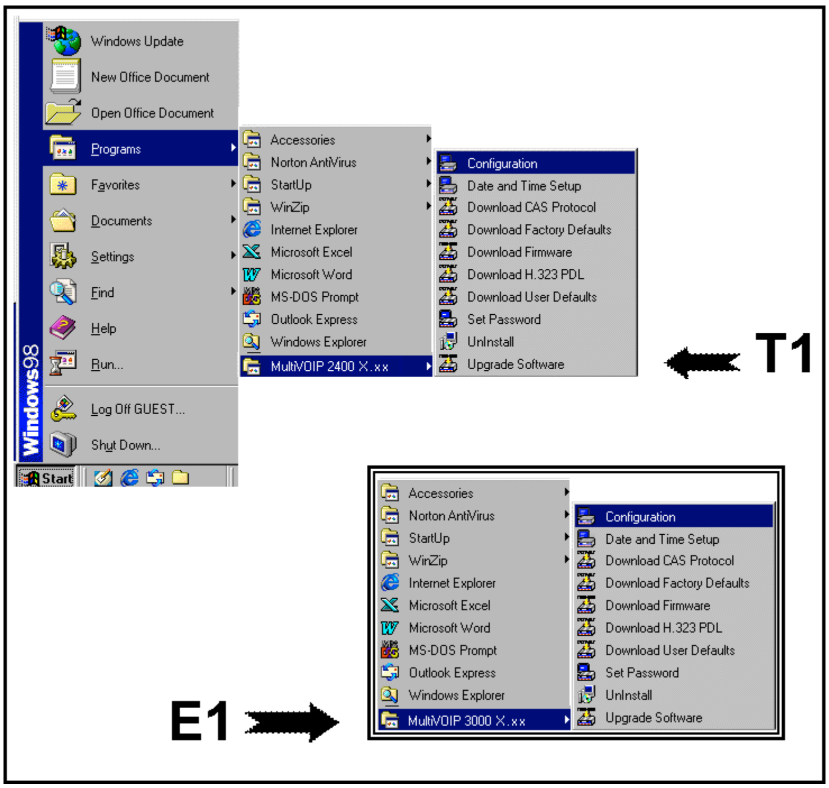
Local Configuration Procedure (Detailed)

You can begin the configuration process as a continuation of the MultiVOIP software installation. You can establish your configuration or modify it at any time by launching the MultiVOIP program from the Windows **Start** menu.

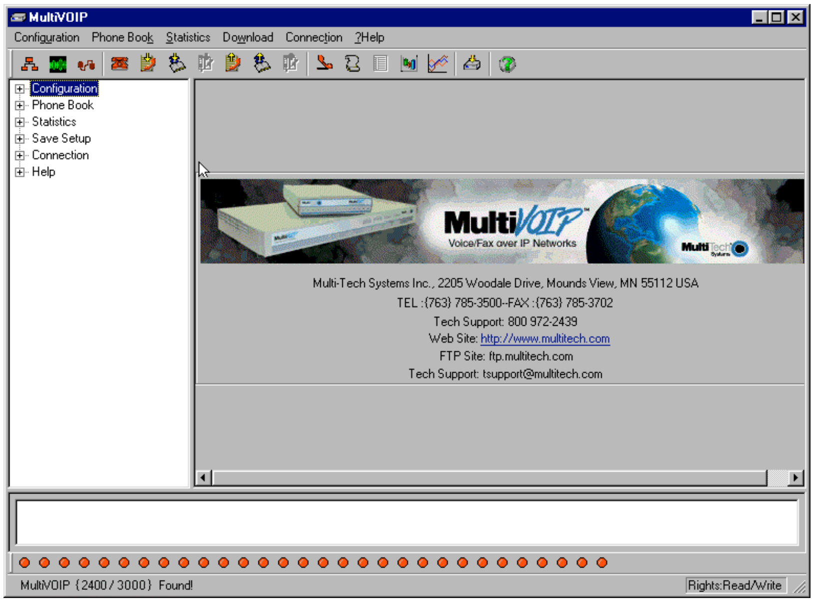
1. **Check Power and Cabling**. Be sure the MultiVOIP is turned on and connected to the computer via the MultiVOIP's Command Port (DB9 connector at computer's COM port; RJ45 connector at MultiVOIP).

   You must allow the MultiVOIP to finish booting before you launch the MultiVOIP Configuration Program. The RED boot LED turns itself off when the booting process is completed.

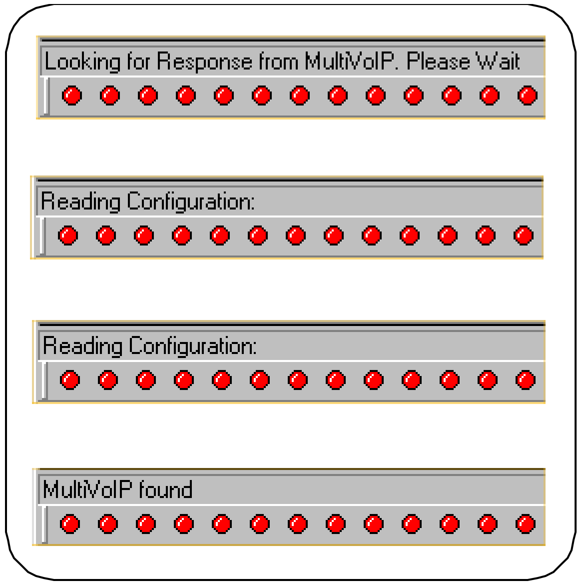
2. **Start MultiVOIP Configuration Program**. Launch the MultiVOIP program from the Windows **Start** menu (from the folder location determined during installation).



3. **Confirm Connection**. If the MultiVOIP is set for an available COM port and is correctly cabled to the PC, the MultiVOIP main screen will appear. (If the main screen appears *grayed out* and seems inaccessible, go to step 4.)



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. Skip to step 5.

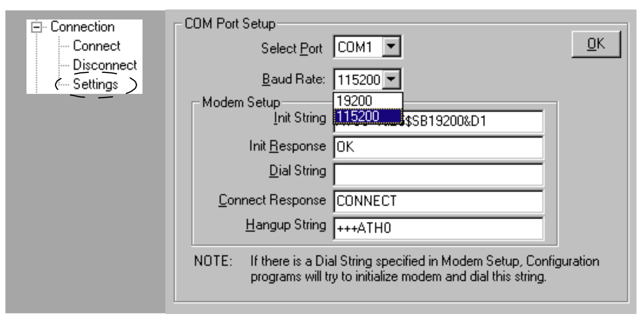


4. **Solving Common Connection Problems**.

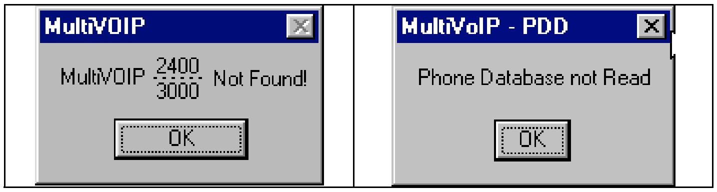
A. Fixing a COM Port Problem. If the MultiVOIP main screen appears but is grayed out and seems inaccessible, the COM port that was specified for its communication with the PC is unavailable and must be changed. An error message will appear.

To change the COM port setting, use the **COM Port Setup** dialog box, which is accessible via the keyboard shortcut **Ctrl + G** or by going to the **Connection** pull-down menu and choosing "Settings." 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.)

Ctrl + G



4B. Fixing a Cabling Problem. If the MultiVOIP cannot be located by the computer, two error messages will appear (saying "Multi-VOIP Not Found" and "Phone Database Not Read").

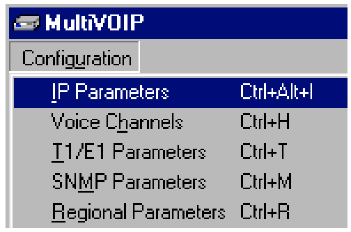
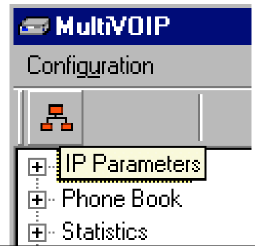
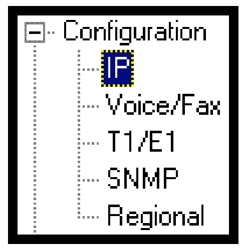


In this case, the MultiVOIP is simply disconnected from the network. For instructions on MultiVOIP cable connections, see the "Cabling" section of Chapter 3.

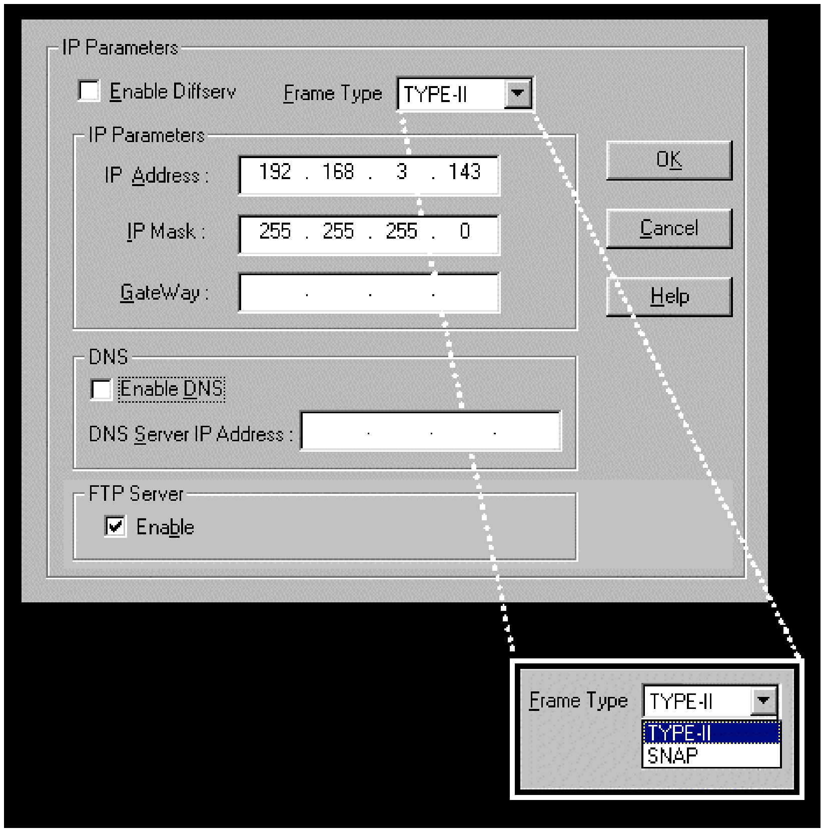
5. **Configuration Parameter Groups: Getting Familiar, Learning About Access**. The first part of configuration concerns IP parameters, Voice/FAX parameters, T1/E1 parameters, SNMP parameters, Regional parameters, SMTP parameters, Supplementary Services parameters, Logs, and System Information. In the MultiVOIP software, these seven types of parameters are grouped together under "Configuration" and each has its own dialog box for entering values.

Generally, you can reach the dialog box for these parameter groups in one of four ways: pulldown menu, toolbar icon, keyboard shortcut, or sidebar..

6. **Set IP Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing "IP Parameters"</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td>
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + Alt + I</code></td><td>
</td></tr>
</table>

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



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

IP Parameter Definitions		
Field Name	**Values**	**Description**
Enable Diffserv	Y/N	Diffserv is used for QoS (quality of service). When enabled, the TOS (Type of Service) bits in the IP header are configured so that routers supporting Diffserv can give priority to the VOIP's IP packets. Disabled by default.
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.
IP Address	4-places, 0-255	The unique LAN IP address assigned to the MultiVOIP.
IP Mask	4-places, 0-255	Subnetwork address that allows for sharing of IP addresses within a LAN.
Gateway	4-places, 0-255	The IP address of the device that connects your MultiVOIP to the Internet.
Enable DNS	Y/N *(feature not yet implemented; for future use)*	Enables Domain Name Space/System function where computer names are resolved using a worldwide distributed database.
DNS Server IP Address	4-places, 0-255. *(feature not yet implemented; for future use)*	IP address of specific DNS server to be used to resolve Internet computer names.
FTP Server Enable	Y/N See "FTP Server File Transfers" in *Operation & Maintenance* chapter.	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.

7. **Enable Web Browser GUI (Optional)**. After an IP address for the MultiVOIP unit has been established, you can choose to do any further configuration of the unit (a) by using the MultiVOIP web browser GUI, or (b) by continuing to use the MultiVOIP Windows GUI. If you want to do configuration work using the web browser GUI, you must first enable it. To do so, follow the steps below.

A. Set IP address of MultiVOIP unit using the MultiVOIP Configuration program (the Windows GUI).

B. Save Setup in Windows GUI.

C. Close Windows GUI.

D. Install Java program from MultiVOIP product CD (required on first use only).

E. Open web browser.

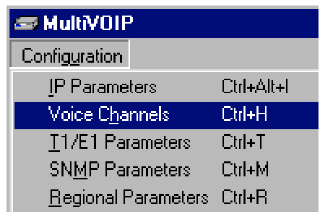
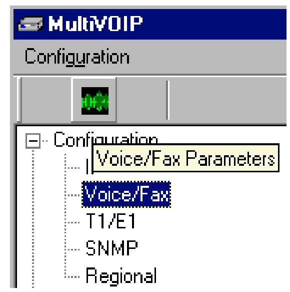
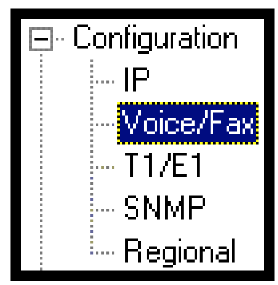
F. Browse to IP address of MultiVOIP unit.

G. If username and password have been established, enter them when when prompted.

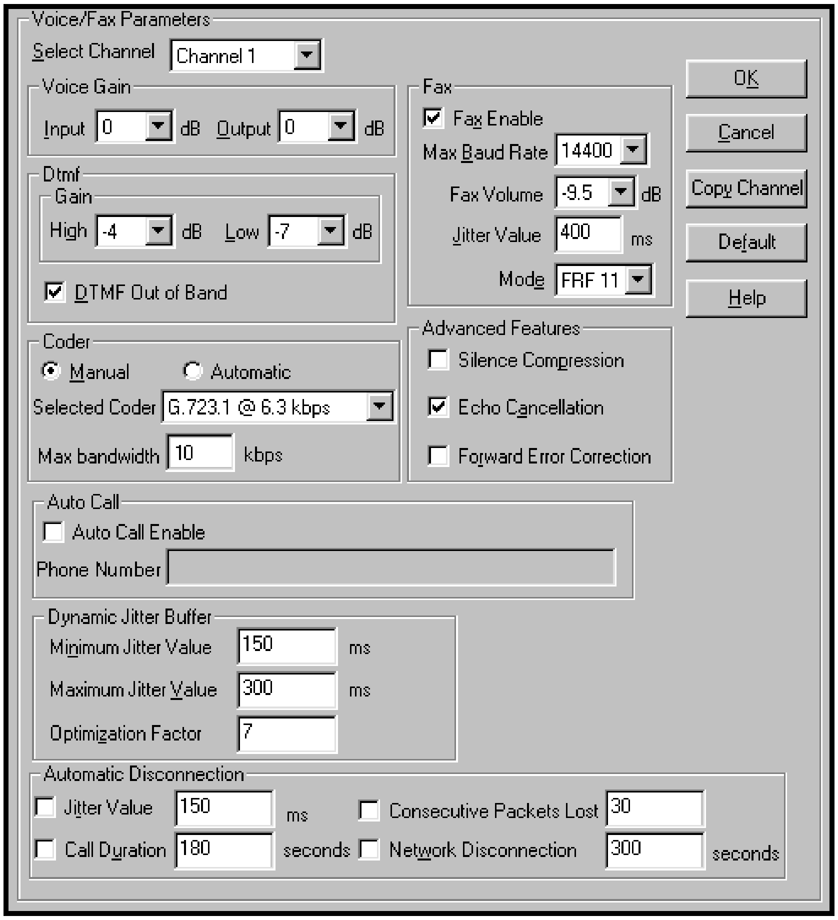
H. Use web browser GUI to configure or operate MultiVOIP unit. The configuration screens in the web browser GUI will have the same content as their counterparts in the Windows GUI; only the graphic presentation will be different.

For more details on enabling the MultiVOIP web GUI, see the "Web Browser Interface" section of the *Operation & Maintenance* chapter of this manual.

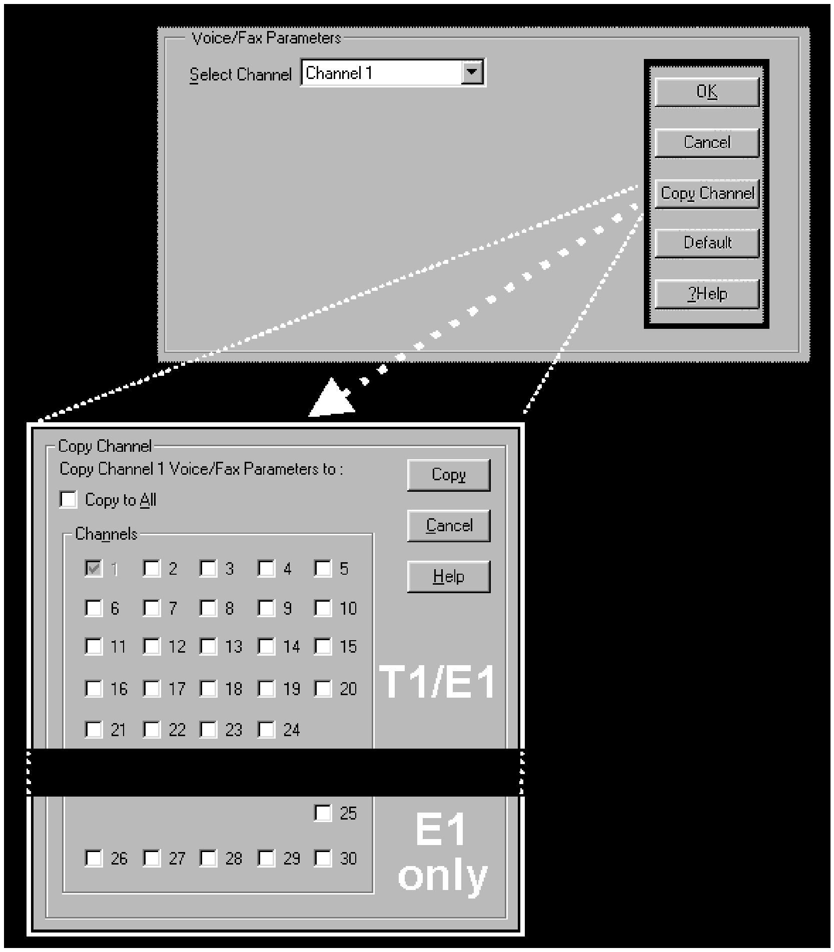
8. **Set Voice/FAX Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

Accessing “Voice/FAX Parameters”	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + H`	

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



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



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

Voice/Fax Parameter Definitions		
Field Name	**Values**	**Description**
Default	--	When this button is clicked, all Voice/FAX parameters are set to their default values.
Select Channel	1-24 (T1) 1-30 (E1)	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 Parameters		
DTMF Gain	--	The **DTMF Gain** (Dual Tone Multi-Frequency) controls the volume level of the digital tones sent out for Touch-Tone dialing.
DTMF Gain, High Tones	+3dB to -31dB & “mute”	Default value: **-4 dB**. Not to be changed except under supervision of MultiTech’s Technical Support.
DTMF Gain, Low Tones	+3dB to -31dB & “mute”	Default value: **-7 dB**. Not to be changed except under supervision of MultiTech’s Technical Support.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
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 (checked), the MultiVOIP detects DTMF tones at its input and regenerates them at its output. When **DTMF Inband** is selected, the DTMF digits are passed through the MultiVOIP unit as they are received.
FAX Parameters		
Fax Enable	Y/N	Enables or disables fax capability for a particular channel.
Max Baud Rate (Fax, bps)	2400, 4800, 7200, 9600, 12000, 14400	Set to match baud rate of fax machine connected to channel (see Fax machine's user manual). Default = 14400 bps.
Fax Volume Default = -9.5 dB	-18.5 dB to –3.5 dB	Controls output level of fax tones. To be changed only under the direction of Multi-Tech's Technical Support.
Jitter Value (Fax)	Default = 400 ms	Defines the inter-arrival packet deviation (in milliseconds) for the fax transmission. A higher value will increase the delay, allowing a higher percentage of packets to be reassembled. A lower value will decrease the delay allowing fewer packets to be reassembled.
Mode (Fax)	FRF 11; T.38 (T.38 not currently sup-ported)	**FRF11** is frame-relay FAX standard using these coders: G.711, G.728, G.729, and G.723.1. **T.38** is an ITU-T standard for storing and forwarding Faxes via email using X.25 packets. It uses T.30 fax standards and includes special provisions to preclude FAX timeouts during IP transmissions.

Voice/Fax Parameter Definitions (cont'd)		
Coder Parameters		
Coder	Manual or Auto-matic	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 are negotiated.
Selected Coder	**G.711** a/u law 64 kbps; **G.726**, @ 16/24/32/40 kbps; **G.727**, @ nine bps rates; **G.723.1** @ 5.3 kbps, 6.3 kbps; **G.729**, 8kbps; **Net Coder** @ 6.4, 7.2, 8, 8.8, 9.6 kbps	Select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling must have the same voice coder selected. Default = G.723.1 @ 6.3 kbps, as required for H.323. Here 64K of digital voice are compressed to 6.3K, allowing several simultaneous conversations over the same bandwidth that would otherwise carry only one. To make selections from the Selected Coder drop-down list, the Manual option must be enabled.
Max bandwidth (coder)	11 – 128 kbps	This drop-down list enables you to select the maximum bandwidth allowed for this channel. The Max Bandwidth drop-down list is enabled only if the Coder is set to Automatic. If coder selected automatically, then enter a value for maximum bandwidth, as directed by VOIP administrator.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Advanced Features		
Silence Compression	Y/N	Determines whether silence compression is enabled (checked) for this voice channel. With **Silence Compression** enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. Default = off.
Echo Cancellation	Y/N	Determines whether echo cancellation is enabled (checked) for this voice channel. **Echo Cancellation** removes echo and improves sound quality. Default = on.
Forward Error Correction	Y/N	Determines whether forward error correction is enabled (checked) for this voice channel. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off
Auto Call Enable	Y/N	The **Auto Call** 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.
Phone No. (Auto Call)	--	Phone number used for Auto Call function. A corresponding phone number must be listed in the Outbound Phonebook.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dynamic Jitter		
Dynamic Jitter Buffer		**Dynamic Jitter** defines a minimum and a maximum jitter value for voice communications. When receiving voice packets from a remote MultiVOIP, varying delays between packets may occur due to network traffic problems. This is called Jitter. To compensate, the MultiVOIP uses a Dynamic Jitter Buffer. The Jitter Buffer enables the MultiVOIP to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An Optimization Factor adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MultiVOIP gateways. The default minimum dynamic jitter buffer of **60** milliseconds is the minimum delay that would be acceptable over a low jitter network. The default maximum dynamic jitter buffer of **300** milliseconds is the maximum delay tolerable over a high jitter network.
Minimum Jitter Value	60 to 400 ms	The default minimum dynamic jitter buffer of **60** milliseconds is the minimum delay that would be acceptable over a low jitter network. Default = 60 msec

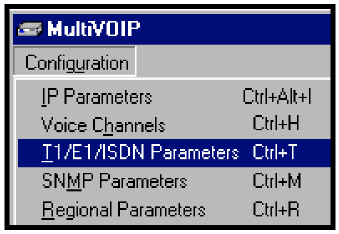
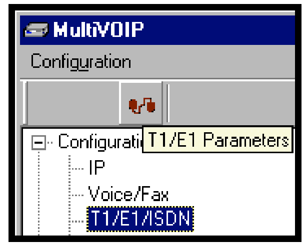
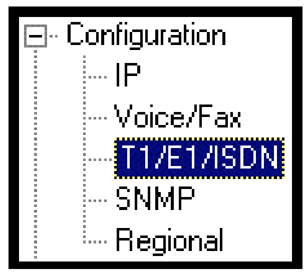
Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dynamic Jitter		
Maximum Jitter Value	60 to 400 ms	The default maximum dynamic jitter buffer of **300** milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 msec
Optimizat-ion Factor	0 to 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**.

Modem Relay

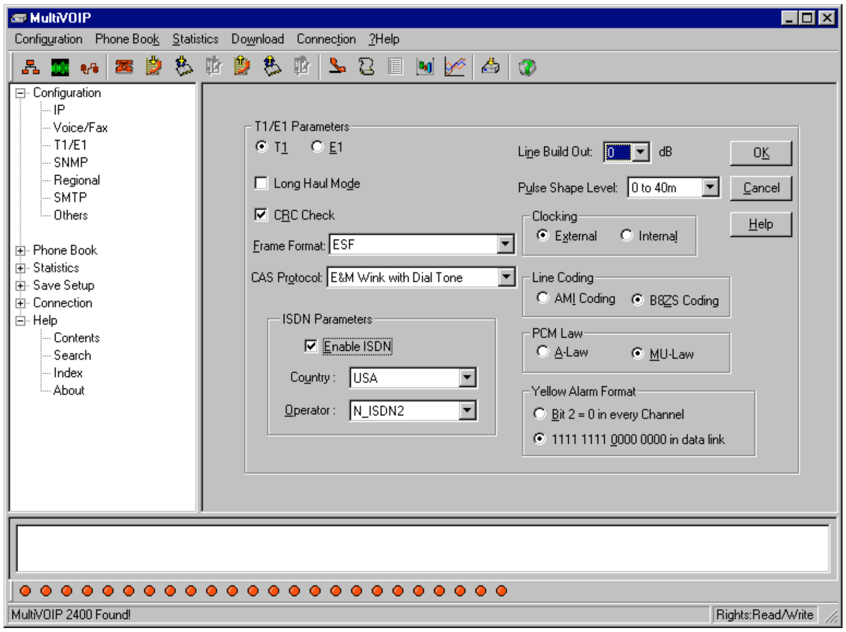
To place modem traffic onto the voip network (an application called "modem relay"), use Coder G.711 mu-law at 64kbps.

Voice/Fax Parameter Definitions (cont'd))		
Field Name	**Values**	**Description**
Auto Disconnect		
Automatic Disconnect-ion	--	The **Automatic Disconnection** group provides four options which can be used singly or in any combination.
Jitter Value	1-65535 milli-seconds	The **Jitter Value** defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. The default is **150** 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 = 150 ms. However, value must equal or exceed Dynamic Minimum Jitter Value.
Call Duration	1-65535 seconds	**Call Duration** defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. Inactive by default. When active, default = **180** sec. This may be too short for most configurations requiring upward adjustment.
Consecutive Packets Lost	1-65535	**Consecutive Packets Lost** defines the number of consecutive packets that are lost after which the call is automatically disconnected. Inactive by default. When active, default = **30**
Network Discon-nection	1 to 65535 seconds; Default = 300 sec.	Specifies how long to wait before disconnecting the call when IP network connectivity with the remote site has been lost.

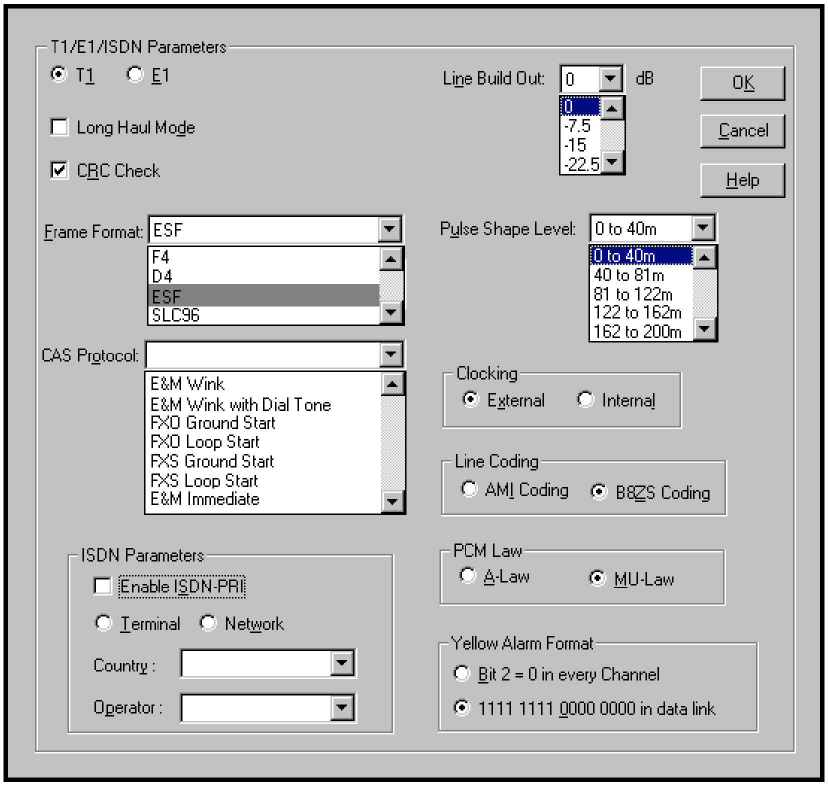
9. **Set T1/E1/ISDN Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “T1/E1/ISDN Parameters”</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td>
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + T</code></td><td>
</td></tr>
</table>

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



T1 Parameters. The parameters applicable to T1 and their values are shown in the figure below. These **T1 Parameter** fields are described in the tables that follow.



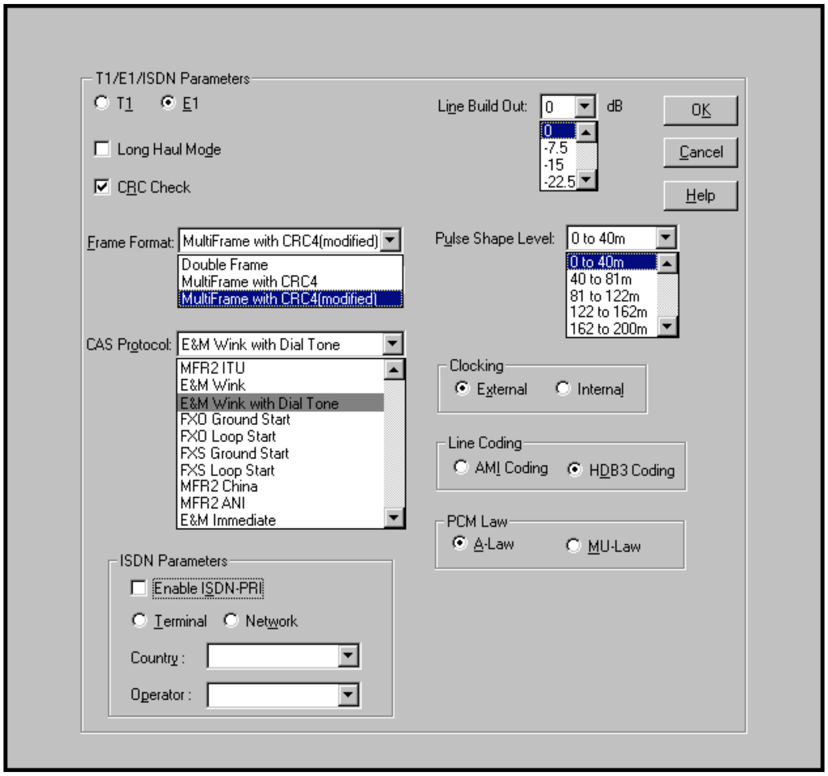
T1 Parameter Definitions		
Field Name	**Values**	**Description**
T1/E1/ISDN	T1	North American standard.
Long-Haul Mode	Y/N	In Long-Haul Mode, the MultiVOIP automatically recovers received signals as low as –36 dB. The maximum reachable length with 22 AWG cable is 2000 meters. When Long-Haul Mode is disabled, signals as low as –10 dB can be received. Default: disabled.
CRC Check (Cyclic Redundancy Check)	Y/N	When enabled, allows generation and checking of CRC bits. If not enabled, all check bits in the transmit direction are set. Only applies to ESF frame format. Default: enabled.
Frame Format	F4, D4, ESF, SLC96	Frame Format of MultiVOIP should match that used by PBX or telco. ESF and D4 are commonly used.

T1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
CAS Protocol	E&M Immed Strt E&M Wink Start E&M Wink with dial tone FXO Ground Strt FXO Loop Start FXS Ground Strt FXS Loop Start	Channel Associated Signaling (CAS) is a method of incorporating telephony signaling info into a T1 voice/data stream. In CAS, the signaling bits (the A, B, C, and D bits) are multiplexed into the signal stream of each T1 channel. (By contrast, in Common Channel Signaling (CCS), one channel handles signaling for all other channels.) Each CAS protocol defines the states of the signaling bits during the various stages of a call (IDLE, SEIZED, ANSWER, RING-ON, RING-OFF). The CAS protocol code allows the VOIP to interact properly with the PBX or central-office switch that it serves. The need to download CAS protocols arises for only a small minority of VOIP users, and only when PBX/switch is found to be incompatible with standard protocols. Match this parameter to the setting of PBX or central-office switch.

T1 Parameter Definitions (cont'd)		
ISDN Parameters		
Field Name	**Values**	**Description**
Enable ISDN-PRI	Y/N	If digital connection is ISDN-PRI type, this box should be checked. When ISDN is enabled, the "CAS Protocols" field is grayed out (ISDN has its own signaling method).
Terminal/ Network	either "Terminal" or "Network"	When "Terminal" is selected, it indicates that the MultiVOIP should emulate the subscriber (terminal) side of the digital connection. When "Network" is selected, it indicates that the MultiVOIP should emulate the central office (network) side of the digital connection. Setting used for MultiVOIP must be opposite to the setting used in the PBX. For example, if the PBX is set to "Terminal," then the MultiVOIP must be set to "Network."
Country	see table, later this chapter	Country in which MultiVOIP is operating with ISDN.
Operator	see table, later this chapter	Indicates phone switch manufacturer/model or refers to telco so as to specify the switching system in question. ISDN is implemented somewhat differently in different switches.
Note on Country & Operator options.	—	[ISDN implementation options are shown, arranged by country, in a table below – soon after E1 Parameter Definitions.]

T1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Line Build Out	0 dB, -7.5 dB, -15 dB, -22.5 dB	To reduce the crosstalk on received signals, a transmit attenuator can be placed in the data path. Transmit attenuation is selectable. **Default: O dB**
Pulse Shape Level	0 to 40 Meters 40 to 81 m 81 to 122 m 122 to 162 m 162 to 200 m	Refers to length of cable between MultiVOIP and PBX/telco in meters. Most common will be 0 to 40m.
Clocking	External/Internal	Set opposite to telco/PBX setting. Example: if telco clocking internal, set VOIP clocking as external.
Line Coding	AMI / B8ZS	Match to PBX or telco.
PCM Law	A-Law/Mu-Law	Match to PBX or telco. " Mu-law" is analog-to-digital compression/expansion standard used in North America. "A-law" is European standard.
Yellow Alarm Format	Bit 2 / 1111…	Depending on the Frame Format used, there are choices of Yellow Alarm format, as follows: **D4**: **-Bit2** = 0 in every speech channel **-FS** bit of frame 12 is forced to one. **ESF**: **-Bit2** = 0 in every speech channel **–1111111100000000** pattern in data link channel. Check with your PBX/telco administrator for the correct setting or use the default value (1111 …).

E1 Parameters. The parameters applicable to E1 and their values are shown in the figure below. These **E1 Parameter** fields are described in the tables that follow.



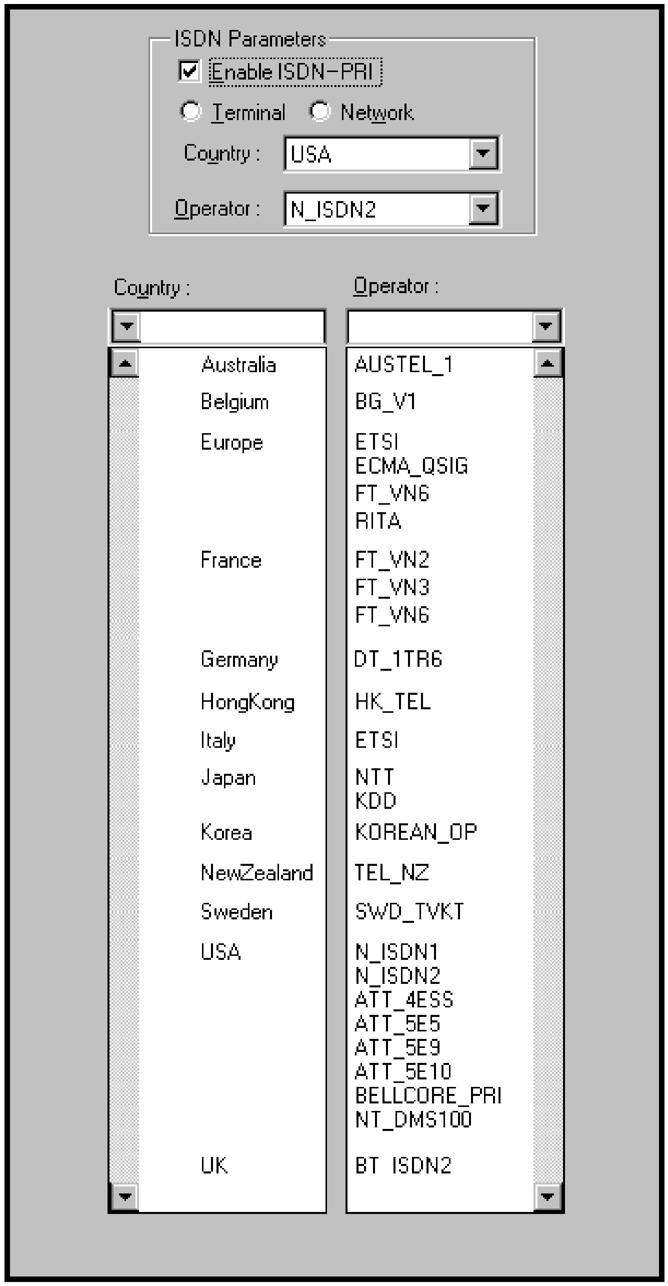
E1 Parameter Definitions		
Field Name	**Values**	**Description**
T1/E1/ISDN	E1	European standard.
Long-Haul Mode	Y/N	In Long-Haul Mode, the MultiVOIP automatically recovers received signals as low as –36 dB. The maximum reachable length with 22 AWG cable is 2000 meters. When Long-Haul Mode is disabled, signals as low as –10 dB can be received. Default: disabled.
CRC Check (Cyclic Redundancy Check)	--	Not applicable to E1.
Frame Format	Double Frame; MultiFrame (with CRC4); MultiFrame (w/CRC4, modified)	Frame Format of MultiVOIP should match that used by PBX or telco.

E1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
CAS Protocol	E&M Immed Strt E&M Wink Start E&M Wink with dial tone FXO Ground Strt FXO Loop Start FXS Ground Strt FXS Loop Start MFR2ITU MFR2 China MFR2 ANI	Channel Associated Signaling (CAS) is a method of incorporating telephony signaling info into an E1 voice/data stream. In CAS, the signaling bits (the A, B, C, and D bits) are multiplexed into the signal stream of each E1 channel. (By contrast, in Common Channel Signaling (CCS), one channel handles signaling for all other channels.) Each CAS protocol defines the states of the signaling bits during the various stages of a call (IDLE, SEIZED, ANSWER, RING-ON, RING-OFF). The CAS protocol code allows the VOIP to interact properly with the PBX or central-office switch that it serves. The need to download CAS protocols arises for only a small minority of VOIP users, and only when PBX/switch is found to be incompatible with standard protocols. Match this parameter to the setting of PBX or central-office switch.

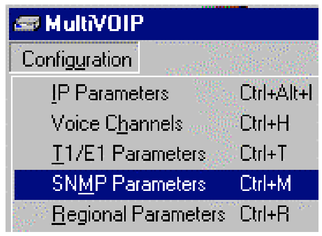
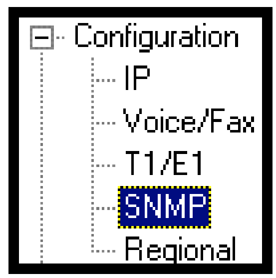
E1 Parameter Definitions (cont'd)		
ISDN Parameters		
Field Name	**Values**	**Description**
Enable ISDN-PRI	Y/N	If digital connection is ISDN-PRI type, this box should be checked. When ISDN is enabled, the "CAS Protocols" field is grayed out (ISDN has its own signaling method).
Terminal/ Network	either "Terminal" or "Network"	When "Terminal" is selected, it indicates that the MultiVOIP should emulate the subscriber (terminal) side of the digital connection. When "Network" is selected, it indicates that the MultiVOIP should emulate the central office (network) side of the digital connection. Setting used for MultiVOIP must be opposite to the setting used in the PBX. For example, if the PBX is set to "Terminal," then the MultiVOIP must be set to "Network."
Country	see table, later this chapter	Country in which MultiVOIP is operating with ISDN.
Operator	see table, later this chapter	Indicates phone switch manufacturer/model or refers to telco so as to specify the switching system in question. ISDN is implemented somewhat differently in different switches.
Note on Country & Operator options.	—	[ISDN implementation options are shown, arranged by country, in a table below – soon after E1 Parameter Definitions.]

E1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Line Build Out	0 dB, -7.5 dB, -15 dB, -22.5 dB	To reduce the crosstalk on received signals, a transmit attenuator can be placed in the data path. Transmit attenuation is selectable. **Default: O dB**
Pulse Shape Level	0 to 40 Meters 40 to 81 m 81 to 122 m 122 to 162 m 162 to 200 m	Refers to length of cable between MultiVOIP and PBX/telco in meters. Most common will be 0 to 40m.
Clocking	External/Internal	Set opposite to telco/PBX setting. Example: if telco clocking internal, set VOIP clocking as external.
Line Coding	AMI / HDB3	Match to PBX or telco.
PCM Law	A-Law/Mu-Law	Match to PBX or telco. "A-law" is analog-to-digital compression/expansion standard used in Europe. "Mu-law" is North American standard.

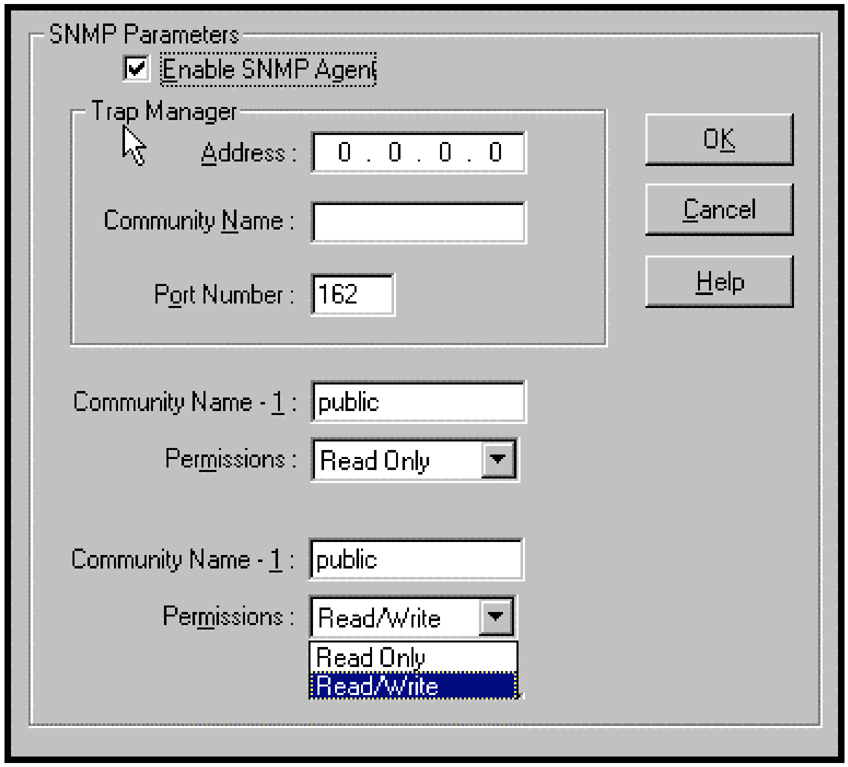
10. **Set ISDN Parameters** (if applicable). These parameters are acces-sible in the **T1/E1/ISDN Parameters** screen. If your T1 or E1 phone line is a Primary Rate Interface ISDN line, enable ISDN-PRI and set it for the particular implementation of ISDN that your telco uses. The ISDN types supported by the digital MultiVOIP units (at press time) are listed below, organized by country.



11. **Set SNMP Parameters** (Remote Voip Management). This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar. To make the MultiVOIP controllable by a remote PC running the MultiVoipManager software, check the “Enable SNMP Agent” box on the **SNMP Parameters** screen.

<table>
<tr><th colspan="2">Accessing “SNMP Parameters”</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + M</code></td><td>
</td></tr>
</table>

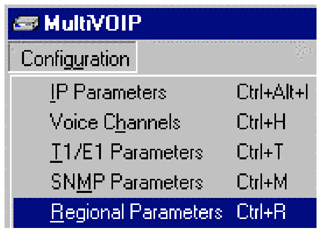
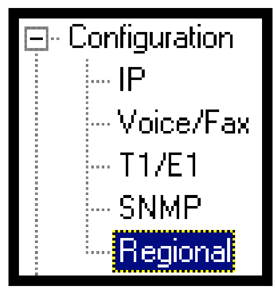
In each field, enter the values that fit your particular system.



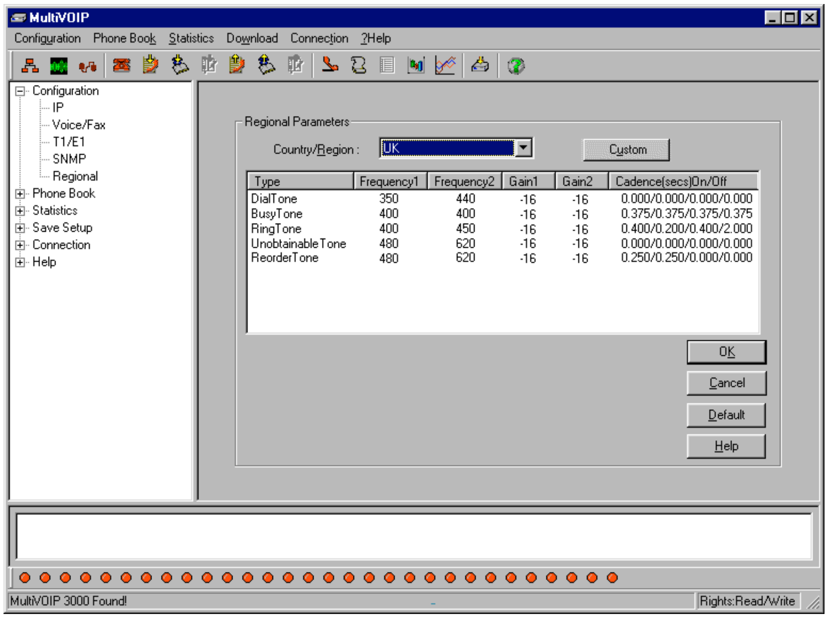
The SNMP Parameter fields are described in the table below.

SNMP Parameter Definitions		
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 MultiVoipManager software. Default: disabled
Trap Manager Parameters		
Address	4 places; n.n.n.n n = 0-255	IP address of MultiVoipManager PC.
Community Name	--	A "community" is a group of VOIP endpoints that can communicate with each other. Often "public" is used to designate a grouping where all end users have access to entire VOIP network. However, 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.

12. **Set Regional Parameters** (Phone Signaling Tones & Cadences)**.**
This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

Accessing "Regional Parameters"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + R`	

The **Regional Parameters** screen will appear. For the country selected, the standard set of frequency pairs will be listed for dial tone, busy tone, 'unobtainable' tone (fast busy or trunk busy), and ring tone.



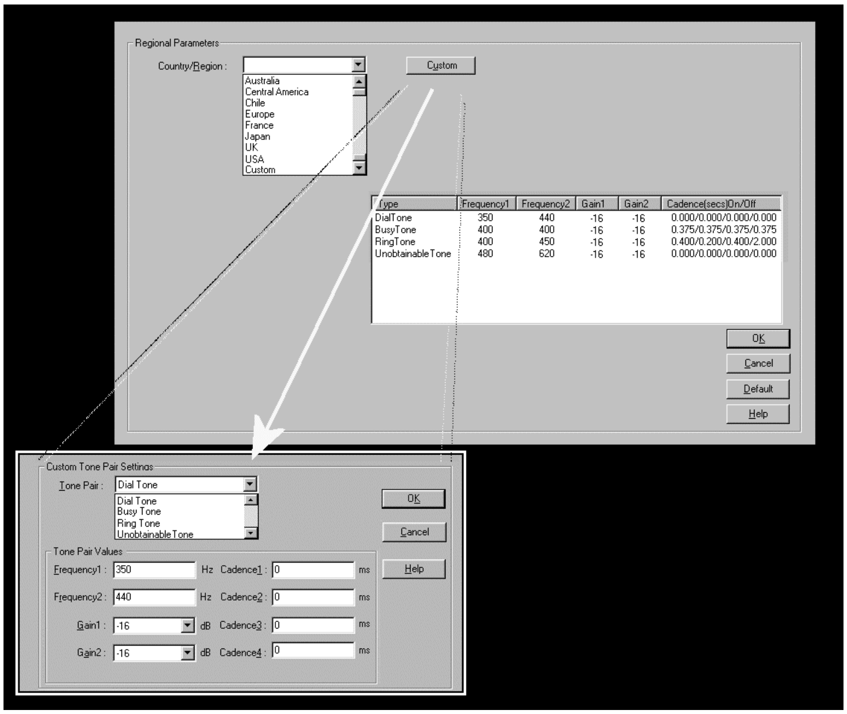
In each field, enter the values that fit your particular system.

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

"Regional Parameter" Definitions		
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) and re-order tone (a tone pattern indicating the need for the user to hang up the phone). In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The "Custom" option (button) assures that any tone-pairing scheme worldwide can be accommodated.
Type column	dial tone, ring tone, busy tone, unobtainable tone (fast busy), & re-order tone	Type of telephony tone-pair for which frequency, gain, and cadence are being presented.
Frequency 1	frequency in Hertz	Lower frequency of pair.
Frequency 2	frequency 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 figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the T1 port. **Default: -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the T1 port. **Default: -16dB**

“Regional Parameter” Definitions (cont’d)		
Field Name	**Values**	**Description**
Cadence (msec) On/Off	n/n/n/n four integer time values in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable (fast busy), and dial tone (continuous and described as “0“). Default values differ for different countries/regions. Although most cadences have only two parts (an “on” duration and an “off” duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.
Custom (button)	--	Click on the “Custom” button to bring up the **Custom Tone Pair Settings** screen. This screen allows the user to specify tone pair attributes that are not found in any of the standard national/regional telephony toning schemes.

13. **Set Custom Tones and Cadences** (optional) . The **Regional Parameters** dialog box has a secondary dialog box that allows you to customize DTMF tone pairs to create unique ring-tones, dial tones, busy-tones "unobtainable" tones (fast busy signal) or "re-order" tones (telling the user that they must hang up an off-hook phone) 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.

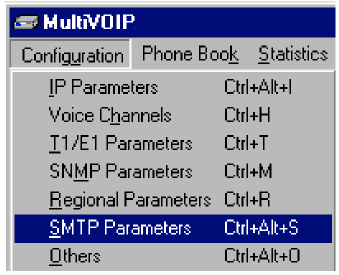
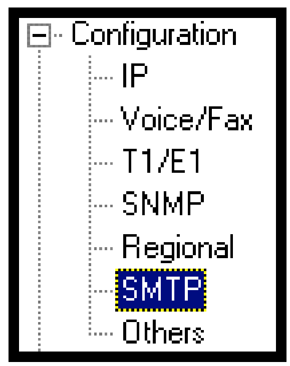


The **Custom Tone-Pair Settings** fields are described in the table below.

Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Tone Pair	dial tone busy tone ring tone, 'unobtainable' & re-order tones	Identifies the type of telephony signaling tone for which frequencies are being specified.
TONE PAIR VALUES		**About Defaults**: US telephony values are used as defaults on this screen. However, since this dialog box is provided to allow custom tone-pair settings, default values are essentially irrelevant.
Frequency 1	frequency in Hertz	Frequency of lower tone of pair. This outbound tone pair enters the MultiVOIP at the T1/E1 port.
Frequency 2	frequency in Hertz	Frequency of higher tone of pair. This outbound tone pair enters the MultiVOIP at the T1/E1 port.
Gain 1	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the T1 port. **Default = -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the T1 port. **Default = -16dB**

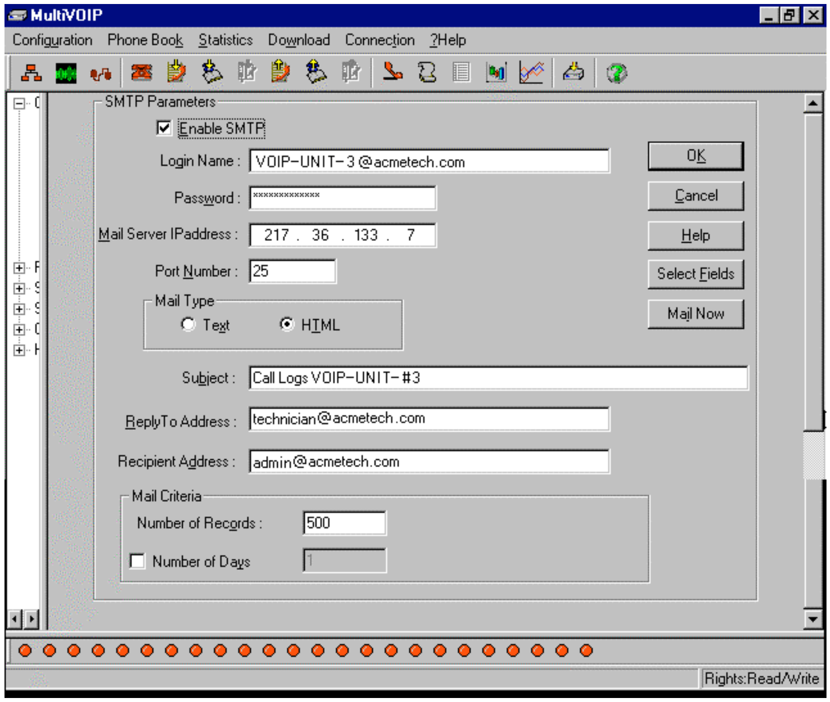
Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Cadence 1	integer time value in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable tone (fast busy), dial tone (which is continuous and described as "0") & re-order tone. Cadence 1 is duration of first period of tone being "on" in the cadence of the telephony signal (which could be ring-tone, busy-tone, unobtainable tone, dial tone, or re-order tone).
Cadence 2	duration in milliseconds	Cadence 2 is duration of first "off" period in signaling cadence.
Cadence 3	duration in milliseconds	Cadence 3 is duration of second "on" period in signaling cadence.
Cadence 4	duration in milliseconds	Cadence 4 is duration of second "off" period in the signaling cadence, after which the 4-part cadence pattern of the telephony signal repeats.

14. **Set 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.). The **SMTP Parameters** screen can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “SMTP Parameters”</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + S</td><td>
</td></tr>
</table>

MultiVOIP as Email Sender. When SMTP is used, the MultiVOIP will actually be given its own email account (with Login Name and Password) on some mail server connected to the IP network. Using this account, the MultiVOIP will then send out email messages containing log report information. The “Recipient” of the log report email is ordinarily the VoIP administrator. Because the MultiVOIP cannot receive email, a “Reply-To” address must also be set up. Ordinarily, the “Reply-To” address is that of 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 main function of the Reply-To address is to receive error or failure messages regarding the emailed reports.

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

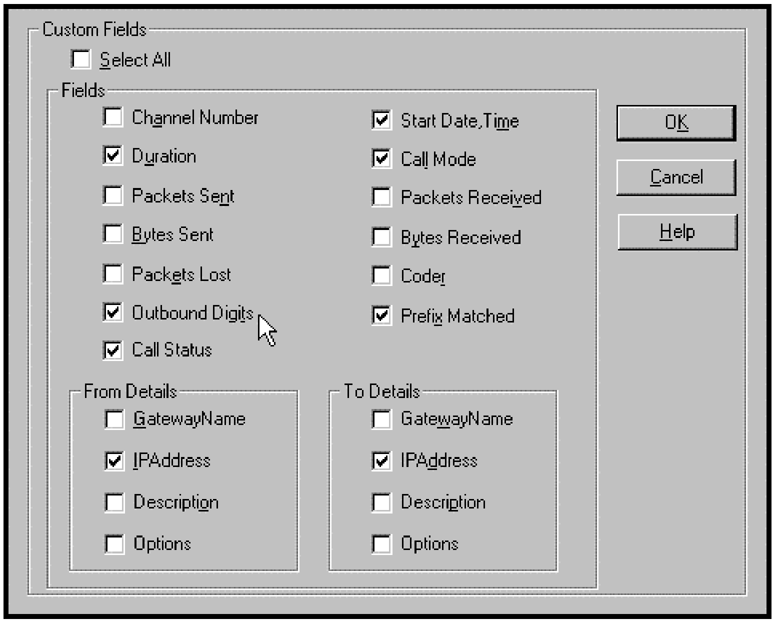


"SMTP Parameters" Definitions		
Field Name	**Values**	**Description**
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select "SMTP" in the **Logs** screen.
Login Name	alpha-numeric, per email domain	This is the User 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 for n= 0 to 255	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.

......

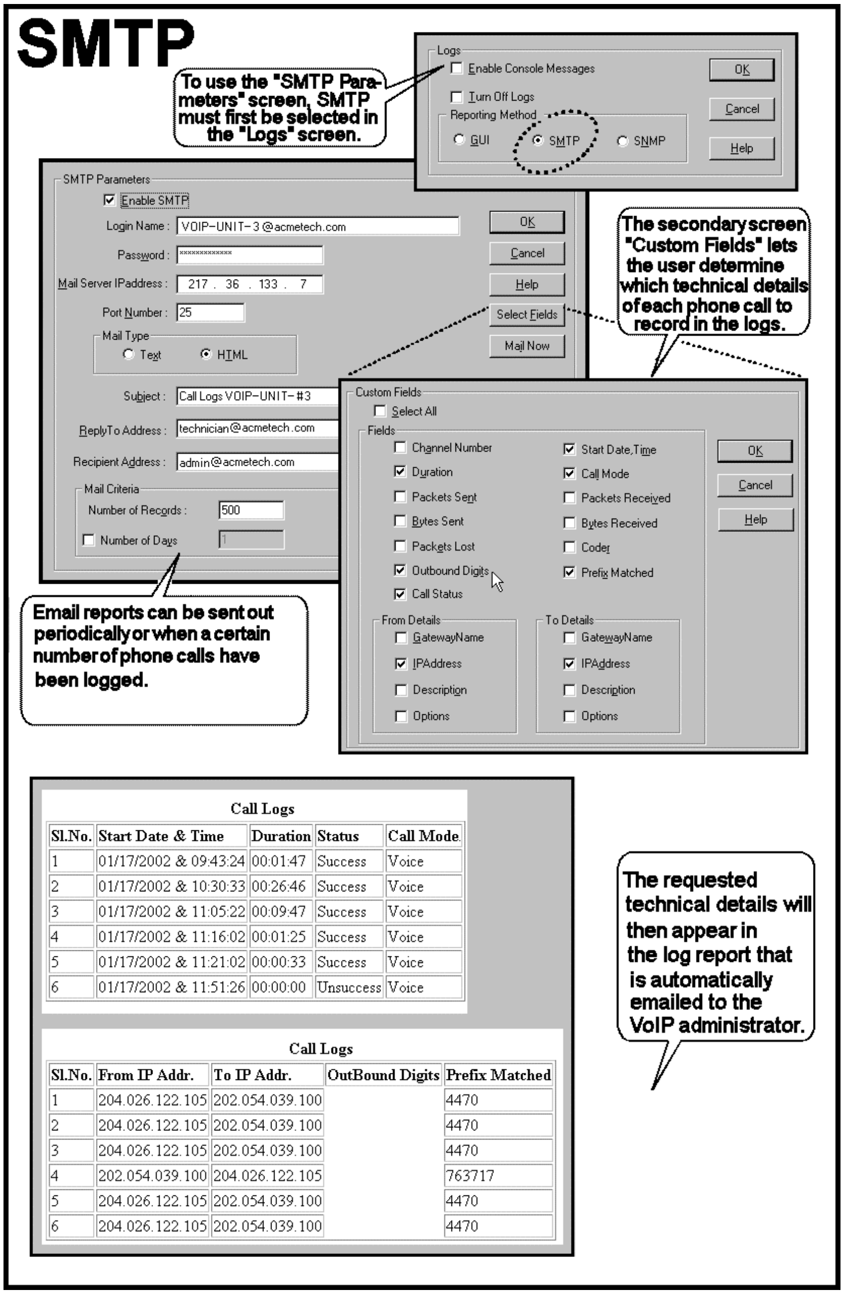
"SMTP Parameters" Definitions (cont'd)		
Field Name	**Values**	**Description**
Mail Type	text or html	Mail type in which log reports will be sent.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP unit.
Reply-To Address	email address	User specified. 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	User specified. Email address at which VOIP administrator will receive log reports.
Mail Criteria		Criteria for sending log summary by email. The log summary email will be sent out either when the user-specified number of log messages has accumulated, or once every day or multiple days, *which ever comes first*.
Number of Records	integer	This is the number of log records that must accumulate to trigger the sending of a log-summary email.
Number of Days	integer	This is 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, **Custom Fields**, that allows you to customize email log messages for the MultiVOIP. 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 Fields" Definitions			
Field	**Description**	**Field**	**Description**
Select All	Log report to include all fields shown.		
Channel Number	Data channel carrying call.	Start Date, Time	Date and time the phone call began.
Duration	Length of call.	Call Mode	Voice or fax.
Packets Sent	Total packets sent in call.	Packets Received	Total packets received in call.
Bytes Sent	Total bytes sent in call.	Bytes Received	Total bytes received in call.
Packets Lost	Packets lost in call.	Coder	Voice Coder /Compression Rate used for call will be listed in log.

"Custom Fields" Definitions (cont'd)			
Field	**Description**	**Field**	**Description**
Outbound Digits	Digits put out by MultiVOIP onto the T1 or E1 line.	Prefix Matched	When selected, the phonebook prefix matched in processing call will be listed in log.
Call Status	Successful or unsuccessful.		
From Details		**To Details**	
Gateway Number	Originating gateway	Gatew N.	Completing or terminating gateway
IP Addr	IP address where call originated.	IP Addr	IP address where call was completed or terminated.
Descript	Identifier of site where call originated.	Descript	Identifier of site where call was completed or terminated.
Options	When selected, log will not use/non-use of Silence Compression and Forward Error Correction by call originator.	Options	When selected, log will not use/non-use of Silence Compression and Forward Error Correction by call terminator.



Call Logs

Sl.No.	Start Date & Time	Duration	Status	Call Mode
1	01/17/2002 & 09:43:24	00:01:47	Success	Voice
2	01/17/2002 & 10:30:33	00:26:46	Success	Voice
3	01/17/2002 & 11:05:22	00:09:47	Success	Voice
4	01/17/2002 & 11:16:02	00:01:25	Success	Voice
5	01/17/2002 & 11:21:02	00:00:33	Success	Voice
6	01/17/2002 & 11:51:26	00:00:00	Unsuccess	Voice

Call Logs

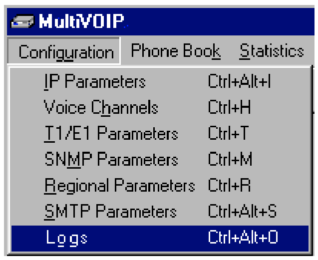
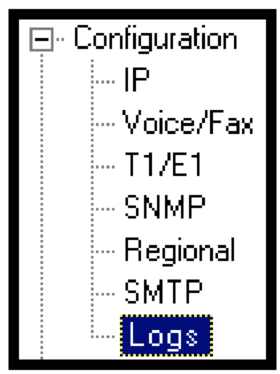
Sl.No.	From IP Addr.	To IP Addr.	OutBound Digits	Prefix Matched
1	204.026.122.105	202.054.039.100		4470
2	204.026.122.105	202.054.039.100		4470
3	204.026.122.105	202.054.039.100		4470
4	202.054.039.100	204.026.122.105		763717
5	204.026.122.105	202.054.039.100		4470
6	204.026.122.105	202.054.039.100		4470

15. **Set Log Reporting Method**. The **Logs** 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:

A. in the MultiVOIP program (GUI),

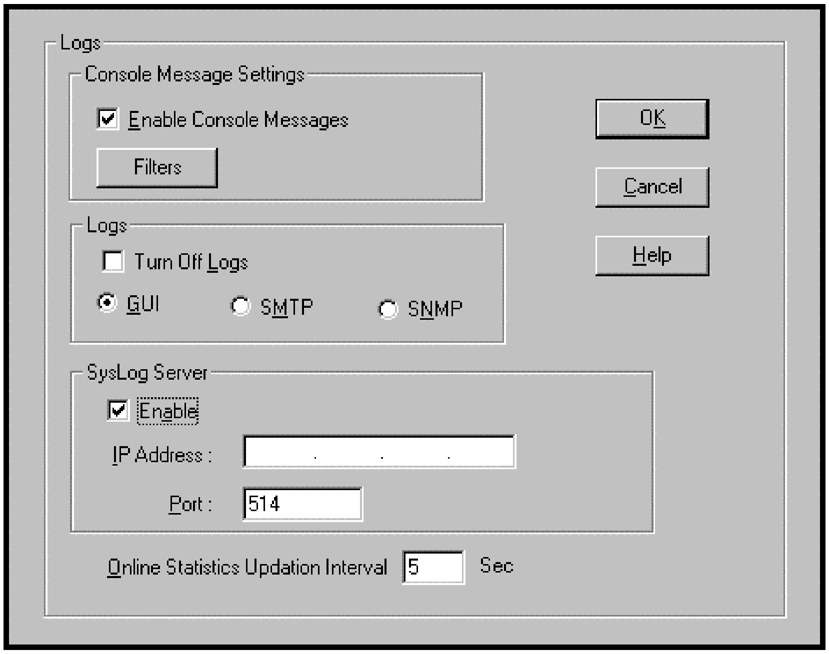
B. via email (SMTP), or

C. at the MultiVoipManager remote voip system management program (SNMP).

Accessing "Logs" Screen	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
Ctrl + Alt + O	

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 (see subsequent page). If you use the logging function, select the logging option that applies to your

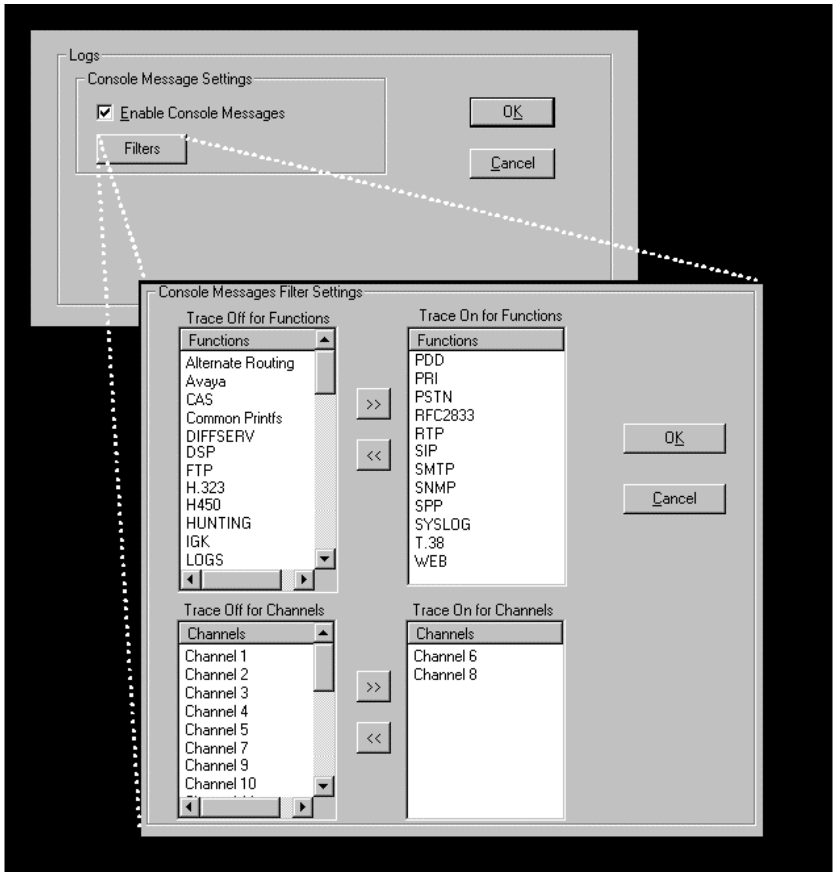
VoIP system design. If you intend to use a SysLog Server program for logging, click in that Enable check box. The common SysLog logical port number is 514. If you intend to use the MultiVOIP web browser GUI for configuration and control of MultiVOIP units, be aware that the web browser GUI does not support logs directly. However, when the web browser GUI is used, log files can still be sent to the voip administrator via email (which requires activating the SMTP logging option in this screen).



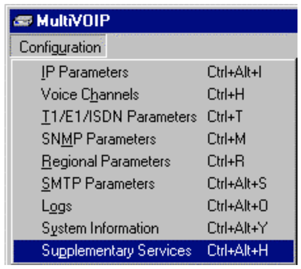
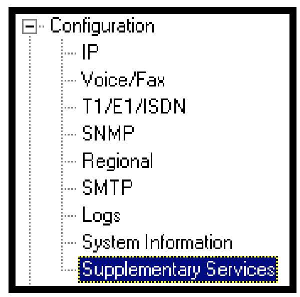
"Logs" Screen Definitions		
Field Name	**Values**	**Description**
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic tele-communications program like HyperTerminal ™ or similar application. Normally, this should be disabled because it consumers MultiVOIP pro-cessing resources. Console messages are meant for use by tech support personnel.

“Logs” Screen Definitions (cont’d)		
Field Name	**Values**	**Description**
Filters (button)		Click to access secondary screen on where console messages can be included/excluded by category and on a per-channel basis. (See the Console Messages Filter Settings screen on subsequent page.)
Turn Off Logs	Y/N	Disables log reporting function.
Logs Buttons		Only one of these three log reporting methods, GUI, SMTP, or SNMP, may be chosen.
GUI	Y/N	User must view logs at the MultiVOIP configuration program.
SNMP	Y/N	Log messages will be delivered to the MultiVoipManager application program.
SMTP	Y/N	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. For more on SysLog Server, see *Operation & Maintenance* chapter.
IP Address	n.n.n.n for n= 0-255	IP address of computer, connected to 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.

To customize console messages by category and/or by channel, click on "Filters" and use the **Console Messages Filters Settings** screen.

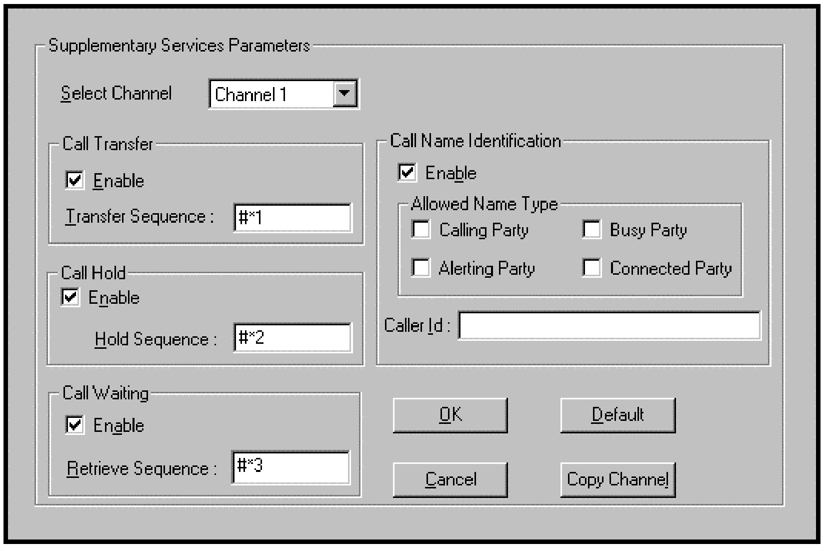


16. **Set Supplementary Services Parameters.** This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

Accessing "Supplementary Services Parameters"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
Ctrl + Alt +H	

Supplementary Services features derive from the H.450 standard, which brings to voip telephony functionality once only available with PSTN or PBX telephony. Supplementary Services features can be used under H.323 only and not under SIP.

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



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**.

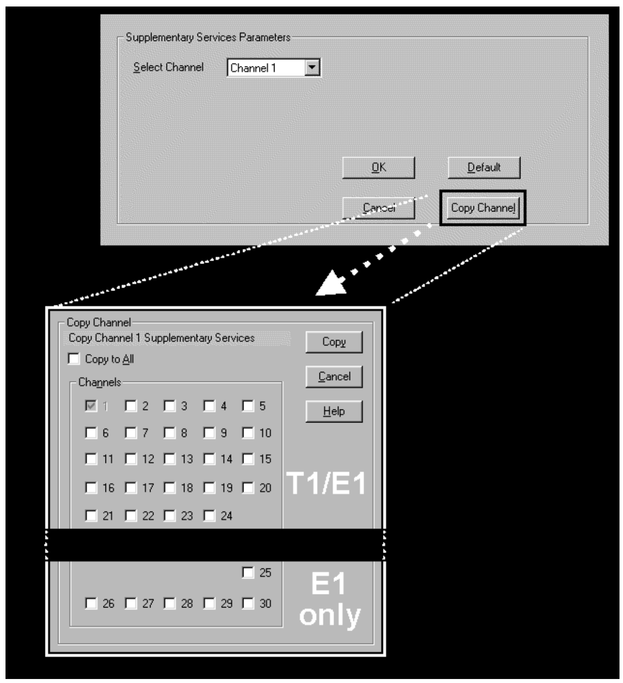
Call Transfer. Call Transfer allows one party to re-connect the party with whom they have been speaking to a third party. The first party is disconnected when the third party becomes connected. Feature is invoked by a programmable phone keypad sequence (for example, #7).

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. Invoked by keypad sequence.

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. Invoked by keypad sequence.

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 (callee) 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 "). These messages appear in the **Statistics – Call Progress** screen of the remote voip.

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



The **Supplementary Services** fields are described in the tables below.

Supplementary Services Parameter Definitions		
Field Name	**Values**	**Description**
Select Channel	1-2 (210); 1-4 (410); 1-8 (810)	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.
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*#.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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 GUI (the 'home voip'), Call Name 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 callee (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.)

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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.

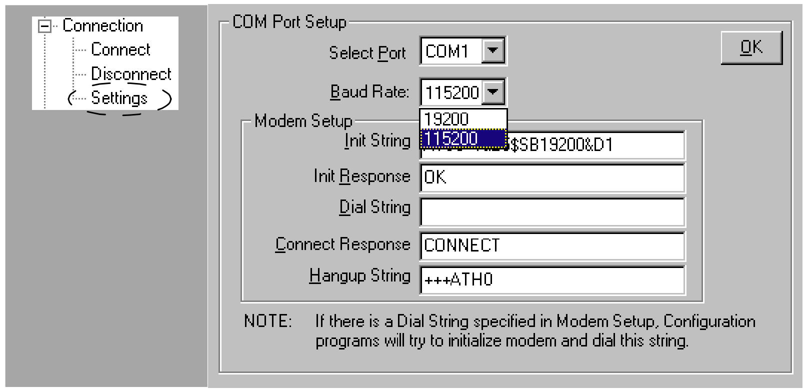
Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Alerting Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving the call and **Alerting Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit 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.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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. When channel 2 of the Omaha voip completes an attempted call from any other voip phone station (for example, the Denver office), the message "Connected 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.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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.

17. **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.

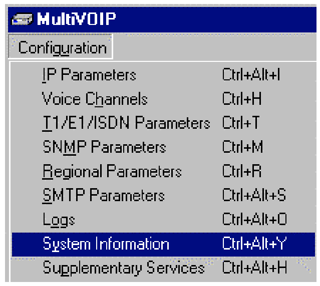
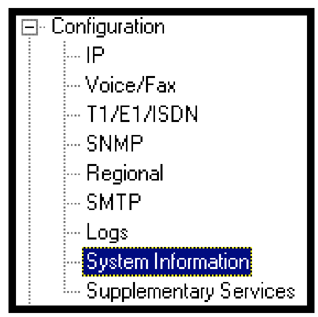


First, it is important to note that the default COM port established by the MultiVOIP program is COM1. ***Do not accept the default value until you have checked the COM port allocation on your PC***. To do this, check for COM port assignments in the system resource dialog box(es) of your Windows operating system. If COM1 is not available, you must change the COM port setting to COM2 or some other COM port that you have confirmed as being available on your PC.

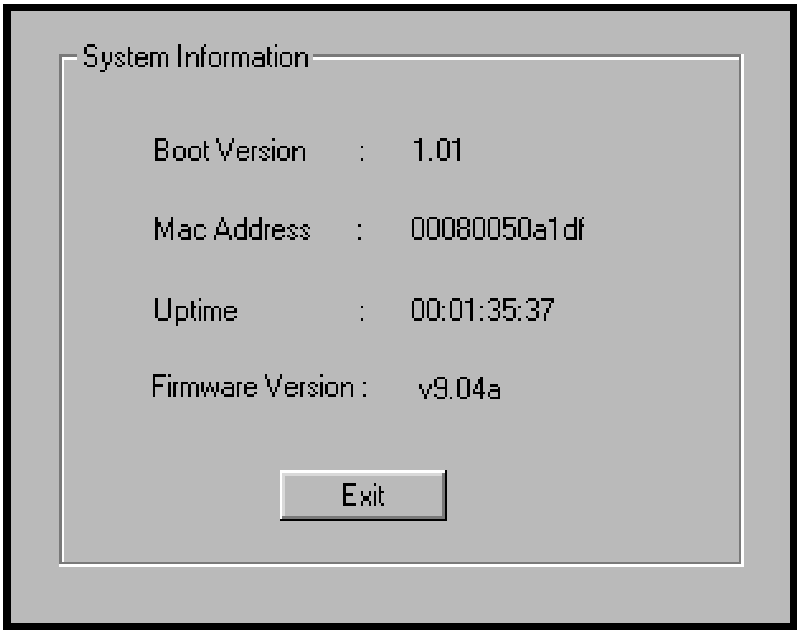
The default baud rate is 115,200 bps.

18. View **System Information** screen and set updating interval (optional).

This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

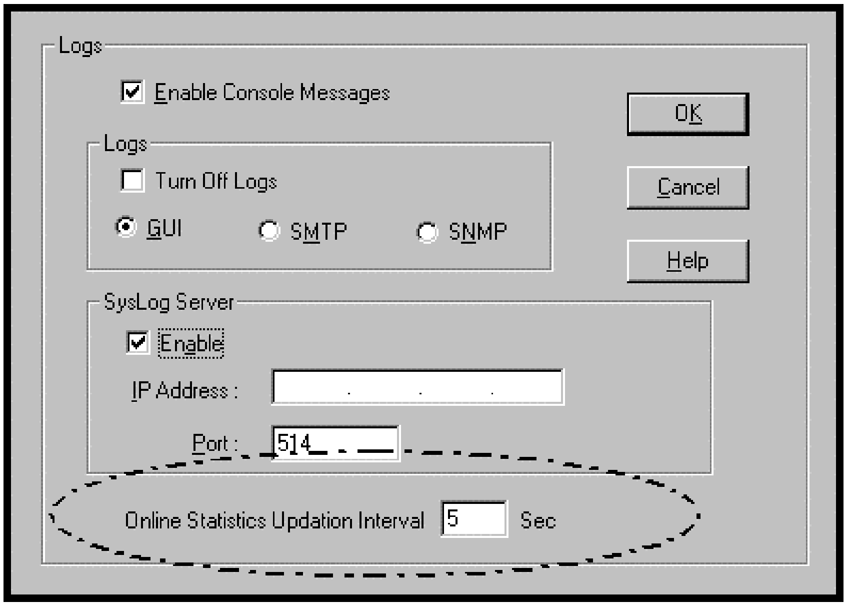
<table>
<tr><th colspan="2">Accessing the “System Information” Screen</th></tr>
<tr><td>Pulldown</td><td>Icon</td></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><td>Shortcut</td><td>Sidebar</td></tr>
<tr><td>Ctrl + Alt +Y</td><td>
</td></tr>
</table>

This screen presents vital system information at a glance. Its primary use is in troubleshooting.

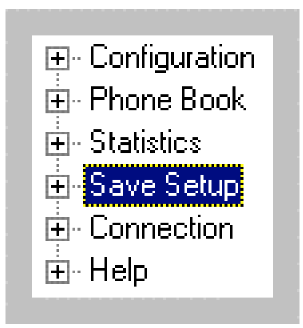


System Information Parameter Definitions		
Field Name	**Values**	**Description**
Boot Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Mac Address	alpha-numeric	Denotes the number assigned 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.
Firmware Version	alpha-numeric	Indicates version of MultiVOIP firmware.

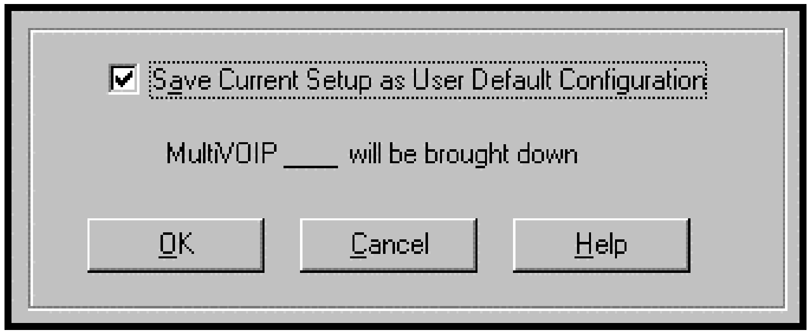
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



19. **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.



20. **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.



Chapter 6: Technical Configuration for Analog/BRI MultiVOIPs (MVP130, MVP-210/210G, MVP-410/410G, MVP-810/810G & MVP-410ST/810ST)

Configuring the Analog/BRI MultiVOIP

There are two ways in which the MultiVOIP must be configured before operation: technical configuration and phonebook configuration.

Technical Configuration. First, the MultiVOIP must be configured to operate with technical parameter settings that will match the equipment with which it interfaces. There are eight types of technical parameters that must be set.

These technical parameters pertain to
(1) its operation in an IP network,
(2) its operation with telephony equipment,
(3) its transmission of voice and fax messages,
(4) its interaction with SNMP (Simple Network Management Protocol) network management software (MultiVoipManager),
(5) certain telephony attributes that are common to particular nations or regions,
(6) its operation with a mail server on the same IP network (per SMTP parameters) such that log reports about VoIP telephone call traffic can be sent to the administrator by email,
(7) implementing some common premium telephony features (Call Transfer, Call Hold, Call Waiting, Call ID – “Supplementary Services”), and
(8) selecting the method by which log reports will be made accessible.

The process of specifying values for the various parameters in these seven categories is what we call “technical configuration” and it is described in this chapter.

Phonebook Configuration. The second type of configuration that is required for the MultiVOIP pertains to the phone number dialing sequences that it will receive and transmit when handling calls. Dialing patterns will be affected by both the PBX/telephony equipment and the other VOIP devices that the MultiVOIP unit interacts with. We call this “Phonebook Configuration,” and, for analog MultiVOIP units, it is described nominally in *Chapter 9: Analog Phonebook Configuration* of this manual. But, in fact, nearly all of the descriptions and examples for analog phonebook configuration are to be found in Chapter 7 if the analog voip is operating under the North American telephony scheme, or in Chapter 8 if the analog voip is operating under a European telephony scheme. Chapter 2, the *Quick Start Instructions*, presents additional examples relevant to the analog voips.

Local/Remote Configuration. The MultiVOIP must be configured locally at first (to establish an IP address for the MultiVOIP unit). But changes to this initial configuration can be done either locally or remotely.

Local configuration is done through a connection between the "Command" port of the MultiVOIP and the COM port of the computer; the MultiVOIP configuration program is used.

Remote configuration is done through a connection between the MultiVOIP's Ethernet (network) port and a computer connected to the same network. The computer could be miles or continents away from the MultiVOIP itself. There are two ways of doing remote configuration and operation of the MultiVOIP unit: (1) using the MultiVoipManager SNMP program, or (2) using the MultiVOIP web browser interface program.

MultiVoipManager. MultiVoipManager is an SNMP agent program (Simple Network Management Protocol) that extends the capabilities of the MultiVOIP configuration program: MultiVoipManager allows the user to manage any number of VOIPs on a network, whereas the MultiVOIP configuration program can manage only the VOIP to which it is directly/locally connected. The MultiVoipManager can configure multiple VOIPs simultaneously, whereas the MultiVOIP configuration program can configure only one at a time.

MultiVoipManager may (but does not need to) reside on the same PC as the MultiVOIP configuration program. The MultiVoipManager program is on the MultiVOIP Product CD. Updates, when applicable, may be posted at on the MultiTech FTP site. To download, go to ftp://ftp.multitech.com/MultiVoip/.

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

Functional Equivalence of Interfaces. The MultiVOIP configuration program is required to do the initial configuration (that is, setting an IP address for the MultiVOIP unit) so that the VOIP unit can communicate with the MultiVoipManager program or with the web browser GUI. Management of the VOIP after that point can be done from any of these three programs since they all offer essentially the same functionality. Functionally, either the MultiVoipManager program or the web browser GUI can replace the MultiVOIP configuration program after the initial configuration is complete (with minor exceptions, as noted).

WARNING: Do not attempt to interface the MultiVOIP unit with two control programs simultaneously (that is, by accessing the MultiVOIP configuration program via the Command Port and either the MultiVoipManager program or the web browser interface via the Ethernet Port). The results of using two programs to control a single VOIP simultaneously would be unpredictable.

Local Configuration

This manual primarily describes local configuration with the Windows GUI. After IP addresses have been set locally using the Windows GUI, most aspects of configuration (logging functions are an exception) can be handled through the web browser GUI, as well (see the *Operation and Maintenance* chapter of this manual). In most aspects of configuration, the Windows GUI and web-browser GUI differ only graphically, not functionally. For information on SNMP remote configuration and management, see the MultiVoipManager documentation.

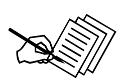
Pre-Requisites

To complete the configuration of the MultiVOIP unit, you ***must*** know several things about the overall system.

Before configuring your MultiVOIP Gateway unit, you must know the values for several IP and telephone parameters that describe the IP network system and telephony system (PBX or telco central office equipment) with which the digital MultiVOIP will interact. If you plan to receive log reports on phone traffic by email (SMTP), you must arrange to have an email address assigned to the VOIP unit on the email server on your IP network.

IP Parameters

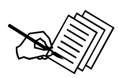
The following parameters must be known about the network (LAN, WAN, Internet, etc.) to which the MultiVOIP will connect:

➨	*Ask your computer network administrator.*	*Info needed to operate:* **all MultiVOIP models.**
	 IP Network Parameters: Record for each VOIP Site in System	
	• IP Address	
	• IP Mask	
	• Gateway	
	• Domain Name Server (DNS) Info *(not implemented; for future use)*	

Write down the values for these IP parameters. You will need to enter these values in the "IP Parameters" screen in the Configuration section of the MultiVOIP software. You must have this IP information about *every* VOIP in the system.

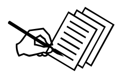
Analog Telephony Interface Parameters (for MVP130/210/410/810)

The following parameters must be known about the PBX or telco central office equipment to which the analog MultiVOIP will connect:

➨	**Analog Phone Parameters** *Ask phone company or telecom manager.*	*Needed for:* **MVP810** **MVP410** **MVP210** **MVP130**
	 Analog Telephony Interface Parameters: Record for this VOIP Site	
	• Which interface type (or "signaling") is used? E&M_____ FXS/FXO_____	
	• If FXS, determine whether the line will be used for a phone, fax, or KTS (key telephone system)	
	• If FXO, determine if line will be an analog PBX extension or an analog line from a telco central office	
	• If E&M, determine these aspects of the E&M trunk line from the PBX: • What is its Type (1, 2, 3, 4, or 5)? • Is it 2-wire or 4-wire? • Is it Dial Tone or Wink?	

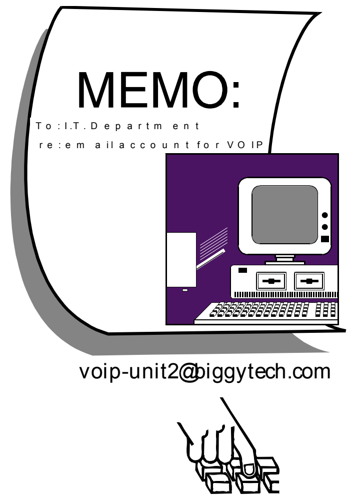
ISDN-BRI Telephony Parameters (for MVP-410ST/810ST)

The following parameters must be known about the PBX or telco central office equipment to which the analog MultiVOIP will connect:

➨	**ISDN-BRI Phone Parameters** *Ask phone company or telecom manager.*	*Needed for:* **MVP810ST** **MVP410ST**
	 ISDN-BRI Telephony Interface Parameters: Record them for this VOIP Site	
	• In which country is this voip installed?	
	• Which operator (switch type) is used?	
	• What type of line coding use required, A-law or u-law?	
	• Determine which BRI ports will be network side and which BRI ports will be terminal side.	

Write down the values for these telephony parameters (whether analog or ISDN-BRI). You will need to enter these values in the “Interface” screen (analog) or “ISDN Parameters” screen (ISDN-BRI) in the Configuration section of the MultiVOIP software.

SMTP Parameters (for email call log reporting)

required if log reports of VOIP call traffic are to be sent by email	**Optional**
SMTP Parameters Preparation Task: 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.	

Local Configuration Procedure (Summary)

After the MultiVOIP configuration software has been installed in the 'Command' PC (which is connected to the MultiVOIP unit), several steps must be taken to configure the MultiVOIP to function in its specific setting. Although the summary below includes all of these steps, some are optional.

1. Check Power and Cabling.

2. Start MultiVOIP Configuration Program.

3. Confirm Connection.

4. Solve Common Connection Problems.

   A. Fixing a COM Port Problem.

   B. Fixing a Cabling Problem.

5. Familiarize yourself with configuration parameter screens and how to access them.

6. Set IP Parameters.

7. Enable web browser GUI (optional).

8. Set Voice/Fax Parameters.

9. Set Telephony Interface Parameters (analog) or ISDN Parameters (ISDN/BRI).

10. Set SNMP Parameters (applicable if MultiVoipManager remote management software is used).

11. Set Regional Parameters (Phone Signaling Tones and Cadences).

12. Set Custom Tones and Cadences (optional).

13. Set SMTP Parameters (applicable if Log Reports are via Email).

14. Set Log Reporting Method (GUI, locally in MultiVOIP Configuration program; SNMP, remotely in MultiVoipManager program; or SMTP, via email).

15. Set Supplementary Services Parameters. The Supplementary Services screen allows voip deployment of features that are normally found in PBX or PSTN systems (e.g., call transfer and call waiting).

16. Set Baud Rate (of COM port connection to 'Command' PC).

17. View System Info screen and set updating interval (optional).

18. Save the MultiVOIP configuration.

19. Create a User Default Configuration (optional).

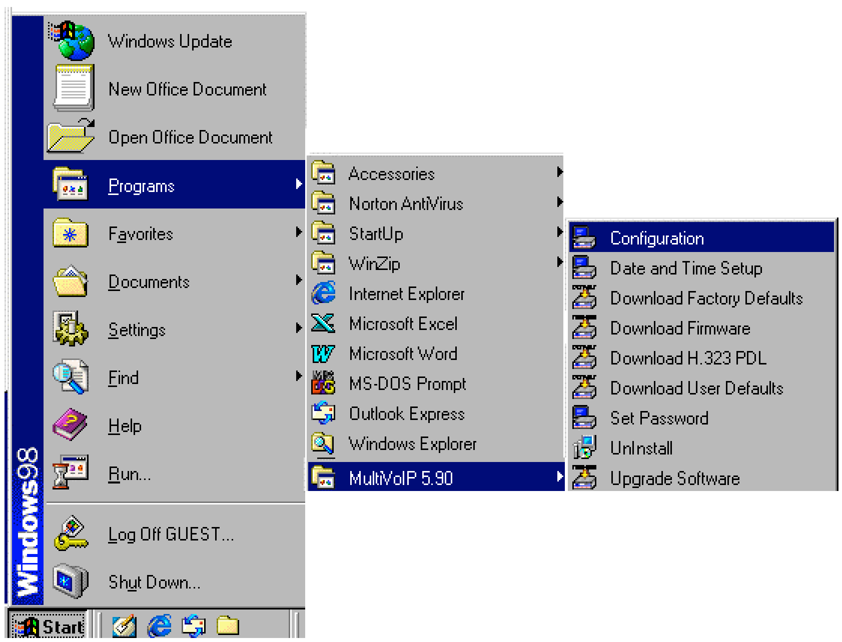
When technical configuration is complete, you will need to configure the MultiVOIP's phonebooks (for all models) and its embedded gatekeeper functionality, if present (for MVP-210G, -410G, and 810G only). This manual has separate chapters describing ***T1 Phonebook Configuration*** for North-American-influenced telephony settings and ***E1 Phonebook Configuration*** for Euro-influenced telephony settings, as well as a separate ***Embedded Gatekeeper*** chapter.

Local Configuration Procedure (Detailed)

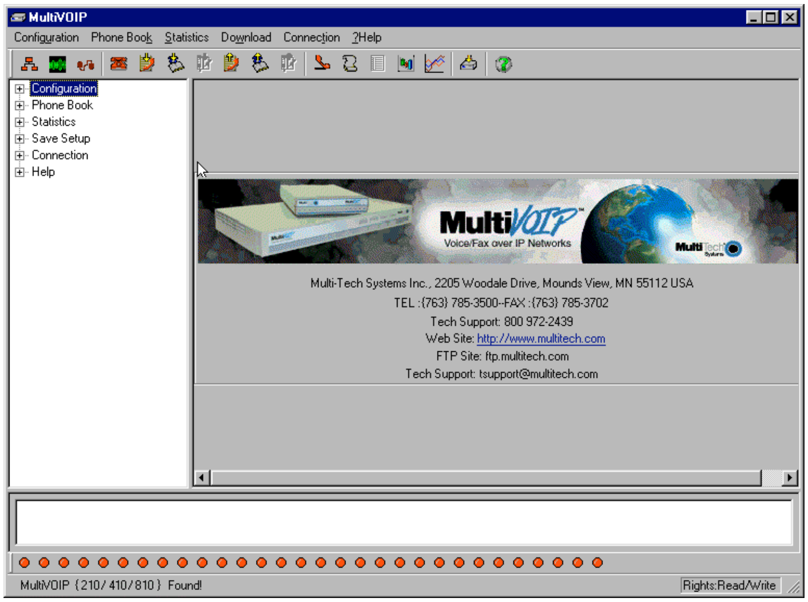
You can begin the configuration process as a continuation of the MultiVOIP software installation. You can establish your configuration or modify it at any time by launching the MultiVOIP program from the Windows **Start** menu.

1. **Check Power and Cabling**. Be sure the MultiVOIP is turned on and connected to the computer via the MultiVOIP's Command Port (DB9 connector at computer's COM port; RJ45 connector at MultiVOIP).

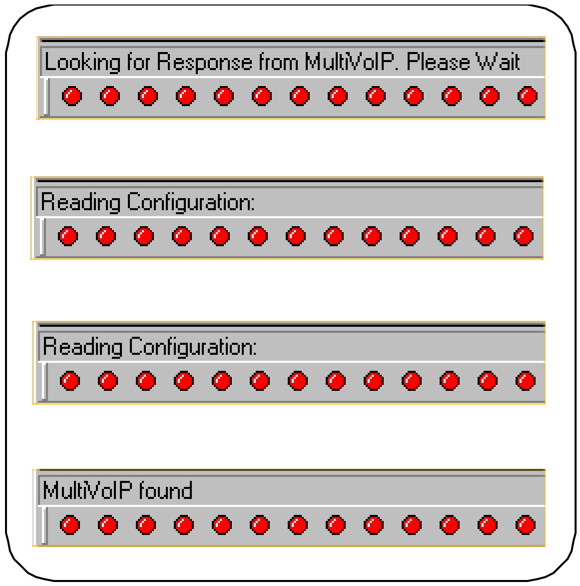
2. **Start MultiVOIP Configuration Program**. Launch the MultiVOIP program from the Windows **Start** menu (from the folder location determined during installation).



3. **Confirm Connection**. If the MultiVOIP is set for an available COM port and is correctly cabled to the PC, the MultiVOIP main screen will appear. (If the main screen appears *grayed out* and seems inaccessible, go to step 4.)



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. Skip to step 5.

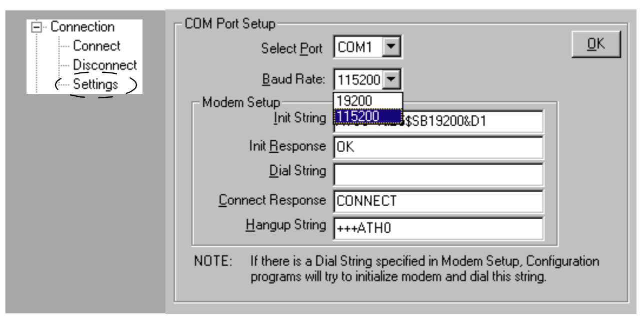


4. **Solving Common Connection Problems**. .

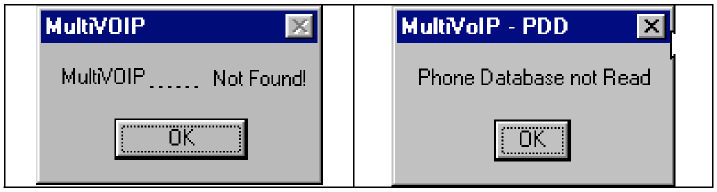
A. Fixing a COM Port Problem. If the MultiVOIP main screen appears but is grayed out and seems inaccessible, the COM port that was specified for its communication with the PC is unavailable and must be changed. An error message will appear.

To change the COM port setting, use the **COM Port Setup** dialog box, which is accessible via the keyboard shortcut **Ctrl + G** or by going to the **Connection** pull-down menu and choosing "Settings." 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.)

Ctrl + G



4B. Fixing a Cabling Problem. If the MultiVOIP cannot be located by the computer, two error messages will appear (saying "Multi-VOIP Not Found" and "Phone Database Not Read").

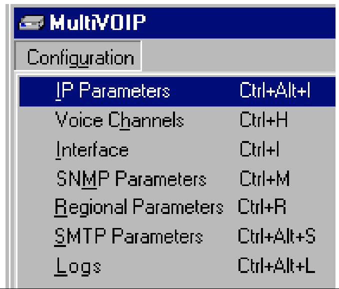
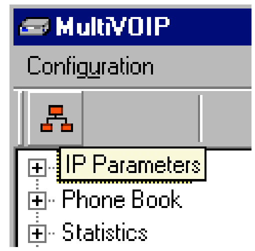
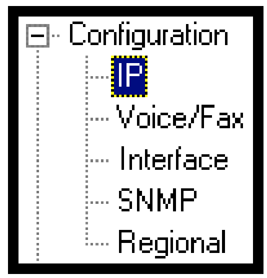


In this case, the MultiVOIP is simply disconnected from the network. For instructions on MultiVOIP cable connections, see the Cabling section of Chapter 3.

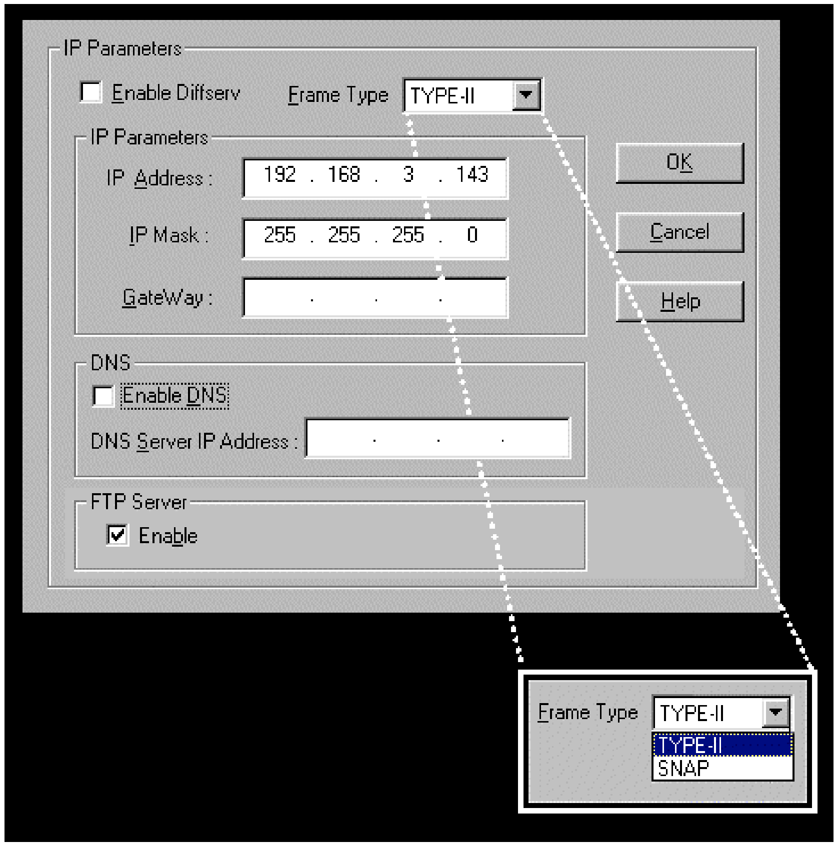
5. **Configuration Parameter Groups: Getting Familiar, Learning About Access**. The first part of configuration concerns IP parameters, Voice/FAX parameters, Telephony Interface parameters, SNMP parameters, Regional parameters, SMTP parameters, Supplementary Services parameters, Logs, and System Information. In the MultiVOIP software, these seven types of parameters are grouped together under "Configuration" and each has its own dialog box for entering values.

Generally, you can reach the dialog box for these parameter groups in one of four ways: pulldown menu, toolbar icon, keyboard shortcut, or sidebar. ..

6. **Set IP Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing "IP Parameters"</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td>
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>**Ctrl + Alt + I**</td><td>
</td></tr>
</table>

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



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

IP Parameter Definitions		
Field Name	**Values**	**Description**
Enable Diffserv	Y/N	Diffserv is used for QoS (quality of service). When enabled, the TOS (Type of Service) bits in the IP header are configured so that routers supporting Diffserv can give priority to the VOIP's IP packets. Disabled by default.
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.
IP Address	4-places, 0-255	The unique LAN IP address assigned to the MultiVOIP.
IP Mask	4-places, 0-255	Subnetwork address that allows for sharing of IP addresses within a LAN.
Gateway	4-places, 0-255.	The IP address of the device that connects your MultiVOIP to the Internet.
Enable DNS	Y/N. *(feature not yet implemented; for future use)*	Enables Domain Name Space/System function where computer names are resolved using a worldwide distributed database.
DNS Server IP Address	4-places, 0-255 *(feature not yet implemented; for future use)*	IP address of specific DNS server to be used to resolve Internet computer names.
FTP Server Enable	Y/N See "FTP Server File Transfers" in *Operation & Maintenance* chapter.	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.

7. **Enable Web Browser GUI (Optional)**. After an IP address for the MultiVOIP unit has been established, you can choose to do any further configuration of the unit (a) by using the MultiVOIP web browser GUI, or (b) by continuing to use the MultiVOIP Windows GUI. If you want to do configuration work using the web browser GUI, you must first enable it. To do so, follow the steps below.

A. Set IP address of MultiVOIP unit using the MultiVOIP Configuration program (the Windows GUI).

B. Save Setup in Windows GUI.

C. Close Windows GUI.

D. Install Java program from MultiVOIP product CD (on first use only).

E. Open web browser.

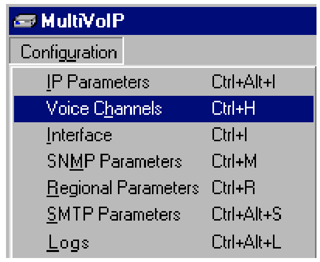
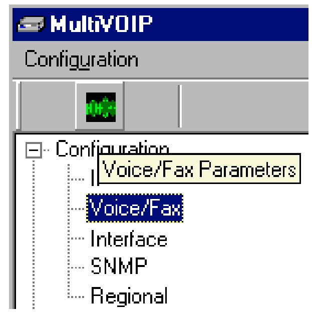
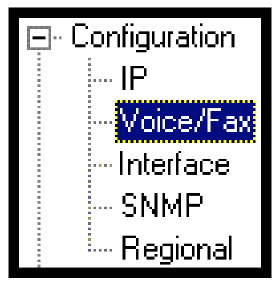
F. Browse to IP address of MultiVOIP unit.

G. If username and password have been established, enter them when when prompted.

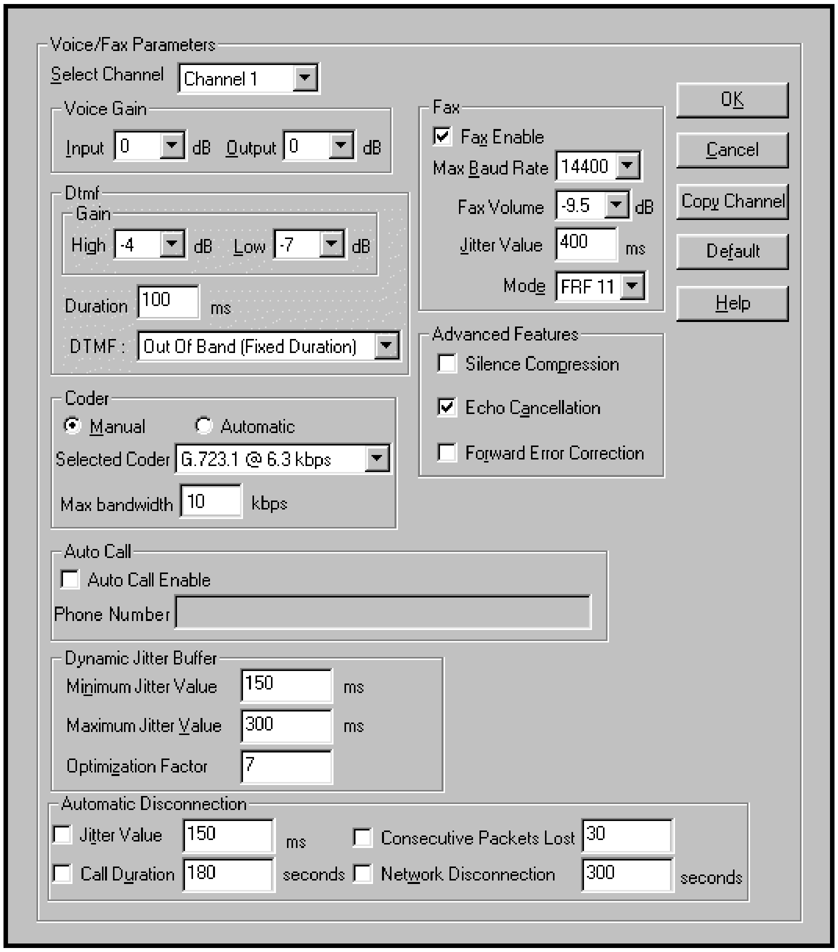
H. Use web browser GUI to configure or operate MultiVOIP unit. The configuration screens in the web browser GUI will have the same content as their counterparts in the Windows GUI; only the graphic presentation will be different.

For more details on enabling the MultiVOIP web GUI, see the "Web Browser Interface" section of the *Operation & Maintenance* chapter of this manual.

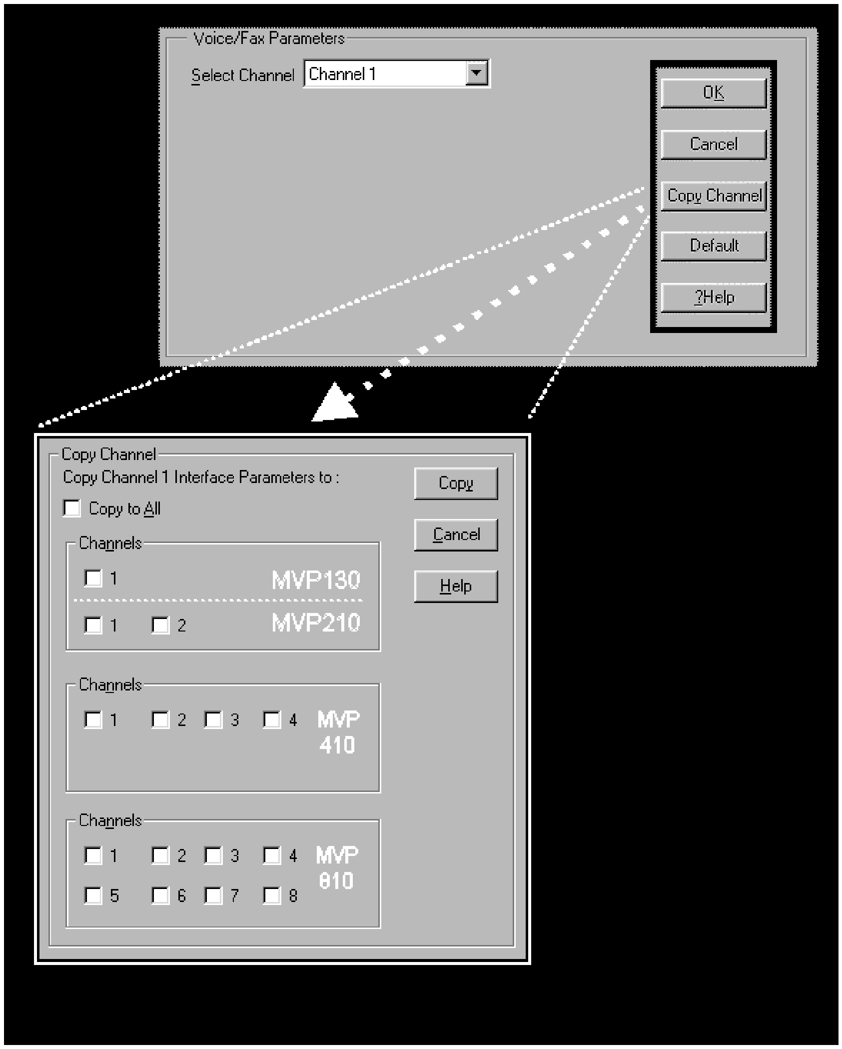
8. **Set Voice/FAX Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

Accessing "Voice/FAX Parameters"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + H`	

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



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



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

Voice/Fax Parameter Definitions		
Field Name	**Values**	**Description**
Default	--	When this button is clicked, all Voice/FAX parameters are set to their default values.
Select Channel	1-2 (210) 1-4 (410) 1-8 (810)	Channel to be configured is selected here.
Copy Channel	--	Copies the Voice/FAX attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.
Voice Gain	--	Signal amplification (or attenuation) in dB.
Input Gain	+31dB to –31dB	Modifies audio level entering voice channel before it is sent over the network to the remote VOIP. The default & recommended value is **0 dB**.
Output Gain	+31dB to –31dB	Modifies audio level being output to the device attached to the voice channel. The default and recommended value is **0 dB**.
DTMF Parameters		
DTMF Gain	--	The **DTMF Gain** (Dual Tone Multi-Frequency) controls the volume level of the digital tones sent out for Touch-Tone dialing.
DTMF Gain, High Tones	+3dB to -31dB & “mute”	Default value: **-4 dB**. Not to be changed except under supervision of MultiTech’s Technical Support.
DTMF Gain, Low Tones	+3dB to -31dB & “mute”	Default value: **-7 dB**. Not to be changed except under supervision of MultiTech’s Technical Support.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
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. Not supported in 5.02c BRI software.
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. In 502c BRI software, "DTMF Out of Band" can be checked or unchecked.
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 (Default = -9.5 dB)	-18.5 dB to –3.5 dB	Controls output level of fax tones. To be changed only under the direction of Multi-Tech's Technical Support.
Jitter Value (Fax)	Default = 400 ms	Defines the inter-arrival packet deviation (in milliseconds) for the fax transmission. A higher value will increase the delay, allowing a higher percentage of packets to be reassembled. A lower value will decrease the delay allowing fewer packets to be reassembled.
Mode (Fax)	FRF 11; T.38 (T.38 not currently sup-ported)	**FRF11** is frame-relay FAX standard using these coders: G.711, G.728, G.729, G.723.1. **T.38** is an ITU-T standard for storing and forwarding FAXes via email using X.25 packets. It uses T.30 fax standards and includes special provisions to preclude FAX timeouts during IP transmissions.

Voice/Fax Parameter Definitions (cont'd)		
Coder Parameters		
Coder	Manual or Auto-matic	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 are negotiated.
Selected Coder	**G.711** a/u law 64 kbps; **G.726**, @ 16/24/32/40 kbps; **G.727**, @ nine bps rates; **G.723.1** @ 5.3 kbps, 6.3 kbps; **G.729**, 8kbps; **Net Coder** @ 6.4, 7.2, 8, 8.8, 9.6 kbps	Select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling must have the same voice coder selected. Default = G.723.1 @ 6.3 kbps, as required for H.323. Here 64K of digital voice are compressed to 6.3K, allowing several simultaneous conversations over the same bandwidth that would otherwise carry only one. To make selections from the Selected Coder drop-down list, the Manual option must be enabled.
Max bandwidth (coder)	11 – 128 kbps	This drop-down list enables you to select the maximum bandwidth allowed for this channel. The Max Bandwidth drop-down list is enabled only if the Coder is set to Automatic. If coder is to be selected automatically ("Auto" setting), then enter a value for maximum bandwidth.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
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 (checked) for this voice channel. **Echo Cancellation** removes echo and improves sound quality. Default = on.
Forward Error Correction	Y/N	Determines whether forward error correction is enabled (checked) for this voice channel. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off
Auto Call Enable	Y/N	The **Auto Call** 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.
Phone No. (Auto Call)	--	Phone number used for Auto Call function. A corresponding phone number must be listed in the Outbound Phonebook.

Voice/Fax Parameter Definitions (cont'd))		
Field Name	**Values**	**Description**
Dynamic Jitter		
Dynamic Jitter Buffer		**Dynamic Jitter** defines a minimum and a maximum jitter value for voice communications. When receiving voice packets from a remote MultiVOIP, varying delays between packets may occur due to network traffic problems. This is called Jitter. To compensate, the MultiVOIP uses a Dynamic Jitter Buffer. The Jitter Buffer enables the MultiVOIP to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An Optimization Factor adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MultiVOIP gateways.
Minimum Jitter Value	60 to 400 ms	The minimum dynamic jitter buffer of **60** milliseconds is the minimum delay that would be acceptable over a low jitter network. Default = 150 msec

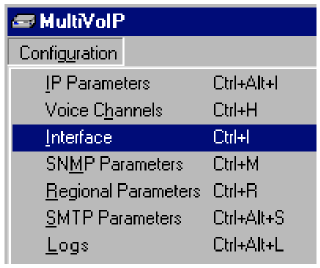
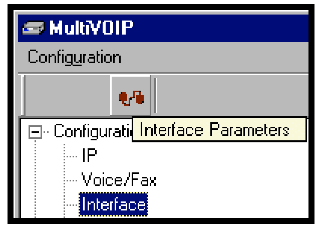
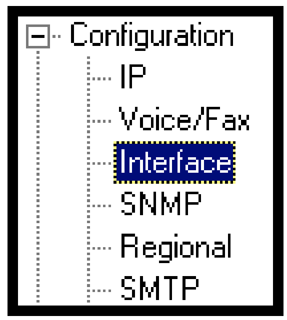
Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dynamic Jitter		
Maximum Jitter Value	60 to 400 ms	The maximum dynamic jitter buffer of **400** milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 msec
Optimizat-ion Factor	0 to 12	The **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**.

Modem Relay

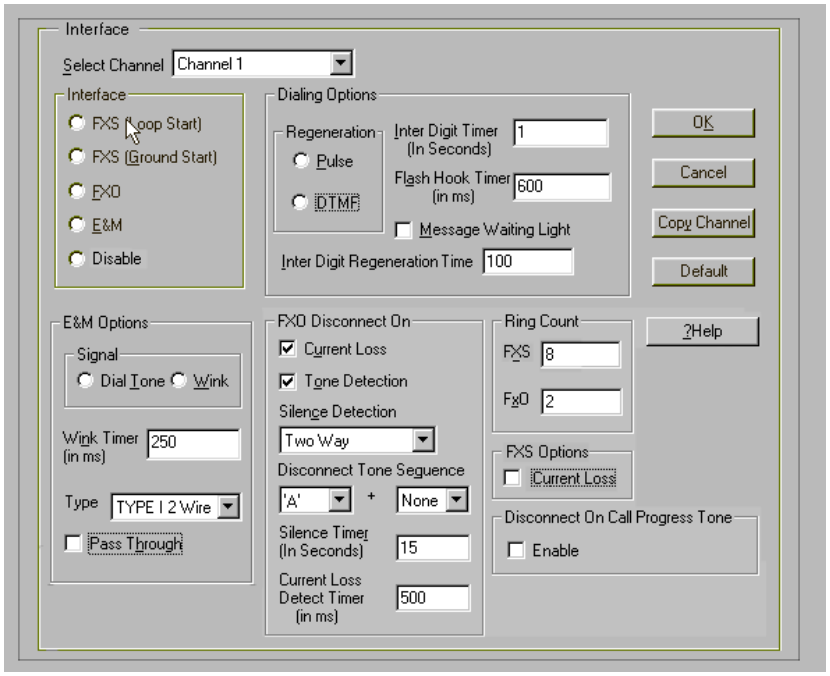
To place modem traffic onto the voip network (an application called "modem relay"), use Coder G.711 mu-law at 64kbps.

Voice/Fax Parameter Definitions (cont'd))		
Field Name	**Values**	**Description**
Auto Disconnect		
Automatic Disconnect-ion	--	The **Automatic Disconnection** group provides four options which can be used singly or in any combination.
Jitter Value	1-65535 milli-seconds	The **Jitter Value** defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. The default is **300** milliseconds. A higher value means voice transmission will be more accepting of jitter. A lower value is less tolerant of jitter. Inactive by default. When active, default = 300 ms. However, value must equal or exceed Dynamic Minimum Jitter Value.
Call Duration	1-65535 seconds	**Call Duration** defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. Inactive by default. When active, default = **180** sec. This may be too short for most configurations, requiring upward adjustment.
Consecutive Packets Lost	1-65535	**Consecutive Packets Lost** defines the number of consecutive packets that are lost after which the call is automatically disconnected. Inactive by default. When active, default = **30**
Network Discon-nection	1 to 65535 seconds; Default = 30 sec.	Specifies how long to wait before disconnecting the call when IP network connectivity with the remote site has been lost.

9a. (**Analog VOIPs**). **Set Telephony Interface Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

Accessing Telephony Interface Parameters	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + I`	

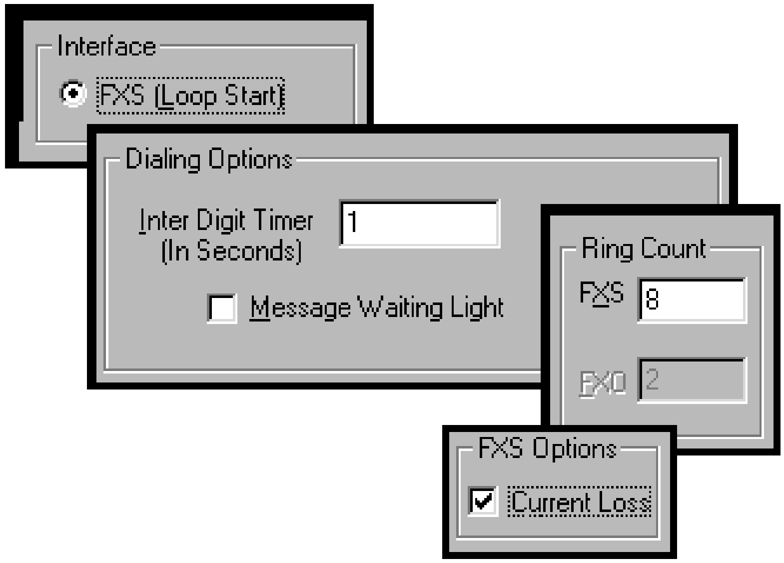
In each field, enter the values that fit your particular network.



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

Interface: Disabled. If the “Disabled” option is selected, the voip channel itself will be disabled, i.e., non-operational.

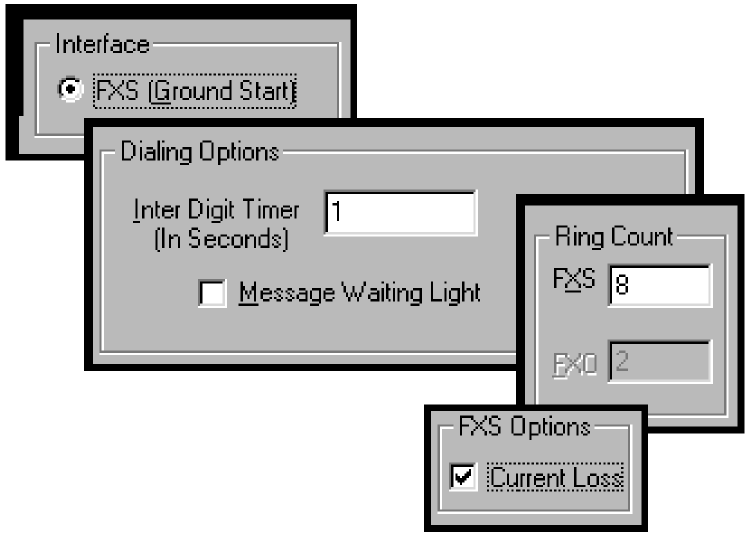
FXS Loop Start Parameters. The parameters applicable to FXS Loop Start are shown in the figure below and described in the table that follows.



FXS Loop Start Interface: Parameter Definitions		
Field Name	**Values**	**Description**
FXS Loop Start	Y/N	Enables FXS Loop Start interface type.
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.

FXS Loop Start Interface: Parameter Definitions		
Field Name	**Values**	**Description**
Message Waiting Light	Y/N	Applicable only when MultiVOIP is used with Avaya Magix PBX units equipped with Merlin Messaging Centralized mail. When enabled, the **Message Waiting Light** feature allows the PBX to send mode-codes and message-waiting indications to another Avaya Magix PBX, which in turn will turn on the message waiting light on a phone station. It also allows Direct Inward Dialing, such that no additional dial tone is needed on voip call.
Ring Count, FXS	integer values	Maximum number of rings that the MultiVOIP will issue before giving up the attempted call.
FXS Options, Current Loss	Y/N	When enabled, the MultiVOIP will interrupt loop current in the FXS circuit to initiate a disconnection. This tells the device connected to the FXS port to hang up. The Multi-VOIP cannot drop the call; the FXS device must go on hook.

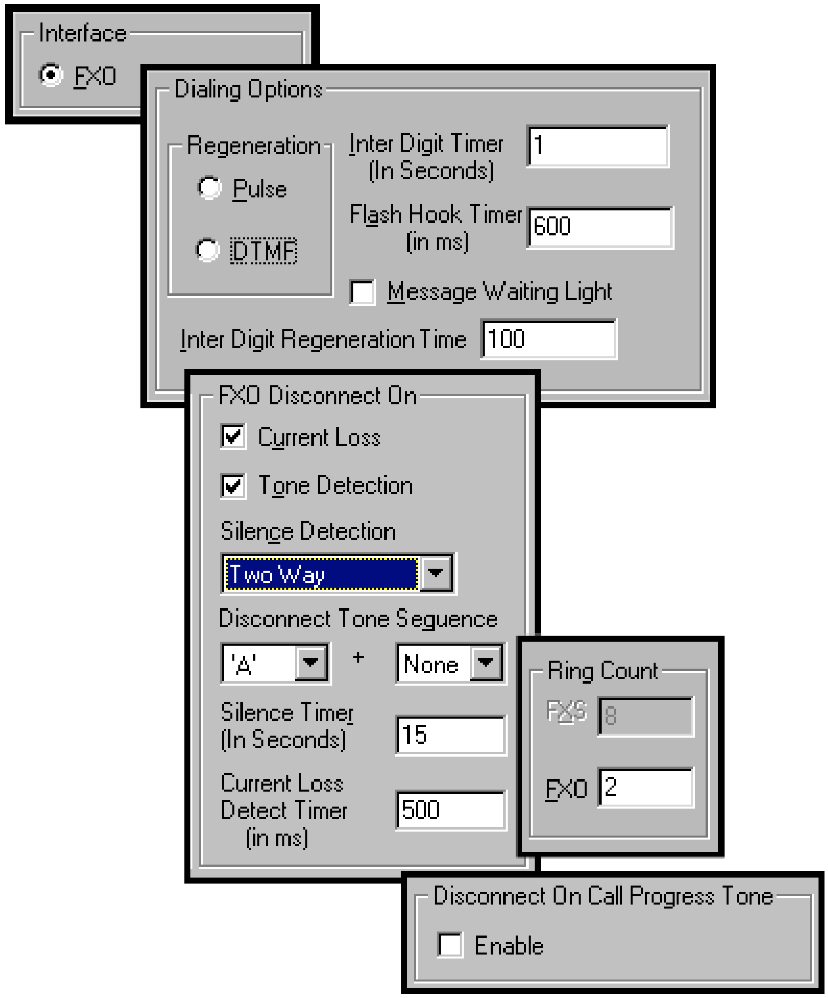
FXS Ground Start Parameters (not supported). The parameters applicable to FXS Ground Start are shown in the figure below and described in the table that follows.



FXS Ground Start Interface: Parameter Definitions		
Field Name	**Values**	**Description**
FXS Ground Start	Y/N	Enables FXS Loop Start interface type.
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.

FXS Ground Start Interface: Parameter Definitions (continued)		
Field Name	**Values**	**Description**
Message Waiting Light	Y/N	Applicable only when MultiVOIP is used with Avaya Magix PBX units equipped with Merlin Messaging Centralized mail. When enabled, the **Message Waiting Light** feature allows the PBX to send mode-codes and message-waiting indications to another Avaya Magix PBX, which in turn will turn on the message waiting light on a phone station. It also allows Direct Inward Dialing, such that no additional dial tone is needed on voip call.
Ring Count, FXS	integer values	Maximum number of rings that the MultiVOIP will issue before giving up the attempted call.
FXS Options, Current Loss	Y/N	When enabled, the MultiVOIP will interrupt loop current in the FXS circuit to initiate a disconnection. This tells the device connected to the FXS port to hang up. The Multi-VOIP cannot drop the call; the FXS device must go on hook.

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



FXO Interface: Parameter Definitions		
Field Name	**Values**	**Description**
Interface, FXO	Y/N	Enables FXO functionality
Dialing Options		
Regeneration	Pulse, DTMF	Determines whether digits generated and sent out will be pulse tones or DTMF.
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.
Flash Hook Timer	integer values, in 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.
Message Waiting Light	Y/N	Applicable only when MultiVOIP is used with Avaya Magix PBX units equipped with Merlin Messaging Centralized mail. When enabled, the **Message Waiting Light** feature allows the PBX to send mode-codes and message-waiting indications to another Avaya Magix PBX, which in turn will turn on the message waiting light on a phone station. It also allows Direct Inward Dialing, such that no additional dial tone is needed on voip call.

FXO Interface: Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dialing Options (cont'd)		
Inter Digit Regeneration Time	milliseconds	The length of time between the outputting of DTMF digits. Default = 100 ms.
FXO Disconnect On		There are three possible criteria for disconnection under FXO: current loss, tone detection, and silence detection. Disconnection can be triggered by more than one of the three criteria.
Current Loss	Y/N	Disconnection to be triggered by loss of current. That is, when Current Loss is enabled ("Y"), the MultiVOIP will hang up the call when it detects a loss of current initiated by the attached device.
FXO Current Detect Timer	integer values (in milliseconds)	The minimum time required for detecting the current loss signal on the FXO interface. In other words, this is the minimum length of time the current must be absent to validate 'current loss' as a disconnection criterion. Default = 500 ms.
Tone Detection	Y/N	Disconnection to be triggered by a tone sequence.

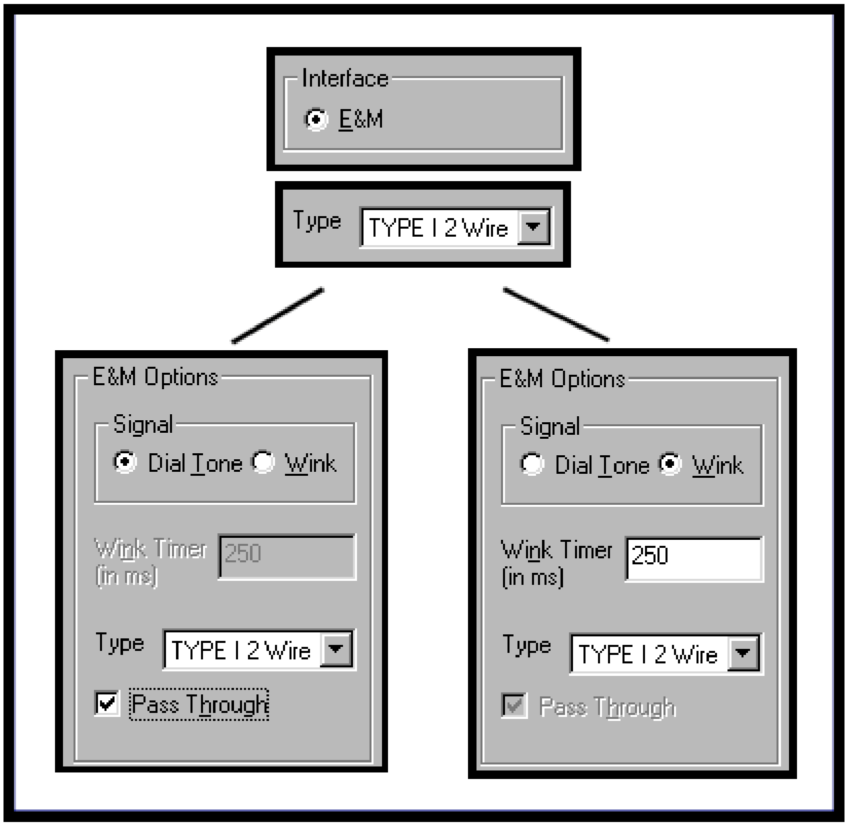
FXO Interface: Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
FXO Disconnect On (cont'd)		
Disconnect Tone Sequence	1st tone pair + 2nd tone pair	These are DTMF tone pairs. Values for first tone pair are: *, #, 0, 1-9, and A-D. Values for second tone pair are: none, 0, 1-9, A-D, *, 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.

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	

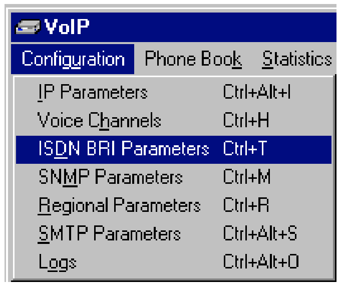
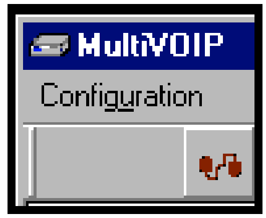
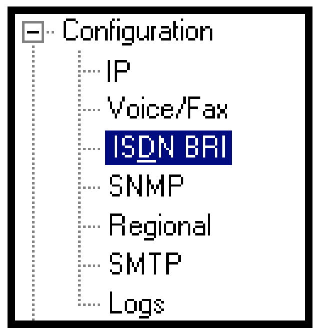
Silence Detection	One-Way or Two-Way	Disconnection to be triggered by silence in one direction only or in both directions simultaneously.
Silence Timer in seconds	integer value	Duration of silence required to trigger disconnection.
Disconnect on Call Progress Tone	Y/N	Allows call on FXO port to be disconnected when a PBX issues a call-progress tone denoting that the phone station on the PBX that has been involved in the call has been hung up.
Ring Count, FXO	integer value	Number of rings required before the MultiVOIP answers the incoming call.

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

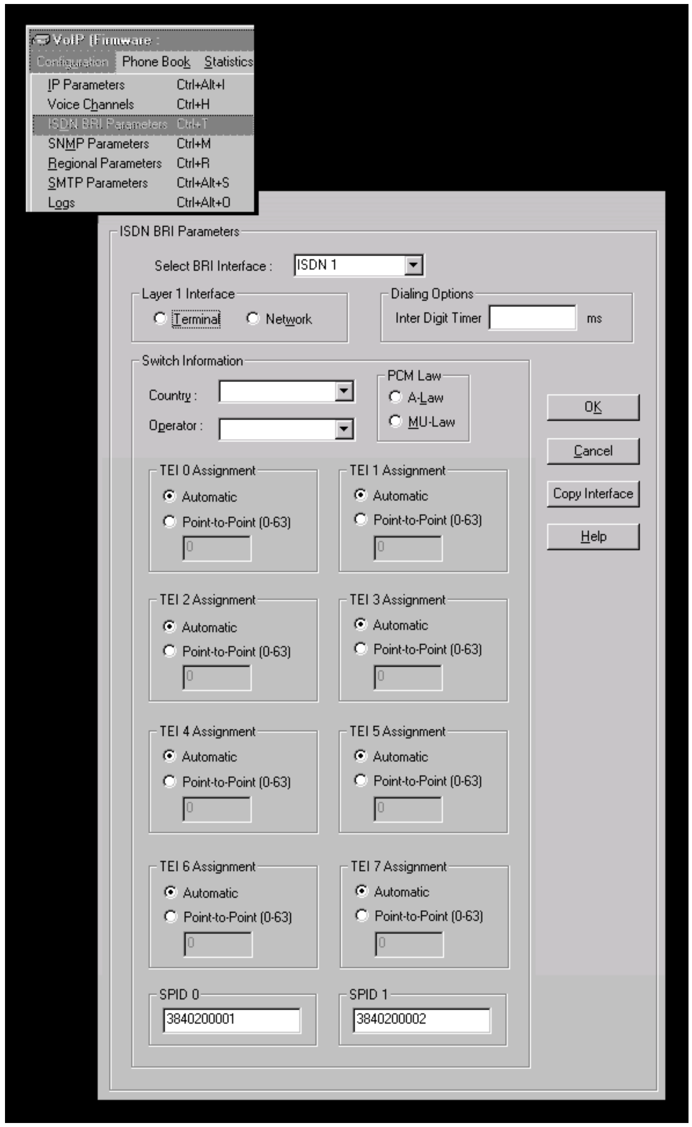


E&M Interface Parameter Definitions		
Field Name	**Values**	**Description**
Interface	E&M	enables E&M functionality
Type	Types 1-5. Each type can be 2-wire or 4-wire.	Refers to the type of E&M interface being used.
Signal	Dial Tone or Wink	When Dial Tone is selected, no wink is required on the E lead or M lead in the call initiation or setup. When Wink is selected, a wink is required during call setup.
Wink Timer (in ms)	integer values, in milliseconds	This is the length of the wink for wink signaling. Applicable only when **Signal** parameter is set to "Wink."
Pass Through	Y/N	When enabled ("Y"), this feature is used to create an open audio path for 2- or 4-wire. The E&M leads are passed through the voip transparently. Applicable only for E&M Signaling with Dial Tone.

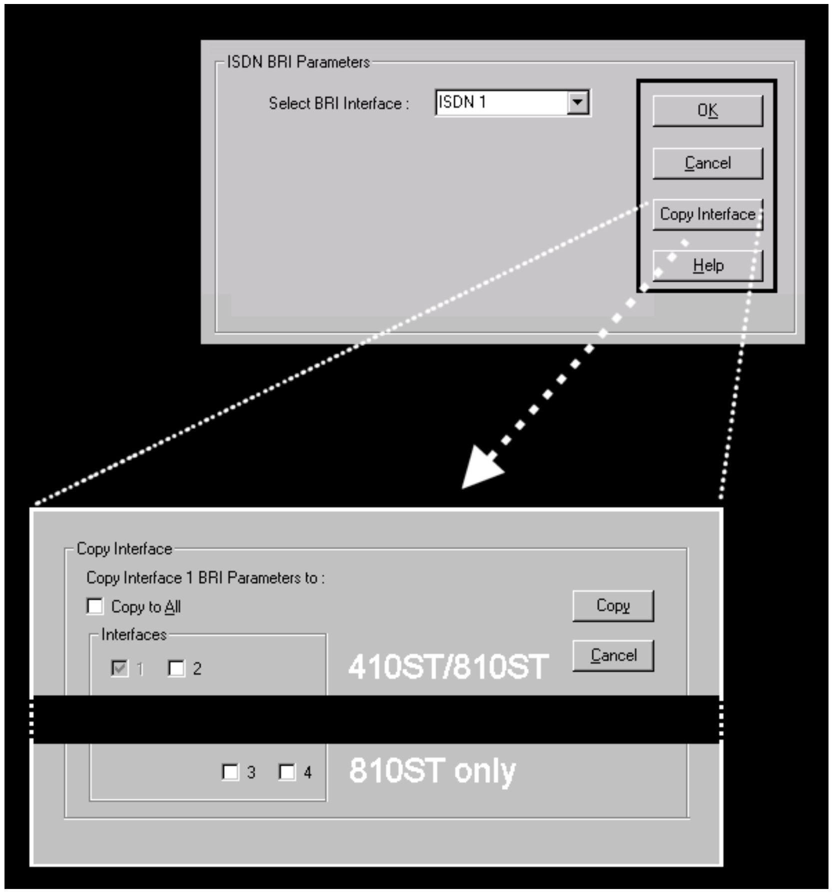
9b. (**for ISDN-BRI MultiVOIP units**). **Set ISDN Parameters**. This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing ISDN (BRI) Parameters</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td>
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + T</code></td><td>
</td></tr>
</table>

In the **ISDN BRI Parameters** screen, select one of the BRI interfaces and configure it for the particular implementation of ISDN that you will use. Configure each BRI interface per the requirements of your voip system. The MVP410ST has two ISDN-BRI interfaces and four channels; the MVP810ST has four ISDN-BRI interfaces and eight channels.



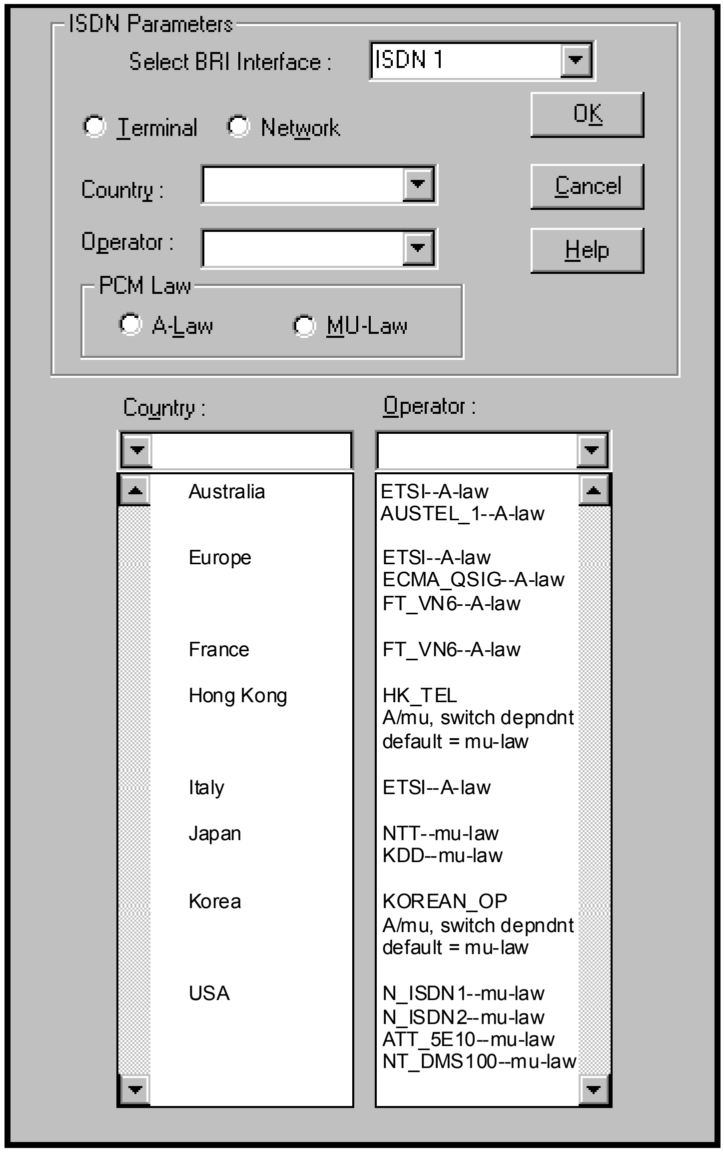
Note that ISDN BRI parameters are applied on an interface-by-interface basis. However, once you have established a set of ISDN BRI parameters for a particular interface, you can apply this entire set of parameters to another interface by using the **Copy Interface** button and its dialog box. To copy a set of ISDN BRI parameters to all interfaces, select "Copy to All" and click **Copy**.



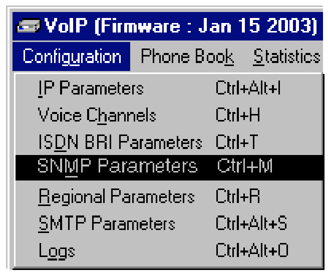
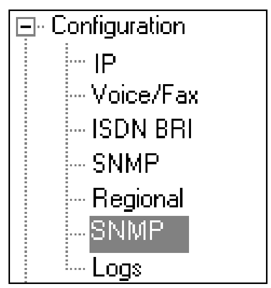
ISDN-BRI Parameter Definitions		
Field Name	**Values**	**Description**
Select BRI Interface	ISDN*n* for n= 1-2 (410ST) for n=1-4 (810ST)	In this field, you will choose which ISDN port you are configuring. The 410ST has two ISDN –BRI ports (or "interfaces"); the 810ST has four ISDN-BRI ports (or "interfaces"). Each port has two channels.
Layer 1 Interface	either "Terminal" or "Network"	When "Terminal" is selected, it indicates that the MultiVOIP should emulate the subscriber (terminal) side of the digital connection. When "Network" is selected, it indicates that the MultiVOIP should emulate the central office (network) side of the digital connection. If connecting to a telco or PBX then choose "Terminal." If connecting to an ISDN phone or terminal adapter, then choose "Network." Default = Terminal.
Dialing Options	Inter Digit Timer (value in milliseconds)	Dialing options are relevant when the MultiVOIP provides dial tone either during an overlap receiving mode or providing a second dial tone. Default is 2000, which is 2 seconds. Range 250 ms to 10000 ms (1/4 sec to 10 sec).
Switch Information		
Country	see table below	Country in which MultiVOIP is operating with ISDN.
Operator	see table below	Indicates phone switch manufacturer/model or refers to telco so as to specify the switching system in question. ISDN is implemented somewhat differently in different switches (different software stacks are used).

ISDN-BRI Parameter Definitions (continued)		
Field Name	**Values**	**Description**
Switch Information		
PCM Law	a-law or mu-law	"A-law" is an analog-to-digital compression/expansion standard used in Europe. "Mu-law" is the North American standard. See the table below of PCM-Law defaults based on country and operator.
TEI *n* Assignment (for n= 0-7)	Automatic or Point-to-Point	
SPID 0	numeric, 3 to 20 digits	
SPID 1	numeric, 3 to 20 digits	
"Copy Interface" button		Copies the ISDN-BRI attributes of one interface to another interface. Attributes can be copied to multiple interfaces or to all interfaces at once.

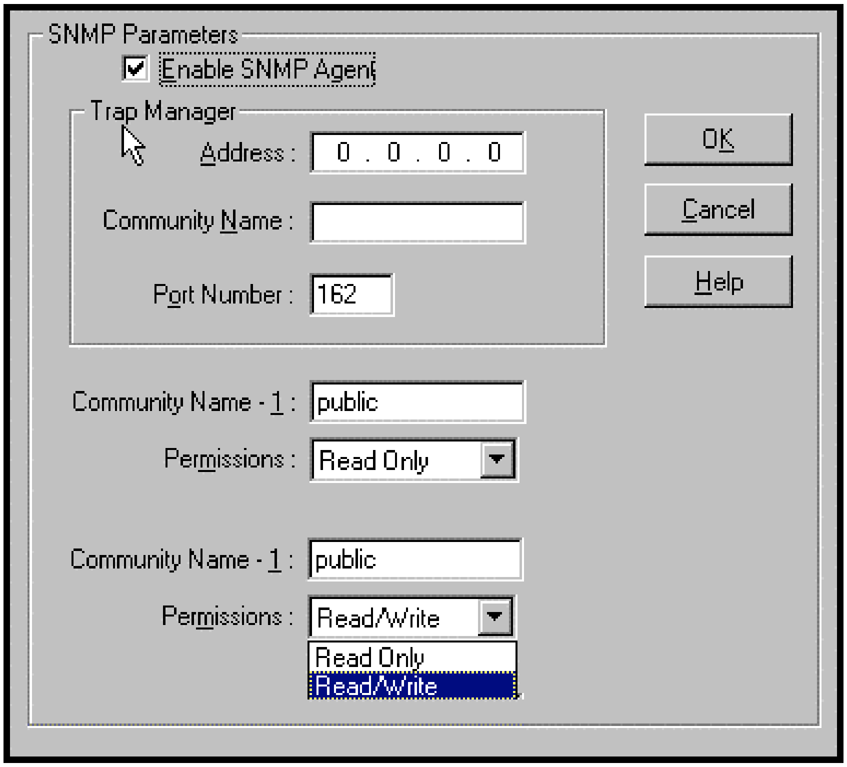
Country and Operator options for the MVP-410ST/810ST voip units are listed below.



10. **Set SNMP Parameters** (Remote Voip Management). This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar. To make the MultiVOIP controllable by a remote PC running the MultiVoipManager software, check the "Enable SNMP Agent" box on the **SNMP Parameters** screen.

<table>
<tr><th colspan="2">Accessing "SNMP Parameters"</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + M</td><td>
</td></tr>
</table>

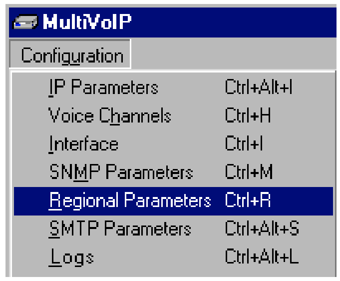
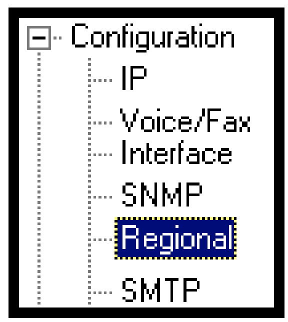
In each field, enter the values that fit your particular system.



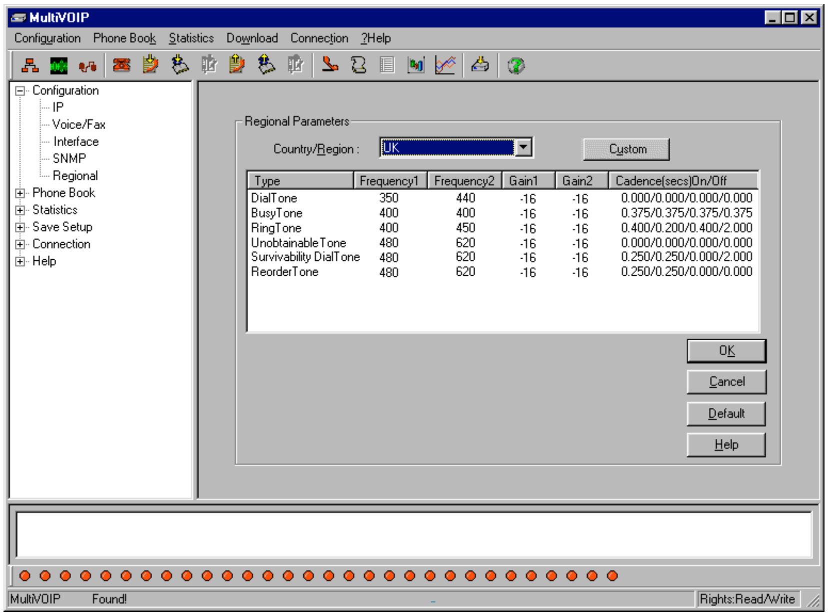
The SNMP Parameter fields are described in the table below.

SNMP Parameter Definitions		
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 MultiVoipManager software. Default: disabled
Trap Manager Parameters		
Address	4 places; n.n.n.n n = 0-255	IP address of MultiVoipManager PC.
Community Name	--	A “community” is a group of VOIP endpoints that can communicate with each other. Often “public” is used to designate a grouping where all end users have access to entire VOIP network. However, 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.

11. **Set Regional Parameters** (Phone Signaling Tones & Cadences).).
This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “Regional Parameters”</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + R</code></td><td>
</td></tr>
</table>

The **Regional Parameters** screen will appear. For the country selected, the standard set of frequency pairs will be listed for dial tone, busy tone, 'unobtainable' tone (fast busy or trunk busy), and ring tone.



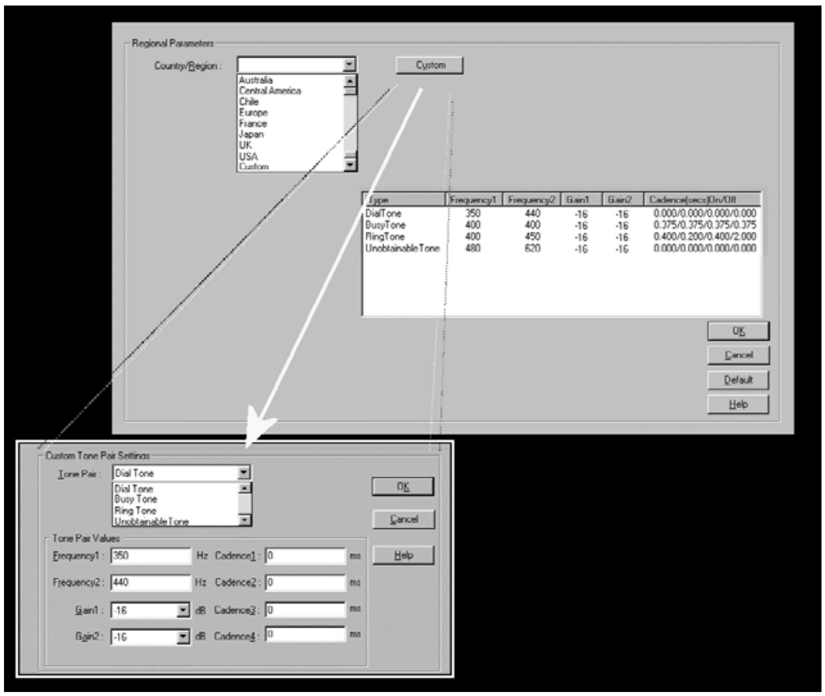
In each field, enter the values that fit your particular system.

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

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

“Regional Parameter” Definitions (cont’d)		
Field Name	**Values**	**Description**
Cadence (msec) On/Off	n/n/n/n four integer time values in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable (fast busy), dial tone (“0” indicates continuous tone), survivability, and re-order. Default values differ for different countries/regions. Although most cadences have only two parts (an “on” duration and an “off” duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.
Custom (button)	--	Click on the “Custom” button to bring up the **Custom Tone Pair Settings** screen. (The “Custom” button is active only when “Custom” is selected in the **Country/Region** field.) This screen allows the user to specify tone pair attributes that are not found in any of the standard national/regional telephony toning schemes.

12. **Set Custom Tones and Cadences** (optional). The **Regional Parameters** dialog box has a secondary dialog box that allows you to customize DTMF tone pairs to create unique ring-tonesdial-tones, busy-tones or “unobtainable” tones (fast busy signal) or “re-order” tones (telling the user that she must hang up an off-hook phone) or “survivability” tones (an indication of call-routing redundancy in MultiVantage systems only) 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.)

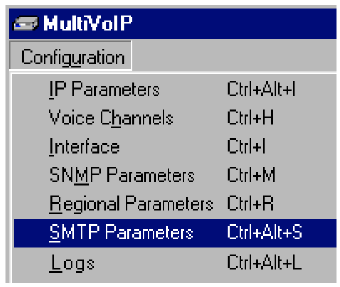
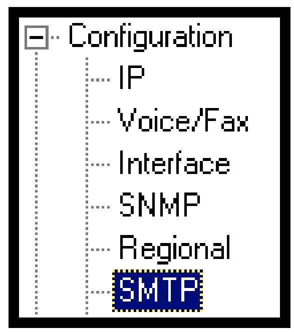


The **Custom Tone-Pair Settings** fields are described in the table below.

Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Tone Pair	dial tone, busy tone, ring tone, 'unobtainable' tone, survivability tone, re-order tone	Identifies the type of telephony signaling tone for which frequencies are being specified.
TONE PAIR VALUES		**About Defaults**: US telephony values are used as defaults on this screen. However, since this dialog box is provided to allow custom tone-pair settings, default values are essentially irrelevant.
Frequency 1	frequency in Hertz	Frequency of lower tone of pair. This outbound tone pair enters the MultiVOIP at the input port.
Frequency 2	frequency in Hertz	Frequency of higher tone of pair. This outbound tone pair enters the MultiVOIP at the input port.
Gain 1	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the input port. **Default = -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the input port. **Default = -16dB**

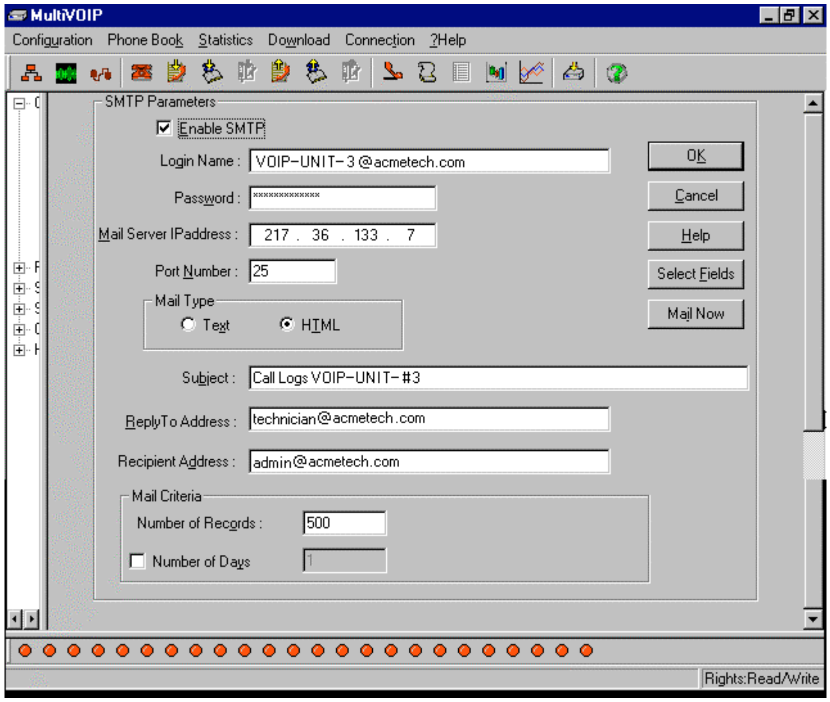
Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Cadence 1	integer time value in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, dial tone (“0” indicates continuous tone) survivability and re-order. Cadence 1 is duration of first period of tone being “on” in the cadence of the telephony signal (which could be ring-tone, busy-tone, unobtainable-tone, or dial tone).
Cadence 2	duration in milliseconds	Cadence 2 is duration of first “off” period in signaling cadence.
Cadence 3	duration in milliseconds	Cadence 3 is duration of second “on” period in signaling cadence.
Cadence 4	duration in milliseconds	Cadence 4 is duration of second “off” period in the signaling cadence, after which the 4-part cadence pattern of the telephony signal repeats.

13. **Set 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.). The **SMTP Parameters** screen can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing "SMTP Parameters"</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + Alt + S</code></td><td>
</td></tr>
</table>

MultiVOIP as Email Sender. When SMTP is used, the MultiVOIP will actually be given its own email account (with Login Name and Password) on some mail server connected to the IP network. Using this account, the MultiVOIP will then send out email messages containing log report information. The "Recipient" of the log report email is ordinarily the VoIP administrator. Because the MultiVOIP cannot receive email, a "Reply-To" address must also be set up. Ordinarily, the "Reply-To" address is that of 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 main function of the Reply-To address is to receive error or failure messages regarding the emailed reports.

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

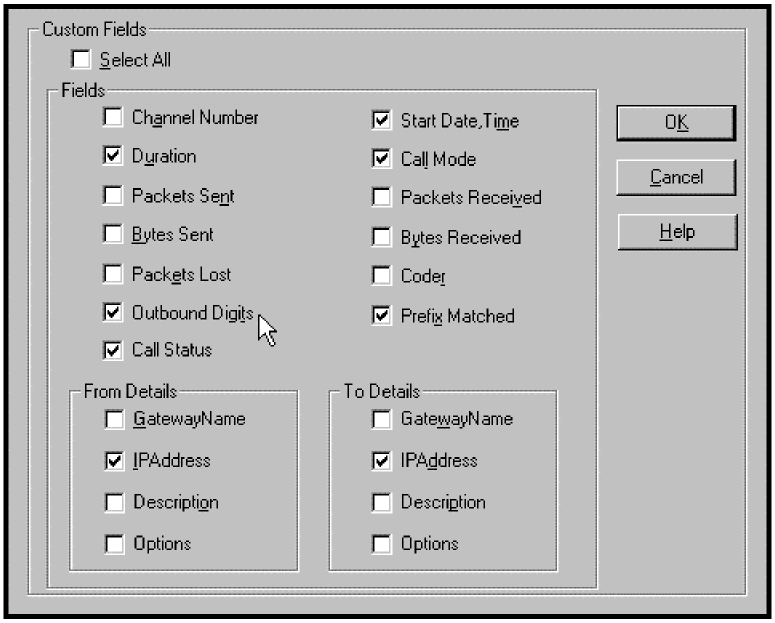


"SMTP Parameters" Definitions		
Field Name	**Values**	**Description**
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select "SMTP" in the **Logs** screen.
Login Name	alpha-numeric, per email domain	This is the User 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 for n= 0 to 255	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.

......

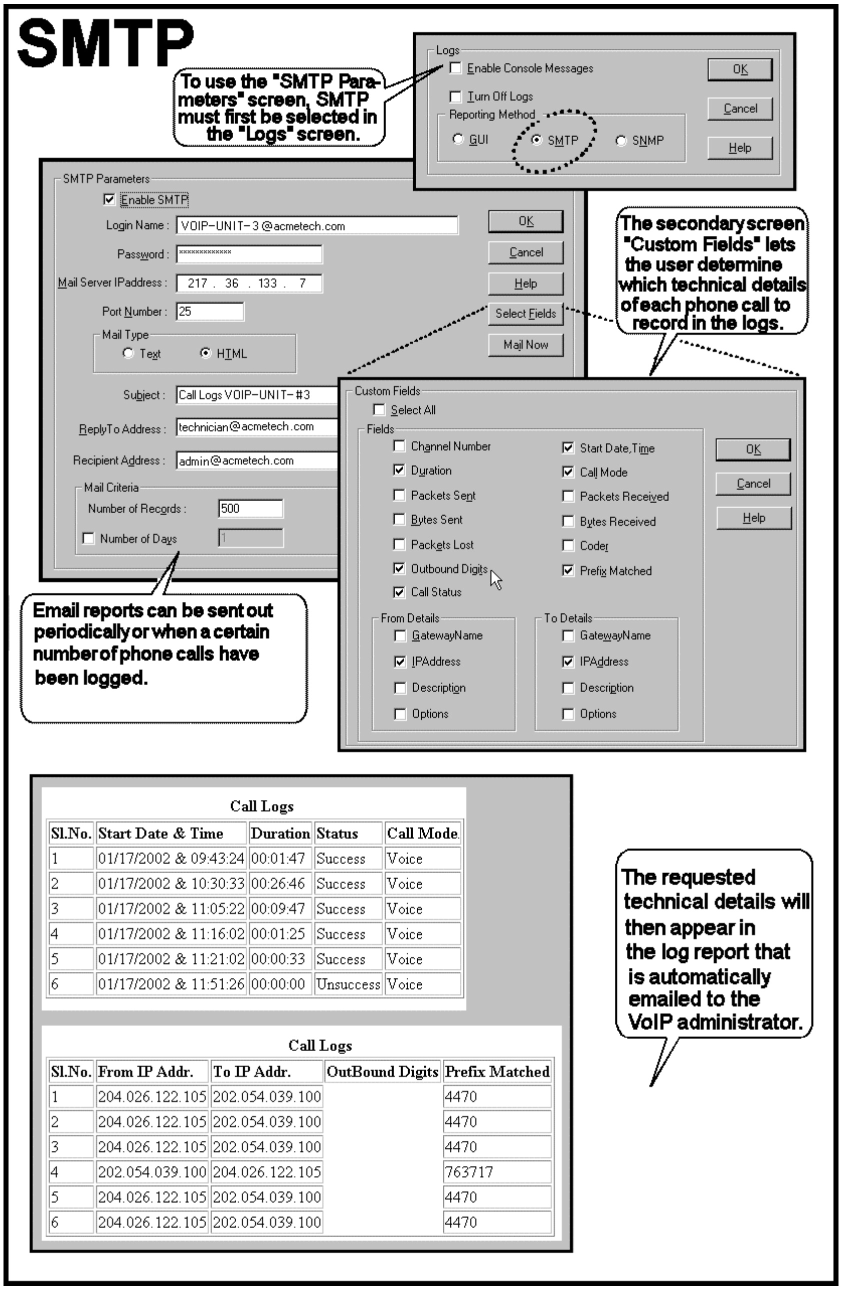
“SMTP Parameters” Definitions (cont’d)		
Field Name	**Values**	**Description**
Mail Type	text or html	Mail type in which log reports will be sent.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP unit.
Reply-To Address	email address	User specified. 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	User specified. Email address at which VOIP administrator will receive log reports.
Mail Criteria		Criteria for sending log summary by email. The log summary email will be sent out either when the user-specified number of log messages has accumulated, or once every day or multiple days, *which ever comes first*.
Number of Records	integer	This is the number of log records that must accumulate to trigger the sending of a log-summary email.
Number of Days	integer	This is 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, **Custom Fields**, that allows you to customize email log messages for the MultiVOIP. 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 Fields” Definitions			
Field	**Description**	**Field**	**Description**
Select All	Log report to include all fields shown.		
Channel Number	Data channel carrying call.	Start Date, Time	Date and time the phone call began.
Duration	Length of call.	Call Mode	Voice or fax.
Packets Sent	Total packets sent in call.	Packets Received	Total packets received in call.
Bytes Sent	Total bytes sent in call.	Bytes Received	Total bytes received in call.
Packets Lost	Packets lost in call.	Coder	Voice Coder /Compression Rate used for call will be listed in log.

"Custom Fields" Definitions (cont'd)			
Field	**Description**	**Field**	**Description**
Outbound Digits	Digits put out by MultiVOIP onto the phone line.	Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Status	Successful or unsuccessful.		
From Details		**To Details**	
Gateway Number	Originating gateway	Gatew N.	Completing or answering gateway
IP Addr	IP address where call originated.	IP Addr	IP address where call was completed or answered.
Descript	Identifier of site where call originated.	Descript	Identifier of site where call was completed or answered.
Options	When selected, log will not use/non-use of Silence Compression and Forward Error Correction by call originator.	Options	When selected, log will not use/non-use of Silence Compression and Forward Error Correction by party answering call.



Call Logs

Sl.No.	Start Date & Time	Duration	Status	Call Mode
1	01/17/2002 & 09:43:24	00:01:47	Success	Voice
2	01/17/2002 & 10:30:33	00:26:46	Success	Voice
3	01/17/2002 & 11:05:22	00:09:47	Success	Voice
4	01/17/2002 & 11:16:02	00:01:25	Success	Voice
5	01/17/2002 & 11:21:02	00:00:33	Success	Voice
6	01/17/2002 & 11:51:26	00:00:00	Unsuccess	Voice

Call Logs

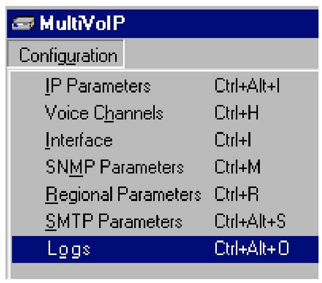
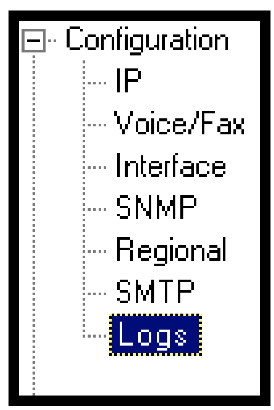
Sl.No.	From IP Addr.	To IP Addr.	OutBound Digits	Prefix Matched
1	204.026.122.105	202.054.039.100		4470
2	204.026.122.105	202.054.039.100		4470
3	204.026.122.105	202.054.039.100		4470
4	202.054.039.100	204.026.122.105		763717
5	204.026.122.105	202.054.039.100		4470
6	204.026.122.105	202.054.039.100		4470

14. **Set Log Reporting Method**. The **Logs** 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:

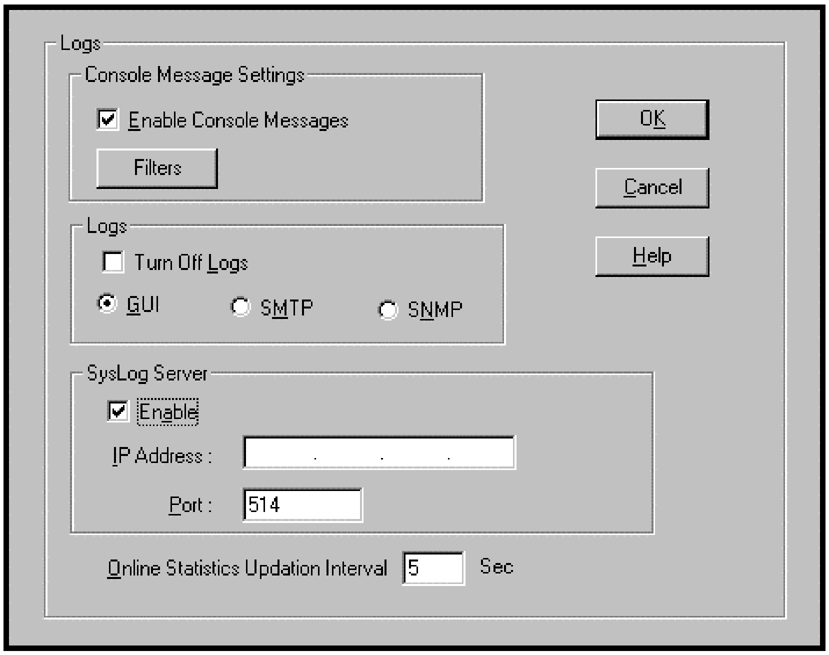
A. in the MultiVOIP program (GUI),

B. via email (SMTP), or

C. at the MultiVoipManager remote voip system management program (SNMP).

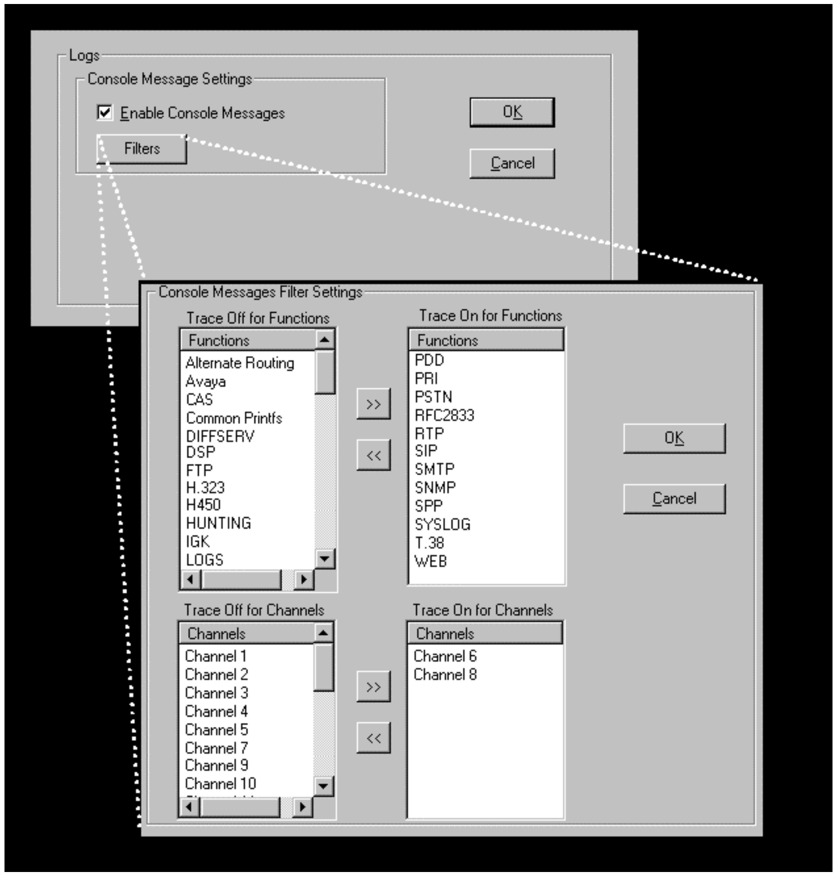
Accessing "Logs" Screen	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
Ctrl + Alt + O	

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 (see subsequent page). 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 GUI for configuration and control of MultiVOIP units, be aware that the web browser GUI does not support logs directly. However, when the web browser GUI is used, log files can still be sent to the voip administrator via email (which requires activating the SMTP logging option in this screen).

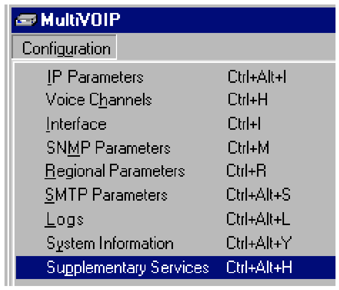
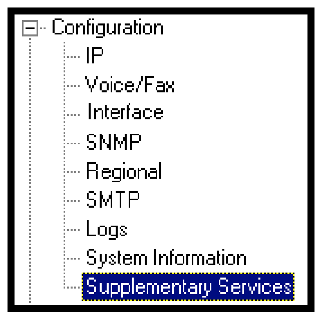


"Logs" Screen Definitions		
Field Name	**Values**	**Description**
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic terminal program like HyperTerminal ™ or equivalent. Normally, this should be disabled because it uses MultiVOIP processing resources. Console messages are meant for tech 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. (See the Console Messages Filter Settings screen on subsequent page.) Not supported in BRI 5.02c software.
Turn Off Logs	Y/N	Check to disable log-reporting function. Not supported in BRI 5.02c software.
Logs Buttons		Only one of these three log reporting methods, GUI, SMTP, or SNMP, may be chosen.
GUI	Y/N	User must view logs at the MultiVOIP configuration program.
SNMP	Y/N	Log messages will be delivered to the MultiVoipManager application program.
SMTP	Y/N	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. For more on SysLog Server, see *Operation & Maintenance* chapter. Not supported in BRI 5.02c software.
IP Address	n.n.n.n for n= 0-255	IP address of computer, connected to voip network, on which SysLog Server program is running. Not supported in BRI 5.02c software.
Port	514	Logical port for SysLog Server. 514 is commonly used. Not supported in BRI 5.02c software.
Online Statistics Updation Interval	integer	Set the interval (in seconds) at which logging information will be updated. Not supported in BRI 5.02c software.

To customize console messages by category and/or by channel, click on "Filters" and use the **Console Messages Filters Settings** screen.

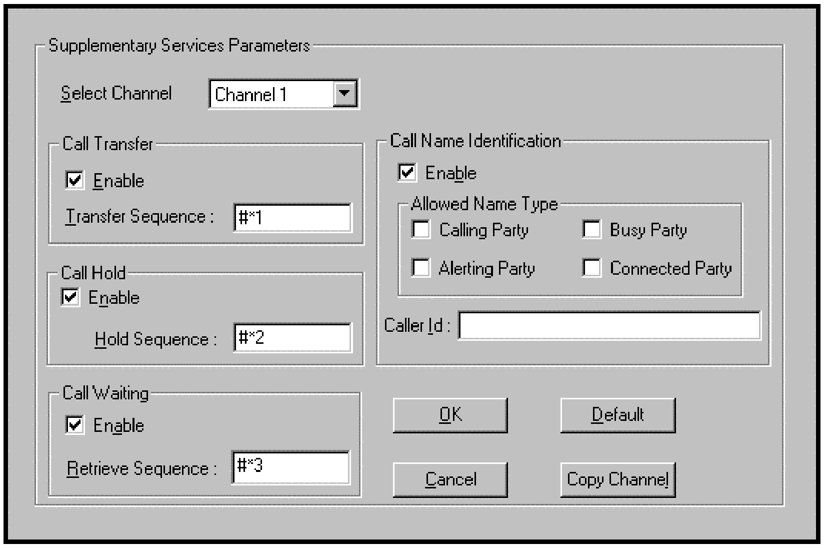


15. **Set Supplementary Services Parameters.** This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar. This screen is not supported in BRI 5.02c software.

Accessing "Supplementary Services" Parameters	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
Ctrl + Alt +H	

Supplementary Services features derive from the H.450 standard, which brings to voip telephony functionality once only available with PSTN or PBX telephony. Supplementary Services features can be used under H.323 only and *not* under SIP.

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



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**.

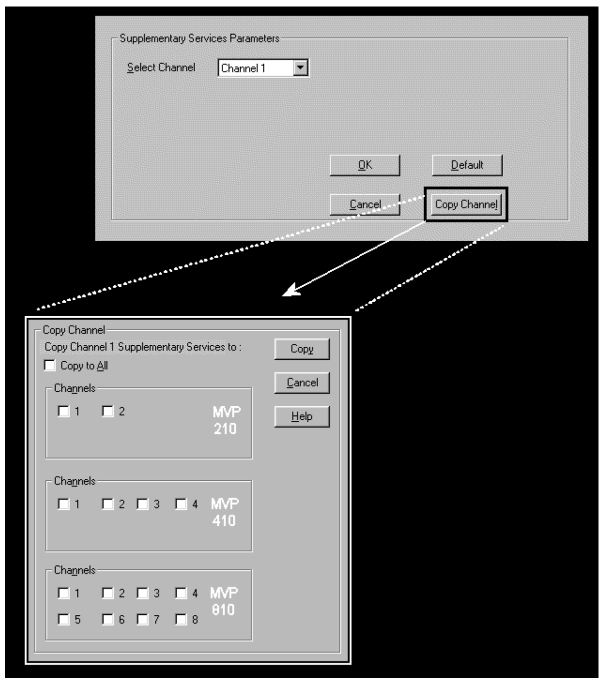
Call Transfer. Call Transfer allows one party to re-connect the party with whom they have been speaking to a third party. The first party is disconnected when the third party becomes connected. Feature is invoked by a programmable phone keypad sequence (for example, #7).

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. Invoked by keypad sequence.

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. Invoked by keypad sequence.

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 (callee) 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.

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



The **Supplementary Services** fields are described in the tables below.

Supplementary Services Parameter Definitions (Not supported in BRI 5.02c software.)		
Field Name	**Values**	**Description**
Select Channel	1-2 (210); 1-4 (410); 1-8 (810)	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.
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*#.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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 GUI (the 'home voip'), Call Name 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 callee (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.)

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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.

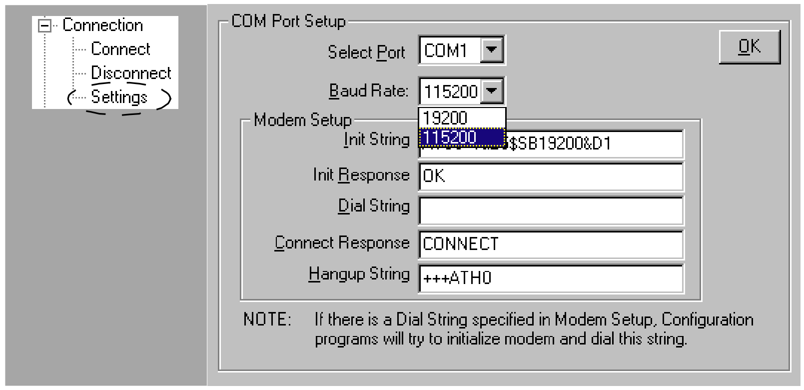
Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Alerting Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving the call and **Alerting Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit 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.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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. 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.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
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.

16. **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.

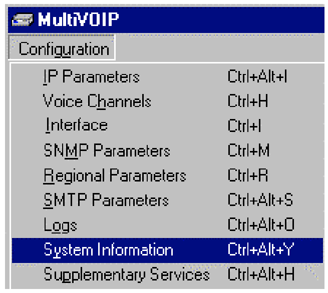
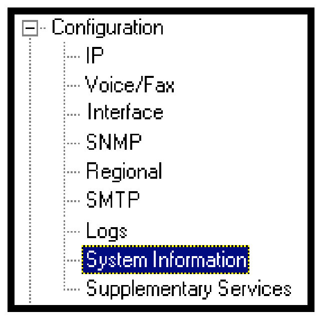


First, it is important to note that the default COM port established by the MultiVOIP program is COM1. ***Do not accept the default value until you have checked the COM port allocation on your PC***. To do this, check for COM port assignments in the system resource dialog box(es) of your Windows operating system. If COM1 is not available, you must change the COM port setting to COM2 or some other COM port that you have confirmed as being available on your PC.

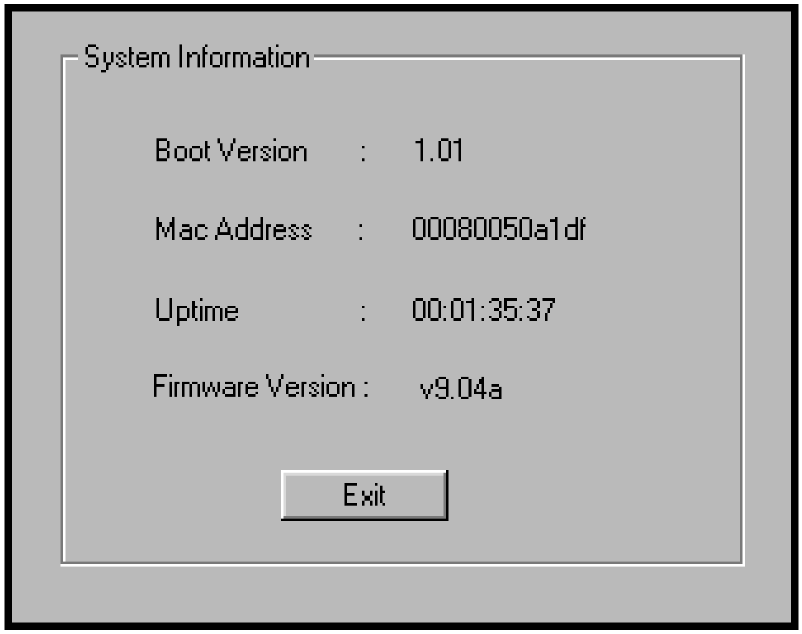
The default baud rate is 115,200 bps.

17. View **System Information** screen and set updating interval (optional). The System Information screen is not supported in BRI 5.02c software.

This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

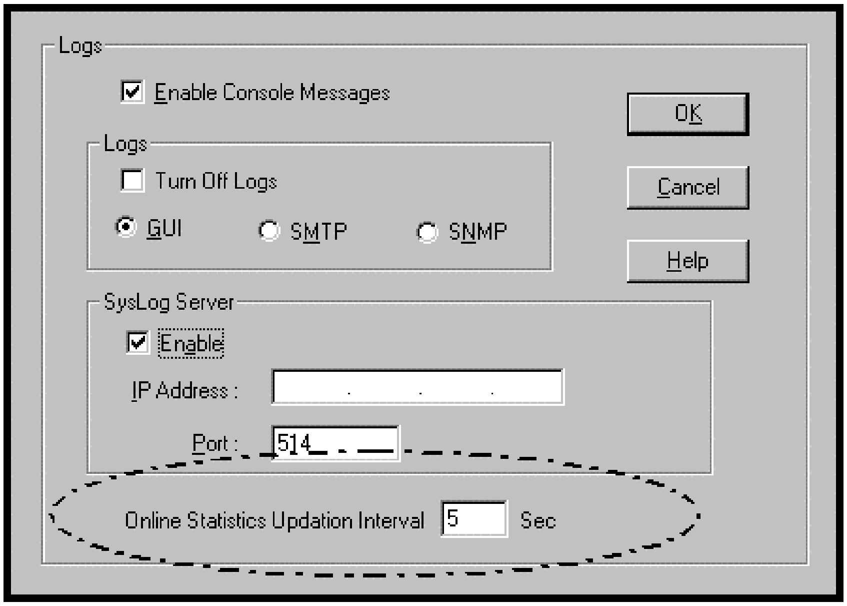
Accessing "System Information" Screen	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
Ctrl + Alt +Y	

This screen presents vital system information at a glance. Its primary use is in troubleshooting.

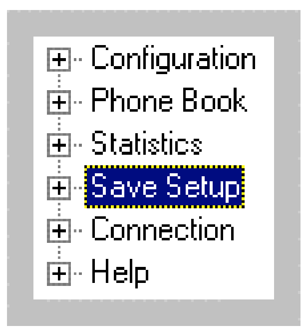


System Information Parameter Definitions		
Field Name	**Values**	**Description**
Boot Code Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Mac Address	alpha-numeric	Denotes the number assigned 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.
Firmware Version	alpha-numeric	Indicates the version of the MultiVOIP firmware.

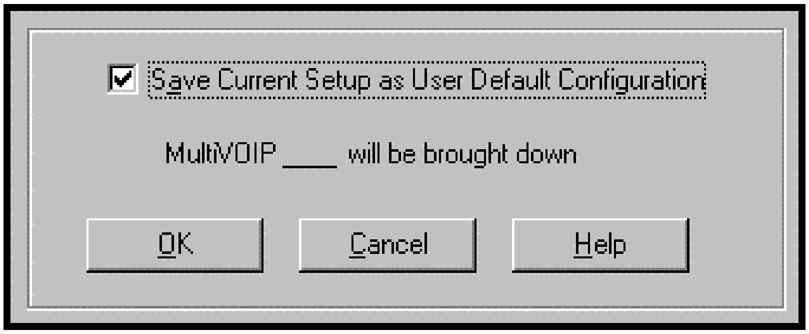
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



18. **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.



19. **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.



Chapter 7: T1 Phonebook Configuration

(North American Telephony Standards)

Configuring the MVP2400/2410 MultiVOIP Phonebooks

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" phonebooks 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.* (Of course, the phone numbers are not literally "listed" individually, but are, instead, described by rule.)

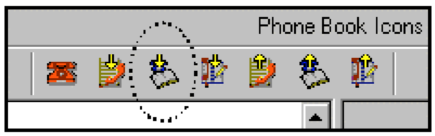
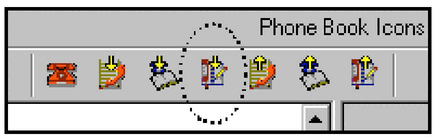
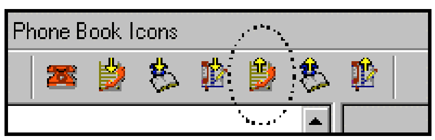
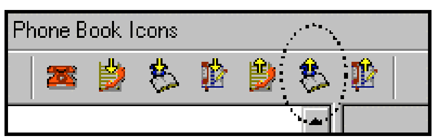
Consider two types of calls in the three-city system described above: (1) calls originating from the Miami office and terminating in the New York (Manhattan) office, and (2) calls originating from the Miami office and terminating in New York City but off the company's premises in an adjacent area code, an area code different than the company's office but still a local call from that office (e.g., Staten Island).

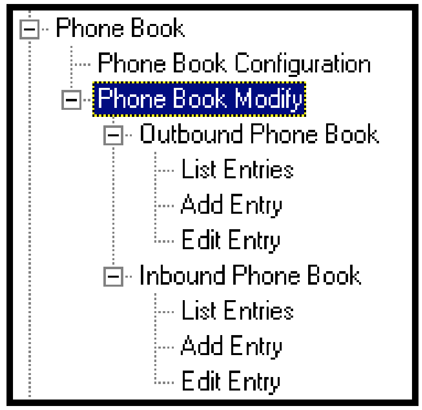
The first type of call requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound phonebook of the New York VOIP. These entries would allow the Miami caller to dial the New York office as if its phones were extensions on the Miami PBX.

The second type of call similarly requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound Phonebook of the New York VOIP. However, these entries will be longer and more complicated. Any Miami call to New York City local numbers will be sent through the VOIP system rather than through the regular toll public phone system (PSTN). But the phonebook entries can be arranged so that the VOIP system is transparent to the Miami user, such that even though that Miami user dials the New York City local number just as they would through the public phone system, that call will still be completed through the VOIP system.

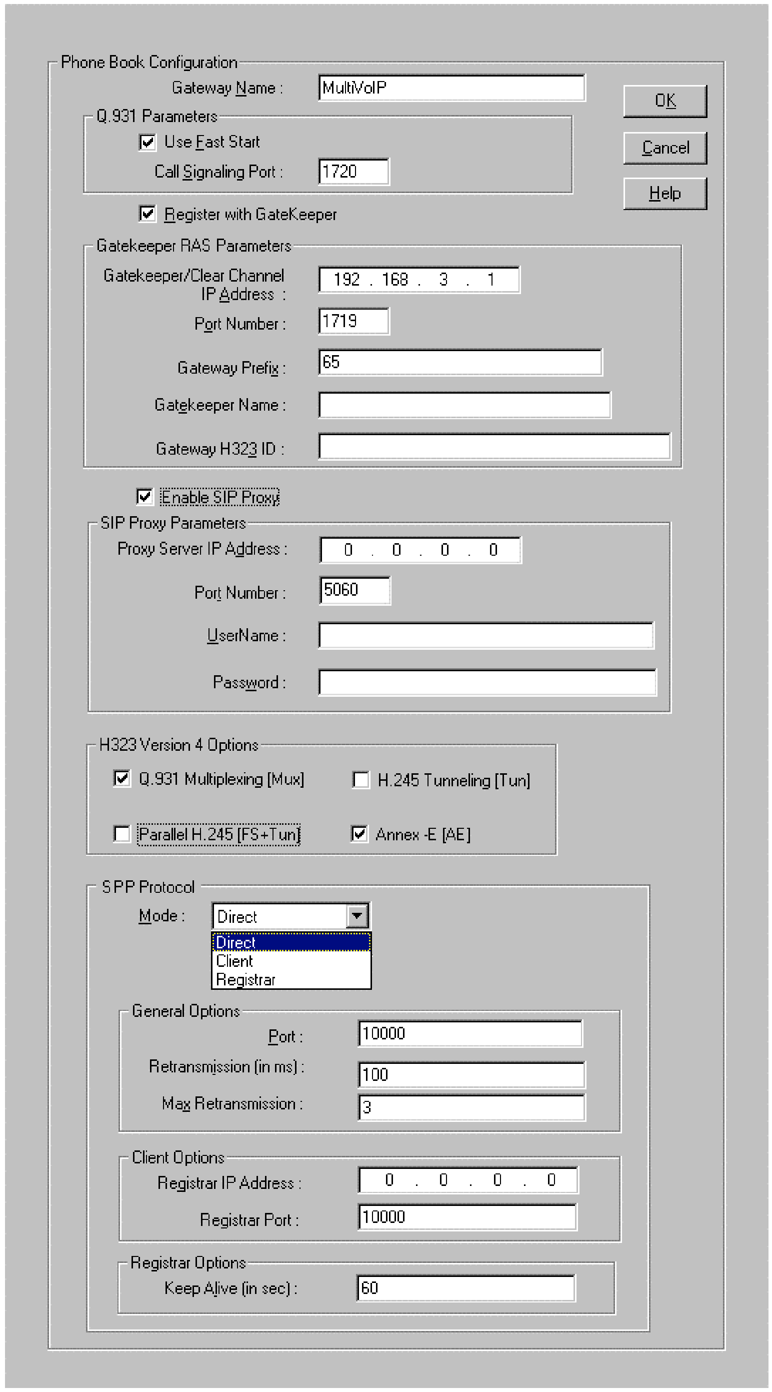
This PhoneBook Configuration procedure is brief, but it is followed by an example case. For many people, the example case may be easier to grasp than the procedure steps. Configuration is not difficult, but all phone number sequences and other information must be entered exactly; otherwise connections will not be made.

Phonebook configuration screens can be accessed using icons or the sidebar menu.

Phonebook Icons	Description
	Phonebook Configuration
	Inbound Phonebook Entries List
	Add Inbound Phonebook Entry
	Edit selected Inbound Phonebook Entry
	Outbound Phonebook Entries List
	Add Outbound Phonebook Entry
	Edit selected Outbound Phonebook Entry

Phonebook Sidebar Menu	
	

1. Go to the **PhoneBook Configuration** screen (using either the sidebar or drop-down menu).



In consultation with your VOIP administrator, enter the Gateway Name and values for Q.931 parameters and Gatekeeper RAS parameters. Determine whether your voip system will operate with a proxy server. Determine which H.323 version 4 functions you will implement. (They are not always applicable. See field description for each parameter.) If the SPP protocol is used, values for another group of parameters must be specified, as well.

The table below describes all fields in the general **PhoneBook Configuration** screen.

PhoneBook Configuration Parameter Definitions		
Field Name	**Values**	**Description**
Gateway Name	Y/N	This field allows you to specify a name for this MultiVOIP. When placing a call, this name is sent to the remote MultiVOIP for display in Call Progress listings, Logs, etc.
Q.931 Parameters		
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.
Call Signaling Port	port number	Default: 1720 (H.323)
GateKeeper RAS Parameters		
Gatekeeper / Clear Channel IP Address		IP address of the GateKeeper.
Port Number		Well-known port number for GateKeepers. Must match port number of GateKeeper, 1719.
Gateway Prefix		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.
Gatekeeper Name	*alpha-numeric string*	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register.
Gateway H.323 ID		The H.323 ID is used to register this particular MultiVOIP with the GateKeeper.

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
SIP Proxy Parameters		
Enable Proxy	Y/N	Allows the MultiVOIP to work in conjunction with a proxy server.
Proxy Server IP Address	n.n.n.n where n=0-255	Network address of the proxy server that the voip is using.
Port Number		Logical port number for proxy communications.
User Name	Values: alphnumeric	Description: Identifier used when proxy server is used in network. If a proxy server is used in a SIP voip network, all clients must enter both a User Name and a Password before being allowed to make a call.
Password	Values: alphanumeric	Description: Password for proxy server function. See "User Name" description above.

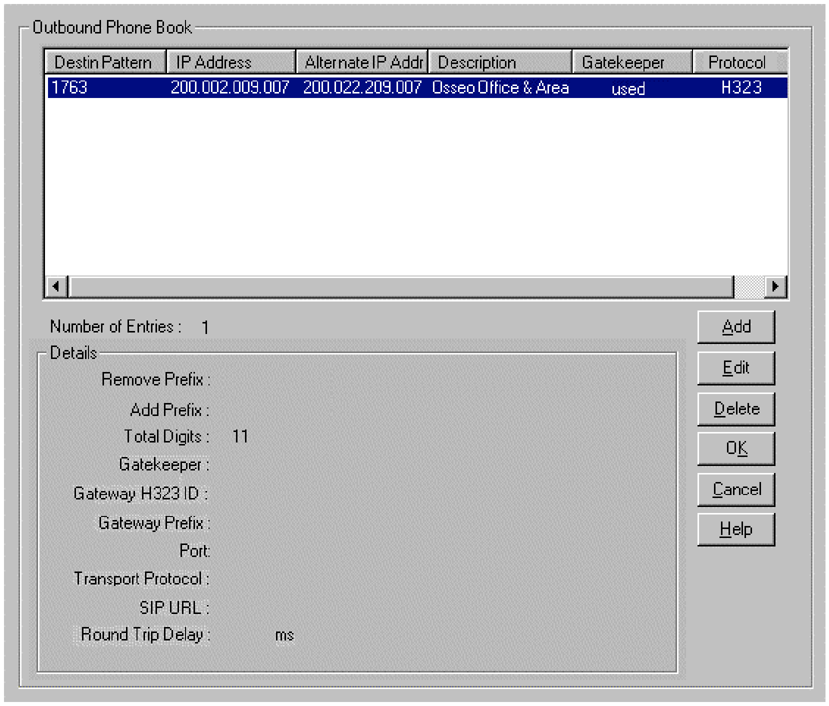
PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
Q.931 Multiplexing (Mux)	Y/N	Signaling for multiple phone calls can be carried on a single port rather than opening a separate signaling port for each call. This conserves bandwidth resources.
H.245 Tunneling (Tun)	Values: Y/N Description: 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.	

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
Parallel H.245 (FS + Tun)	Values: Y/N Description: FS (Fast Start or Fast Connect) is a Q.931 feature of H.323v2 to hasten call setup as well as 'pre-opening' the media channel before the CONNECT message is sent. This pre-opening is a requirement for certain billing activities. Under Parallel H.245 FS + Tun, this Fast Connect feature can operate simultaneously with H.245 Tunneling (see description above).	
Annex –E (AE)	Values: Y/N Description: Multiplexed UDP call signaling transport. Annex E is helpful for high-volume voip system endpoints. Gateways with lesser volume can afford to use TCP to establish calls. However, for larger volume endpoints, the call setup times and system resource usage under TCP can become problematic. Annex E allows endpoints to perform call-signaling functions under the UDP protocol, which involves substantially streamlined overhead. (This feature should not be used on the public Internet because of potential problems with security and bandwidth usage.)	

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP)		
Mode	Direct, Client, or Registrar	SPP voip systems can operate in two modes: in the **direct mode**, where all voip gateways have static IP addresses assigned to them; or in the **registrar/client mode**, where 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		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-trans-mission (in ms)		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 value = 2000 milliseconds.)
Max Re-trans-mission		Number of times the voip will re-transmit a lost packet (if no acknowledgment has been received). (Default value = 3)

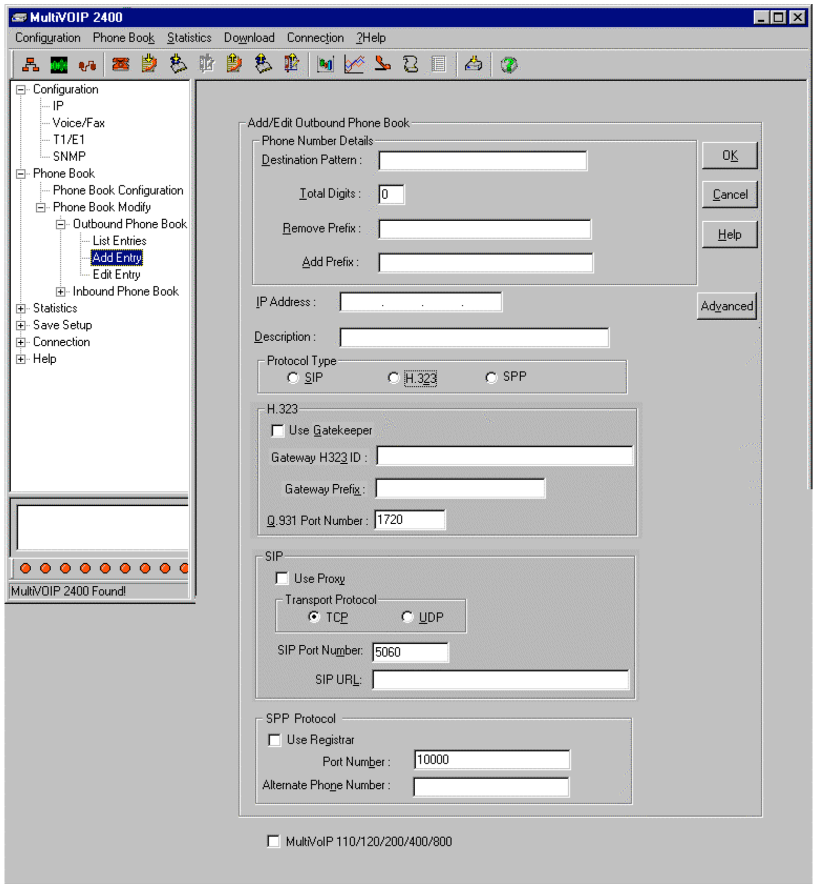
PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP) [continued]		
Client Options		Client Option fields are active only in registrar/client mode and only for client voip units.
Registrar IP Address		This is the IP address of the registrar voip to which this client is assigned. (Default value = 0.0.0.0; effectively, there is no useful default value.)
Registrar Port		This is the port number of the registrar voip to which this client is assigned. (Default port number = 10000.)
Registrar Options		Registrar Option fields are active only in registrar/client mode and only for registrar voip units.
Keep Alive (in sec.)		Time-out duration before a registrar will unregister 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.

2. Select **PhoneBook Modify** and then select **Outbound Phone Book/List Entries**.



Click **Add**.

3. The **Add/Edit Outbound PhoneBook** screen appears.



Enter Outbound PhoneBook data for your MVP2400/2410. 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).

The fields of the **Add/Edit Outbound Phone Book** screen are described in the table below.

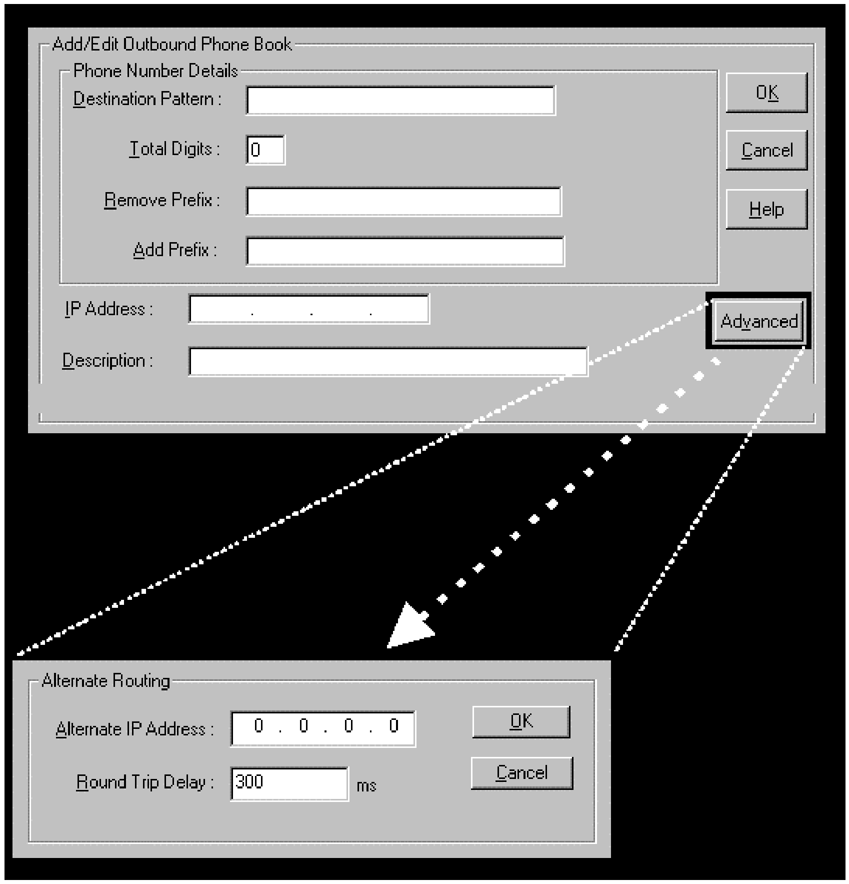
Add/Edit Outbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	number of digits the phone user must dial to reach specified destination
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 for n = 0-255	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.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 fields		
Use Gatekeepr	Y/N	Indicates whether or not gatekeeper is used.
H.323 ID		The H.323 ID assigned to the destination MultiVOIP. Only valid if "Use Gatekeeper" is enabled for this entry.
Gateway Prefix		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.
Q.931 Port Number	1720	Q.931 is the 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, the port number 1720 must be chosen.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
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 RFC3087 ("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 callee 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.

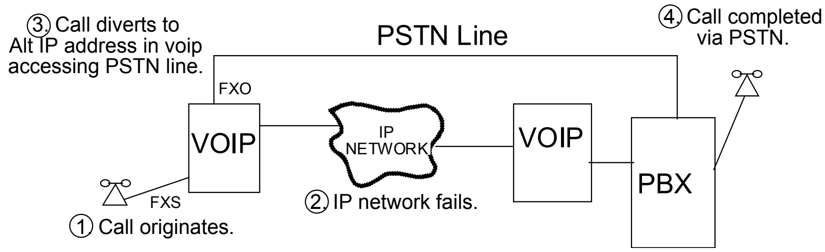
Add/Edit Outbound Phone Book: Field Def'ns (cont'd)		
Field Name	**Values**	**Description**
SPP Fields		
Use Registrar	**Values:** Y/N **Description:** Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set in Phonebook Configuration screen) has a static IP address and all other voips (clients) point to the registar'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. Leave this checkbox unselected if your overall voip system is operating in the "Direct" SPP mode. In this mode, all voips in system are peers and each has its own static IP address.	
Port Number	**Values:** numeric **Description:** 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.
MultiVOIP 110/120/200/400/800	**Values:** Y/N **Description:** Select if any gateways of these model types are included in voip system and are operating in H.323 mode.	
Advanced button	**Values:** N/A **Description:** Gives access to secondary screen where an Alternate IP Route can be specified for backup or redundancy of signal paths. See discussion on next page. For SIP & H.323 operation only.	

Clicking on the **Advanced** button brings up the **Alternate Routing** secondary screen. This feature 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.



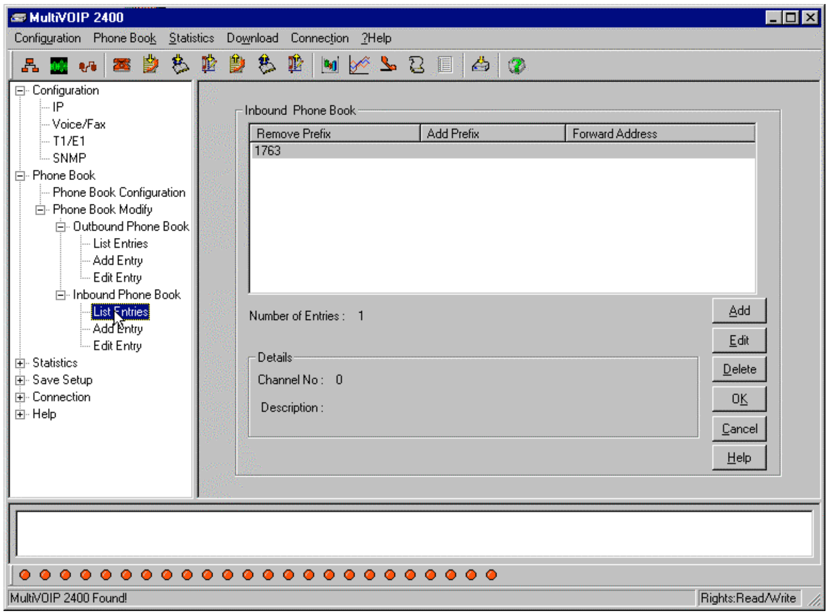
Alternate Routing Field Definitions		
Field Name	**Values**	**Description**
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when 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).

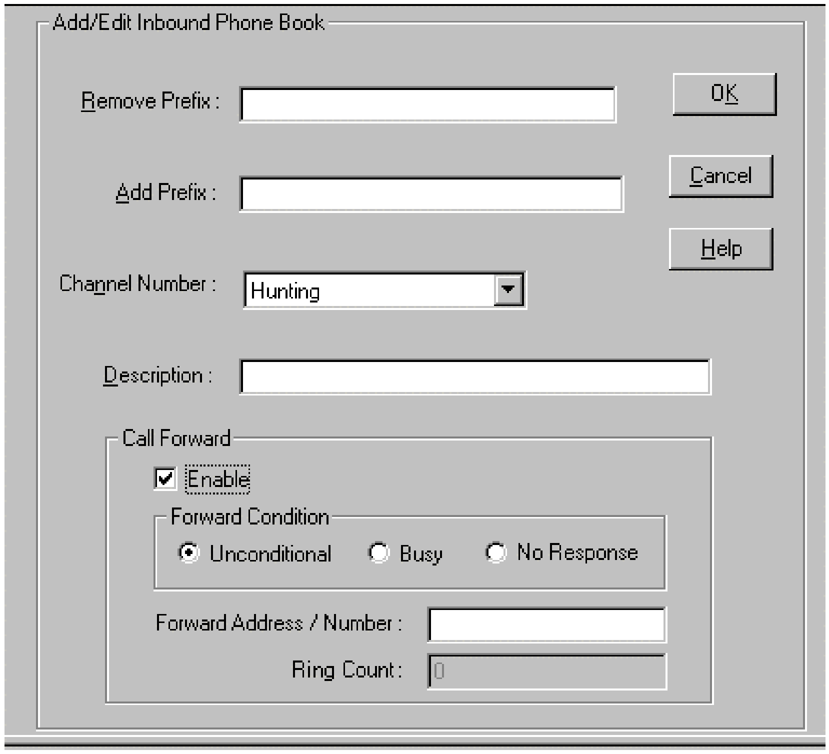


PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

4. Select **PhoneBook Modify** and then select **Inbound PhoneBook | List Entries**.



5. The **Add/Edit Inbound PhoneBook** screen appears.



Enter Inbound PhoneBook data for your MultiVOIP. The fields of the Add/Edit Inbound PhoneBook screen are described in the table below.

Add/Edit Inbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
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)
Channel Number	1-24, or "Hunting"	T1 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.

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
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	Uncondit.; Busy No Resp.	**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.
Forward Address/ Number	IP addr. or phone number	Phone number or IP address to which calls will be directed.
Ring Count	integer	When No Response is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.

6. When your 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.

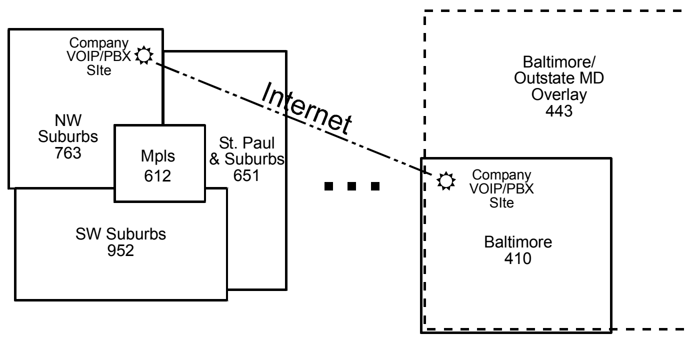
Remember that the initial MVP2400/2410 setup must be done locally using the MultiVOIP program. However, after the initial configuration is complete, all of the MVP2400/2410 units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVoipManager software program.

T1 Phonebook Examples

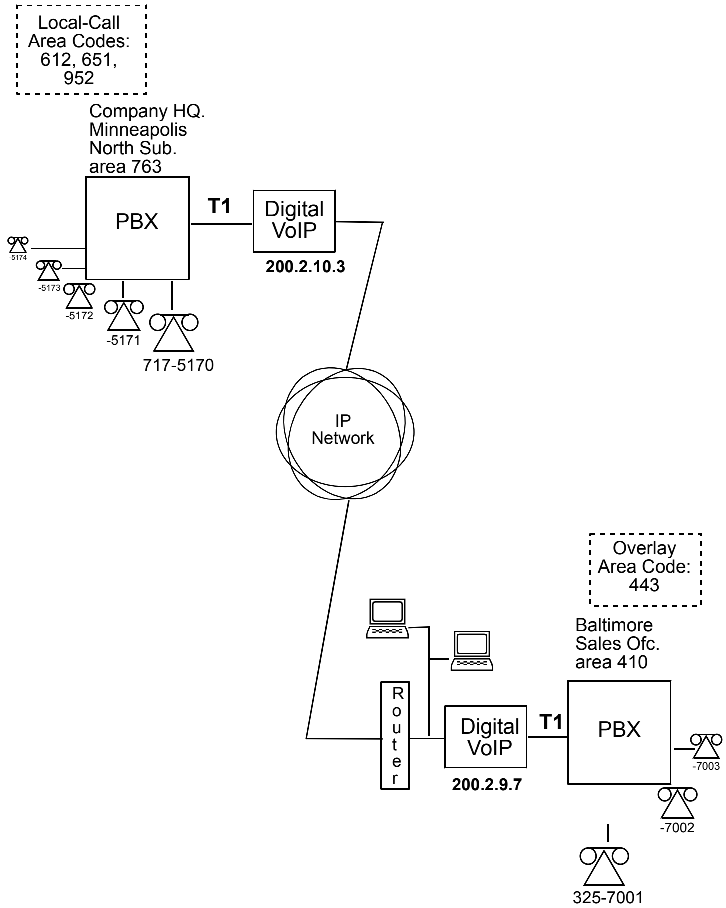
The following example demonstrates how Outbound and Inbound PhoneBook entries work in a situation of multiple area codes. Consider a company with offices in Minneapolis and Baltimore.

3 Sites, All-T1 Example

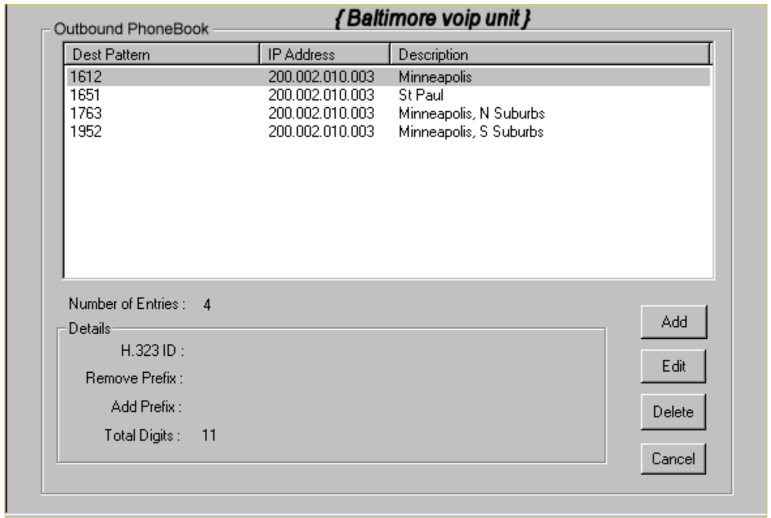
Notice first the area code situation in those two cities: Minneapolis's local calling area consists of multiple adjacent area codes; Baltimore's local calling area consists of a base area code plus an overlay area code.



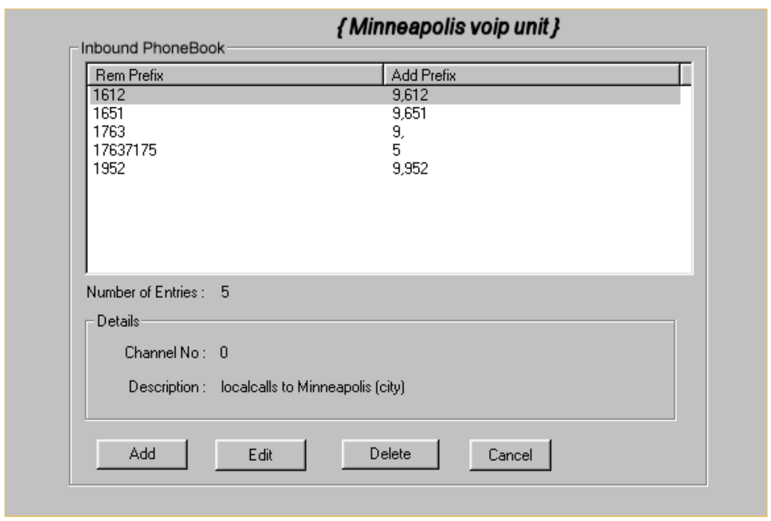
An outline of the equipment setup in both offices is shown below.



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



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



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.)

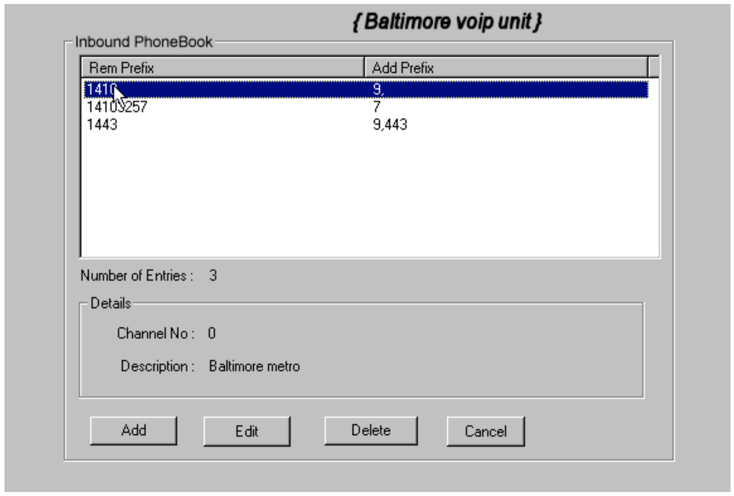
If a Baltimore employee dials any phone number in the 612 area code, the call will automatically be handled by the company’s voip system. Upon receiving such a call, the Minneapolis voip will remove the digits “1612”. But before the suburban-Minneapolis voip can complete the call to the PSTN of the Minneapolis 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 cal 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.

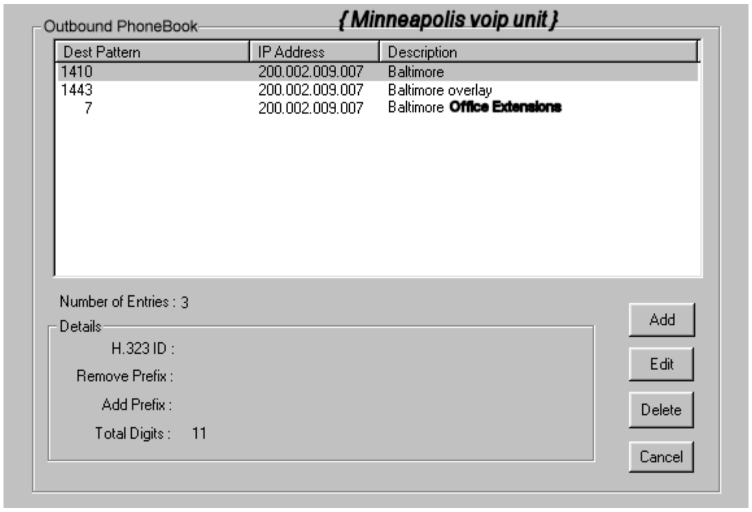
Similarly, the Inbound PhoneBook for the Baltimore VOIP (shown first below) generally matches the Outbound PhoneBook of the Minneapolis VOIP (shown second below).



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.

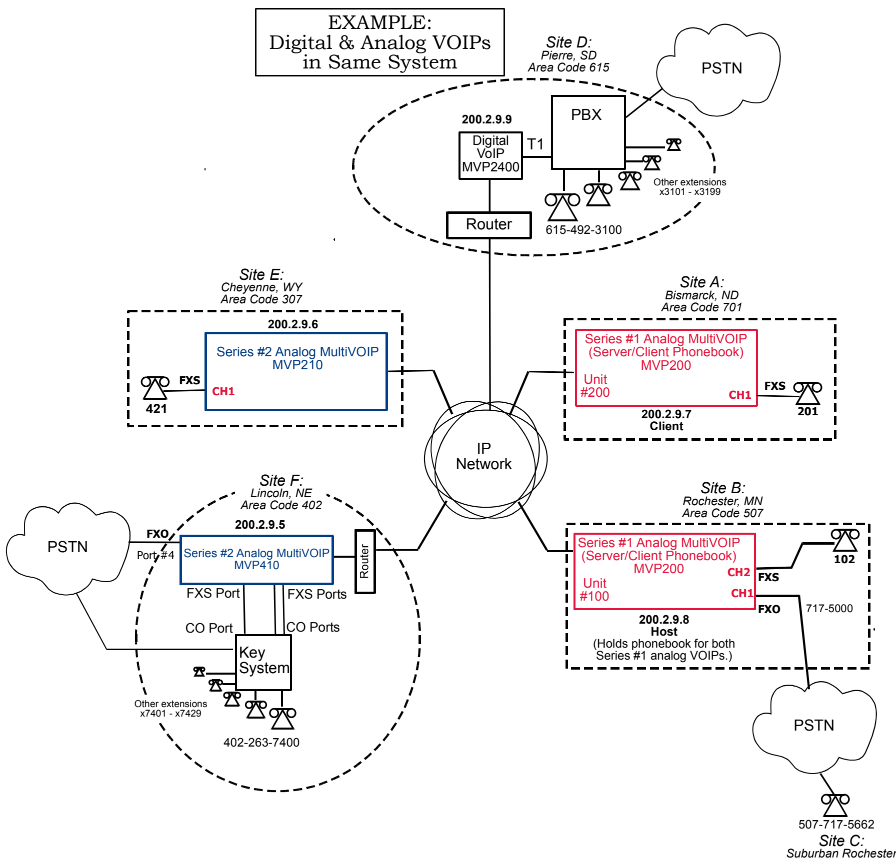
Note also that a comma (as in the entry 9,443) denotes a delay in dialing. A one-second delay is commonly used to allow a second dial tone to be generated for calls going outside of the facility's PBX system.

The Outbound PhoneBook 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" .



Configuring Mixed Digital/Analog VOIP Systems

The MVP2400/2410 digital MultiVOIP unit is compatible with analog VOIPs. In many cases, digital and analog VOIP units will appear in the same telephony/IP system. In addition to MVP-210/410/810 MultiVOIP units (Series II units), legacy analog VOIP units (Series I units made by MultiTech) may be included in the system, as well. When legacy VOIP units are included, the VOIP administrator must handle two styles of phonebooks in the same VOIP network. The diagram below shows a small-scale system of this kind: one digital VOIP (the MVP2400) operates with two Series II analog VOIPs (an MVP210 and an MVP410), and two Series I legacy VOIPs (two MVP200 units).



The Series I analog VOIP phone book resides in the “Host” VOIP unit at Site B. It applies to both of the Series I analog VOIP units.

Each of the Series II analog MultiVOIPs (the MVP210 and the MVP410) requires its own inbound and outbound phonebooks. The MVP2410 digital MultiVOIP requires its own inbound and outbound phonebooks, as well.

These seven phone books are shown below.

Phone Book for Series I Analog VOIP Host Unit (Site B)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
102	200.2.9.8	2	Site B, FXS channel.
101	200.2.9.8	1	Site B, FXO channel.
421	200.2.9.6	0	Site E FXS channel.
201	200.2.9.7	1	Site A, FXS channel.
1615 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to local PSTN of Site D (Pierre, SD, area code 615).
3xxx (Note 1.)	200.2.9.9	0	Allows remote voip users to call all PBX extensions at Site D (Pierre, SD) using only four digits.
1402	200.2.9.5	0	Gives remote voip users access to local PSTN of Site F (Lincoln, NE; area code 402).
140226374 (Note 1) (Note 3)	200.2.9.5	0	Gives remote voip users access to key phone system extensions at Site F (Lincoln).

Note 1. The "x" is a wildcard character.

Note 2. By specifying "Channel 0," we instruct the MVP2400/2410 to choose any available data channel to carry the call.

Note 3. Note that Site F key system has only 30 extensions (x7400-7429). This destination pattern (140226374) actually directs calls to 402-263-74**30** through 402-263-74**99** into the key system, as well.
This means that such calls, which belong on the PSTN, cannot be completed. In some cases, this might be inconsequential because an entire exchange (fully used or not) might have been reserved for the company or it might be unnecessary to reach those numbers. However, to specify only the 30 lines actually used by the key system, the destination pattern 140226374 would have to be replaced by three other destination patterns, namely 140226374**0**, 140226374**1**, and 140226374**2**. In this way, calls to 402-263-74**30** through 402-263-74**99** would be properly directed to the PSTN. In the Site D outbound phonebook, the 30 lines are defined exactly, that is, without making any adjacent phone numbers unreachable through the voip system.

Outbound Phone Book for MVP2400 Digital VOIP (Site D)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Bismarck).
1507	1507	101# Note 3.	200.2.9.8	To originate calls to Rochester local PSTN using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP.
421			200.2.9.6	Calls to Site E (Cheyenne).
1402			200.2.9.5	Calls to Lincoln area local PSTN (via FXO channel, CH4, of the Site F VOIP).
1402 263 740			200.2.9.5	Calls to extensions (thirty) of key system at Site F (Lincoln). Human operator or auto-attendant is needed to complete these calls.
1402 263 741			200.2.9.5	
1402 263 742			200.2.9.5	
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP2400/2410 Digital VOIP (Site D)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
1615	9, Note 4. Note 5.	0	Allows phone users at remote voip sites to call non-toll numbers within the Site D area code (615; Pierre, SD) over the VOIP network.
1615 49231	31	0	Allows voip calls directly to employees at Site D (at extensions x3101 to x3199).
Note 4. "9" gives PBX station users access to outside line. Note 5. The comma represents a one-second pause, the time required for the user to receive a dial tone on the outside line (PSTN). The comma is only allowed in the Inbound phonebook.			

Outbound Phone Book for MVP410 Analog VOIP (Site F)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Bismarck).
1507	1507	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Rochester area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Rochester).
421			200.2.9.6	Calls to Site E (Cheyenne).
1615			200.2.9.9	Calls to Pierre area PSTN via Site D PBX.
31		1615 492	200.2.9.9	Calls to Pierre PBX extensions with four digits.
Note 3. The pound sign (“#”) is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP410 Analog VOIP (Site F)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
1402		4	Access to Lincoln local PSTN by users at remote VOIP locations via FXO port at Site F.
1402 263740	740	0	Gives remote voip users access to extension of key phone system at Site F (Lincoln). Because call is completed at key system, abbreviated dialing (4 digits) is not workable. Human operator or auto-attendant is needed to complete these calls.
1402 263741	741	0	
1402 263742	742	0	

Outbound Phone Book for MVP210 Analog VOIP (Site E)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A.
1507	1507	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Rochester area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP.
1402			200.2.9.5	Calls to Lincoln area PSTN (via FXO channel, CH4, of the Site F VOIP).
7		1402 263	200.2.9.5	Calls to Lincoln key extensions with four digits.
1615			200.2.9.9	Calls to Pierre area PSTN via Site D PBX.
31		1615 492	200.2.9.9	Calls to Pierre PBX extensions with four digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP210 Analog VOIP (Site E)			
Remove Prefix	Add Prefix	Channel Number	Comment
421		1	

Call Completion Summaries

Site A calling Site C, Method 1

1. Dial 101.
2. Hear dial tone from Site B.
3. Dial 7175662.
4. Await completion. Talk.

Site A calling Site C, Method 2

1. Dial 101#7175662
2. Await completion. Talk.

Note: Some analog VOIP gateways will allow completion by Method 2. Others will not.

Site C calling Site A

1. Dial 7175000.
2. Hear dial tone from Site B VOIP.
3. Dial 201.
4. Await completion. Talk.

Site D calling Site C

1. Dial 9,15077175662.
2. “9” gets outside line. On some PBXs, an “8” may be used to direct calls to the VOIP, while “9” directs calls to the PSTN. However, some PBX units can be programmed to identify the destination patterns of all calls to be directed to the VOIP.
3. PBX at Site D is programmed to divert all calls made to the 507 area code and exchange 717 into the VOIP network. (It would also be possible to divert all calls to all phones in area code 507 into the VOIP network, but it may not be desirable to do so.)
4. The MVP2400/2410 removes the prefix “1507” and adds the prefix “101#” for compatibility with the analog MultiVOIP’s phonebook scheme. The “#” is a delimiter separating the analog VOIP’s phone number from the digits that the analog VOIP must dial onto its local PSTN to complete the call. The digits “101#7175662” are forwarded to the Site B analog VOIP.
5. The call passes through the IP network (in this case, the Internet).
6. The call arrives at the Site B VOIP. This analog VOIP receives this dialing string from the MVP2400/2410: 101#7175662. The analog VOIP, seeing the “101” prefix, uses its own channel #1 (an FXO port) to connect the call to the PSTN. Then the analog VOIP dials its local phone number 7175662 to complete the call.

Site D calling Site F

A voip call from Pierre PBX to extension 7424 on the key telephone system in Lincoln, Nebraska.

A. The required entry in the Pierre Outbound Phonebook to facilitate origination of the call, would be 1402263742. The call would be directed to the Lincoln voip's IP address, 200.2.9.5.
(Generally on such a call, the caller would have to dial an initial "9." But typically the PBX would not pass the initial "9" to the voip. If the PBX *did* pass along that "9" however, its removal would have to be specified in the local Outbound Phonebook.)

B. The corresponding entry in the Lincoln Inbound Phonebook to facilitate completion of the call would be

1402263742	for calls within the office at Lincoln
1402	for calls to the Lincoln local calling area (PSTN).

Call Event Sequence

1. Caller at Pierre dials 914022637424.
2. Pierre PBX removes "9" and passes 14022637424 to voip.
3. Pierre voip passes remaining string, 14022637424 on to the Lincoln voip at IP address 200.2.9.5.
4. The dialed string matches an inbound phonebook entry at the Lincoln voip, namely 1402263742.
5. The Lincoln voip rings one of the three FXS ports connected to the Lincoln
   key phone system.
6. The call will be routed to extension 7424 either by a human receptionist/ operator or to an auto-attendant (which allows the caller to specify the extension to which they wish to be connected).

Site F calling Site D

A voip call from a Lincoln key extension to extension 3117 on the PBX in Pierre, South Dakota.

A. The required entry in the Lincoln Outbound Phonebook to facilitate origination of the call, would be “31”. The string “1615492” would have to be added as a prefix. The call would be directed to the Pierre voip’s IP address, 200.2.9.9.

B. The corresponding entry in the Pierre Inbound Phonebook to facilitate completion of the call would be 1615492.

1. Caller at Lincoln picks up phone receiver, presses button on key phone set. This button has been assigned to a particular voip channel (any one of the three FXS ports).
2. The caller at Lincoln hears dial tone from the Lincoln voip.
3. The caller at Lincoln dials 3117.
4. The Lincoln voip adds the prefix 1615492 and sends the entire dialing string, 16154923117, to the Pierre voip at IP address 200.2.9.9.
5. The Pierre voip matches the called digits 16154923117 to its Inbound Phonebook entry “1615492” .
6. The Pierre PBX dials extension 3117 in the office at Pierre.

Variations in PBX Characteristics

The exact dialing strings needed in the Outbound and Inbound Phonebooks of the MVP2400/2410 will depend on the capabilities of the PBX. Some PBXs require trunk access codes (like an “8” or “9” to access an outside line or to access the VOIP network). Other PBXs can automatically distinguish between intra-PBX calls, PSTN calls, and VOIP calls.

Some PBX units can also insert digits automatically when they receive certain dialing strings from a phone station. For example, a PBX may be programmable to insert automatically the three-digit VOIP identifier strings into calls to be directed to analog VOIPs.

The MVP2400/2410 offers complete flexibility for inter-operation with PBX units so that a coherent dialing scheme can be established to connect a company’s multiple sites together in a way that is convenient and intuitive for phone users. When working together with modern PBX units, the presence of the MVP2400/2410 can be completely transparent to phone users within the company.

Chapter 8: E1 Phonebook Configuration

(European Telephony Standards)

MVP3010 Inbound and Outbound MultiVOIP Phonebooks

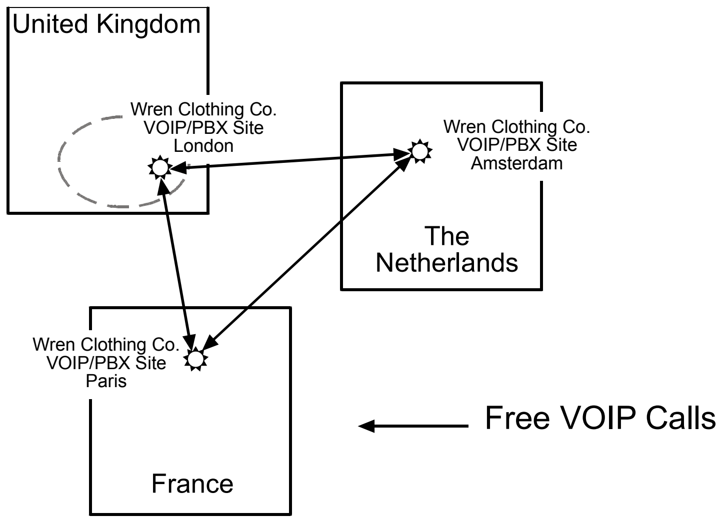
Important Definition:	The MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook describes the dialing sequences that can be used to call that MultiVOIP and how those calls will be directed.

When a VOIP serves a PBX system, the operation of the VOIP should be transparent to the telephone end user and savings in long-distance calling charges should be enjoyed. Use of the VOIP should not require the dialing of extra digits to reach users elsewhere on the VOIP network. 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. More importantly, the VOIP system should be configured to maximize savings in long-distance calling charges. To achieve both of these objectives, ease of use and maximized savings, the VOIP phonebooks must be set correctly.

NOTE: VOIPs are commonly used for another reason, as well: VOIPs allow an organization to integrate phone and data traffic onto a single network. Typically these are private networks.

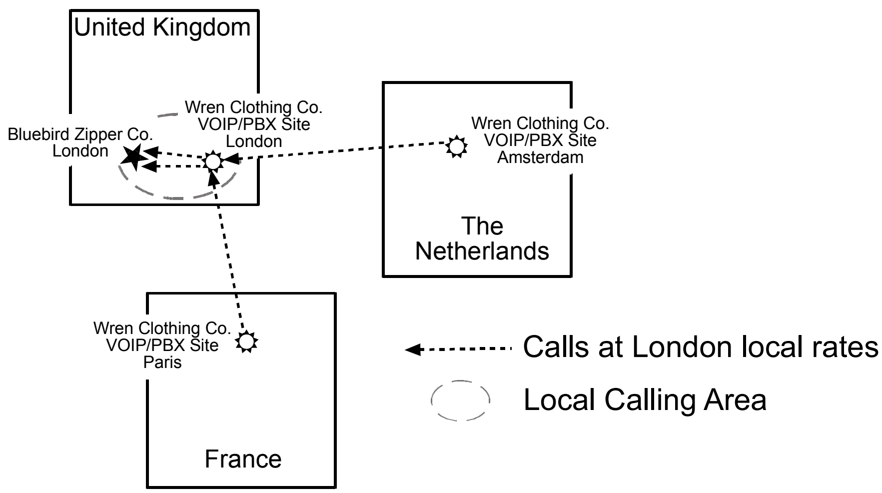
Free Calls: One VOIP Site to Another

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.

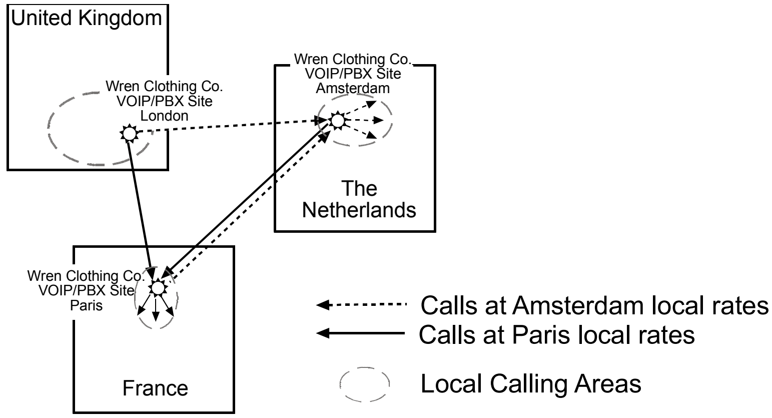


Local Rate Calls: Within Local Calling Area of Remote VOIP

In the second 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 (which includes both Inner London and Outer London). 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.)

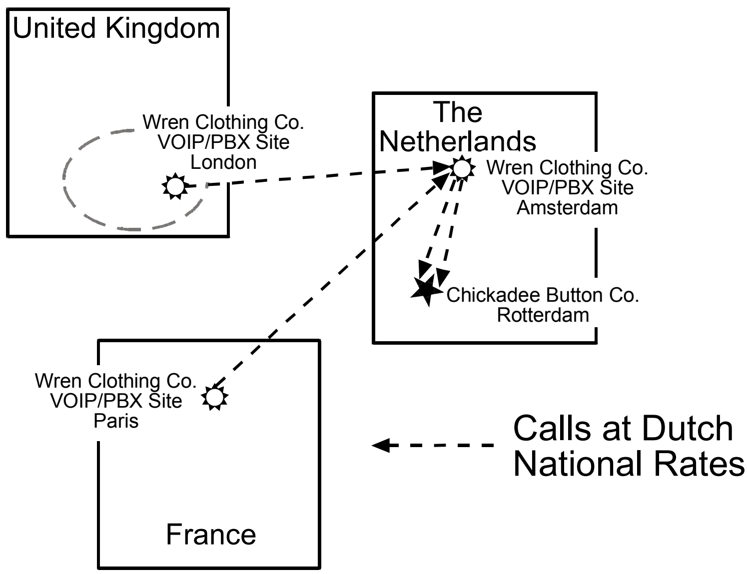


Similarly, the VOIP system allows Wren Clothing employees in London and Amsterdam to call anywhere in Paris at local rates; it allows Wren Clothing employees in Paris and London to call anywhere in Amsterdam at local rates.

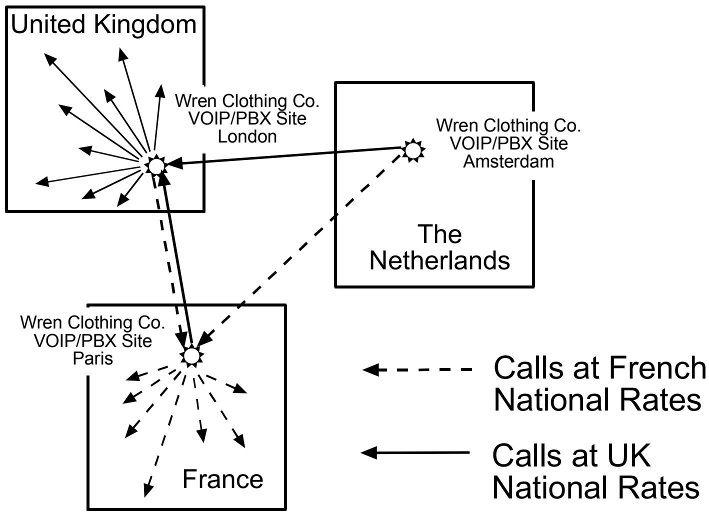


National Rate Calls: Within Nation of Remote VOIP Site

In the third use of the VOIP system, the national 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 national calling rates. Again, significant savings are possible. For example, suppose that the Wren Clothing Company buys its buttons from the Chickadee Button Company in the Dutch city of Rotterdam. In that case, Wren Clothing personnel in both London and Paris could call the Chickadee Button Company without paying international long-distance rates; only Dutch national calling rates would be charged. This applies to calls completed anywhere in The Netherlands.



Similarly, the VOIP system allows Wren Clothing employees in London and Amsterdam to call anywhere in France at French national rates; it allows Wren Clothing employees in Paris and Amsterdam to call anywhere in the United Kingdom at its national rates.



Inbound versus Outbound Phonebooks

To make the VOIP system transparent to phone users and to allow all possible free and reduced-rate calls, the VOIP administrator must 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 calls terminating at points beyond the remote VOIP site.

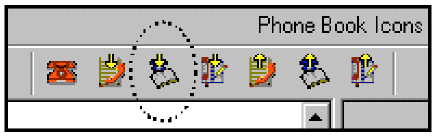
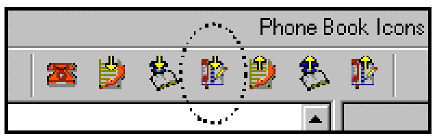
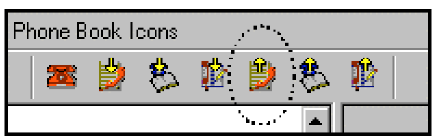
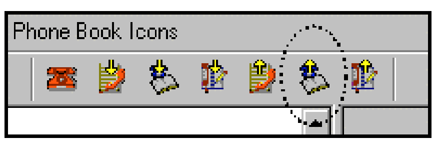
The "Inbound" phonebook for a particular VOIP unit describes the dialing sequences required for a call to originate remotely from any other VOIP sites in the system, and to terminate on that particular VOIP.

Briefly stated, *the MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook lists the dialing sequences that can be used to call that MultiVOIP*. (Of course, the phone numbers are not literally "listed" individually.) The phone stations that can originate or complete calls over the VOIP system are described by numerical rules called "destination patterns." These destination patterns generally consist of country codes, area codes or city codes, and local phone exchange numbers.

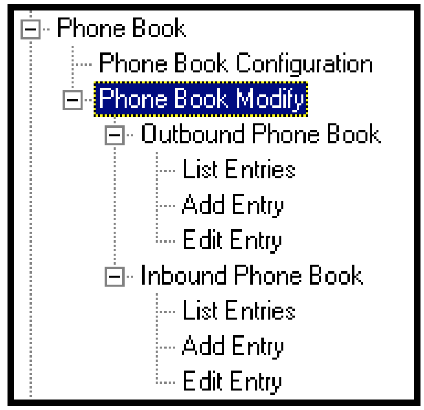
In order for any VOIP phone call to be made, there must be both an Inbound Phonebook entry and an Outbound Phonebook entry that describe the end-to-end connection. The phone station originating the call must be connected to the VOIP system. The Outbound Phonebook for that VOIP unit must have a destination pattern entry that includes the 'called' phone (that is, the phone completing the call). The Inbound Phonebook of the VOIP where the call is completed must have a destination pattern entry that includes the digit sequence dialed by the originating phone station.

The PhoneBook Configuration procedure below is brief, but it is followed by an example case. For many people, the example case may be easier to grasp than the procedure steps. Configuration is not difficult, but all phone number sequences, destination patterns, and other information must be entered exactly; otherwise connections will not be made.

Phonebook configuration screens can be accessed using icons or the sidebar menu.

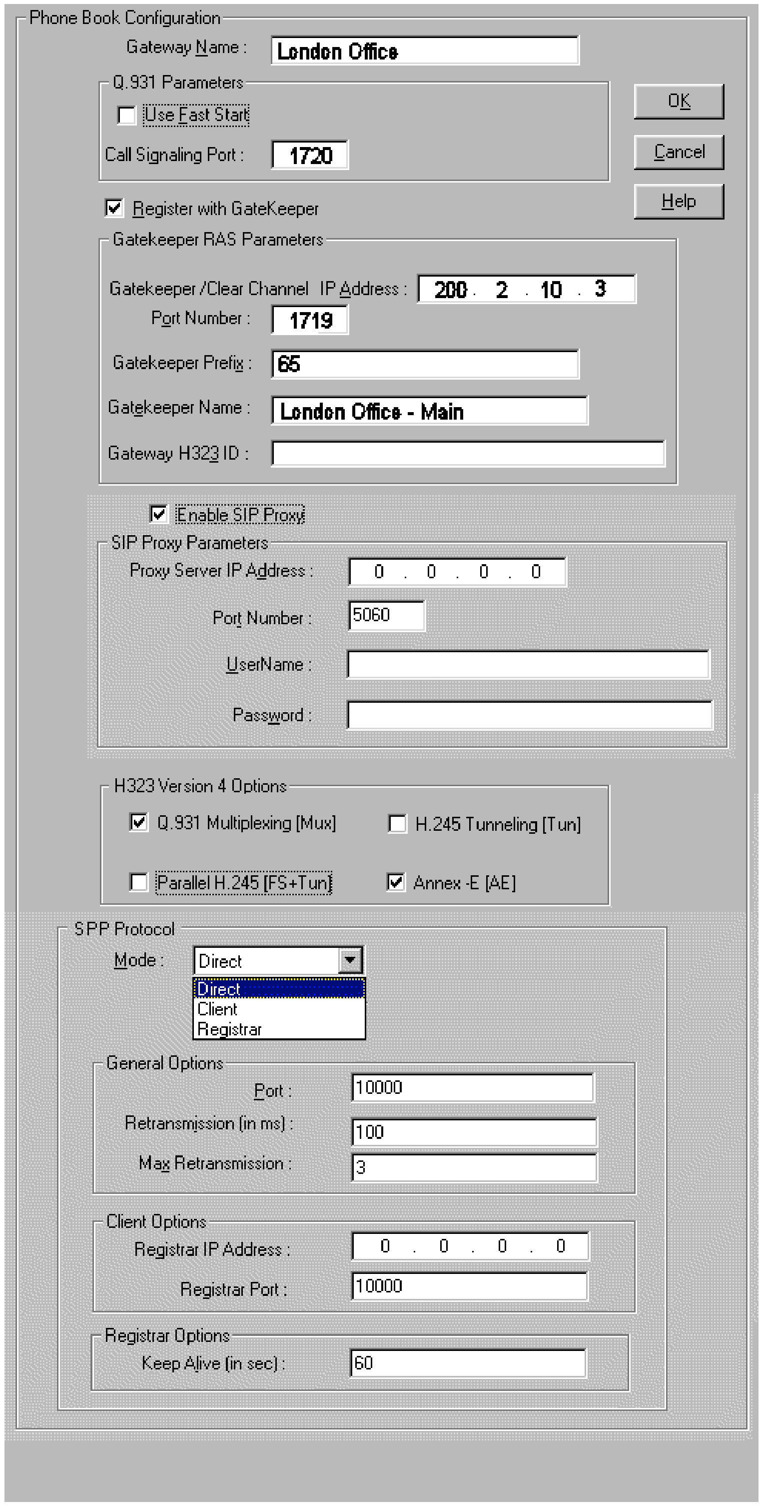
Phonebook Icons	Description
	Phonebook Configuration
	Inbound Phonebook Entries List
	Add Inbound Phonebook Entry
	Edit selected Inbound Phonebook Entry
	Outbound Phonebook Entries List
	Add Outbound Phonebook Entry
	Edit selected Outbound Phonebook Entry

Phonebook Sidebar Menu



Phonebook Configuration Procedure

1. Go to the **PhoneBook Configuration** screen (using either the sidebar menu, drop-down menu, or icon).


Phone Book Configuration
Gateway Name :
London Office
Q.931 Parameters
Use Fast Start
Call Signaling Port :
1720
OK
Cancel
Help
Register with GateKeeper
Gatekeeper RAS Parameters
Gatekeeper /Clear Channel IP Address :
200 . 2 . 10 . 3
Port Number :
1719
Gatekeeper Prefix :
65
Gatekeeper Name :
London Office - Main
Gateway H323 ID :
Enable SIP Proxy
SIP Proxy Parameters
Proxy Server IP Address :
0 . 0 . 0 . 0
Port Number :
5060
UserName :
Password :
H323 Version 4 Options
Q.931 Multiplexing [Mux]
H.245 Tunneling [Tun]
Parallel H.245 [FS+Tun]
Annex -E [AE]
SPP Protocol
Mode :
Direct
Direct
Client
Registrar
General Options
Port :
10000
Retransmission (in ms) :
100
Max Retransmission :
3
Client Options
Registrar IP Address :
0 . 0 . 0 . 0
Registrar Port :
10000
Registrar Options
Keep Alive (in sec) :
60

In consultation with your VOIP administrator, enter the Gateway Name and values for Q.931 parameters and Gatekeeper RAS parameters. Determine whether your voip system will operate with a proxy server. Determine which H.323 version 4 functions you will implement. (They are not always applicable. See field description for each parameter.) If the SPP protocol is used, values for another group of parameters must be specified, as well.

The table below describes all fields in the **PhoneBook Configuration** screen.

PhoneBook Configuration Parameter Definitions		
Field Name	**Values**	**Description**
Gateway Name	Y/N	This field allows you to specify a name for this MultiVOIP. When placing a call, this name is sent to the remote MVP3000 for display in Call Progress listings, Logs, etc.
Q.931 Parameters		
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.
Call Signaling Port	port number	Default: 1720 (H.323)
GateKeeper RAS Parameters		
Gatekeeper / Clear Channel IP Address		IP address of the GateKeeper.
Port Number		Well-known port number for GateKeepers. Must match port number of GateKeeper, 1719.
Gateway Prefix		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.
Gatekeeper Name	*alpha-numeric string*	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register.
Gateway H.323 ID		The H.323 ID is used to register this particular MultiVOIP with the GateKeeper. H.323 ID is an alias entry sent to the GateKeeper, made of alphanumeric characters. For NetMeeting endpoints, numbers are preferred over letters. The H.323 ID identifies the IP calling sequence that the GateKeeper must 'dial' to contact the remote VOIP.

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
SIP Proxy Parameters		
Enable Proxy	Y/N	
Proxy Server IP Address	n.n.n.n where n=0-255	Network address of the proxy server that the voip is using.
Port Number		Logical port number for proxy communications.
User Name		Identifier used when proxy server is used in network. If a proxy server is used in a SIP voip network, all clients must enter both a User Name and a Password before being allowed to make a call.
Password		Password for proxy server function. Password for proxy server function. See "User Name" description above.

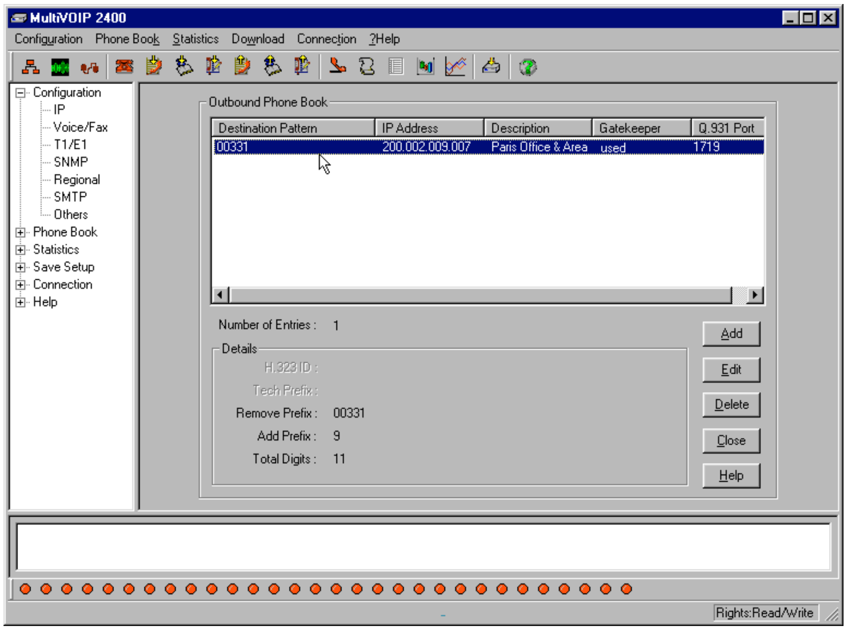
PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
Q.931 Multiplexing (Mux)	Y/N	Signaling for multiple phone calls can be carried on a single port rather than opening a separate signaling port for each call. This conserves bandwidth resources.
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.

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
Parallel H.245 (FS + Tun)	Y/N	FS (Fast Start or Fast Connect) is a Q.931 feature of H.323v2 to hasten call setup as well as 'pre-opening' the media channel before the CONNECT message is sent. This pre-opening is a requirement for certain billing activities. Under Parallel H.245 FS + Tun, this Fast Connect feature can operate simultaneously with H.245 Tunneling (see description above).
Annex –E (AE)	Y/N	Multiplexed UDP call signaling transport. Annex E is helpful for high-volume voip system endpoints. Gateways with lesser volume can afford to use TCP to establish calls. However, for larger volume endpoints, the call setup times and system resource usage under TCP can become problematic. Annex E allows endpoints to perform call-signaling functions under the UDP protocol, which involves substantially streamlined overhead. (This feature should not be used on the public Internet because of potential problems with security and bandwidth usage.)

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP)		
Mode	Direct, Client, or Registrar	SPP voip systems can operate in two modes: in the **direct mode**, where all voip gateways have static IP addresses assigned to them; or in the **registrar/client mode**, where 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		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-trans-mission (in ms)		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 value = 2000 milliseconds.)
Max Re-trans-mission		Number of times the voip will re-transmit a lost packet (if no acknowledgment has been received). (Default value = 3)

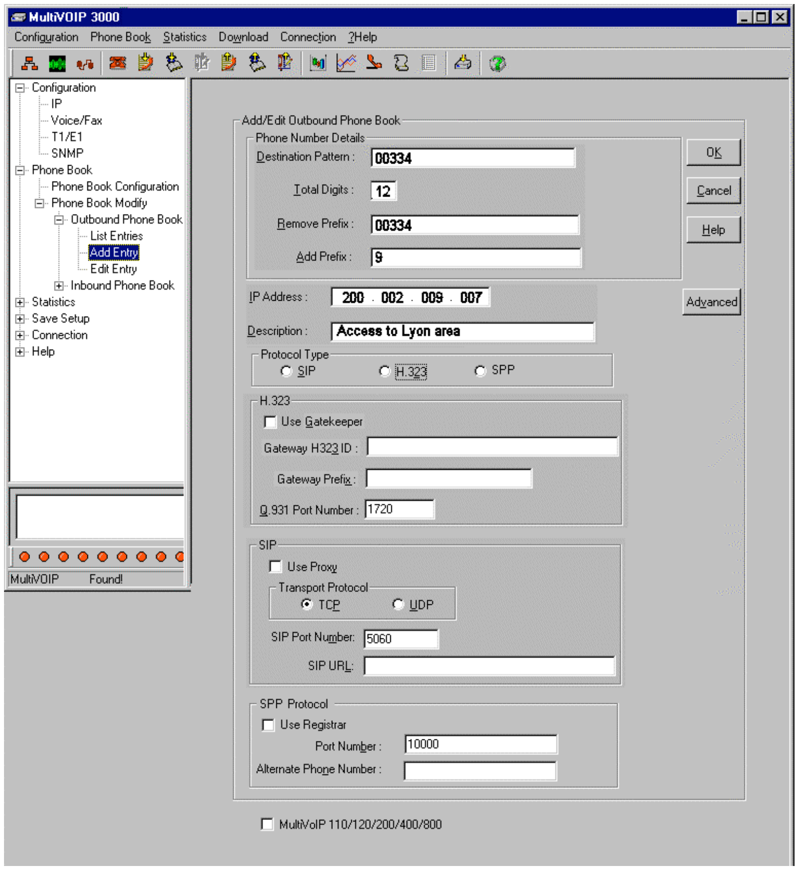
PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP) [cont'd]		
Client Options		Client Option fields are active only in registrar/client mode and only for client voip units.
Registrar IP Address		This is the IP address of the registrar voip to which this client is assigned. (Default value = 0.0.0.0; effectively, there is no useful default value.)
Registrar Port		This is the port number of the registrar voip to which this client is assigned. (Default port number = 10000.)
Registrar Options		Registrar Option fields are active only in registrar/client mode and only for registrar voip units.
Keep Alive (in sec.)		Time-out duration before a registrar will unregister 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.

2. Select **PhoneBook Modify** and then select
Outbound Phone Book/List Entries.



Click **Add**.

3. The **Add/Edit Outbound PhoneBook** screen appears.



Enter Outbound PhoneBook data for your MVP3010. 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).

The fields of the **Add/Edit Outbound Phone Book** screen are described in the table below.

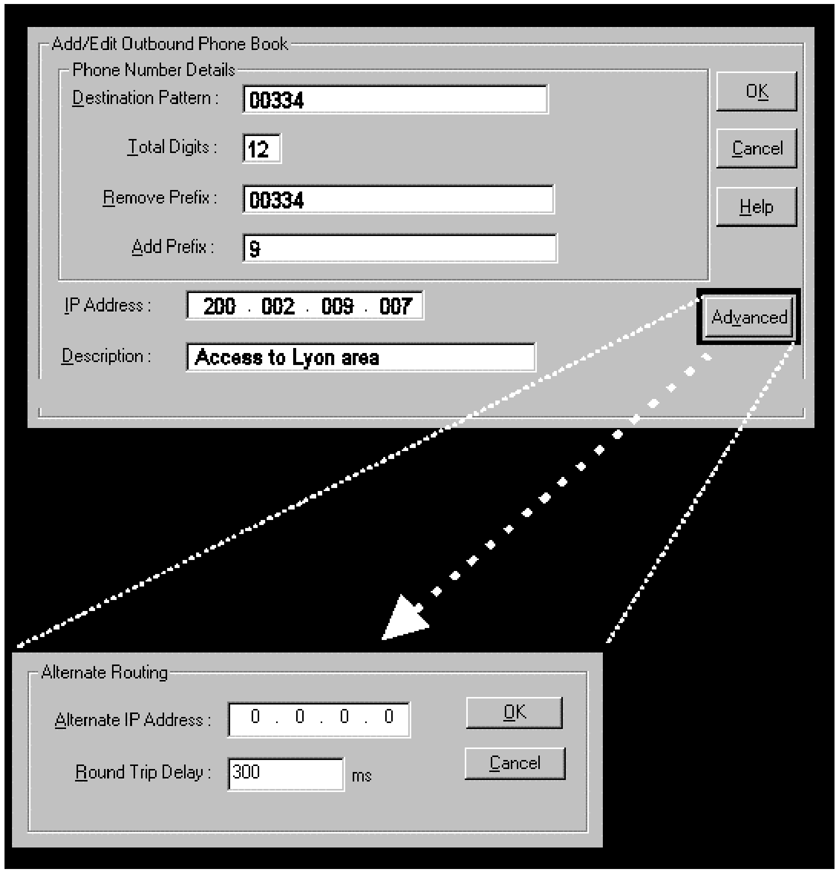
Add/Edit Outbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	number of digits the phone user must dial to reach specified destination
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 for = 0-255	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, H.323, or SPP	Indicates protocol to be used in outbound transmission.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 fields		
Use Gatekeepr	Y/N	Indicates whether or not gatekeeper is used.
H.323 ID		The H.323 ID assigned to the destination MultiVOIP. Only valid if "Use Gatekeeper" is enabled for this entry.
Gateway Prefix		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.
Q.931 Port Number Q.931 Port Number	1720	Q.931 is the 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, the port number 1720 must be chosen.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
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 RFC3087 ("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 callee 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.

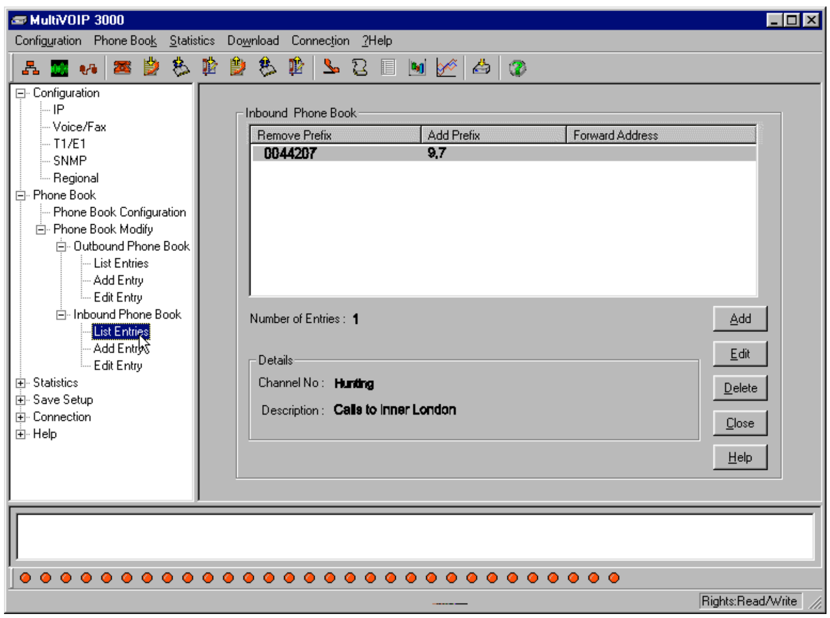
Add/Edit Outbound Phone Book: Field Def'ns (cont'd)		
Field Name	**Values**	**Description**
SPP Fields		
Use Registrar	**Values:** Y/N **Description:** Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set in Phonebook Configuration screen) has a static IP address and all other voips (clients) point to the registar'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. Leave this checkbox unselected if your overall voip system is operating in the "Direct" SPP mode. In this mode, all voips in system are peers and each has its own static IP address.	
Port Number	**Values:** numeric **Description:** 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.
MultiVOIP 110/120/200/400/800	**Values:** Y/N **Description:** Select if any gateways of these model types are included in voip system and are operating in H.323 mode.	
Advanced button	**Values:** N/A **Description:** Gives access to secondary screen where an Alternate IP Route can be specified for backup or redundancy of signal paths. See discussion on next page. For SIP & H.323 operation only.	

Clicking on the **Advanced** button brings up the **Alternate Routing** secondary screen. This feature 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.

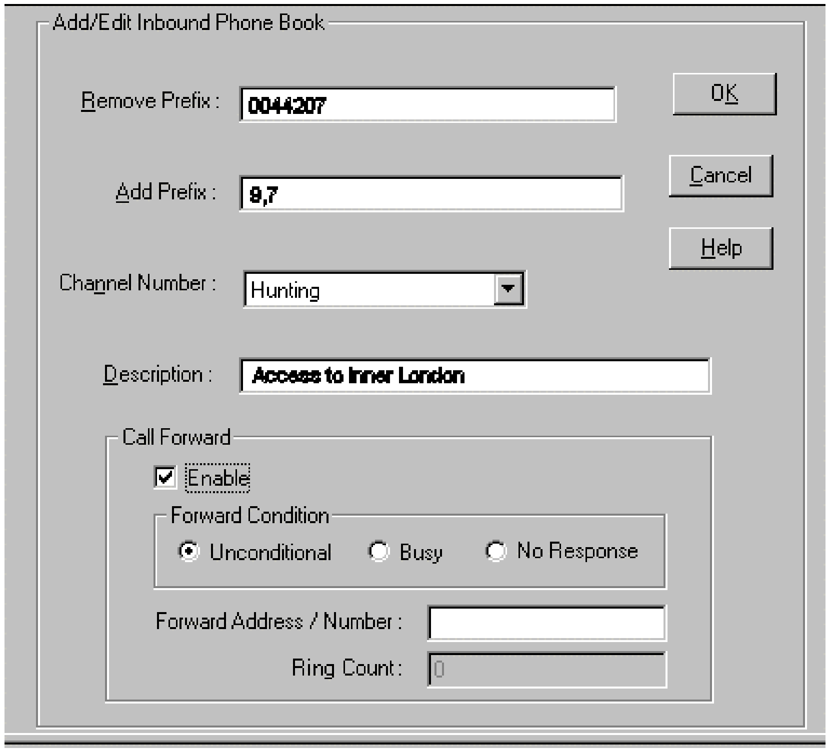


Alternate Routing Field Definitions		
Field Name	**Values**	**Description**
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when 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.

4. Select **PhoneBook Modify** and then select **Inbound PhoneBook/List Entries**.



5. The **Add/Edit Inbound PhoneBook** screen appears.



Enter Inbound PhoneBook data for your MVP3010. The fields of the Add/Edit Inbound PhoneBook screen are described in the table below.

Add/Edit Inbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
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)
Channel Number	1-30, or "Hunting"	E1 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.

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
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	Uncondit.; Busy No Resp.	**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.
Forward Address/ Number	IP addr. or phone number	Phone number or IP address to which calls will be directed.
Ring Count	integer	When No Response is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.

6. When your 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 MVP3010 setup must be done locally using the MultiVOIP program. However, after the initial configuration is complete, all of the MVP3010 units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVoipManager software program.

E1 Phonebook Examples

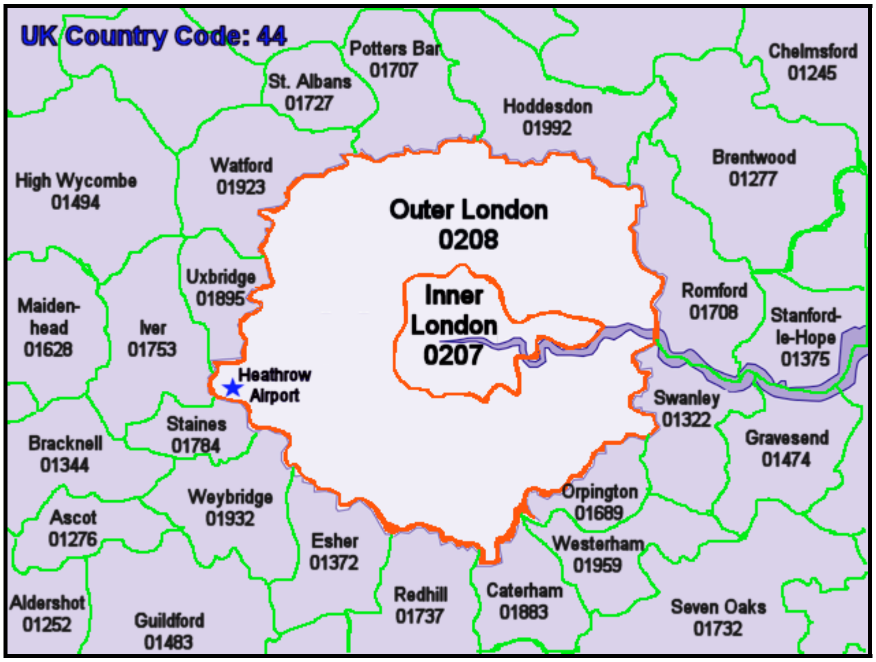
To demonstrate how Outbound and Inbound PhoneBook entries work in an international VOIP system, we will re-visit our previous example in greater detail. It's an international company with offices in London, Paris, and Amsterdam. In each office, a MVP3010 has been connected to the PBX system.

3 Sites, All-E1 Example

The VOIP system will have the following features:

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

Note that the phonebook entries for Series II analog MultiVOIP used in Euro-type telephony settings will be the same in format as entries for the MVP3010.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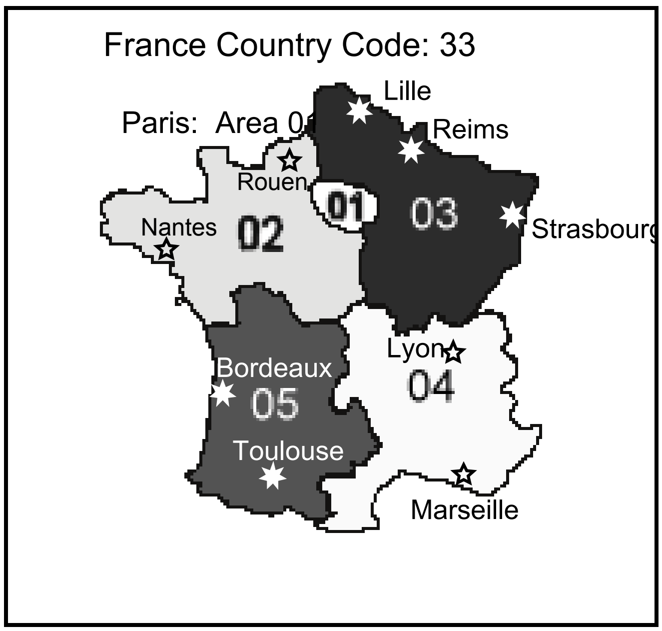
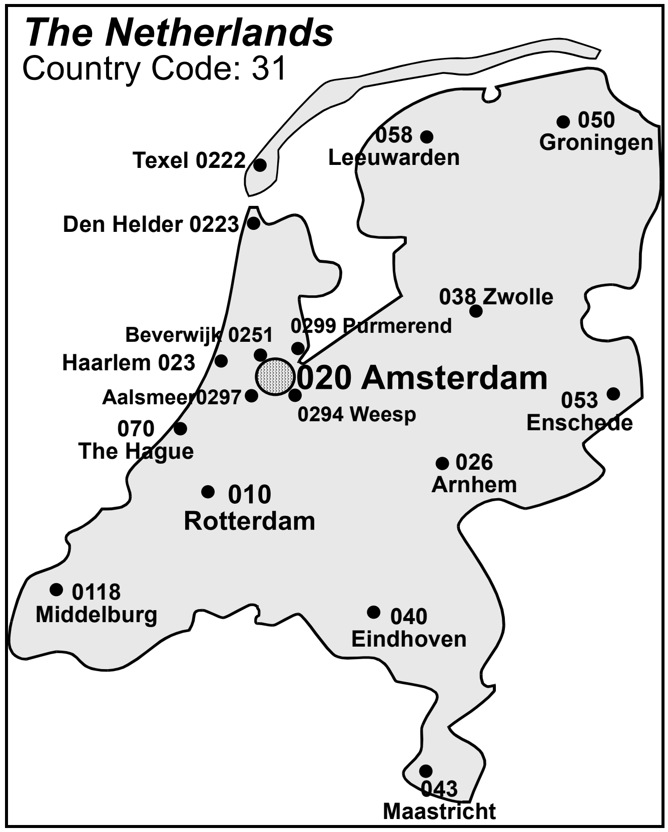
Potters Bar 01707
St. Albans 01727
Chelmsford 01245
Hoddesdon 01992
Brentwood 01277
High Wycombe 01494
Watford 01923
Outer London 0208
Inner London 0207
Uxbridge 01895
Romford 01708
Maiden-head 01628
Iver 01753
Stanford-le-Hope 01375
Heathrow Airport
Swanley 01322
Staines 01784
Bracknell 01344
Gravesend 01474
Weybridge 01932
Orpington 01689
Ascot 01276
Esher 01372
Westerham 01959
Redhill 01737
Caterham 01883
Aldershot 01252
Guildford 01483
Seven Oaks 01732


France Country Code: 33
Lille
Paris: Area 01
Reims
Rouen
01
03
Strasbourg
Nantes
02
Lyon
04
Bordeaux
05
Toulouse
Marseille


The Netherlands
Country Code: 31
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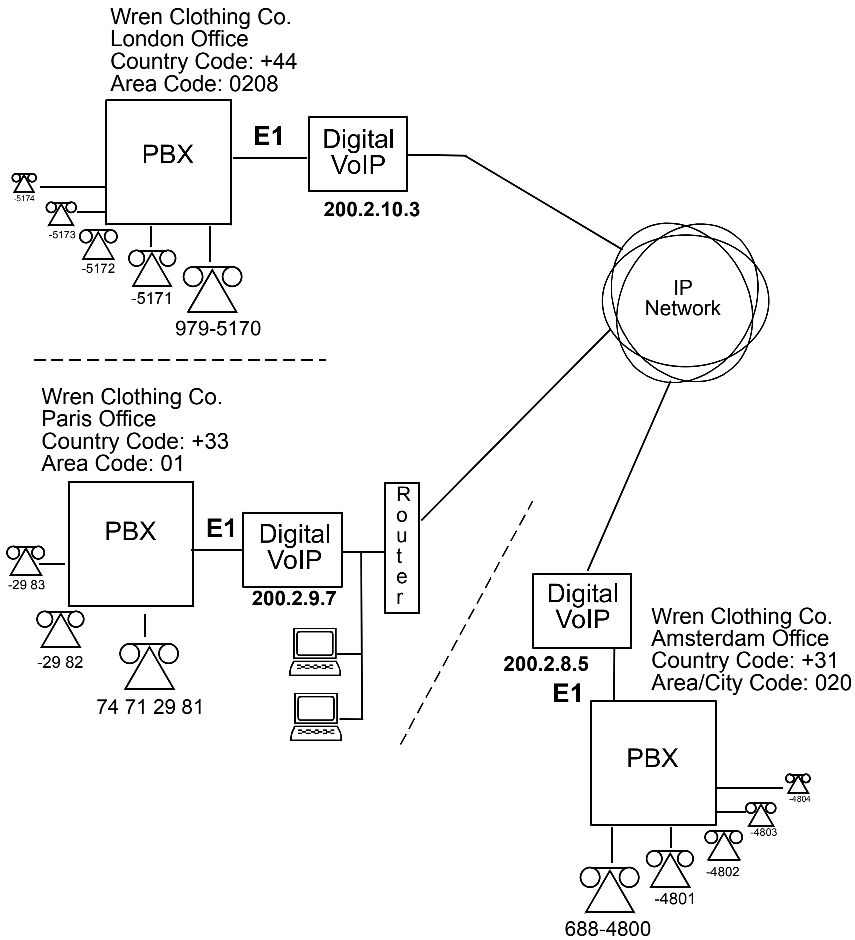
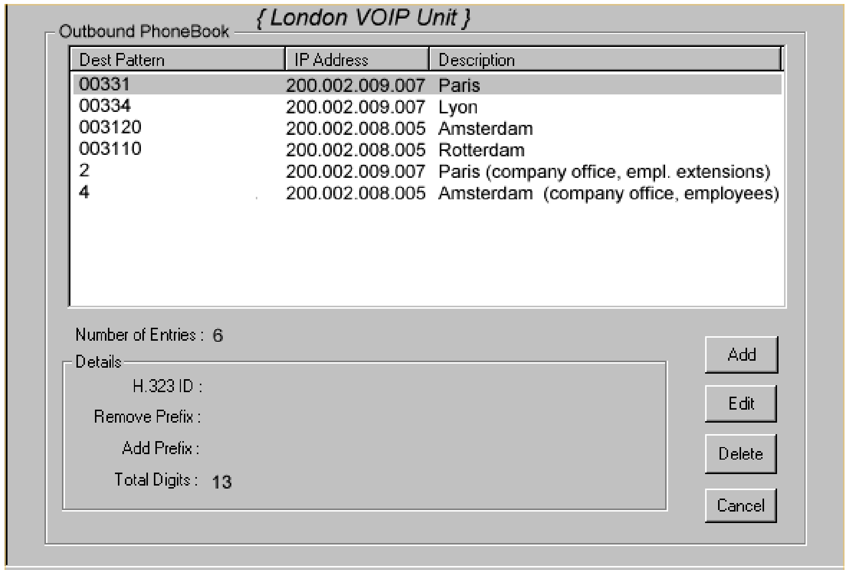
058 Leeuwarden
050 Groningen
Den Helder 0223
038 Zwolle
Beverwijk 0251
0299 Purmerend
Haarlem 023
020 Amsterdam
Aalsmeer 0297
0294 Weesp
053 Enschede
070 The Hague
026 Arnhem
010 Rotterdam
0118 Middelburg
040 Eindhoven
043 Maastricht

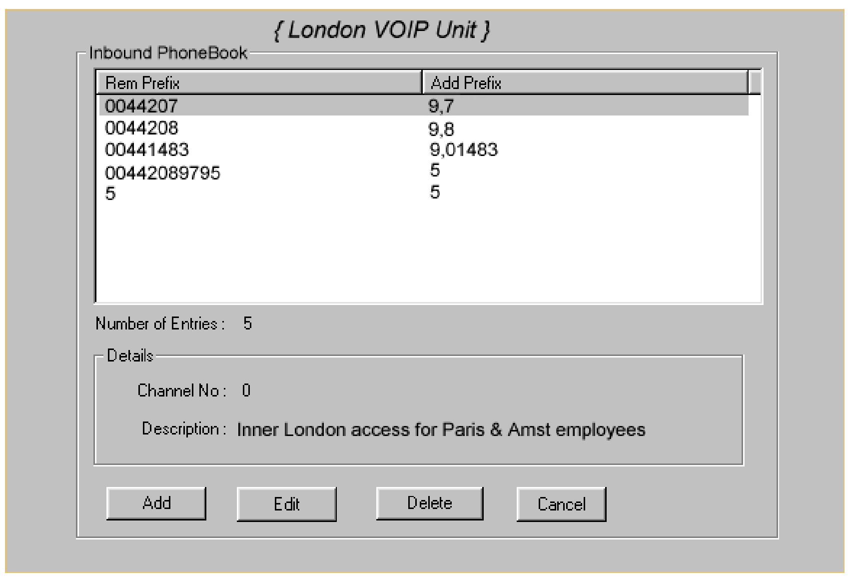
An outline of the equipment setup in these three offices is shown below.



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

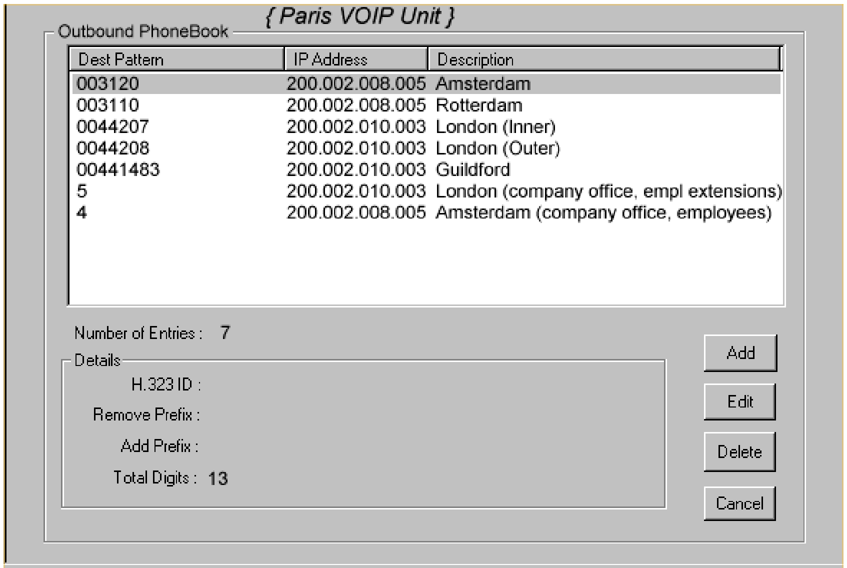


The Inbound PhoneBook for the London VOIP is shown below.

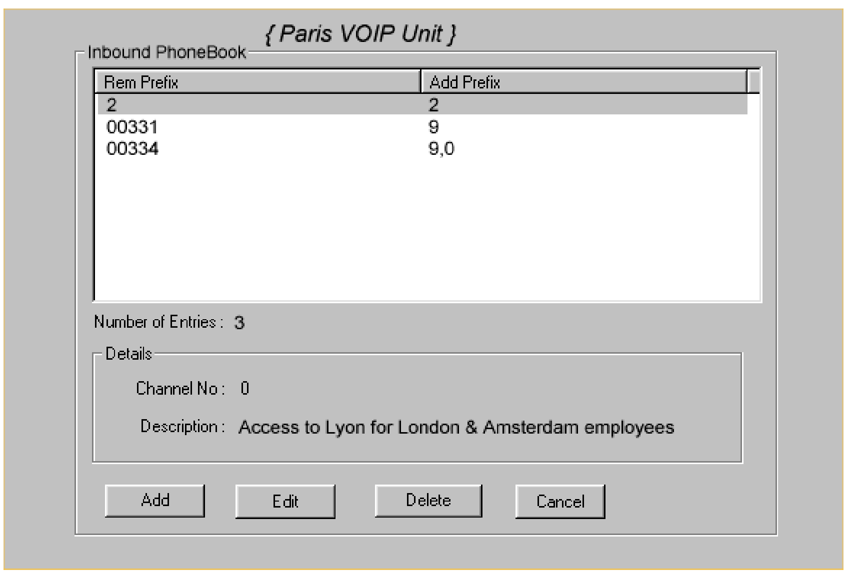


NOTE: Commas are allowed in the Inbound Phonebook, but **not** in the Outbound Phonebook. Commas denote a brief pause for a dial tone, allowing time for the PBX to get an outside line.

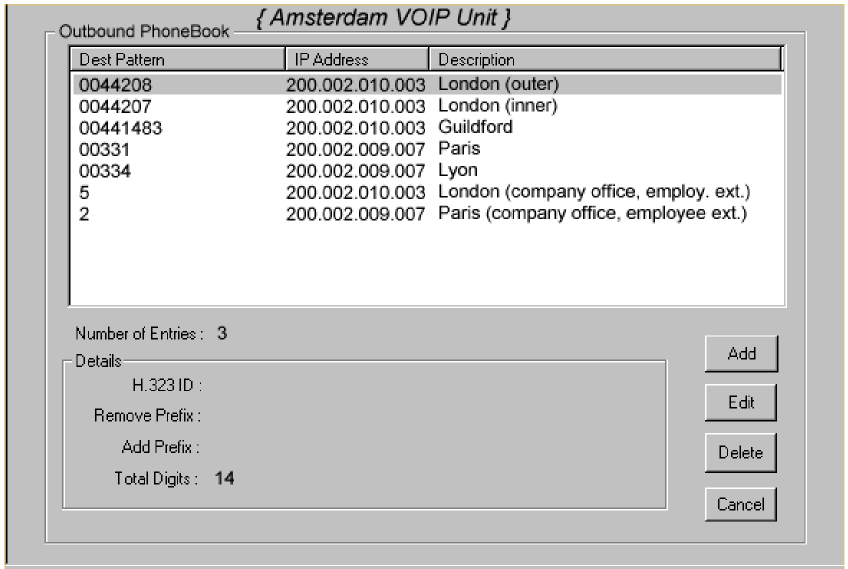
The screen below shows Outbound PhoneBook entries for the VOIP located in the company's Paris facility.



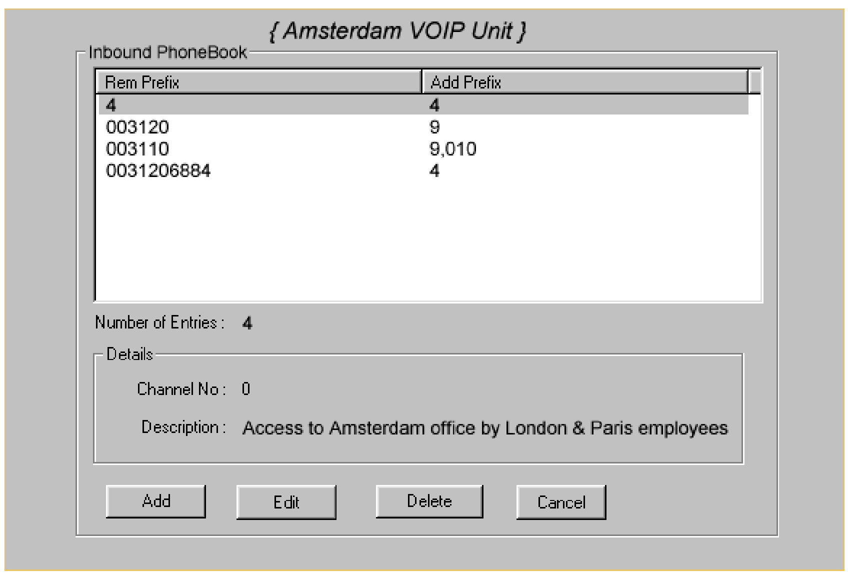
The Inbound PhoneBook for the Paris VOIP is shown below.



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

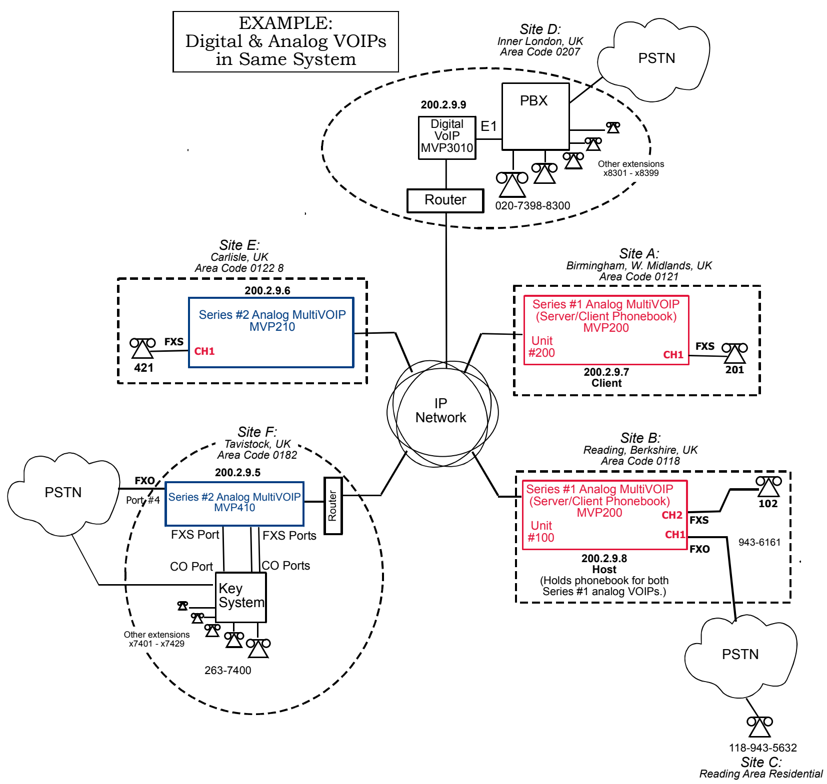


The Inbound PhoneBook for the Amsterdam VOIP is shown below.



Configuring Digital & Analog VOIPs in Same System

The MVP3010 digital MultiVOIP unit is compatible with analog VOIPs. In many cases, digital and analog VOIP units will appear in the same telephony/IP system. In addition to MVP-210/410/810 MultiVOIP units (Series II units), legacy analog VOIP units (Series I units made by MultiTech) may be included in the system, as well. When legacy VOIP units are included, the VOIP administrator must handle two styles of phonebooks in the same VOIP network. The diagram below shows a small-scale system of this kind: one digital VOIP (the MVP3010) operates with two Series II analog VOIPs (an MVP210 and an MVP410), and two Series I legacy VOIPs (two MVP200 units).



The Series I analog VOIP phone book resides in the “Host” VOIP unit at Site B. It applies to both of the Series I analog VOIP units.

Each of the Series II analog MultiVOIPs (the MVP210 and the MVP410) requires its own inbound and outbound phonebooks. The MVP3010 digital MultiVOIP requires its own inbound and outbound phonebooks, as well.

These **seven** phone books are shown below.

Phone Book for Analog VOIP Host Unit (Site B)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
102	200.2.9.8	2	Site B, FXS channel. (Reading, UK)
101	200.2.9.8	1	Site B, FXO channel. (Reading, UK)
201	200.2.9.7	1	Site A, FXS channel. (Birmingham)
421	200.2.9.6	0	Site E, FXS channel. (Carlisle, UK)
018226374 Note 3.	200.2.9.5	0	Gives remote voip users access to key phone system extensions at Tavistock office (Site F). The key system might be arranged either so that calls go through a human operator or through an auto-attendant (which prompts user to dial the desired extension).
0182	200.2.9.5	4	Gives remote voip users access to Tavistock PSTN via FXO port (#4) at Site F.
3xx	200.2.9.9	0 (Note 1.)	Allows remote voip users to call all PBX extensions at Site D (Inner London) using only three digits.

Phone Book for Analog VOIP Host Unit (Site B) (continued)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
0207 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to phone numbers in 0207 area code (Inner London) in which Site D is located.
0208 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to phone numbers in 0208 area code (Outer London) for which calls are local from Site D (Inner London).

Note 1. The "x" is a wildcard character.

Note 2. By specifying "Channel 0," we instruct the MVP3010 to choose any available data channel to carry the call.

Note 3. Note that Site F key system has only 30 extensions (x7400-7429). This destination pattern (018226374) actually directs calls to 402-263-74**30** through
402-263-74**99** into the key system, as well.
This means that such calls, which belong on the PSTN, cannot be completed. In some cases, this might be inconsequential because an entire exchange (fully used or not) might have been reserved for the company or it might be unnecessary to reach those numbers. However, to specify only the 30 lines actually used by the key system, the destination pattern 018226374 would have to be replaced by three other destination patterns, namely 018226374**0**, 018226374**1**, and 018226374**2**. In this way, calls to 0182-263-74**30** through 0182-263-74**99** would be properly directed to the PSTN. In the Site D outbound phonebook, the 30 lines are defined exactly, that is, without making any adjacent phone numbers unreachable through the voip system.

The Outbound PhoneBook of the MVP3010 is shown below.

Outbound Phone Book for MVP3010 Digital VOIP (Site D)				
Destin. Pattern	**Remove Prefix**	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Birmingham).
901189	901189	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Reading area using the FXO channel (channel #1) of the Site B VOIP (Reading, UK).
421	--	--	200.2.9.6	Calls to Site E (Carlisle).
90182				Calls to Tavistock local PSTN (Site F) could be arranged by operator or possibly by auto-attendant.
90182 263 740	9	--	200.2.9.5	Calls to extensions of key phone system at Tavistock office.
90182 263 741	9	--	200.2.9.5	
90182 263 742	9	--	200.2.9.5	
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Reading).
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

The Inbound PhoneBook of the MVP3010 is shown below.

Inbound Phone Book for MVP3010 Digital VOIP (Site D)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comments**
0207	9,7 Note 4. Note 5.	0	Allows phone users at remote voip sites to call local numbers (those within the Site D area code, 0207, Inner London) over the VOIP network.
0208	9,8 Note 4. Note 5.	0	Allows phone users at remote voip sites to call local numbers (those in Outer London) over the VOIP network.
0207 39883	3	0	Allows phone users at remote voip sites to call extensions of the Site D PBX using three digits, beginning with “3” .
Note 4. “9” gives PBX station users access to outside line. Note 5. The comma represents a one-second pause, the time required for the user to receive a dial tone on the outside line (PSTN). Commas can be used in the Inbound Phonebook, but **not** in the Outbound Phonebook.			

Outbound Phone Book for MVP410 Analog VOIP (Site F)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Birmingham).
01189	0118	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Reading area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Reading).
421			200.2.9.6	Calls to Site E (Carlisle).
0207			200.2.9.9	Calls to Inner London area PSTN via Site D PBX.
0208			200.2.9.9	Calls to Inner London area PSTN via Site D PBX.
3	--	0207 398 8	200.2.9.9	Calls to Inner London PBX extensions with three digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP410 Analog VOIP (Site F)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
01822	2	4	Calls to Tavistock local PSTN through FXO port (Port #4) at Site F.
0182 263 740	740.	0	Gives remote voip users, access to extensions of key phone system atTavistock office.
0182 263 741	741.	0	Because call is completed at key system, abbreviated dialing (3-digits) is not workable.
0182 263 742	742	0	Human operator or auto-attendant is needed to complete these calls.

Outbound Phone Book for MVP210 Analog VOIP (Site E)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Birmingham).
01189	0118	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Reading area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Reading).
01822	01822	--	200.2.9.5	Calls to Tavistock area PSTN (via FXO channel of the Site F VOIP).
0182 26374			200.2.9.5	Calls to Tavistock key system operator or auto-attendant.
0207	0207		200.2.9.9	Calls to London area PSTN via Site D PBX.
8		0207 398	200.2.9.9	Calls to London PBX extensions with four digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP210 Analog VOIP (Site E)			
Remove Prefix	Add Prefix	Channel Number	Comment
421		1	

Call Completion Summaries

Site A calling Site C, Method 1

1. Dial 101.
2. Hear dial tone from Site B.
3. Dial 9435632.
4. Await completion. Talk.

Site A calling Site C, Method 2

5. Dial 101#9435632
6. Await completion. Talk.

Note: Some analog VOIP gateways will allow completion by Method 2. Others will not.

Site C calling Site A

1. Dial 9436161.
2. Hear dial tone from Site B VOIP.
3. Dial 201.
4. Await completion. Talk.

Site D calling Site C

1. Dial 901189435632.
2. "9" gets outside line. On some PBXs, an "8" may be used to direct calls to the VOIP, while "9" directs calls to the PSTN. However, some PBX units can be programmed to identify the destination patterns of all calls to be directed to the VOIP.
3. PBX at Site D is programmed to divert all calls made to the 118 area code and exchange 943 into the VOIP network. (It would also be possible to divert *all* calls to all phones in area code 118 into the VOIP network, but it may not be desirable to do so.)
4. The MVP3010 removes the prefix "0118" and adds the prefix "101#" for compatibility with the analog MultiVOIP's phonebook scheme. The "#" is a delimiter separating the analog VOIP's phone number from the digits that the analog VOIP must dial onto its local PSTN to complete the call. The digits "101#9435632" are forwarded to the Site B analog VOIP.
5. The call passes through the IP network (in this case, the Internet).
6. The call arrives at the Site B VOIP. This analog VOIP receives this dialing string from the MVP3010: 101#9435632. The analog VOIP, seeing the "101" prefix, uses its own channel #1 (an FXO port) to connect the call to the PSTN. Then the analog VOIP dials its local phone number 9435632 to complete the call.

NOTE: In the case of Reading, Berkshire,, England, both "1189" and "1183" are considered local area codes. This is, in a sense however, a matter of terminology. It simply means that numbers of the form 9xx-xxxx and 3xx-xxxx are both local calls for users at other sites in the VOIP network.

Site D calling Site F

A voip call from Inner London PBX to extension 7424 on the key telephone system in Tavistock, UK.

A. The required entry in the London Outbound Phonebook to facilitate origination of the call, would be 90182263742. The call would be directed to the Tavistock voip's IP address, 200.2.9.5. (Generally on such a call, the caller would have to dial an initial "9". But typically the PBX would not pass the initial "9" dialed to the voip. If the PBX *did* pass along that "9" however, its removal would have to be specified in the local Outbound Phonebook.)

B. The corresponding entry in the Tavistock Inbound Phonebook to facilitate completion of the call would be

0182263742	for calls within the office at Tavistock
01822	for calls to the Tavistock local calling area (PSTN).

Call Event Sequence

1. Caller in Inner London dials 901822637424.
2. Inner London voip removes "9" .
3. Inner London voip passes remaining string, 01822637424on to the Tavistock voip
   at IP address 200.2.9.5.
4. The dialed string matches an inbound phonebook entry at the Tavistock voip, namely 0182263742.
5. The Tavistock voip rings one of the three FXS ports connected to the Tavistock
   key phone system.
6. The call will be routed to extension 7424 either by a human receptionist/ operator or to an auto-attendant (which allows the caller to specify the extension to which they wish to be connected).

Site F calling Site D

A voip call from a Tavistock key extension to extension 3117 on the PBX in Inner London.

A. The required entry in the Tavistock Outbound Phonebook to facilitate origination of the call, would be "3". The string 02073988 is added, preceding the "3". The call would be directed to the Inner London voip's IP address, 200.2.9.9.

B. The corresponding entry in the Inner-London Inbound Phonebook to facilitate completion of the call would be 020739883.

1. The caller in Tavistock picks up the phone receiver, presses a button on the key phone set. This button has been assigned to a particular voip channel.
2. The caller in Tavistock hears dial tone from the Tavistock voip.
3. The caller in Tavistock dials 02073983117.
4. The Tavistock voip sends the entire dialed string to the Inner-London voip at IP address 200.2.9.9.
5. The Inner-London voip matches the called digits 02073983117to its Inbound Phonebook entry "020739883, " which it removes. Then it adds back the "3" as a prefix.
6. The Inner-London PBX dials extension 3117 in the office in Inner London.

Variations in PBX Characteristics

The exact dialing strings needed in the Outbound and Inbound Phonebooks of the MVP3010 will depend on the capabilities of the PBX. Some PBXs require trunk access codes (like an "8" or "9" to access an outside line or to access the VOIP network). Other PBXs can automatically distinguish between intra-PBX calls, PSTN calls, and VOIP calls.

Some PBX units can also insert digits automatically when they receive certain dialing strings from a phone station. For example, a PBX may be programmable to insert automatically the three-digit VOIP identifier strings into calls to be directed to analog VOIPs.

The MVP3010 offers complete flexibility for inter-operation with PBX units so that a coherent dialing scheme can be established to connect a company's multiple sites together in a way that is convenient and intuitive for phone users. When working together with modern PBX units, the presence of the MVP3010 can be completely transparent to phone users within the company.

International Telephony Numbering Plan Resources

Due to the expansion of telephone number capacity to accommodate pagers, fax machines, wireless telephony, and other new phone technologies, numbering plans have been changing worldwide. Many new area codes have been established; new service categories have been established (for example, to accommodate GSM, personal numbering, corporate numbering, etc.). Below we list several web sites that present up-to-date information on the telephony numbering plans used around the world. While we find these to be generally good resources, we would note that URLs may change or become nonfunctional, and we cannot guarantee the quality of information on these sites.

URL	Description
http://phonebooth.interocitor.net /wtng	The World Telephone Numbering Guide presents excellent international numbering info that is both broad and detailed. This includes info on re-numbering plans carried out worldwide in recent years to accommodate new technologies.
http://www.oftel.gov.uk/numbers /number.htm	UK numbering plan from the Office of Telecommunications, the UK telephony authority.
http://www.itu.int/home/index.html	The International Telecommunications Union is an excellent source and authority on international telecom regulations and standards. National and international number plans are listed on this site.

URL	Description
http://kropla.com/phones.htm	Guide to international

	use of modems.
http://www.numberplan.org/	National and international numbering plans based on direct input from regulators worldwide. Includes lists of telecom carriers per country.
http://www.eto.dk/	European Telecommunications Office. Primarily concerned with mobile/wireless radiotelephony, GSM, etc.
http://www.eto.dk/ETNS.htm	European Telephony Numbering Space. Resources for pan-European telephony services, standards, etc. Part of ETO site.
http://www.regtp.de/en/reg_tele/start/fs_05.html	List of European telecom regulatory agencies by country (from German telecom authority).

Chapter 9: Analog/BRI Phonebook Configuration

Phonebooks for Series II analog MultiVOIP units (MVP130, MVP210, MVP210G, MVP410, MVP410G, MVP810, and MVP810G) and BRI MultiVOIP units (MVP410ST/810ST) are, in principle, configured the same as phonebooks for digital MultiVOIP products that would operate in the same environment (under either North American or European telephony standards, T1 or E1).

Therefore, if you are operating an analog MultiVOIP unit in a North American telephony environment, you will find useful phonebook instructions and examples in *Chapter 7: T1 Phonebook Configuration*. If you are operating an analog MultiVOIP unit in a European telephony environment, you will find useful phonebook instructions and examples in *Chapter 8: E1 Phonebook Configuration*.

Most of the examples in Chapters 7 and 8 describe systems containing both digital and analog MultiVOIP units.

You will also find useful information in *Chapter 2: Quick Start Guide*. See especially these sections:

Phonebook Starter Configuration
Phonebook Tips
Phonebook Example (One Common Situation)

Chapter 2 also contains a “Phonebook Worksheet” section. You may want to print out several worksheet copies. Paper copies can be very helpful in comparing phonebooks at multiple sites at a glance. This will assist you in making the phonebooks clear and consistent and will reduce ‘surfing’ between screens on the configuration program.

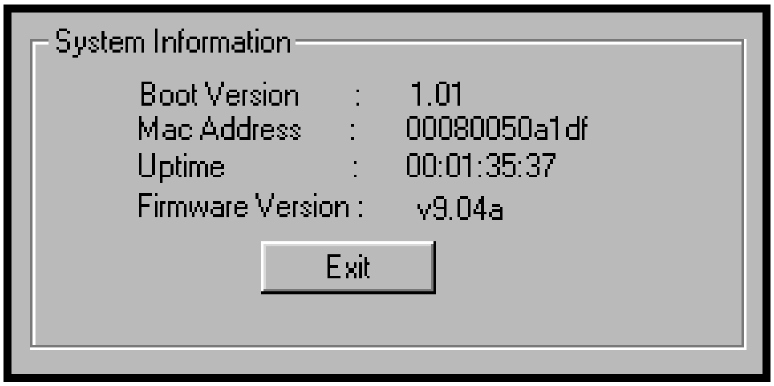
Chapter 10: Operation and Maintenance

Operation and Maintenance

Although most Operation and Maintenance functions of the software are in the **Statistics** group of screens, an important summary appears in the **System Information** of the **Configuration** screen group.

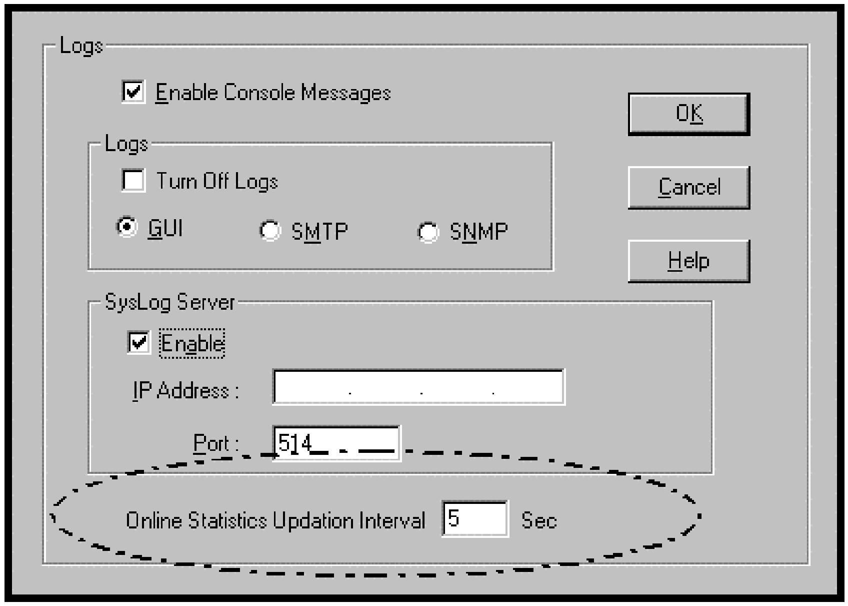
System Information screen

This screen presents vital system information at a glance. Its primary use is in troubleshooting. This screen is accessible via the **Configuration** pulldown menu, the **Configuration** sidebar menu, or by the keyboard shortcut **Ctrl + Alt + Y**. However, the System Information screen is not supported in the BRI 5.02c software.



System Information Parameter Definitions (not supported in BRI 5.02c software)		
Field Name	**Values**	**Description**
Boot Version	nn.nn	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.
Mac Address	alpha-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.
Firmware Version	alpha-numeric	Indicates the version of the MultiVOIP firmware.

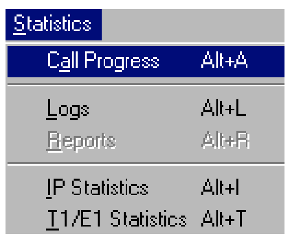
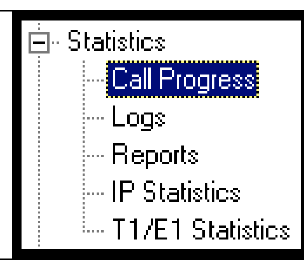
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



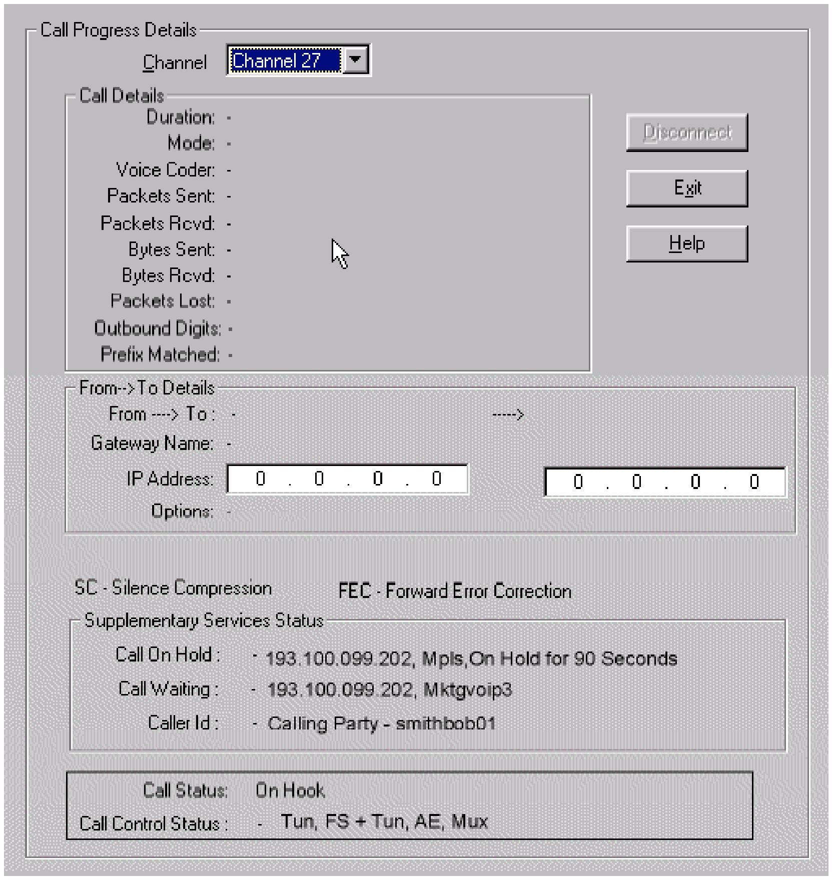
Statistics Screens

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.

About Call Progress

Accessing Call-Progress Statistics	
Channel Icons (Main Screen Lower Left)	
Channel icons are green when data traffic is present, red when idle.	
In the web GUI, call progress details can be viewed by clicking on an icon (one for each channel) arranged similarly on the web-browser screen.	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Alt + A`	

The Call Progress Details Screen



Call Progress Details: Field Definitions		
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	Hours: Minutes: Seconds	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, etc.	The voice coder being used on this call.
Packets Sent	integer value	The number of data packets sent over the IP network in the course of this call.
Packets Rcvd	integer value	The number of data packets received over the IP network in the course of this call.
Bytes Sent	integer value	The number of bytes of data sent over the IP network in the course of this call.
Bytes Rcvd	integer value	The number of bytes of data received over the IP network in the course of this call.
Packets Lost	integer value	The number of voice packets from this call that were lost after being received from the IP network.
Outbound Digits	0-9, #, *	The digits transmitted by the MultiVOIP to the PBX/telco for this call.
Prefix Matched		Displays the dialed digits that were matched to a phonebook entry.

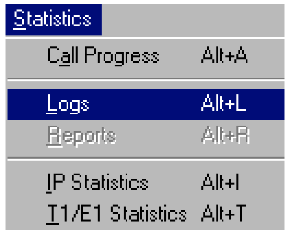
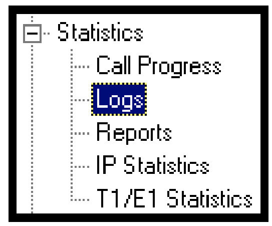
Call Progress Details: Field Definitions (cont'd)		
From – To Details		**Description**
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that handled this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address from which the call was received.
Options	SC, FEC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.
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

Call Progress Details: Field Definitions (cont'd)		
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 comes 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 comes from **Gateway Name** field in **Phone Book Configuration** screen of remote voip.

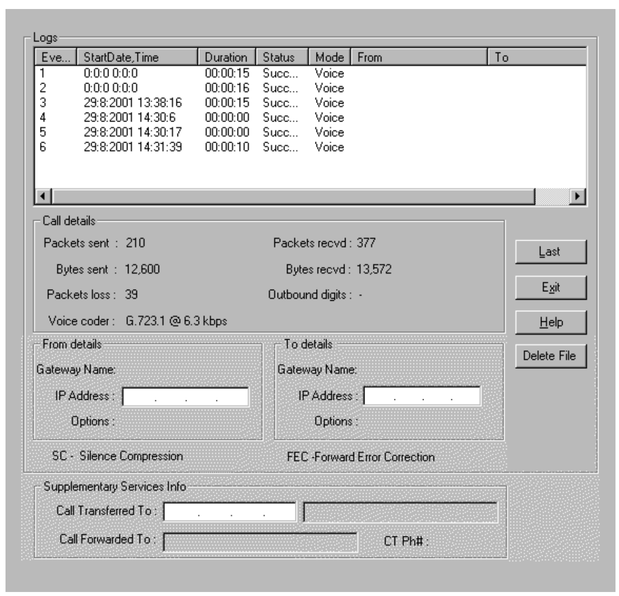
Call Progress Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Supplementary Services Status		
Caller ID	There are four values: "Calling Party + *identifier"*; "Alerting Party + *identifier";* "Busy Party + *identifier";* and "Connected Party + *identifier*"	This field shows 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.
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). See **Phonebook Configuration Parameters** (in T1 or E1 chapters) for more on H.323v4 features.

About Logs

The Logs

Accessing "Statistics: Logs"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Alt + L`	

The Logs Screen



Logs Screen Details: Field Definitions		
Field Name	**Values**	**Description**
Event # 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 (event). The date is presented as a day expression of one or two digits, a month expression of one or two digits, and a four-digit year. This is followed by a time-of-day expression presented as a two-digit hour, a two-digit minute, and a two-digit seconds value. (statistics, logs) field
Duration column	hh:mm:ss	This describes how long the call (event) lasted in hours, minutes, and seconds.
Status column	success or failure	Displays the status of the call, i.e., whether the call was completed successfully or not.
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		
Last		Displays last log entry.
Delete File		Deletes selected log file.
Call Details		
Packets sent	integer value	The number of data packets sent over the IP network in the course of this call.
Bytes sent	integer value	The number of bytes of data sent over the IP network in the course of this call.

Logs Screen Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Details (cont'd)		
Packets loss (lost)	integer value	The number of voice packets from this call that were lost after being received from the IP network.
Voice coder	G.723, G.729, G.711, etc.	The voice coder being used on this call.
Packets received	integer value	The number of data packets received over the IP network in the course of this call.
Bytes received	integer value	The number of bytes of data received over the IP network in the course of this call.
Outbound digits	0-9, #, *	The digits transmitted by the MultiVOIP to the PBX/telco for this call.
FROM Details		
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that originated this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	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 string	Identifier for the VOIP gateway that completed (terminated) this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address of the VOIP gateway at which the call was completed (terminated).
Options		Displays VOIP transmission options used by the VOIP gateway terminating the call. These may include Forward Error Correction or Silence Compression.

Logs Screen Details: Field Definitions (cont'd)		
Supplementary Services Info (Not supported in BRI 502c software.)		
Call Transferred To	phone number string	Number of party called in transfer.
Call Forwarded To	phone number string	Number of party called in forwarding.
CT Ph#	phone number string	Call Transfer phone number.

About Reports

This feature not implemented as of this writing.

About IP Statistics

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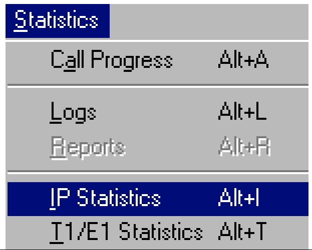
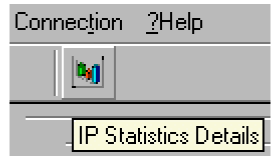
<tr><td>
</td><td>
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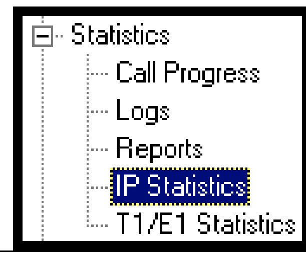
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Alt + I</td><td>
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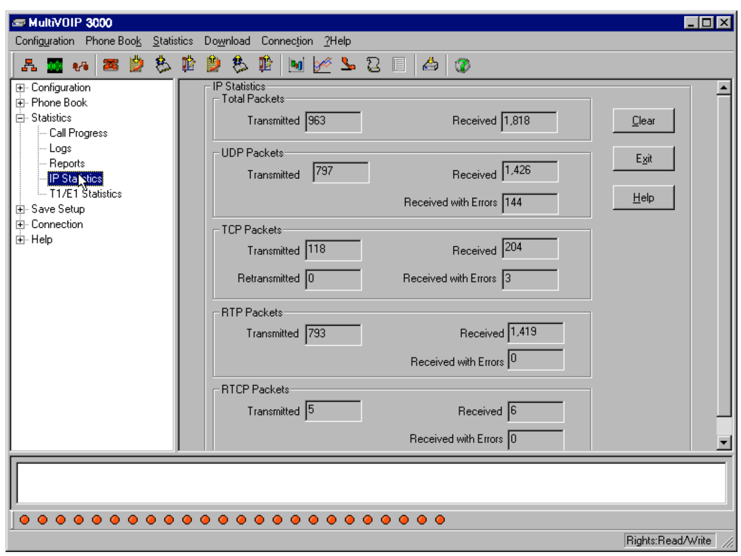
IP Statistics Screen



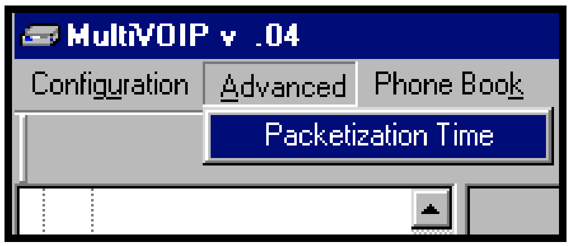
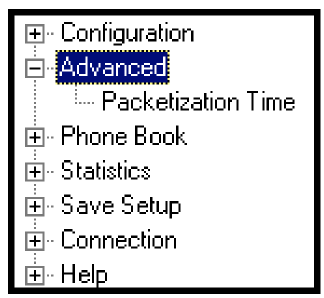
IP Statistics: Field Definitions		
Field Name	**Values**	**Description**
		UDP versus TCP. (User Datagram Protocol versus Transmission Control Protocol). UDP provides unguaranteed, connectionless transmission of data across an IP network. By contrast, TCP provides reliable, connection-oriented transmission 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 unretrievable; 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 appear as static).
"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.

IP Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Total Packets (cont'd)		Sum of data packets of all types.
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.

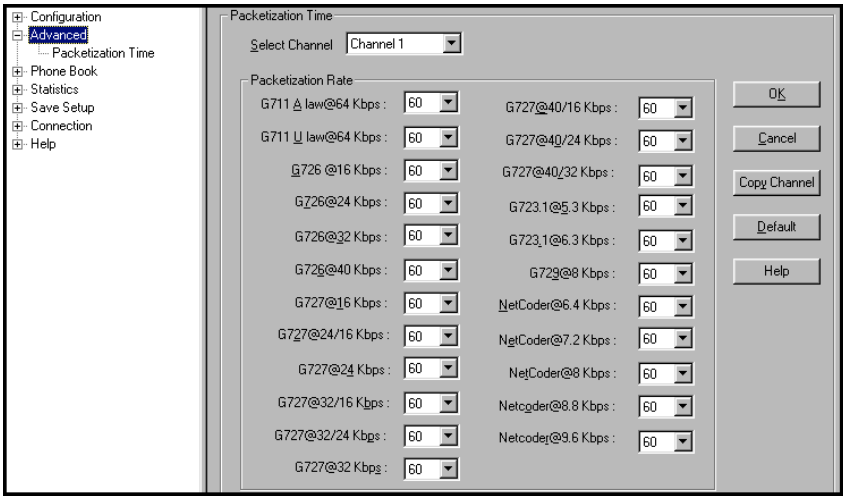
IP Statistics: Field Definitions (cont'd)		
RTP Packets		Voice signals are transmitted in Realtime Transport Protocol packets. RTP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the 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.

About 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.

<table>
<tr><th colspan="2">Accessing Packetization Time</th></tr>
<tr><th colspan="2">Pulldown</th></tr>
<tr><td colspan="2">
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut/Icon</th><th>Sidebar</th></tr>
<tr><td>none/none</td><td>
</td></tr>
</table>

Packetization Time Screen

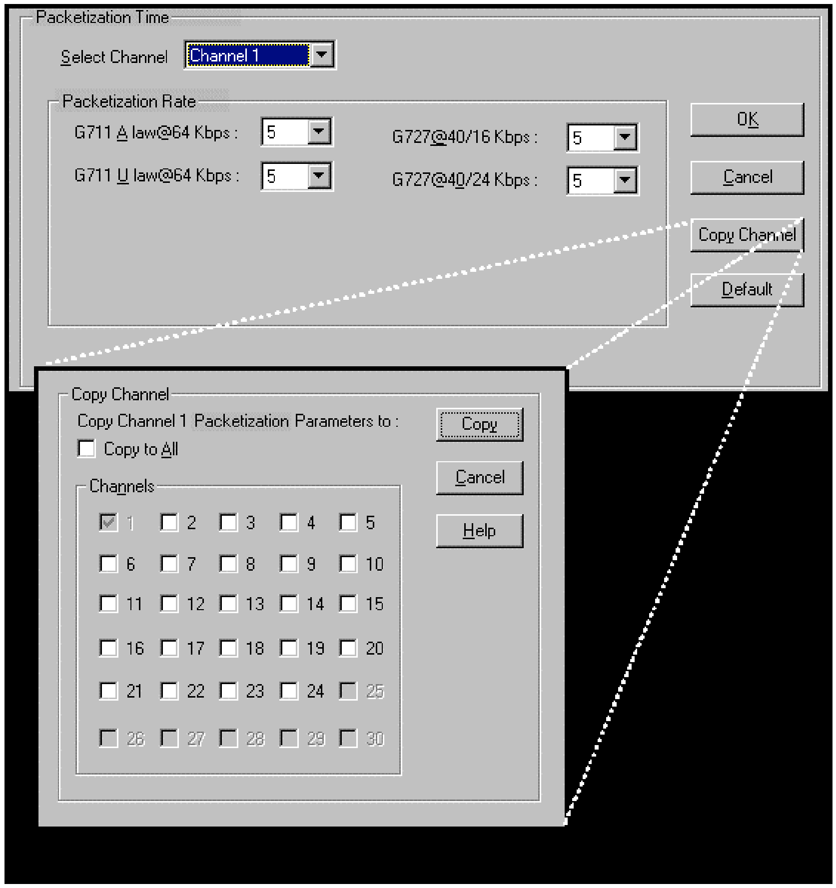


Packetization rates can be set separately for each channel.

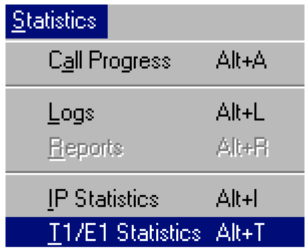
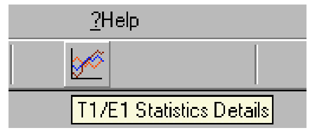
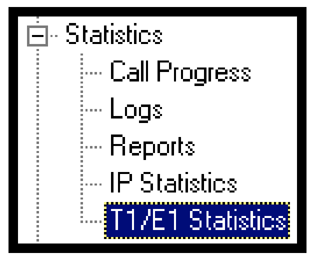
The table below presents the ranges and increments for packetization rates.

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

Once the packetization rate has been set for one channel, it can be copied into other channels.



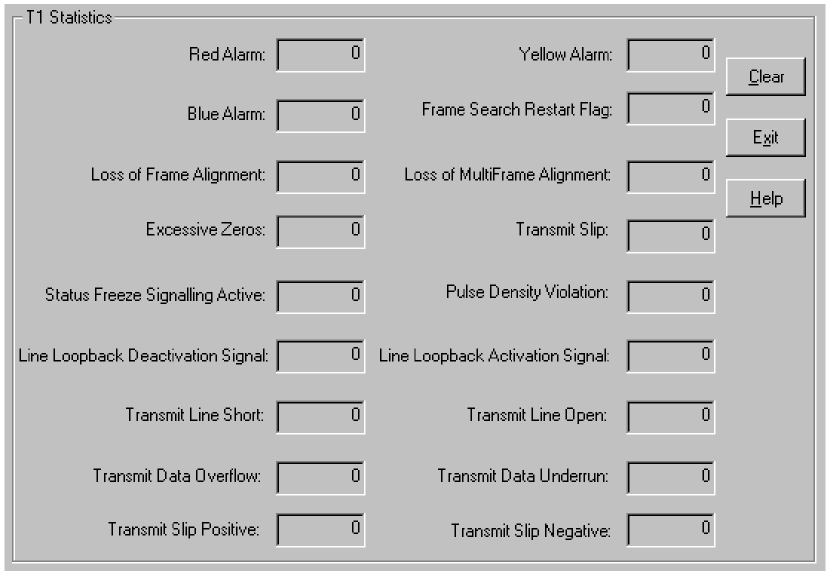
About T1/E1 and BRI Statistics

Accessing T1 Statistics	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
Alt + T	

The T1 and E1 Statistics screens are only accessible and applicable for the MVP2400, MVP2410, and MVP3010.

The BRI statistics screens are only accessible and applicable for the MVP410ST and MVP810ST .

T1 Statistics Screen

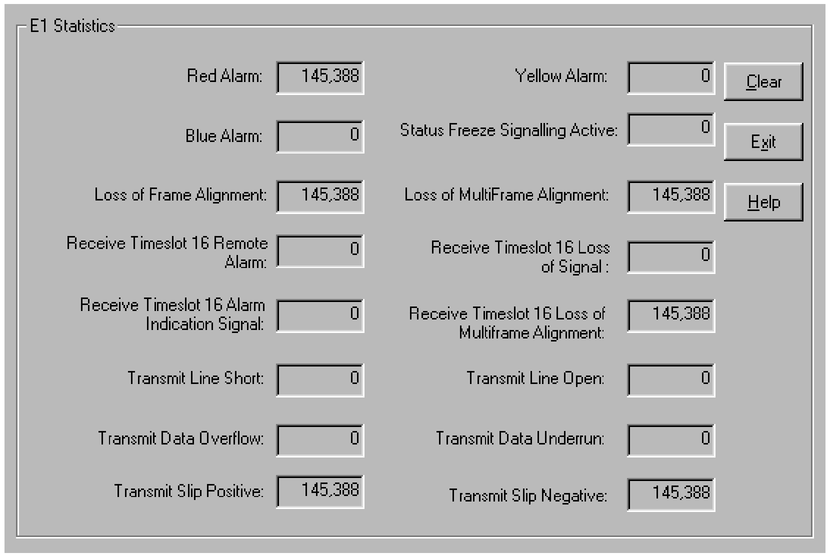


T1 Statistics: Field Definitions		
Field Name	**Values**	**Description**
Red Alarm	Integer tally of alarms counted since last reset.	The alarm condition declared when a device receives no signal or cannot synchronize to the signal being received. A Red Alarm is generated if the incoming data stream has no transitions for 176 consecutive pulse positions.
Blue Alarm	Tally since last reset.	Alarm signal consisting of all 1's (including framing bit positions) which indicates disconnection or failure of attached equipment.
Loss of Frame Alignment	Tally since last reset.	Loss of data frame synchronization.
Excessive Zeroes	Tally since last reset.	Displayed value will increment if consecutive zeroes beyond a set threshold are detected. I.e., tally increments if more than 7 consecutive zeroes in the received data stream are detected under B8ZS line coding, or if 15 consecutive zeroes are detected under AMI line coding.
Status Freeze Signaling Active		Signaling has been frozen at the most recent values due to loss of frame alignment, loss of multiframe alignment or due to a receive slip.
Line Loopback Deactivation Signal		Line loopback deactivation signal has been detected in the receive bit stream.
Transmit Line Short		A short exists between the transmit pair for at least 32 consecutive pulses.
Transmit Data Overflow		For use by MTS Technical Support personnel.
Transmit Slip Positive		The frequency of the transmit clock is less than the frequency of the transmit system interface working clock. A frame is repeated.

T1 Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Yellow Alarm	Tally since last reset.	The alarm signal sent by a remote T1/E1 device to indicate that it sees no receive signal or cannot synchronize on the receive signal.
Frame Search Restart Flag		[To be supplied.]
Loss of MultiFrame Alignment	Tally since last reset.	In D4 or ESF mode, displayed value will increment if multiframe alignment has been lost or if loss of frame alignment has been detected.
Transmit Slip	Tally since last reset.	Slip in transmitted data stream. Slips indicate a clocking mismatch (or lack of synchronization) between T1/E1 devices. When slips occur, data may be lost or repeated.
Pulse Density Violation		The pulse density of the received data stream is below the requirement defined by ANSI T1.403 or more than 15 consecutive zeros are detected.
Line Loopback Activation Signal		The line loopback activation signal has been detected in the received bit stream.
Transmit Line Open		At least 32 consecutive zeros were transmitted.
Transmit Data Underrun		For use by MTS Technical Support Personnel.
Transmit Slip Negative		The frequency of the transmit clock is greater than the frequency of the transmit system interface working clock. A frame is skipped.

T1 Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Bipolar Violation	Integer tally of violation count since last reset.	Two successive pulses of the same polarity have been received and these pulses are not part of zero substitution. On an AMI-encoded line, this represents a line error. On a B8ZS line, this may represent the substitution for a string of 8 zeroes.
Receive Slip	Tally since last reset.	A receive slip (positive or negative) has occurred. Slips indicate a clocking mismatch (or lack of synchronization) between T1/E1 devices. When slips occur, data may be lost or repeated.

E1 Statistics Screen

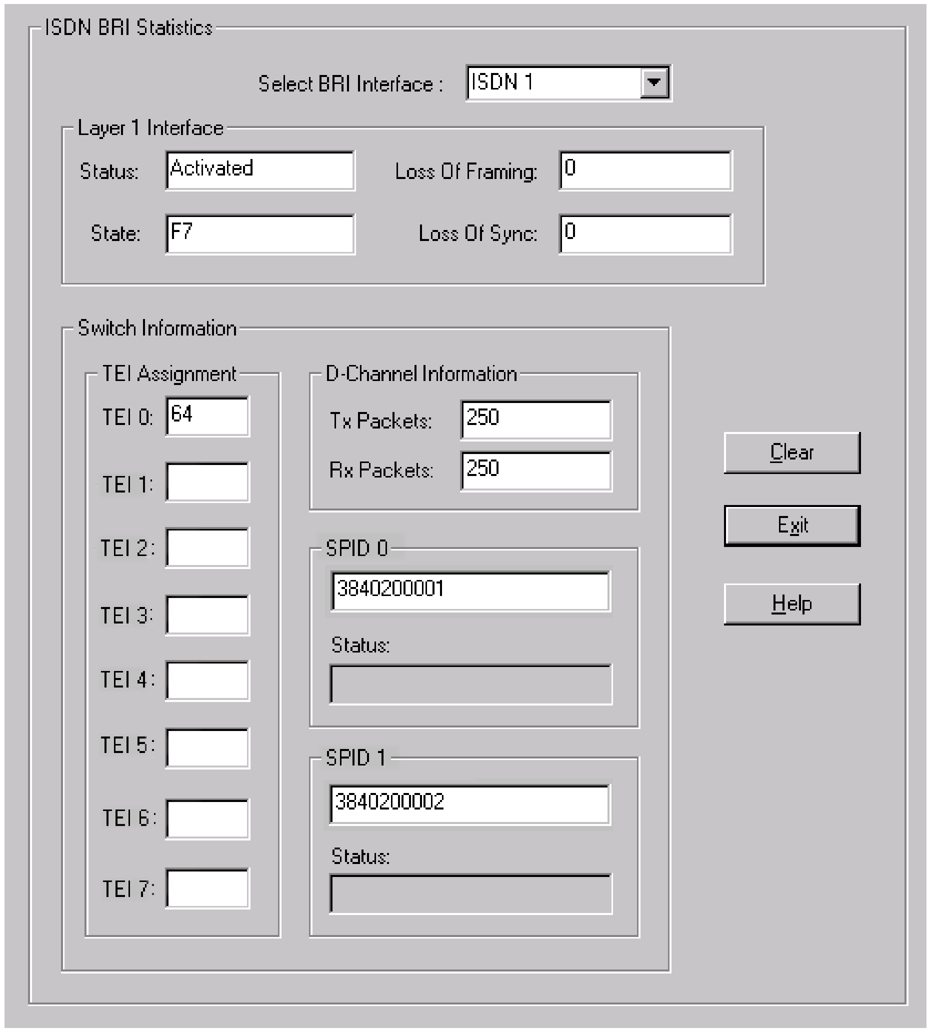


E1 Statistics: Field Definitions		
Field Name	**Values**	**Description**
Red Alarm	Integer tally of alarms counted since last reset.	The alarm condition declared when a device receives no signal or cannot synchronize to the signal being received. A Red Alarm is generated if the incoming data stream has no transitions for 176 consecutive pulse positions.
Blue Alarm	Tally since last reset.	Alarm signal consisting of all 1's (including framing bit positions) which indicates disconnection or failure of attached equipment.
Loss of Frame Alignment	Tally since last reset.	Loss of data frame synchronization.

E1 Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Receive Timeslot 16 Alarm Indication Signal		Detected alarm indication signal in timeslot 16 according to ITU-T G.775. Indicates the incoming time slot 16 contains less than 4 zeros in each of two consecutive time slot 16 multiframe periods.
Transmit Line Short		A short exists between the transmit pair for at least 32 consecutive pulses.
Transmit Data Overflow		For use by MTS personnel.
Transmit Slip Positive		The frequency of the transmit clock is less than the frequency of the transmit system interface working clock. A frame is repeated.
Yellow Alarm	Tally since last reset.	The alarm signal sent by a remote T1/E1 device to indicate that it sees no receive signal or cannot synchronize on the receive signal.
Status Freeze Signaling Active		Signaling has been frozen at the most recent values due to loss of frame alignment, loss of multiframe alignment or due to a receive slip.
Loss of MultiFrame Alignment	Tally since last reset.	In D4 or ESF mode, displayed value will increment if multiframe alignment has been lost or if loss of frame alignment has been detected.
Receive Timeslot 16 Loss of Signal		The time slot 16 data stream contains all zeros for at least 16 contiguously received time slots.

E1 Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Receive Timeslot 16 Loss of MultiFrame Alignment		The framing pattern '0000' in 2 consecutive CAS multiframes were not found or in all time slot 16 of the previous multiframe all bits were reset.
Transmit Line Open		At least 32 consecutive zeroes were transmitted.
Transmit Data Underrun		For use by MTS Technical Support Personnel.
Transmit Slip Negative		The frequency of the transmit clock is greater than the frequency of the transmit system interface working clock. A frame is skipped.
Bipolar Violation	Integer tally of violation count since last reset.	Bipolar Violation (or BPV) refers to two successive pulses of the same polarity on the E1 line. On an AMI-encoded line, this represents a line error. On a B8ZS line, this may represent the substitution for a string of 8 zeroes.
Excessive Zeroes	Tally since last reset.	Displayed value will increment if consecutive zeroes beyond a set threshold are detected. I.e., tally increments if more than 7 consecutive zeroes in the received data stream are detected under B8ZS line coding, or if 15 consecutive zeroes are detected under AMI line coding.
Transmit Slip	Tally since last reset.	Slip in transmitted data stream. Slips indicate a clocking mismatch (or lack of synchronization) between T1/E1 devices. When slips occur, data may be lost or repeated.
Receive Slip	Tally since last reset.	Slip in received data stream. Slips indicate a clocking mismatch (or lack of synchronization) between T1/E1 devices. When slips occur, data may be lost or repeated.

ISDN BRI Statistics Screen



ISDN BRI Statistics: Field Definitions		
Field Name	**Values**	**Description**
Select BRI Interface	ISDNn For n=1-2 (410ST) For n-1-4 (810ST)	In this field, you can choose the ISDN port for which you want to view the status. The 410ST has two ISDN –BRI ports (or "interfaces"); the 810ST has four ISDN-BRI ports (or "interfaces"). Each interface has two channels.
Layer 1 Interface		
Status	inactive (F1), sensing (F2), deactivated (F3), awaiting signal (F4), identifying input (F5), synchronized (F6), activated (F7), lost framing (F8), deactive (G1), pending activation (G2), active (G3), pending deactivation (G4)	Shows the current Layer 1 status of the ISDN connection. Each status description (inactive, sensing, etc.) corresponds to a particular "state" label (F1-F8 and G1-G4).
State	F1-F8 (for Terminal mode ports), G1-G4 (for Network mode ports)	Shows the I.430 state name for Layer 1. An "F" state name indicates this port is in Terminal mode (F1-F8), as set in the **ISDN BRI Parameters** screen. A "G" state name indicates that this port is in Network mode (G1-G4), as set in the **ISDN BRI Parameters** screen.
Loss Of Framing	integer	Shows the number of lost-framing events on the ISDN physical layer.
Loss of Sync	integer	Shows the number of lost-synchronization events on the ISDN physical layer.

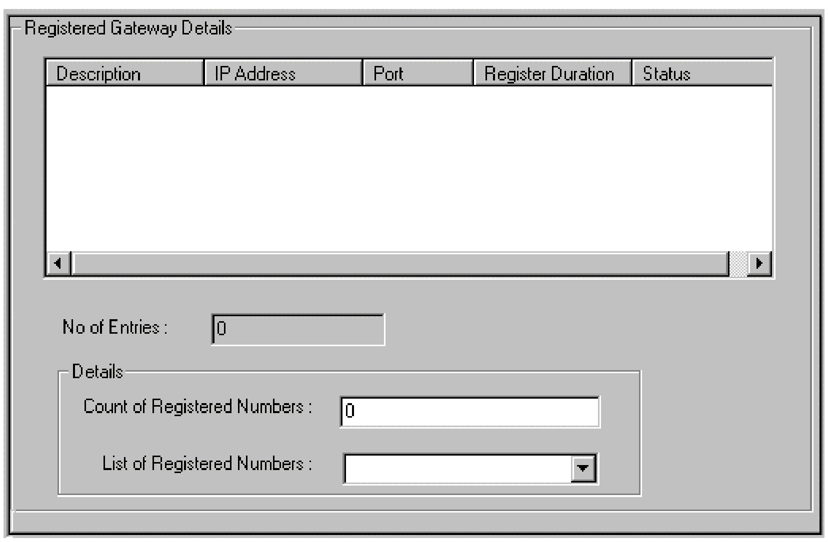
ISDN BRI Statistics: Field Definitions (continued)		
Field Name	**Values**	**Description**
Switch Information: TEI Assignment		
TEI 0 through TEI 7	0-63 (point-to-point assignments) 64-126 (automatic assignments)	Displays the value for each TEI assigned to the BRI port. The TEI (Terminal Endpoint Identifier) uniquely identifies each device connected to the ISDN physical layer.
Switch Information: D-Channel Information		
Tx Packets	0 to 4294967295	Shows the number of packets transmitted on the channel. When the value exceeds 4294967295 packets, it will reset to zero and continue counting.
Rx Packets	0 to 4294967295	Shows the number of packets received on the channel. When the value exceeds 4294967295 packets, it will reset to zero and continue counting.
Switch Information: SPID 0		
(SPID 0 *number*)	numeric, 3 to 20 digits	A SPID (Service Profile Identifier) is assigned by the ISDN provider and pertains to one channel of the BRI interface (port), in this case channel 0. The SPID identifies an ISDN terminal uniquely. The SPID associates a set of services (features) with the terminal. (In Terminal mode the provider is a telco or PBX. In Network mode MultiVOIP is the provider.) A SPID is only used when the "Country" field is set to "USA" in the **ISDN BRI Parameters** screen.
Status	Not Checked, Correct, Incorrect	Indicates whether SPID0 is correct, incorrect, or not being checked.

ISDN BRI Statistics: Field Definitions (continued)		
Field Name	**Values**	**Description**
Switch Information: **SPID 1**		
(SPID 1 *number*)	numeric	SPID for channel 1 of the BRI interface. Otherwise, same as SPID0 description above.
Status	Not Checked, Correct, Incorrect	Indicates whether SPID1 is correct, incorrect, or not being checked.
"Clear" button		Clears (sets to zero) all **ISDN BRI Statistics** fields with numeric tally values (these are Loss of Framing, Loss of Sync, Tx Packets, Rx Packets).

About 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 **PhoneBook Configuration** screen and in the **Add/Edit Outbound PhoneBook** screen.

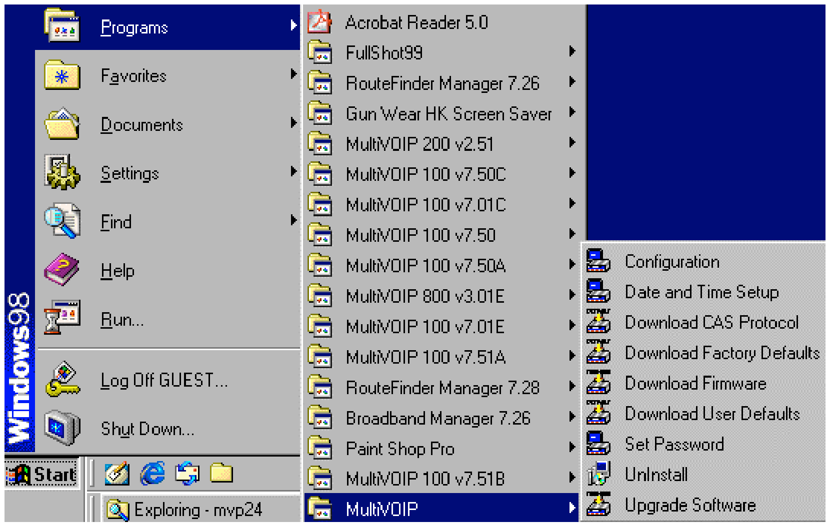
Accessing Registered Gateway Details	
Pulldown	**Icon**
Shortcut	**Sidebar**
	



Registered Gateway Details: Field Definitions		
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 (e.g., city, building, etc.) and perhaps even its location in an equipment rack.
IP Address	n.n.n.n, for n = 0-255	The RAS address for the gateway.
Port		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		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.

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 ____** | …). In this section, we describe the software functions available on this menu.



Several basic software functions are accessible from the MultiVoip software menu, as shown below.

MultiVOIP Program Menu	
Menu Selection	**Description**
Configuration	Select this to enter the Configuration program where values for IP, telephony, and other parameters are set.
Date and Time Setup	Select this for access to set calendar/clock used for data logging.
Download CAS Protocol	Telephony CAS files are for Channel Associated Signaling. There are many CAS files, some labeled for specific functionality, others for countries or regions where certain telephony attributes are standard.

MultiVOIP Program Menu (cont'd)	
Menu Selection	**Description**
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 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 GUI, web browser GUI, & FTP server). Only the FTP Server function *requires* a password for access. The FTP Server function also requires that a username be established 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 invoked).
Upgrade Software	Loads firmware (including H.323 stack) and factory default settings from the controller PC to the MultiVOIP unit.

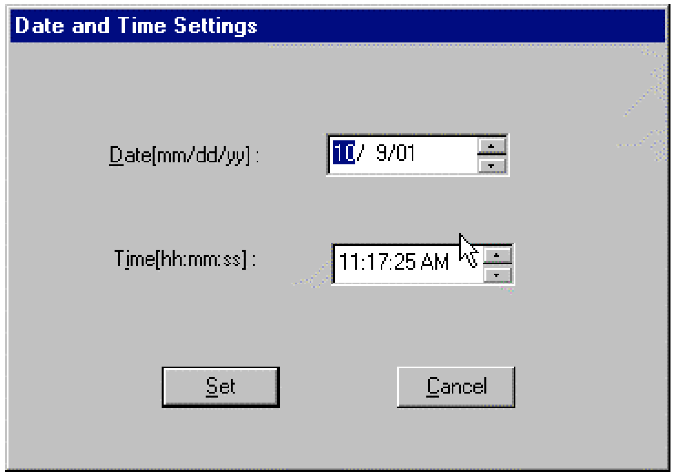
"Downloading" here refers to transferring program files from the PC to the nonvolatile "flash" memory of the MultiVOIP. Such transfers are made via the PC's serial port. This can be understood as a "download" from the perspective of the MultiVOIP unit.

When new versions of the MultiVoip software become available, they will be posted on MultiTech's web or FTP sites. Although transferring updated program files from the MultiTech web/FTP site 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, updated firmware must be downloaded from the MultiTech web/FTP site to the PC before it can be loaded from the PC to the MultiVOIP.

Date and Time Setup

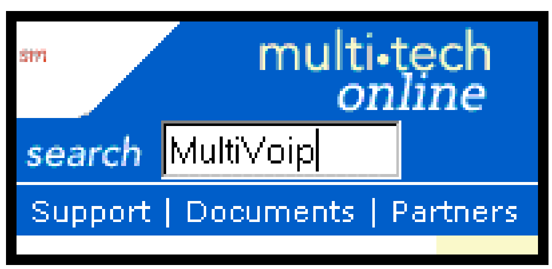
The dialog box below allows you to set the time and date indicators of the MultiVOIP system.



Obtaining Updated Firmware

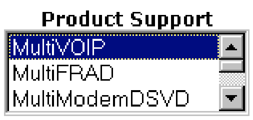
Generally, updated firmware must be downloaded from the MultiTech web/FTP site to the user's PC before it can be downloaded from that PC to the MultiVOIP.

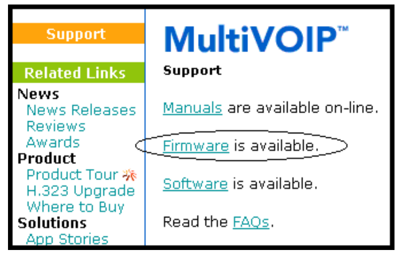
Note that the structure of the MultiTech web/FTP site may change without notice. However, firmware updates can generally be found using standard web techniques. For example, you can access updated firmware by doing a search or by clicking on **Support**.



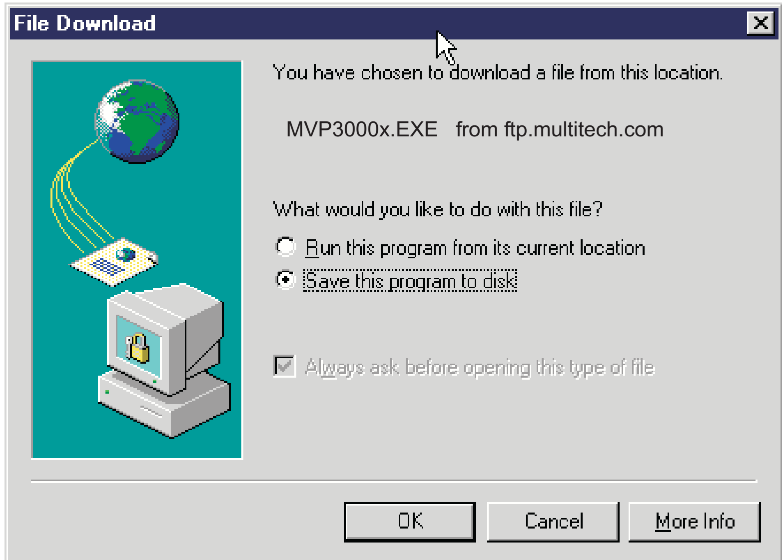
If you conduct a search, for example, on the word "MultiVoip," you will be directed to a list of firmware that can be downloaded.

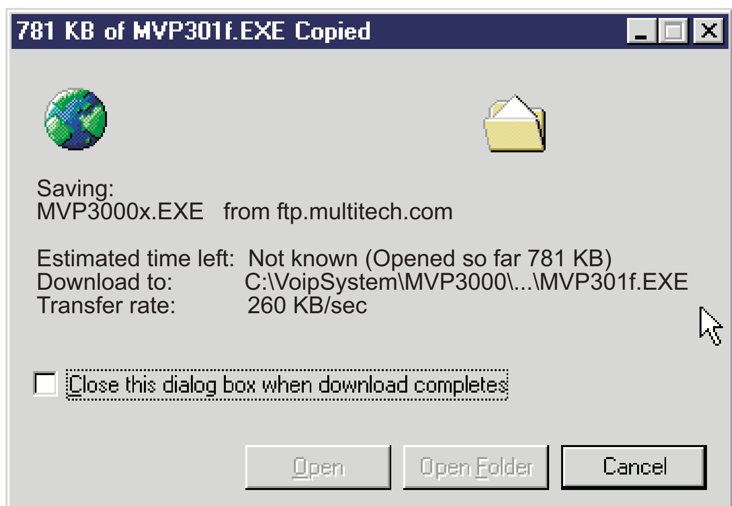
If you choose **Support**, you can select "MultiVoip" in the **Product Support** menu and then click on **Firmware** to find MultiVOIP resources.



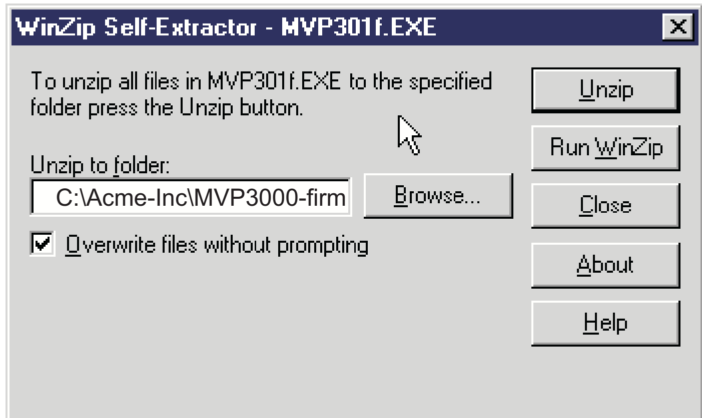


Once the updated firmware has been located, it can be downloaded from the web/ftp site using normal PC/Windows procedures. While the next 3 screens below pertain to the MVP3010, similar screens will appear for any MultiVOIP model described in this manual.





Generally, the firmware file will be a self-extracting compressed file (with .zip extension), which must be expanded (decompressed, or “unzipped”) on the user’s PC in a user-specified directory.



Implementing a Software Upgrade

Beginning with the 4.03/6.03 software release, MultiVOIP software can be upgraded locally using a single command at the MultiVOIP Windows GUI, 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 GUI, firmware and factory default settings can also be transferred from controller PC to MultiVOIP piecemeal using separate commands.

When using the MultiVOIP web browser GUI 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 piecemeal software upgrade (whether from the Windows GUI or web browser GUI), follow these steps in order:

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

When upgrading firmware, the software commands "Download Firmware," and "Download Factory Defaults" must be implemented in order, else 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 ____ x.xx**. The final expression, x.xx, is the firmware version number. In the illustration below, the firmware version is 4.00a, made for the E1 MultiVOIP (MVP3010).

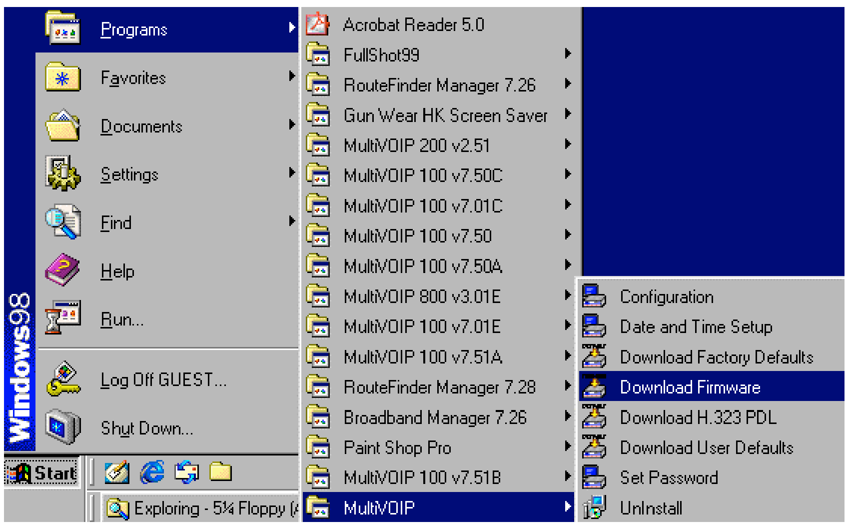
When a new firmware version is installed, the MultiVOIP software can be upgraded in one step using the **Upgrade Software** command, or piecemeal using the **Download Firmware** command and the **Download Factory Defaults** command.

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

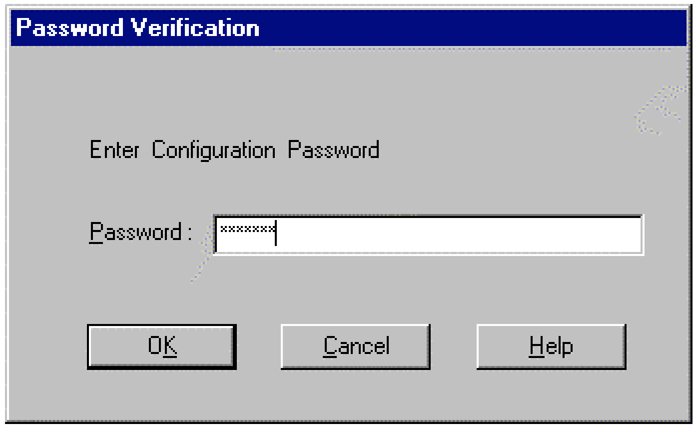
Downloading Firmware

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 invoke the Download Factory Defaults command, go to **Start | Programs | MVP____ x.xx | Download Firmware**.

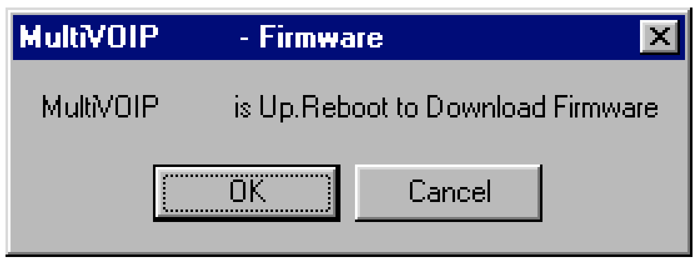


3. If a password has been established, the **Password Verification** screen will appear.



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

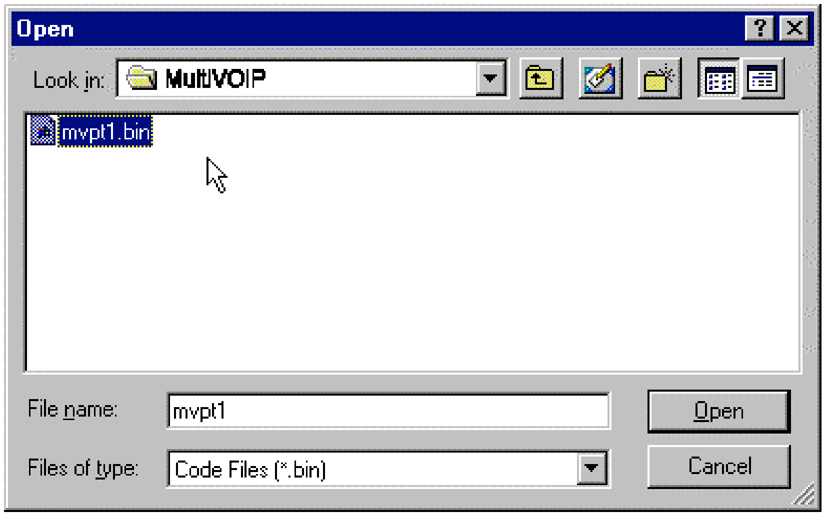
4. The **MultiVOIP ___- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"



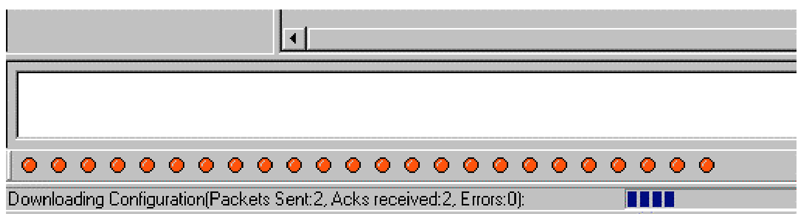
Click **OK** to download the firmware.

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.



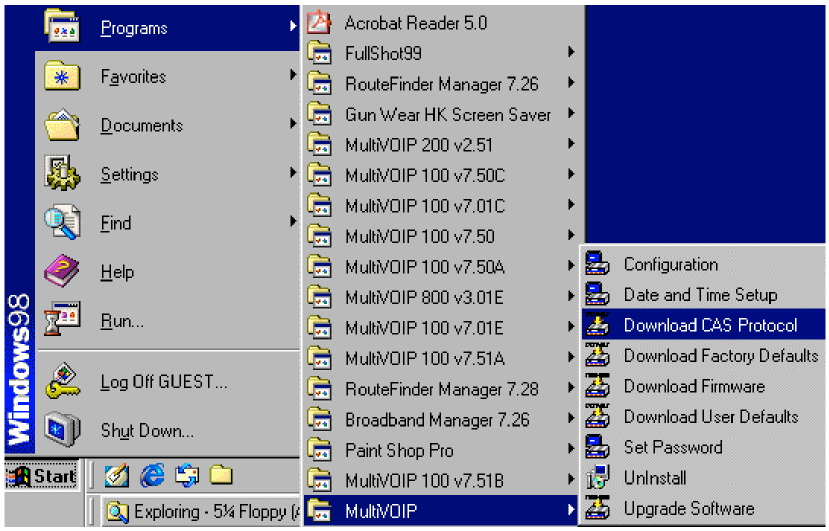
The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

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

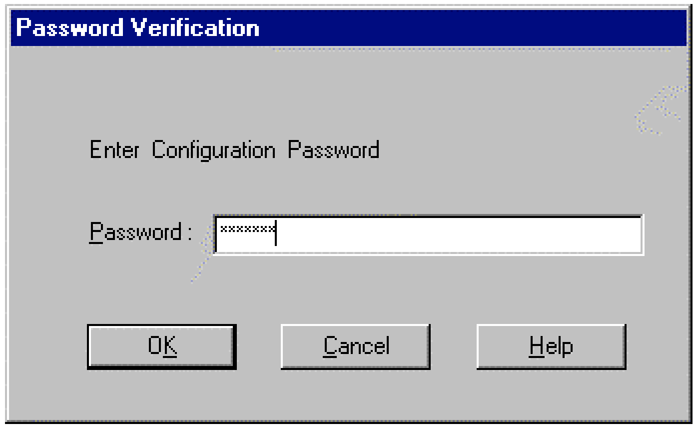
Downloading CAS Protocols

1. The MultiVoip Configuration program must be off when invoking the **Download CAS Protocol** command. If it is on, the command will not work.

2. To invoke the **Download H.323 PDL** command, go to **Start | Programs | MVP____ x.xx | Download H.323 PDL**.

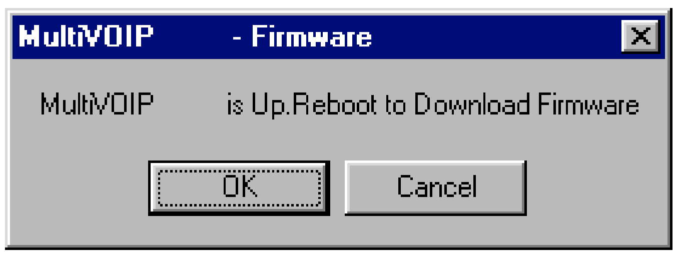


3. If a password has been established, the **Password Verification** screen will appear.



Type in password and click **OK**.

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



Click **OK** to download the CAS Protocol file(s) to the MultiVOIP.

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

5. The program will locate the CAS protocol file in the MultiVOIP directory. Highlight the correct (newest) file and click **Open**.

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

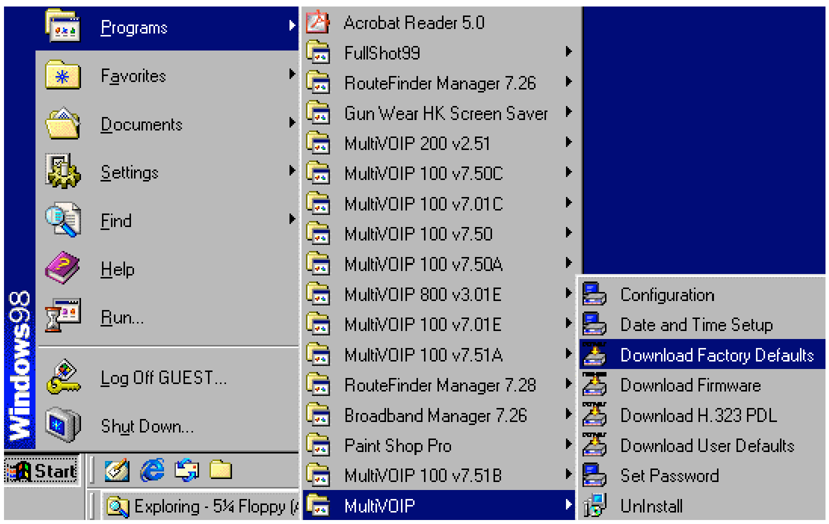
   The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

7. The **Download CAS Protocol** procedure is complete.

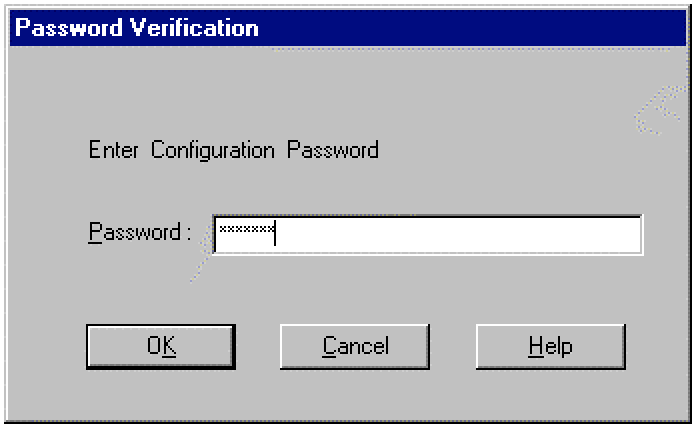
Downloading Factory Defaults

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 invoke the **Download Factory Defaults** command, go to **Start | Programs | MVP____ x.xx | Download Factory Defaults**.

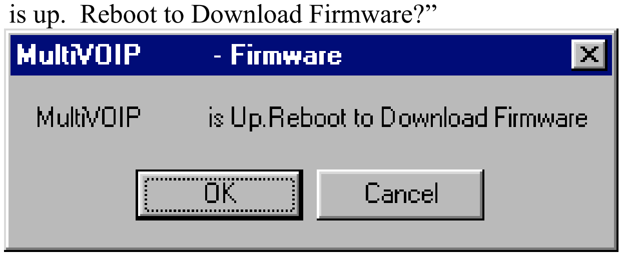


3. If a password has been established, the **Password Verification** screen will appear.



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

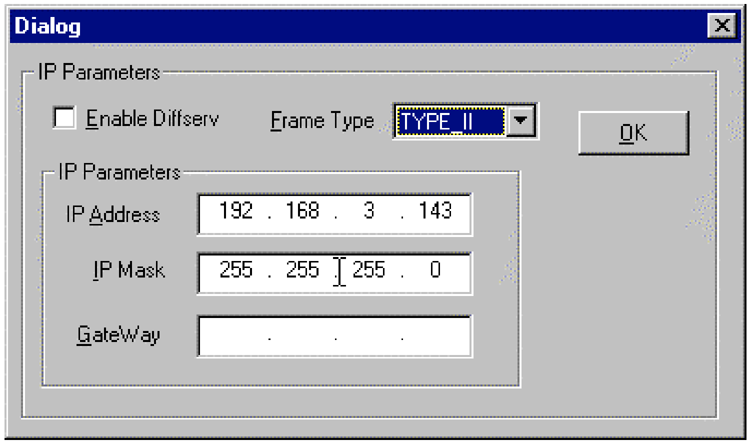
4. The **MVP____- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"



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

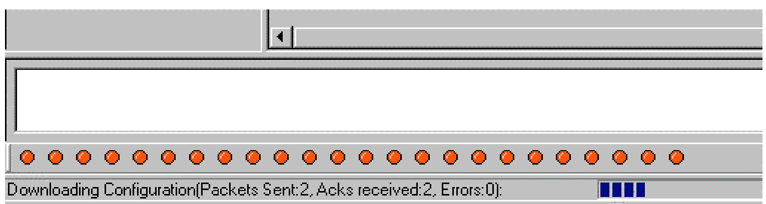
The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

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.

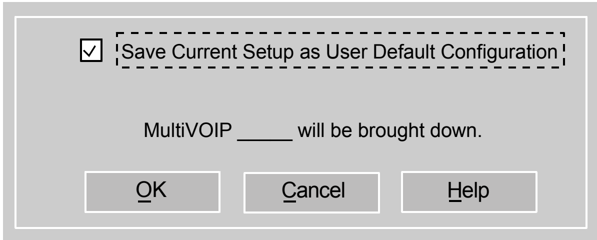
7. The **Download Factory Defaults** procedure is complete.

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.

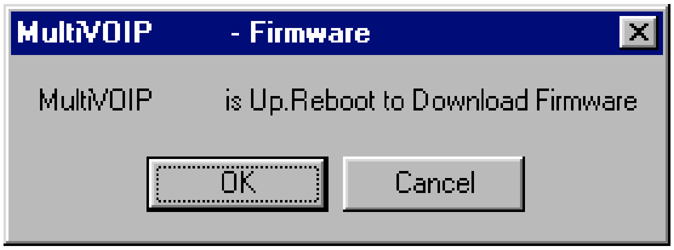
1. Before you can invoke the Download User Defaults command, you must first save a set of configuration parameters by using the **Save Setup** command in the sidebar menu of the MultiVOIP software.

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**.



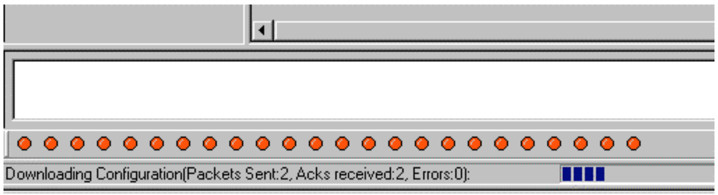
A user default file will be created.

3. The **MVP____- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"

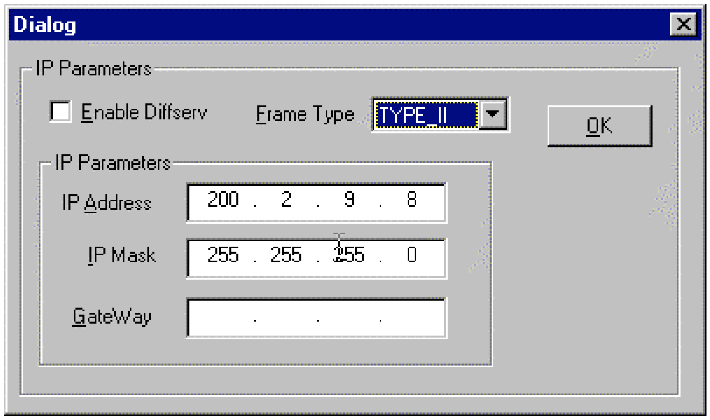


Click **OK** to download the factory defaults. The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

4. Progress bars will appear during the file transfer process.



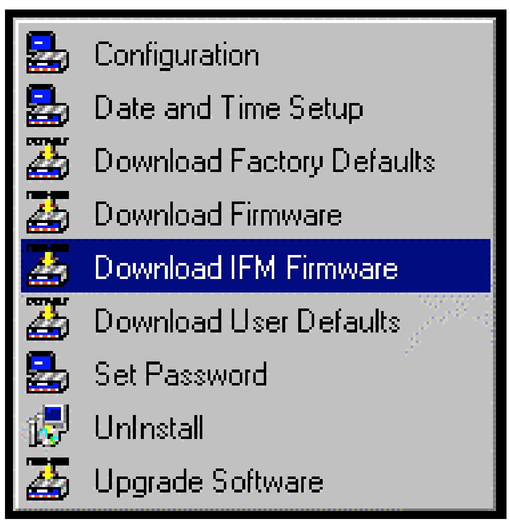
5. When the file transfer process is complete, the **Dialog-- IP Parameters** screen will appear.



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

Downloading IFM Firmware

The Download IFM Firmware command applies only to the MVP210/410/810 and MVP210G/410G/810G models. This command transfers firmware to the telephony interface modules of each voice channel. These firmware modules handle the physical interface (FXS, FXO and E&M) to the attached analog telephony equipment.



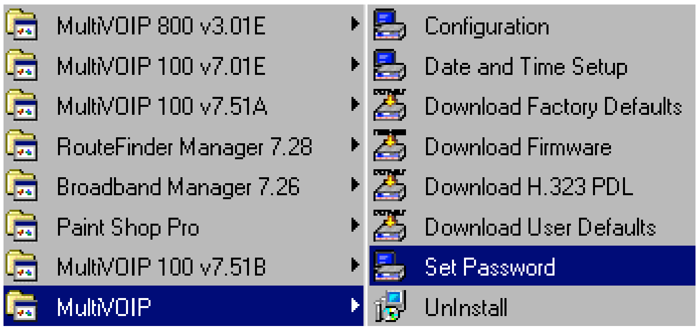
Setting a Password (Windows GUI)

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 GUI.

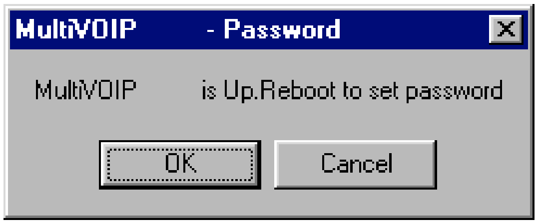
NOTE: Record your user name and password in a safe place. If the password is lost, forgotten, or unretrievable, the user must contact MultiTech Tech Support in order to resume use of the MultiVOIP unit.

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 invoke the **Set Password** command, go to **Start | Programs | MVP____ x.xx | Set Password**.



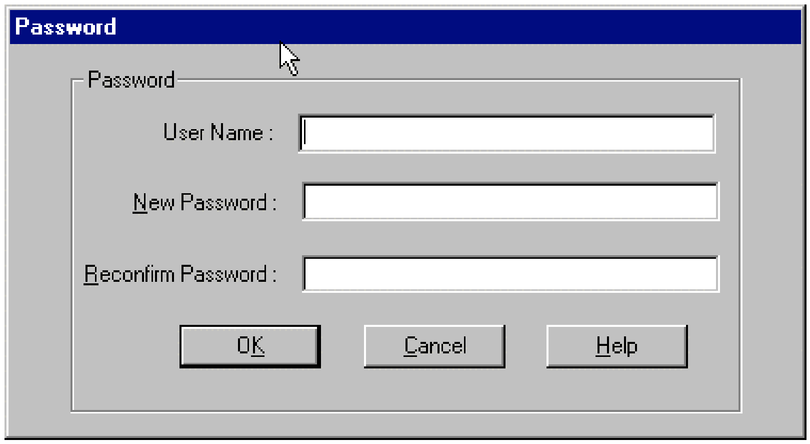
3. You will be prompted to confirm that you want to establish a password, which will entail rebooting the MultiVOIP (which is done automatically).



Click **OK** to proceed with establishing a password.

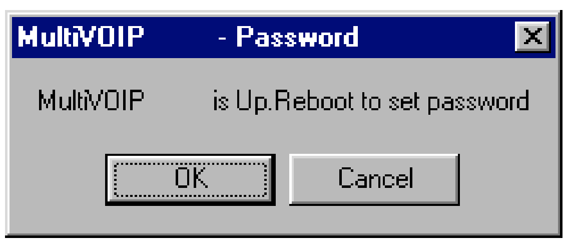
4. The **Password** screen will appear. If you intend to use the FTP Server function that is built into the MultiVOIP, enter a user name. (A User Name is not needed to access the local Windows GUI, the web browser GUI, or the commands in the **Program** group.) Type your password in the **Password**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

NOTE: Be sure to write down your password in a convenient but secure place. If the password is forgotten, contact MultiTech Technical Support for advice.



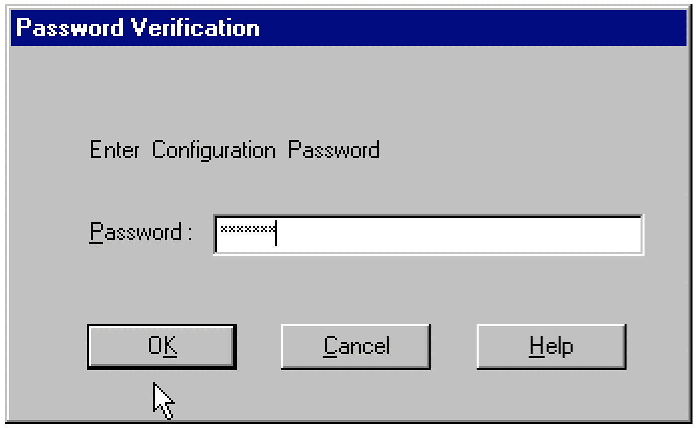
Click **OK**.

5. A message will appear indicating that a password has been set successfully.



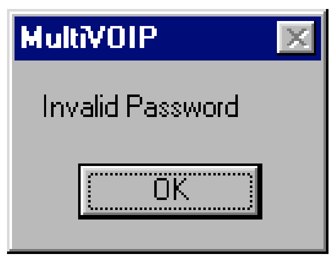
After the password has been set successfully, the MultiVOIP will re-boot itself and, in so doing, its **BOOT** LED will light up.

6. After the password has been set, the user will be required to enter the password to gain access to the web browser GUI 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 at launch of program, the program will simply shut down if **CANCEL** is selected.

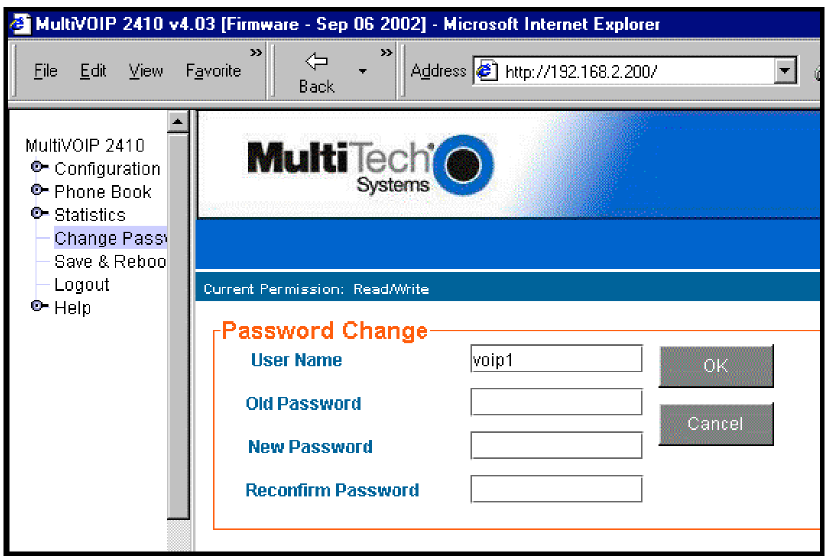
The MultiVOIP program will produce an error message if an invalid password is entered.



Setting a Password (Web Browser GUI)

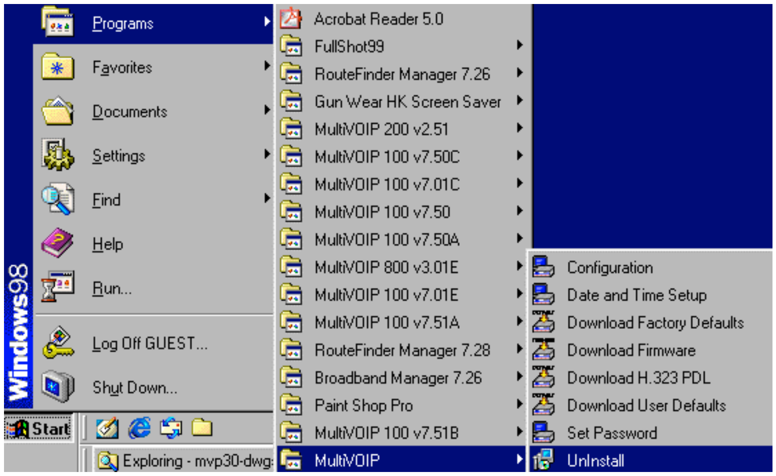
Setting a password is optional when using the MultiVOIP web browser GUI. Only one password can be assigned and it works for all MultiVOIP software functions (Windows GUI, web browser GUI, FTP server, and all Program menu commands, e.g., Upgrade Software – only the FTP Server function requires a User Name in addition to the password). After a password has been set, that password is required to access the MultiVOIP web browser GUI.

NOTE: Record your user name and password in a safe place. If the password is lost, forgotten, or unretrievable, the user must contact MultiTech Tech Support in order to resume use of the MultiVOIP web browser GUI.

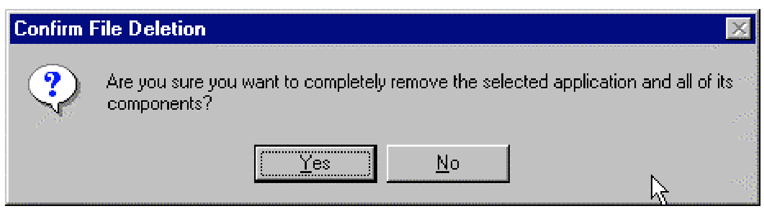


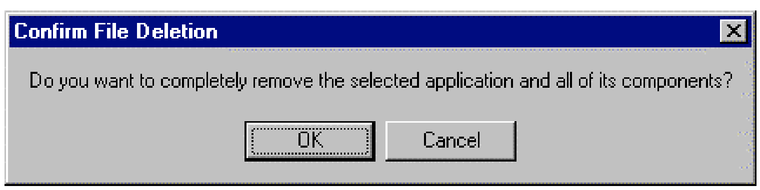
Un-Installing the MultiVOIP Software

1. To un-install the MultiVOIP configuration software, go to **Start** | **Programs** and locate the MultiVOIP entry. Select **Uninstall MVP____ vx.xx** (versions may vary).

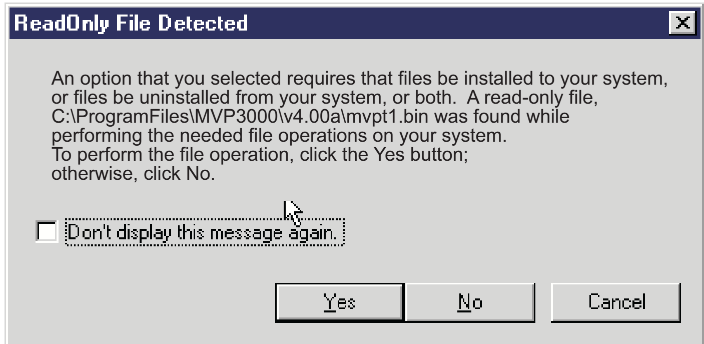


2. Two confirmation screens will appear. Click **Yes** and **OK** when you are certain you want to continue with the uninstallation process.

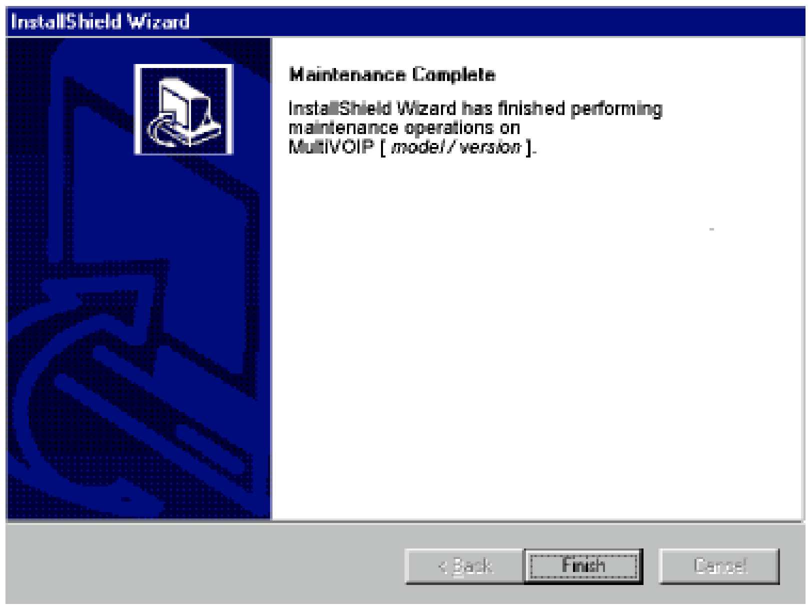




3. A special warning message similar to that shown below may appear for the MultiVOIP software's ".bin" file. Click **Yes**.



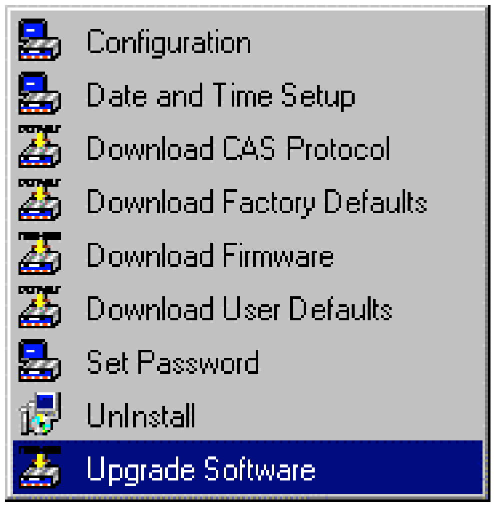
4. A completion screen will appear.



Click **Finish**.

Upgrading Software

As noted earlier (see the section *Implementing a Software Upgrade* above), the Upgrade Software command transfers, from the controller PC to the MultiVOIP unit, firmware (including the H323 stack) and factory default configuration settings. As such, **Upgrade Software** implements the functions of both **Download Firmware** and **Download Factory Defaults** in a single command.



FTP Server File Transfers ("Downloads")

With the 4.03/6.03 software release, MultiTech has built an FTP server into the MultiVOIP unit. Therefore, file transfers from the controller PC to the voip unit can be done using an FTP client program or even using a browser (e.g., Internet Explorer or Netscape, used in conjunction with Windows Explorer).

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, i.e., 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.

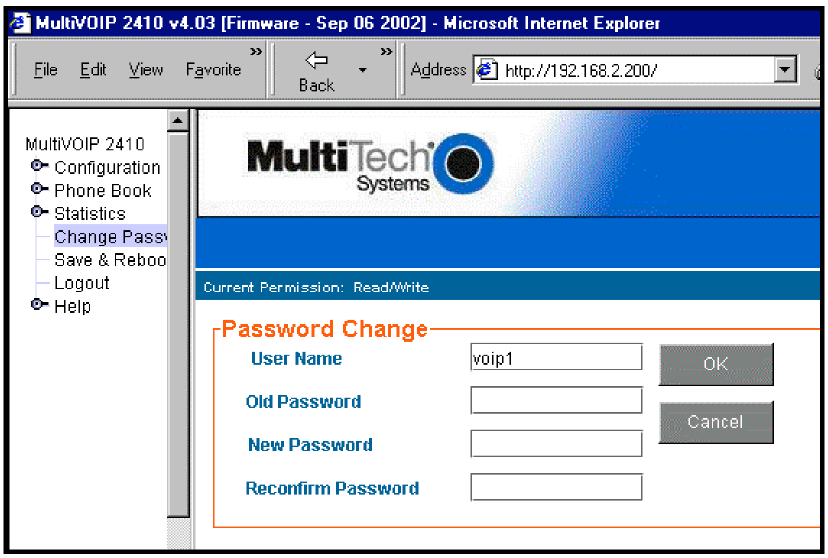
The phonebook downloading feature greatly reduces the data-entry required to establish inbound and outbound phonebooks for the voip units within a system. Although each MultiVOIP unit will require some unique phonebook entries, most will be common to the entire voip system. After the phonebooks for the 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 functionality in the MultiVOIP, follow these directions.

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.

IP Address of Control PC	____ . ____ . ____ . ____
IP Address of voip unit #1	____ . ____ . ____ . ____
:	: : : :
IP address of voip unit #n	____ . ____ . ____ . ____

2. **Establish User Name and Password**. You must establish a user name and (optionally) a password for contacting the voip over the IP network. (When connection is made via a local serial connection between the PC and the voip unit, no user name is needed.)

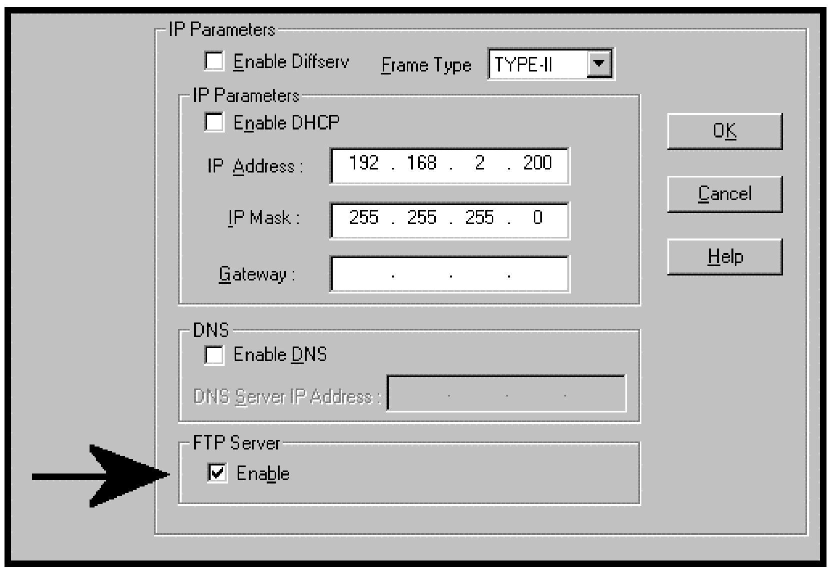


As shown above, the username and password can be set in the web GUI as well as in the Windows GUI.

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 (e.g., Netscape or Internet Explorer) in conjunction with a local Windows browser a (e.g., 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 MultiTech 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. Examples here show use of both programs.

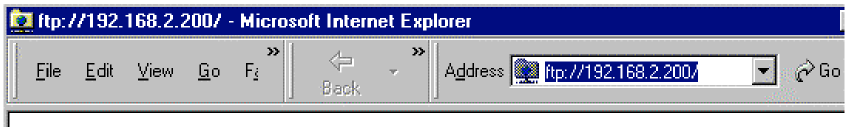
4. **Enable FTP Functionality**. Go to the **IP Parameters** screen and click on the “FTP Server: Enable” box.



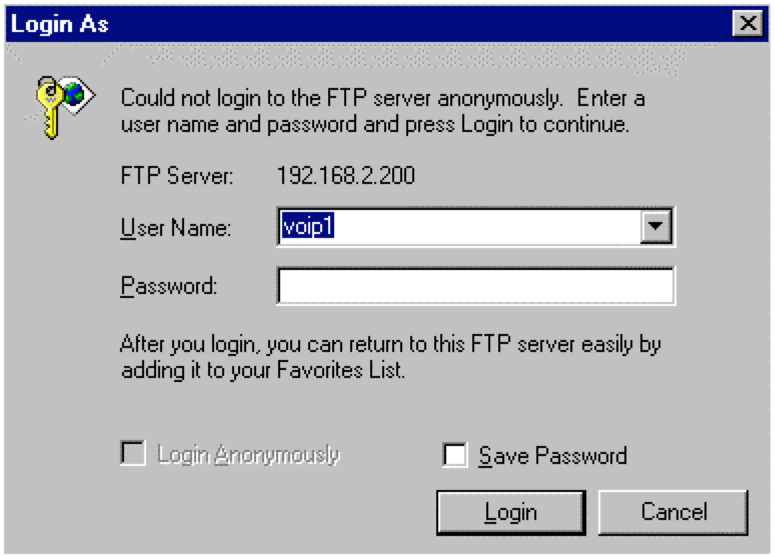
5. **Identify Files to be Updated**. Determine which files you want to update. Six types of files can be updated using the FTP feature. In some cases, the file to be transferred will have "Ftp" as the part of its filename just before the suffix (or extension). So, for example, the file "mvpt1Ftp.bin" 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.

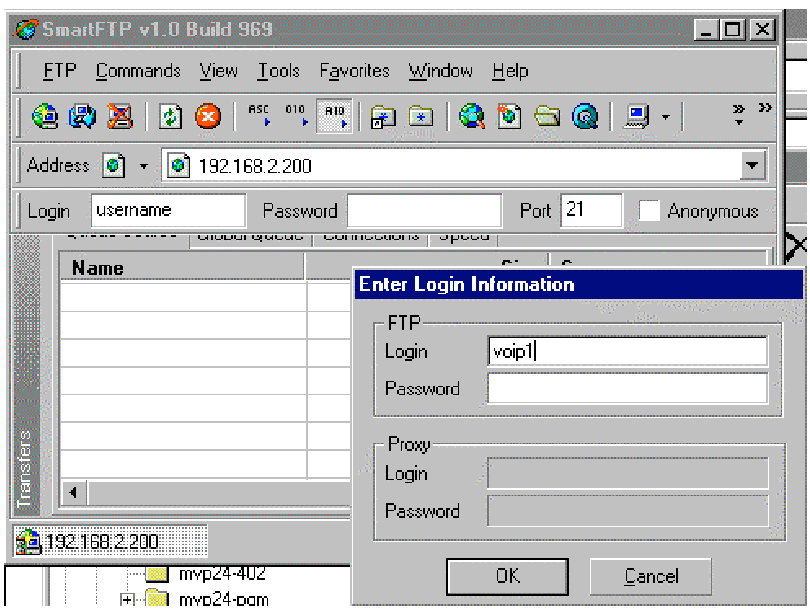
File Type	File Names	Description
firmware "bin" file	mvpt1Ftp.bin	This is the MultiVOIP firmware file. Only one file of this type will be in the directory.
factory defaults	fdefFtp.cnf	This file contains factory default settings for user-changeable configuration parameters. Only one file of this type 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.
H323 PDL file		This file is specific to the particular version of the H.323 standard being used. This file rarely needs to be updated.
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 GUI within the MultiVOIP unit).



7. **Log In**. Use the User Name and password established in item #2 above. The login screens will differ depending on whether the FTP file transfer is to be done with a web browser (see first screen below) or with an FTP client program (see second screen below).



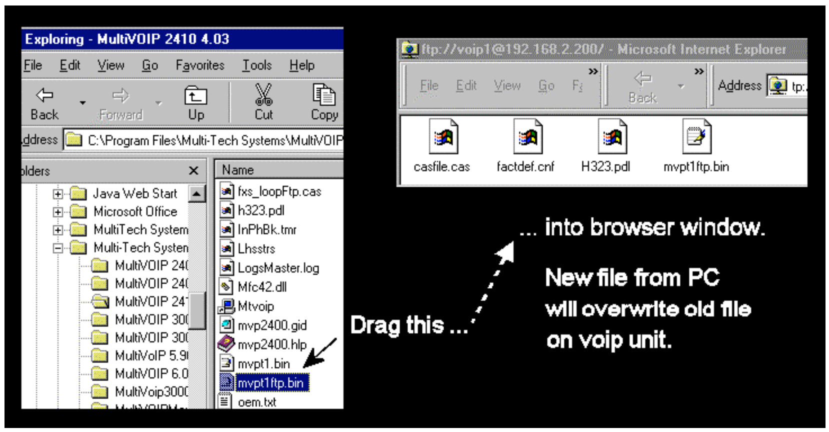


8. **Invoke Download**. Downloading can be done with a web browser or with an FTP client program.

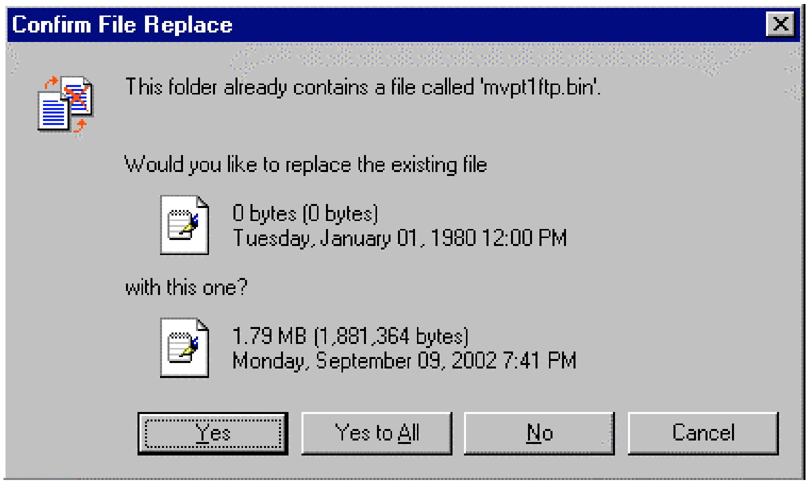
8A. Download with Web Browser.

8A1. 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).

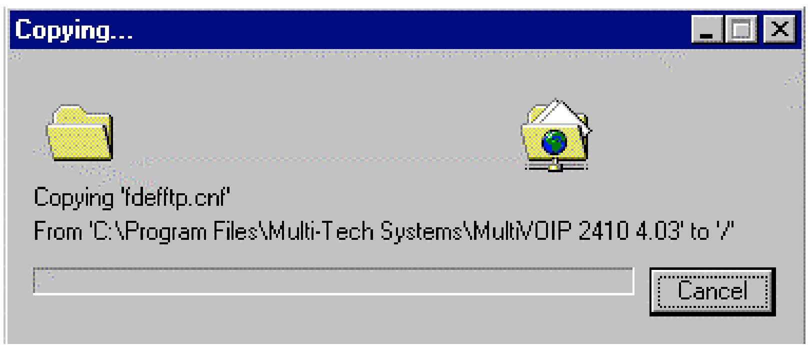
8A2. Drag-and-drop files from the local Windows browser (e.g., Windows Explorer) to the web browser.



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



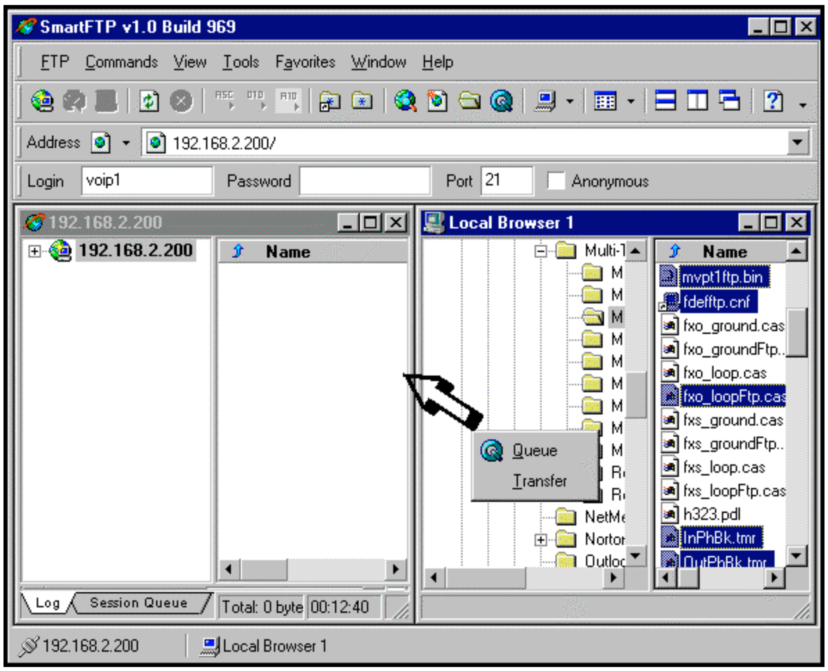
File transfer between PC and voip will look like transfer within voip directories.



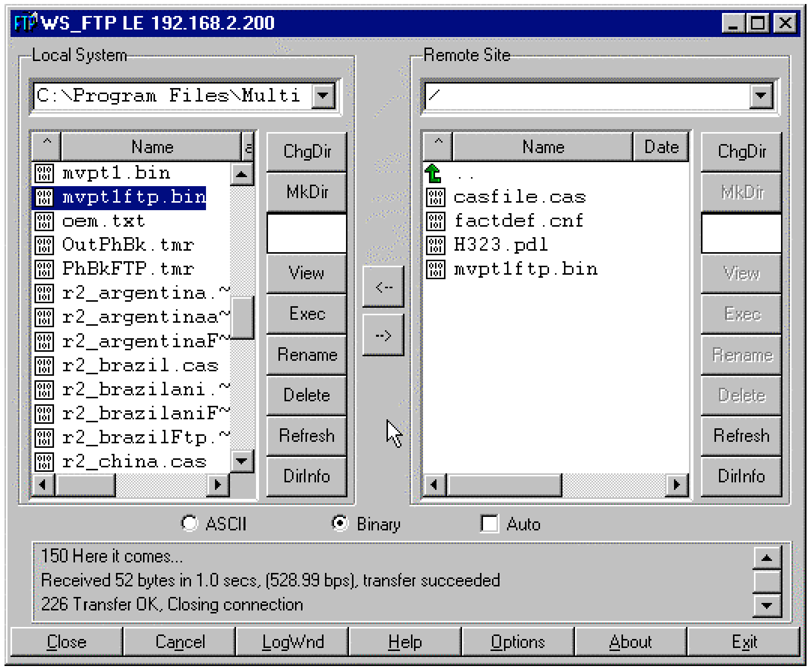
8B. Download with FTP Client Program.

8B1. 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).

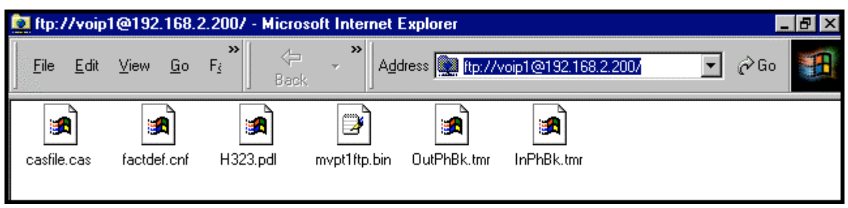
8B2. In the FTP client program window, drag-and-drop files from the local browser pane to the pane for the MultiVOIP FTP server. FTP client GUI operations vary. In some cases, you can choose between immediate and queued transfer. In some cases, there may be automated capabilities to transfer to multiple destinations with a single command.



Some FTP client programs are more graphically oriented (see previous screen), while others (like the "WS-FTP" client) are more text oriented.

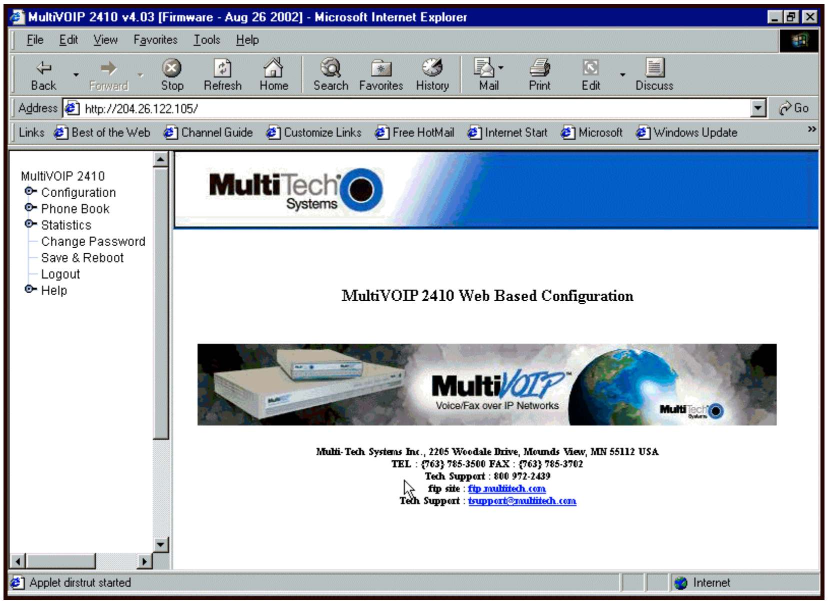


9. **Verify Transfer**. The files transferred will appear in the directory of the MultiVOIP.



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 GUI.

Web Browser Interface

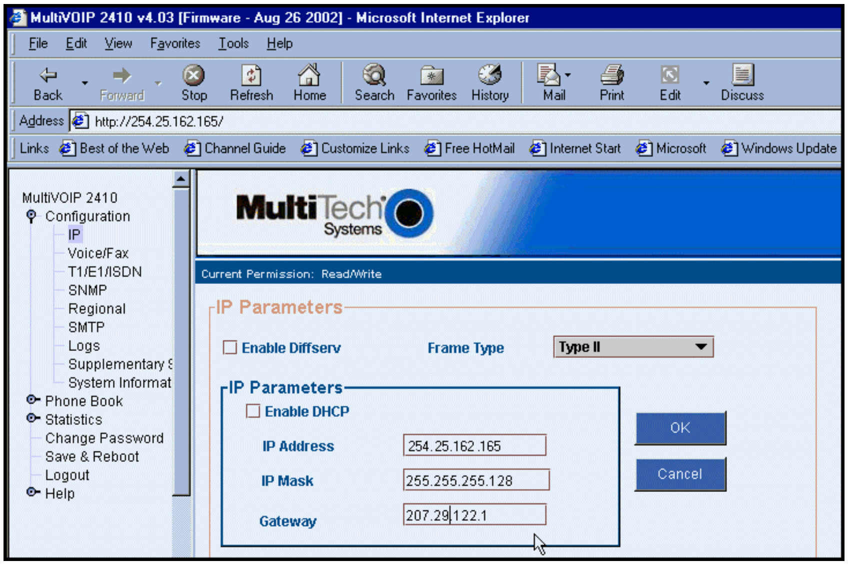


Beginning with the 4.03/6.03 software release, you can control the MultiVOIP unit with a graphic user interface (GUI) based on the common web browser platform. Qualifying browsers are InternetExplorer6 and Netscape6.

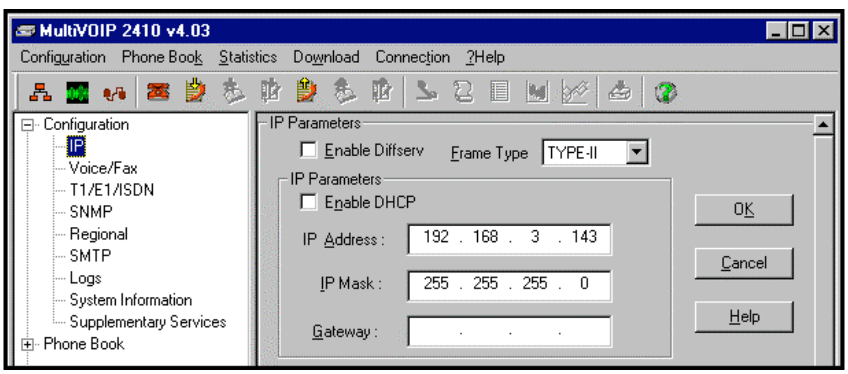
MultiVOIP Web Browser GUI Overview	
Function	Remote configuration and control of MultiVOIP units.
Configuration Prerequisite	Local Windows GUI must be used to assign IP address to MultiVOIP.
Browser Version Requirement	Internet Explorer 6.0 or higher; or Netscape 6.0 or higher
Java Requirement	Java Runtime Environment version 1.4.0_01 or higher (this application program is included with MultiVOIP)
Video Usability	large video monitor recommended

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

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



The Windows GUI gives access to commands via icons and pulldown menus whereas the web GUI does not.



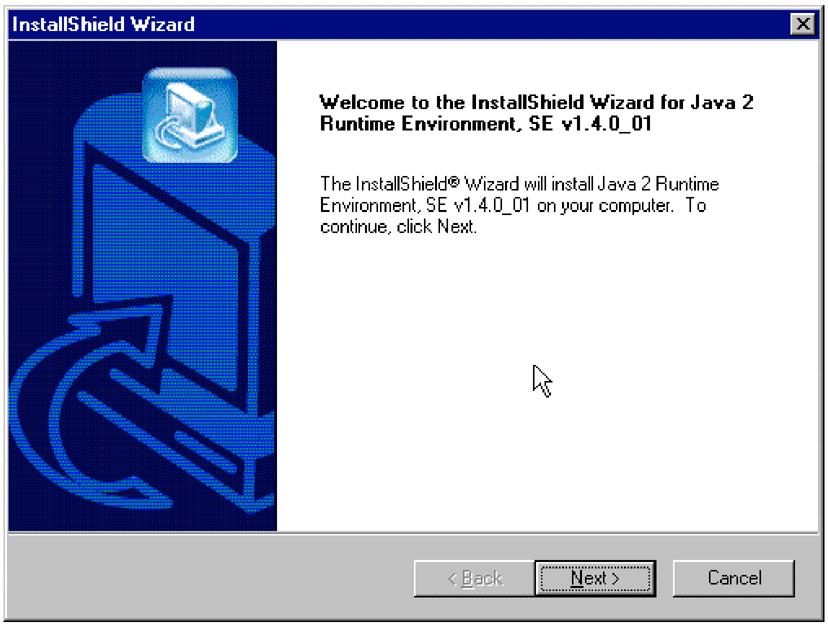
The web GUI, however, cannot perform logging in the same direct mode done in the Windows GUI. However, when the web GUI is used, logging can be done by email (SMTP).

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

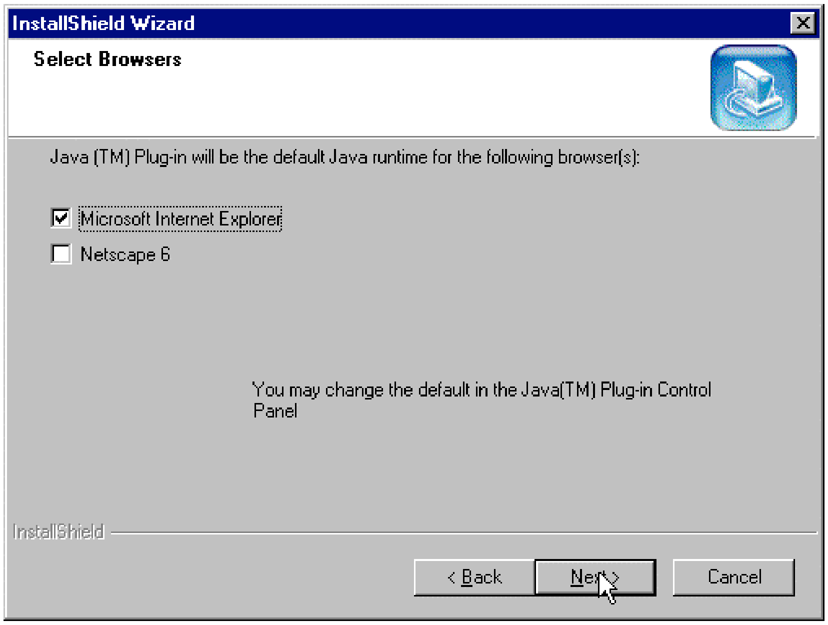
The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

In order to use the web GUI, you must also install a Java application program on the controller PC. This Java program is included on the MultiVOIP product CD.). Java is needed to support drop-down menus and multiple windows in the web GUI.

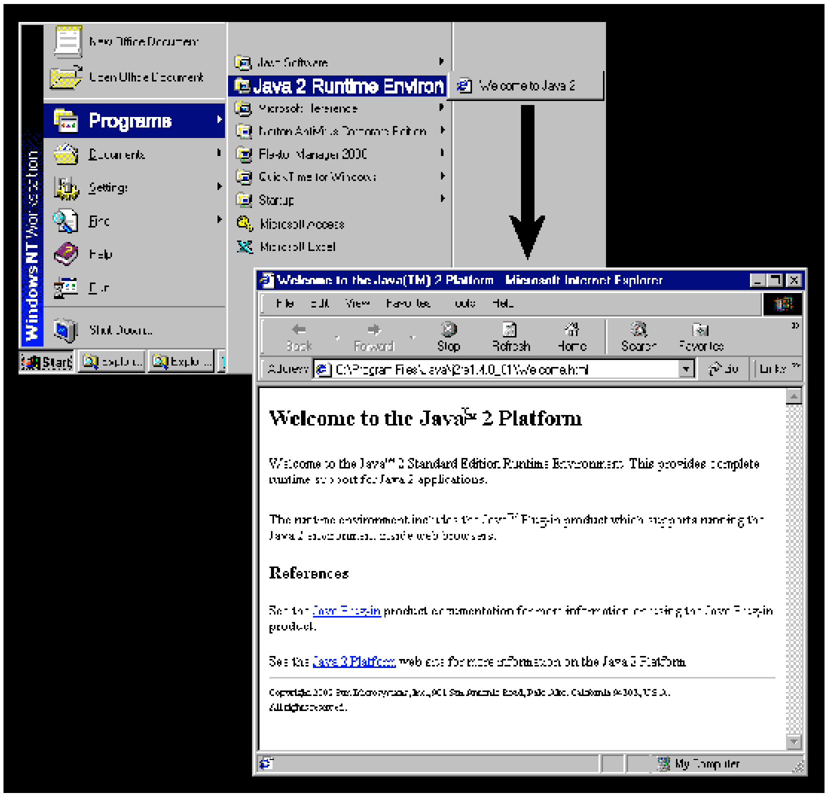
To install the Java program, go to the **Java** directory on the MultiVOIP product CD. Double-click on the EXE file to begin the installation. Follow the instructions on the Install Shield screens.



During the installation, you must specify which browser you'll use in the **Select Browsers** screen.



When installation is complete, the Java program becomes accessible in your **Start | Programs** menu (Java resources are readily available via the web). However, the Java program runs automatically in the background as a plug-in supporting the MultiVOIP web GUI. No overt user actions are required.



After the Java program has been installed, you can access the MultiVOIP using the web browser GUI. Close the MultiVOIP Windows GUI. 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 GUI or for the MultiVOIP's FTP Server function. See "Setting a Password -- Web Browser GUI" earlier in this chapter.) The web browser GUI offers essentially the same control over the voip as can be achieved using the Windows GUI. As noted earlier, logging functions cannot be handled via the web GUI. 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 GUI than with the Windows GUI.

SysLog Server Functions

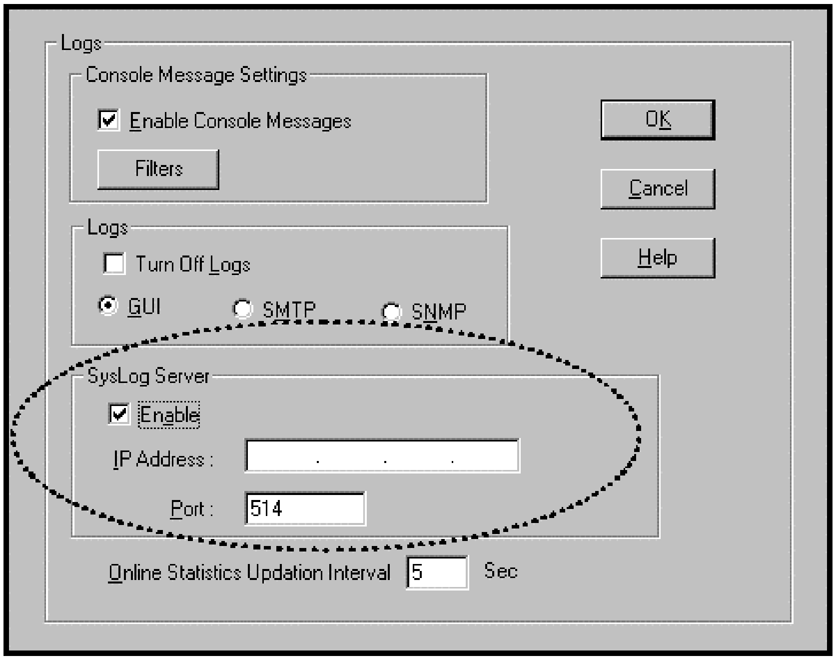
Beginning with the 4.03/6.03 software release, we have built SysLog server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.

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, among other firms. Read the End-User License Agreement carefully and observe license requirements. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by qualified providers should suffice for use with MultiVOIP units. Kiwi’s brief description of their SysLog program is as follows:

> “Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available.”

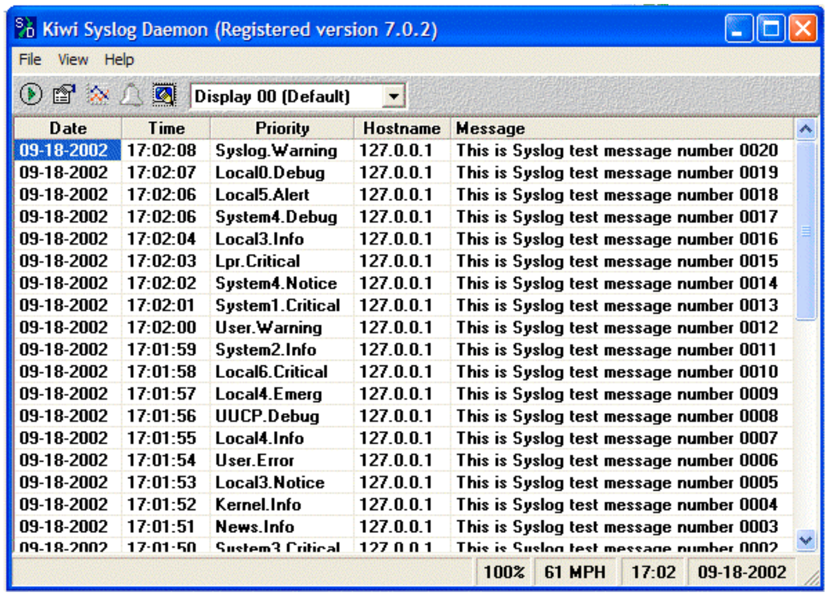
Before a SysLog client program is used, the SysLog functionality must be enabled within the MultiVOIP in the **Logs** menu under **Configuration**.



The IP Address used will be 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 will be saved/archived, opt for interaction with an SNMP system (like MultiVoipManager), 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, etc.). A sample presentation of SysLog info in the Kiwi daemon is shown below. SysLog programs will vary in features and presentation.



Chapter 11: Embedded Gatekeeper (for MVP-210G/410G/810G)

Introduction to Embedded Gatekeeper

This chapter describes how to configure and manage the MultiVOIP Gatekeeper software. The software comes pre-installed on the specially-equipped analog MultiVOIP units, MVP210G, MVP410G, and MVP810G. With gatekeeper functionality, network managers can define and control the flow of H.323 voice traffic across the IP network. In this chapter, we will present both a general description of how gatekeepers work and very specific information on how MultiTech's embedded gatekeeper units operate. In cases where the actual gatekeeper functionality implemented in the current software release differs from theoretically possible gatekeeper functionality, the differences will be noted (i.e., we describe some gatekeeper functionality that will only become available in a later software release and note all such cases).

A gatekeeper unit controls a "zone" on the IP network. (In fact, that is how a H.323 zone is defined; as the set of endpoints controlled by a gatekeeper.) One gatekeeper unit is needed to control a single zone. Therefore, when gatekeeper control is used, it's not necessary that all voip gateways within the system should be gatekeeper equipped – only one per zone is needed.

Network managers can configure, monitor, and manage the activity of registered network endpoints (including voip gateway units like the MVP210G/410G/810G). They can set policies and control bandwidth usage, thus customizing their network for better advantage.

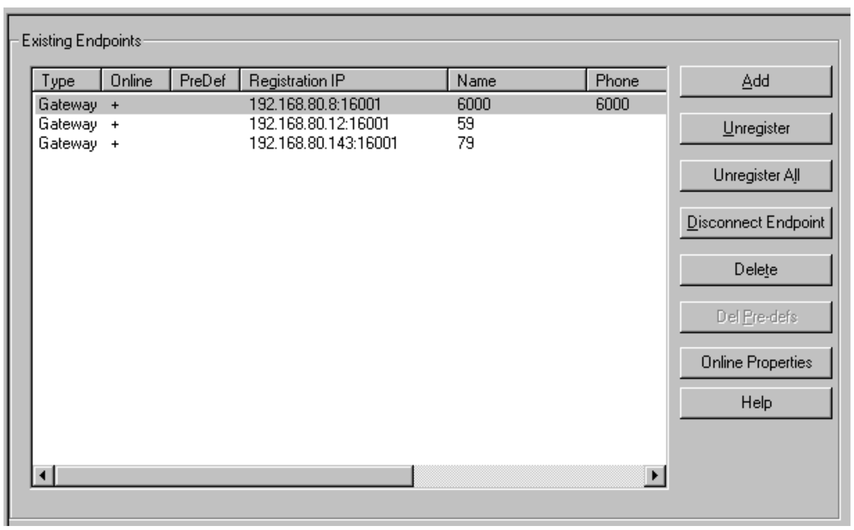
Gatekeeper facilitates interoperability between PBX dial plans and IP-based terminals. With it, call centers can route calls on the basis of need and implement other automatic call distribution features, as well.

Getting Started with the Gatekeeper-Equipped MultiVOIP

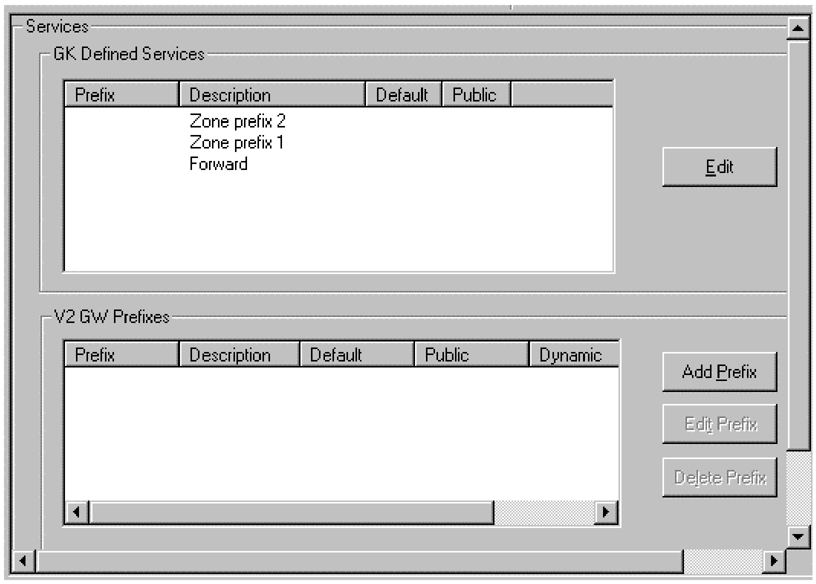
MultiVOIP units equipped with embedded gatekeeper functionality (MVP210G, MVP410G, or MVP810G) require configuration of their gatekeeper parameters before they can control a group of voip gateways. (This configuration is in addition to setting the technical parameters and phonebook parameters that are needed for the *gateway* functionality of these MultiVOIP units.)

Gatekeepers can be configured to enact a wide range of functionality, but they are primarily node points that direct and manage traffic to other endpoints. The essential question of *"whose messages go where?"* can be answered either by a gatekeeper that acts as a coordinating node or clearinghouse for the system or by phonebooks coordinated among the set of peer endpoints (gateways) that make up the system.

In its role as a node point, the gatekeeper directs call traffic between pairs of endpoints engaged in the call. To facilitate this node-point control, all endpoints (voip gateways) must be registered with the gatekeeper. This registration is done in
the **Gatekeeper | Existing Endpoints** screen.



The basic function of directing calls to specified endpoints is done differently in gatekeeper-controlled systems than in systems controlled only by phonebooks. Phonebooks use "destination patterns" like area codes and local prefixes to route calls to specific endpoints. When gatekeepers perform this directive function, they do so by using "services," which one configures in
the **Gatekeeper | Services** screen.



Suppose a voip system consists of three endpoints in three different cities all having different area codes. If this voip system were controlled only by phonebooks, three different *destination patterns* (at least) would be needed; if controlled by a gatekeeper, three different *services* (at least) would be needed.

Matched Settings in Gatekeeper, Phonebook, & Tech Config Screens. Generally, gatekeeper-equipped MultiVOIP units should be configured in this order:

1. Technical Configuration (setup for IP, voice/fax, telephony, etc.)
2. Phonebook Configuration (destination patterns, RAS settings, etc.)
3. Gatekeeper Configuration (listing endpoints, setting up services)

Also, generally, it's best to configure the gatekeeper-equipped MultiVOIP as fully as possible before configuring other gateways in the system. This is so because certain parameters that describe the gatekeeper unit must be entered the configuration screens of the ordinary voip gateway units.

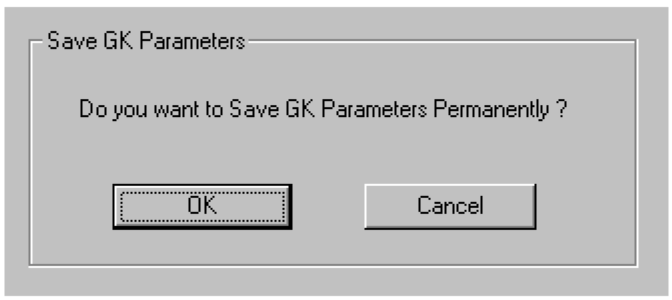
Furthermore and very importantly, several settings needed in the **Gatekeeper | Existing Endpoints** screen and in the

Gatekeeper | Services screen must also be set in the Phonebook Configuration screen. In fact, if the ordered sequence above is followed (tech config, phonebook config, gatekeeper config), the software will automatically transfer several needed phonebook RAS parameters into the fields where they are required in the gatekeeper screens.

Full details on all of the gatekeeper configuration screens are presented in the "MultVOIP Gatekeeper Software Screens" section later in this chapter.

Saving the Gatekeeper Configuration. Just as you must save the technical configuration parameters and the phonebook configuration parameters, so also gatekeeper parameters must be saved in a separate step. In the sidebar menu, go to
Save Setup | Save GK Parameters.

A dialog box will appear to confirm that you want to invoke the 'save' function.



A second dialog box will appear to confirm that the save has been executed successfully.

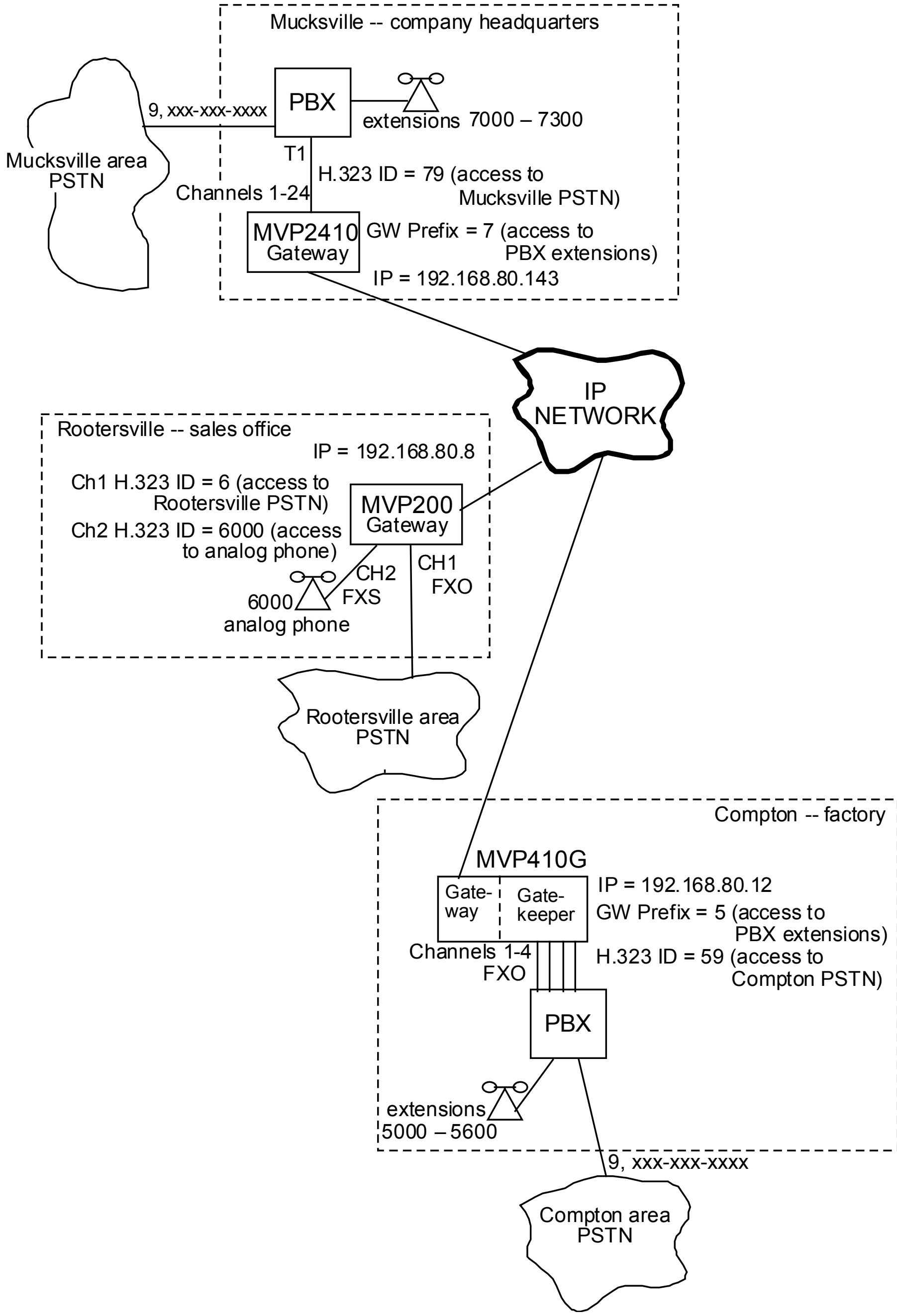
Embedded Gatekeeper System Example

The present example shows a voip system with three gateways, one of whose embedded gatekeeper functionality directs voip traffic in the system. The system design will give phone users at each office toll-free access to both the company employee phones (most are on PBXs) at the remote sites as well as the local PSTNs surrounding the remote sites.

The gatekeeper equipped MultiVOIP is an analog model (MVP410G) whose four channels are all connected (via FXO interface) to a PBX at a company's factory site in "Compton." The second gateway is a T1 digital voip gateway (MVP2410) connected to a PBX at the company's headquarters in "Mucksville." The third gateway, located in one of the company's small sales offices in "Rootersville," is a first-generation MultiTech gateway with two analog channels (MVP200), one serving an analog phone (via FXS interface) and the other giving access to its local area PSTN (via FXO interface).

To implement this configuration, we start with the gatekeeper-equipped MultiVOIP at the Compton site.

1. **MVP410G**. For the MVP410G at Compton, we need first to configure its phonebook with the gatekeeper configuration in mind. (We'll presume that its technical configuration has already been completed. Its IP address would have been set in the **Configuration | IP Parameters** screen and its four channels would have been
set to "FXO" in its **Configuration | Interface** screen.)


Mucksville -- company headquarters
9, xxx-xxx-xxxx
PBX
extensions 7000 – 7300
Mucksville area
PSTN
T1
Channels 1-24
H.323 ID = 79 (access to
Mucksville PSTN)
MVP2410
Gateway
GW Prefix = 7 (access to
PBX extensions)
IP = 192.168.80.143
IP
NETWORK
Rootersville -- sales office
IP = 192.168.80.8
Ch1 H.323 ID = 6 (access to
Rootersville PSTN)
Ch2 H.323 ID = 6000 (access
to analog phone)
MVP200
Gateway
CH2
FXS
CH1
FXO
6000
analog phone
Rootersville area
PSTN
Compton -- factory
MVP410G
Gate-
way
Gate-
keeper
IP = 192.168.80.12
GW Prefix = 5 (access to
PBX extensions)
H.323 ID = 59 (access to
Compton PSTN)
Channels 1-4
FXO
PBX
extensions
5000 – 5600
9, xxx-xxx-xxxx
Compton area
PSTN

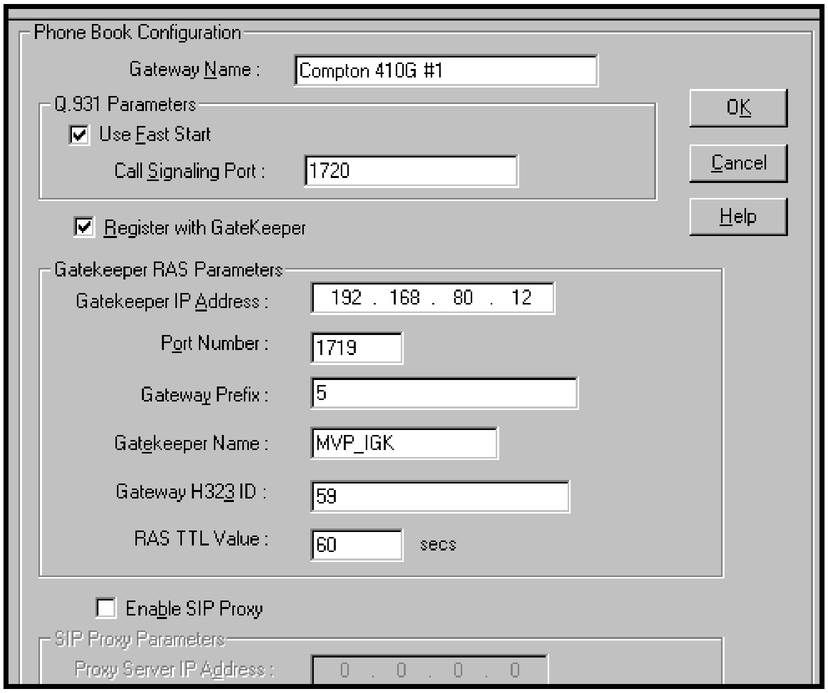
The required MVP410G phonebook configuration is shown below.

“Compton” MVP410G Gateway Functions and Settings			
Function	**PhBk Config Scn Settings** [1]	**Inbound PhoneBook Screen Settings**	**Phone User’s Actions**
Put MVP410G gateway under gatekeeper control	Gatekeeper IP Address = 192.168.80.12	--	--
Give remote users access to Compton factory PBX extensions	Gateway Prefix = 5	Remove Prefix = 5; Add Prefix = 5	Dial 4 digits beginning with “5”
Give remote users access to Compton area PSTN	Gateway H.323 ID = 59	Remove Prefix = 59; Add Prefix= 9	Dial “59” plus Compton local number
		Outbound PhoneBook Screen Settings	
Get access to Mucksville office PBX extensions	--	Destination Pattern = 7 RemovePrefix = 7 Select “Use GateKeeper” Gateway H.323ID = none Gateway Prefix = 7	Dial 4 digits beginning with “7”
Get access to Mucksville area PSTN	--	Destination Pattern = 79 RemovePrefix = none Select “Use GateKeeper” Gateway H.323ID = 79 Gateway Prefix = none	Dial “79” plus Mucksville local number
Get access to Rootersville office phone	--	Destination Pattern = 6000 RemovePrefix = none Select “Use GateKeeper” Gateway H.323ID = 6000 Gateway Prefix = none	Dial 6000.
Get access to Rootersville area PSTN	--	Destination Pattern = 6 RemovePrefix = none Select “Use GateKeeper” Gateway H.323ID = 6 Gateway Prefix = none	Dial “6”; get second dial tone. Dial Hoot #.

1. “PhoneBook Configuration screen settings”

2. **MVP410G**. We begin with the PhoneBook Configuration screen. Because the MVP410G serves as a gatekeeper for its own gateway, the Gatekeeper IP Address is the same as the gateway's regular IP address, as set in the IP Parameters screen.

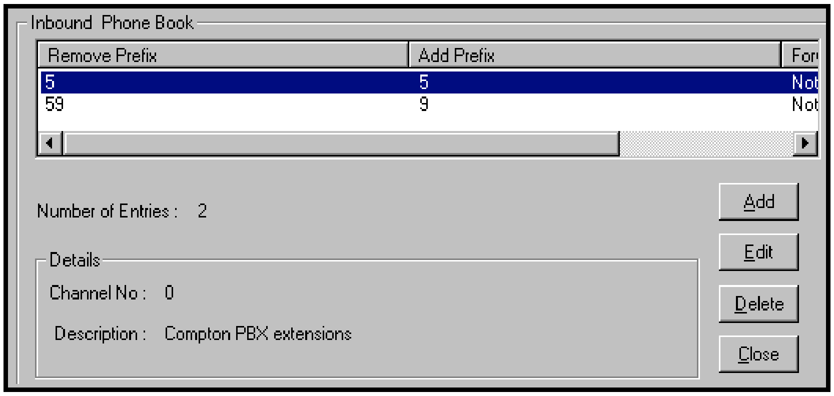
Compton MVP410G MultiVOIP



We have set the Gateway Prefix to 5 to give voip system phone users access to Compton office PBX extensions (this value will appear in the **Gateway | Services | V2 GW Prefixes** screen; see step 8). Because we have set the Gateway Prefix (to "5") in the PhoneBook Configuration screen during the Phonebook Configuration process, it will automatically appear in the Gatekeeper GUI. We have set the Gateway H.323 ID to 59 to give voip system users access to the Compton area PSTN. The Gateway H.323 ID of 59 will need to be added manually to the
GateKeeper | Services screen under "GK Defined Services."
The Gatekeeper Name can be customized for your needs. "MVP_IGK" is the default value.

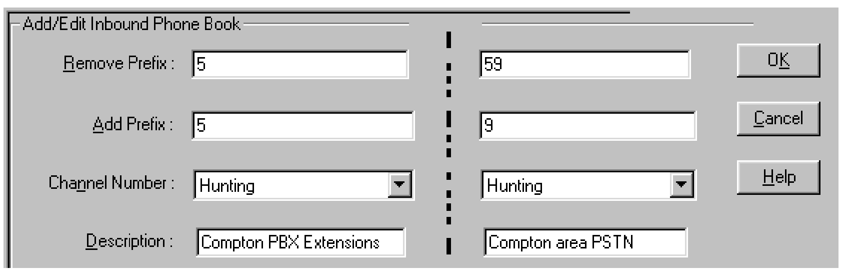
3. **MVP410G**. The Inbound Phonebook of the MVP410G requires two entries, one for access to Compton PBX extensions, another for access to the Compton area PSTN.

Compton MVP410G MultiVOIP



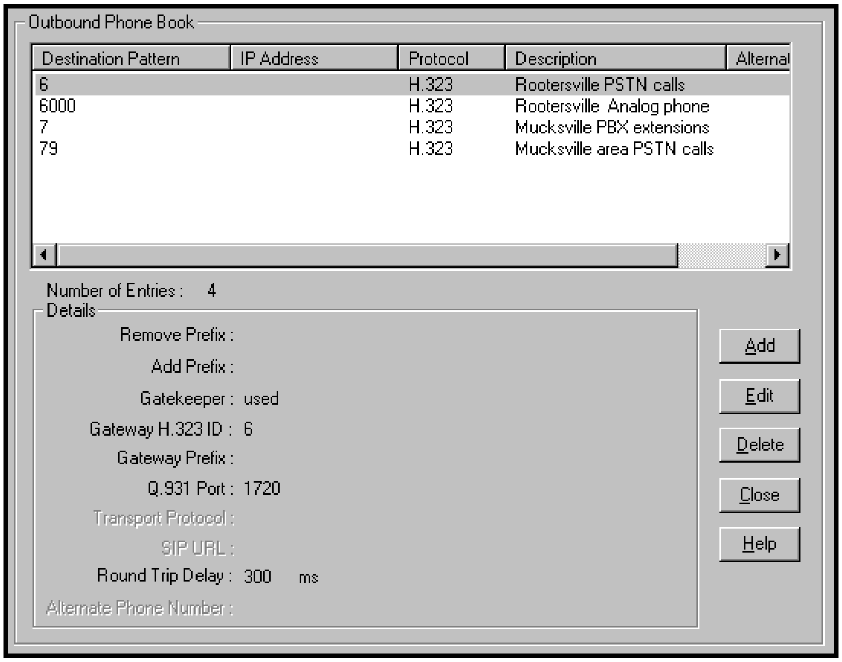
To create each of these entries, you must click on "Add" at the Inbound PhoneBook screen and enter the details for each entry in a separate Add/Edit Inbound PhoneBook screen, as shown below.

Compton MVP410G MultiVOIP: Adding Inbound Phonebook Entries
giving remote users access to local PBX **… and …** *to the local area PSTN*



4. **MVP410G**. The Outbound Phonebook of the MVP410G requires four entries.

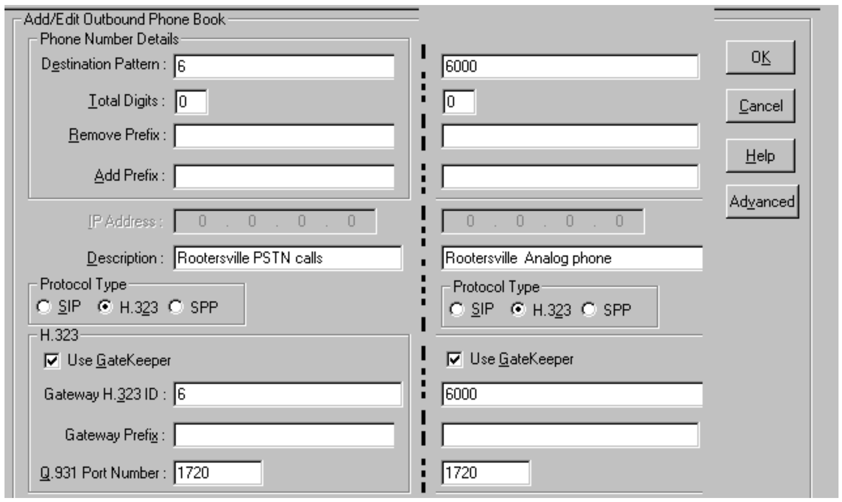
Compton MVP410G MultiVOIP



Two outbound phonebook entries are for Rootersville, one describing access to its local PSTN and the other describing access to its office phone. To create each of these entries, you must click on "Add" at the Outbound PhoneBook screen and enter the details for each entry in a separate Add/Edit Outbound PhoneBook screen.

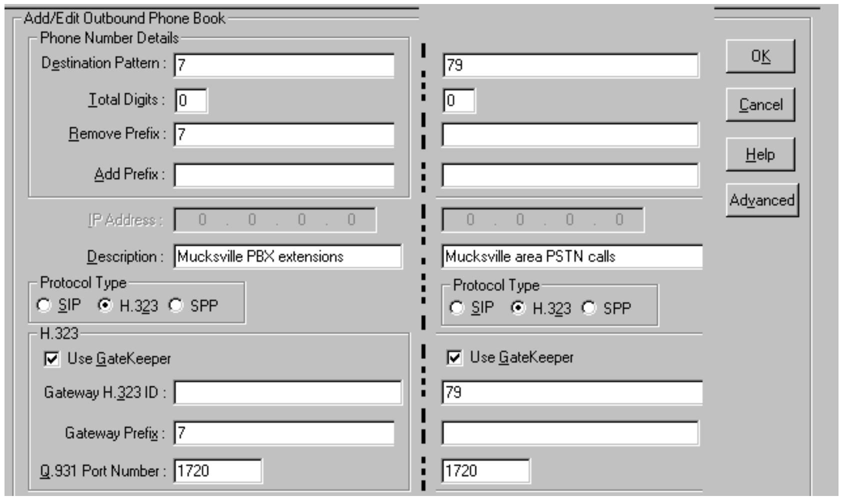
Compton MVP410G MultiVOIP: Adding Outbound Phonebook Entries

gaining access to a remote area PSTN … **and …** *to a remote office phone*



Another two outbound phonebook entries are for Mucksville for access to its PBX extensions and its local PSTN.

Compton MVP410G MultiVOIP: Adding Outbound Phonebook Entries
gaining access to a remote site PBX **… and …** *to a remote area PSTN*



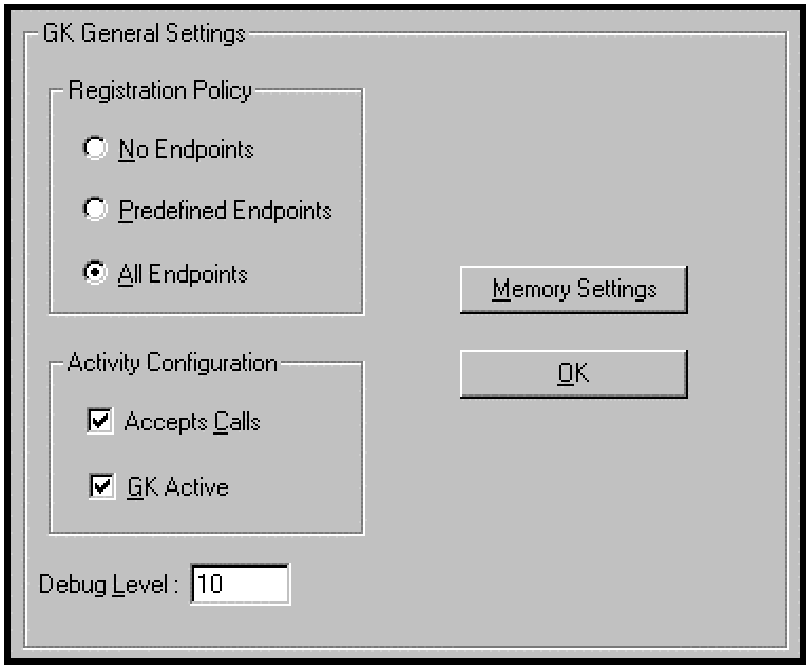
5. **MVP410G**. Save the MVP410G PhoneBook Configuration (the Save Setup command is in the sidebar menu) before proceeding to gatekeeper configuration. Click on **Save & Reboot** and then click **OK** on the screen that will appear directly thereafter.

6. **MVP140G Gatekeeper Function**. We will configure the gatekeeper function of the MVP410G at Compton as summarized in the table below. It is useful to begin the configuration process by listing the functionality that you want to implement in your system.

"Compton" Gatekeeper Functions & Settings			
Function	**GK Services Screen Settings**	**GK General Settings Screen**	**Phone User's Actions**
Activate gatekeeper function of MVP410G	--	Reg Pol. = All Endpts Accepts Calls Y GK Active Y	--
		GK Service Properties Screen Settings	
Access to Compton factory PBX extensions	V2 GW Prefix = TEL:5 As set in PhoneBook Configuration screen, Gateway Prefix field of Compton MVP410G voip.	"Allow as default to online endpoints" = Y "Allow as public for Out-of-Zone Endpoints" = Y	Dial 4 digits beginning with "5"
Access to Compton area PSTN	GK Defined Services Prefix = 59	"Allow as default to online endpoints" = Y	Dial "59" plus Compton local number
Access to Mucksville office PBX extensions	V2 GW Prefix = TEL:7 As set in PhoneBook Configuration screen, Gateway Prefix field of Mucksville MVP2410 voip.	"Allow as default to online endpoints" = Y "Allow as public for Out-of-Zone Endpoints" = Y	Dial 4 digits beginning with "7"
Access to Mucksville area PSTN	GK Defined Services Prefix = 79	"Allow as default to online endpoints" = Y	Dial "79" plus Mucksville local number
Access to Rootersville office phone	GK Defined Services Prefix = 6000	"Allow as default to online endpoints" = Y	Dial 6000.
Access to Rootersville area PSTN	GK Defined Services Prefix = 6	"Allow as default to online endpoints" = Y	Dial "6". Dial local R'ville number.

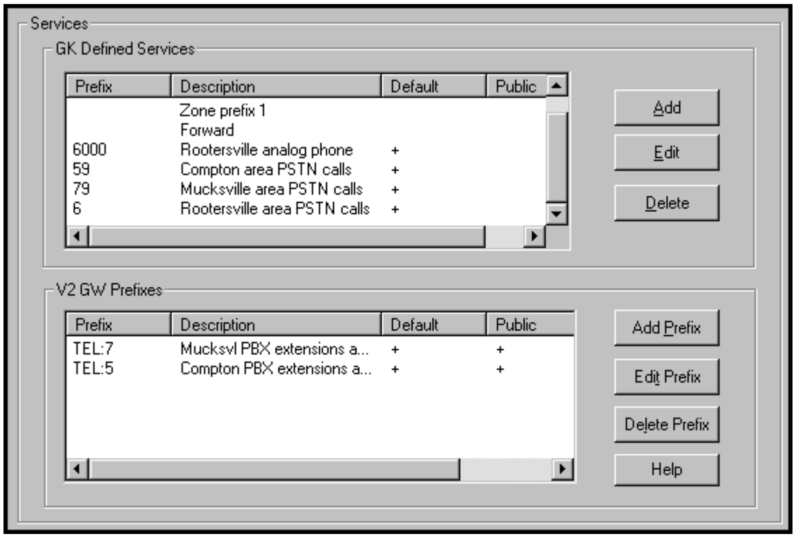
7. **MVP410G**. Begin at the GK General Settings screen. The required settings are default values.

Compton MVP410G MultiVOIP Gatekeeper



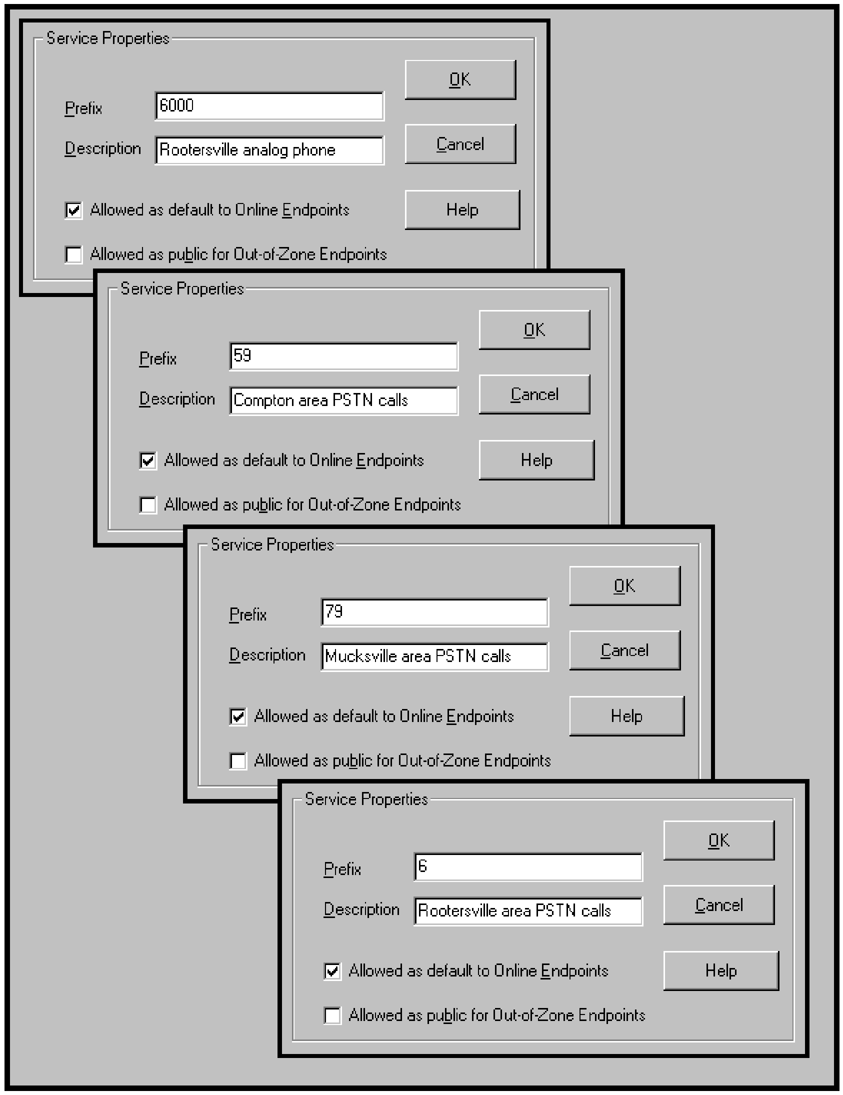
8. **MVP410G**. Adding "services" and "prefixes" in the gatekeeper Services screen fulfills the same role as setting "destination patterns" in outbound phonebook screens. Even though they serve a function similar to destination patterns, the "service" and "prefix" gatekeeper entries do not eliminate the need for phonebook destination patterns; nor do phonebook destination patterns eliminate the need for gatekeeper services and prefixes. They all work together and all must be present for proper operation. (Note also that "Services" constitutes a wider category than we are discussing here. Generally, services can also be, essentially, features, like call forwarding.)

Compton MVP410G MultiVOIP Gatekeeper



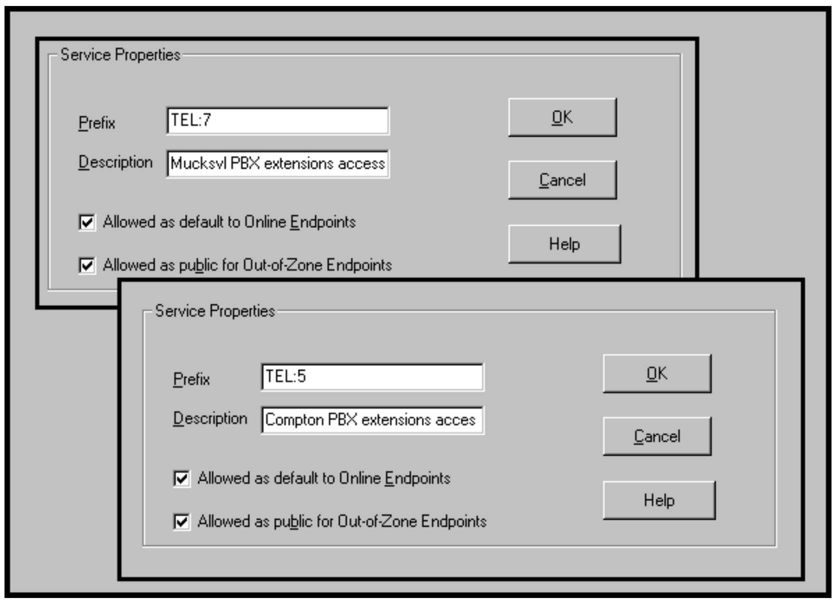
To create each of the four required 'GK-Defined-Services', you must click on "Add" in the Gatekeeper **Services** screen and enter the details for each entry in a separate **Service Properties** screen, as shown below.

Compton MVP410G MultiVOIP Gatekeeper



To give network-wide access to the Compton factory PBX extensions, the Gateway Prefix field of the MVP410G's **PhoneBook Configuration** screen has already been set to 5 (in step 2 above) and this setting appears automatically in the **V2 GW Prefix** screen. (There is no need to add this item manually in the V2 GW Prefixes screen.) Similarly, to give network-wide access to the Mucksville office PBX extensions, the Gateway Prefix of the Mucksville MVP2410's **PhoneBook Configuration** screen must be set to 7. When this setting has been made, and when that voip contacts the MVP410G gatekeeper unit, the setting will appear automatically in the **V2 GW Prefix** screen of the Compton MVP410G gatekeeper/gateway unit. (Again, there is no need to add this item manually in the **Services |V2 GW Prefixes** screen pane.) The **Service Properties** screens for these two V2 GW Prefixes are shown below.

Compton MVP410G MultiVOIP Gatekeeper



9. **MVP410G**. Save the MVP410G gatekeeper configuration before configuring the other gateways in the system (the **Save Setup | Save GK Parameters** command is in the sidebar menu).

10. **MVP200**. A summary of the required MVP200 phonebook configuration is shown below. (We are presuming that the MVP200's IP address has been duly set in the **IP Parameters** screen and that its channels have been set in the **Voice Channels** screen as follows: Ch1 = FXO; CH2 = FXS.) Again, it is useful to begin the configuration process by listing the system functionality that this particular voip unit will have to perform.

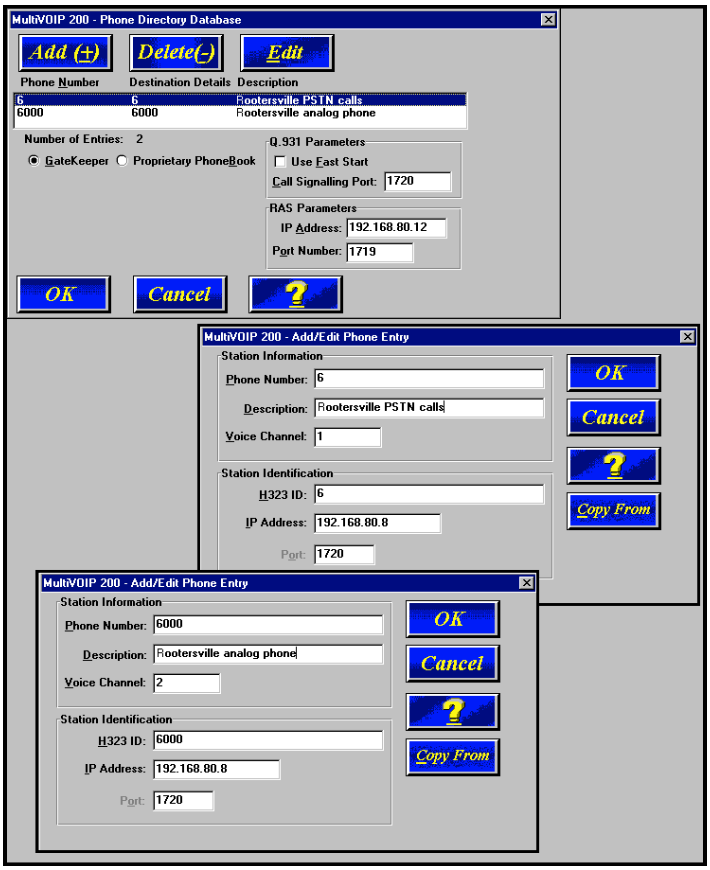
"Rootersville" MVP200 Gateway Functions & Settings			
Function	**Phonebook Directory Data-Base screen settings**	**Add/Edit Phone-Book Entries screen settings**	**Phone User's Actions**
Put MVP200 gateway under gatekeeper control	Select "GateKeeper" radio button. RAS Parameters IP Address = 192.168.80.12;	IP Address = 192.168.80.8	--
Allow remote users access to Rootersville office phone	Phone Number = 6000 Destination Details = 6000	Phone Number = 6000 Ch2 H.323 ID = 6000	Dial "6000"
Allow remote users access to Rootersville area PSTN	Phone Number = 6 Destination Details = 6	Phone Number = 6 Ch1 H.323 ID = 6	Dial "6". Dial local R'ville phone number.
Get access to Compton factory PBX extensions	*These functions are provided by gatekeeper within MVP410G.*		Dial 4 digits beginning with "5"
Get access to Compton area PSTN			Dial "59" plus Compton local number
Get access to Mucksville office PBX extensions			Dial 4 digits beginning with "7"
Get access to Mucksville area PSTN			Dial "79" plus Mucksville local number

11. **MVP200**. From the main MultiVOIP200 screen, select Phone Book.

Rootersville MVP200 MultiVOIP

12. **MVP200**. In the Phone Directory Database screen, click on the "Gatekeeper" radio button to put the MVP200 under the control of the MVP410G gatekeeper. Under "RAS Parameters" in the IP Address field, enter the IP address of the gatekeeper. In this case, since the MVP410G uses a single IP address for both its gateway and its gatekeeper functions, we simply use the MVP410G's regular (and only) IP address (192.168.80.12). Then add the two required destination patterns: **6000** will direct calls to the analog phone in the Rootersville office; **6** will give remote users access to the Rootersville area PSTN (calls can be completed in a single dialing sequence).

Rootersville MVP200 MultiVOIP



13. **MVP200**.When you have completed the configuration, click **OK** on the Phonebook Directory Database screen. Then go to the MultiVOIP 200 main screen and click on **Download Setup** to save the configuration.

14. **MVP2410**. The required MVP2410 phonebook configuration is shown below. We are presuming here that technical configuration is already complete so that the MVP2410's IP address and other technical configuration parameters have already been duly set.

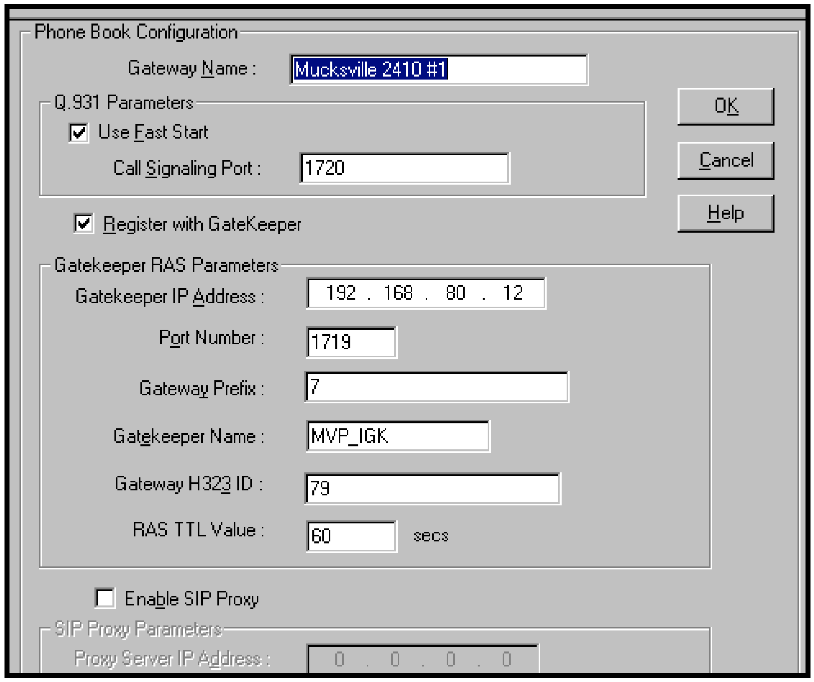
“Mucksville” MVP2410 Gateway Functions and Settings

Function	PhBk Config Scn Settings [1]	Inbound PhoneBook Screen Settings	Phone User's Actions
Put MVP2410 under control of gatekeeper	Gatekeeper IP Address = 192.168.80.12	--	--
Give remote users access to Mucksville office PBX extensions	Gateway Prefix = 7	Remove Prefix = 7; Add Prefix = 7	Dial 4 digits beginning with “7”
Give remote users access to Mucksville area PSTN	Gateway H.323 ID = 79	Remove Prefix = 79; Add Prefix= 9	Dial “79” plus Mucksville local number
		Outbound PhoneBook Screen Settings [3]	
Get access to Compton factory PBX extensions		Destination Pattern = 5 RemovePrefix = 5 Select “Use GateKeeper” Gateway H.323ID = none Gateway Prefix = 5	Dial 4 digits beginning with “5”
Get access to Compton area PSTN		Destination Pattern = 59 RemovePrefix = none Select “Use GateKeeper” Gateway H.323ID = 59 Gateway Prefix = none	Dial “59” plus Compton local number
Get access to Rootersville office phone	--	Destination Pattern = 6000 RemovePrefix = none Select “Use GateKeeper” Gateway H.323ID = 6000 Gateway Prefix = none	Dial 6000.
Get access to Rootersville area PSTN	--	Destination Pattern = 6 RemovePrefix = none Select “Use GateKeeper” Gateway H.323ID = 6 Gateway Prefix = none	Dial “6”. Dial R'ville local phone number.

1. “PhoneBook Configuration screen settings”

15. **MVP2410**. For the MVP2410 at Mucksville, we begin again with the PhoneBook Configuration screen. Because the MVP410G serves as a gatekeeper for the MVP2410, the MVP410G's IP address is the Gatekeeper IP Address for the MVP2410.

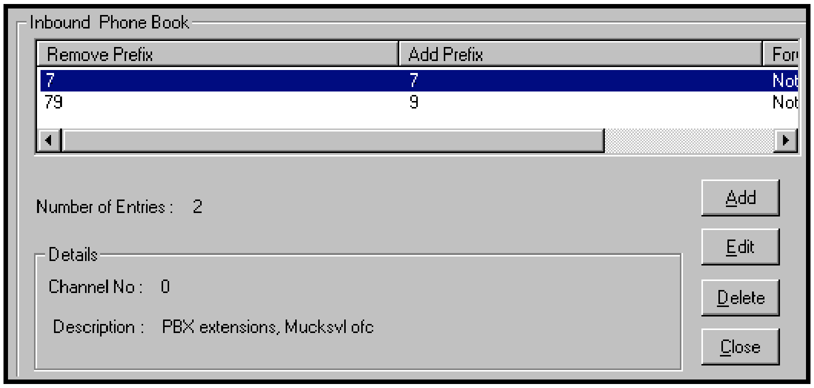
Mucksville MVP2410 MultiVOIP



We have set the Gateway Prefix to 7 to give voip system phone users access to Mucksville office PBX extensions. Because we have set the Gateway Prefix (to "7") in the PhoneBook Configuration screen during the Phonebook Configuration process, it will automatically appear in the Gatekeeper GUI. We have set the Gateway H.323 ID to 79 to give voip system users access to the Mucksville area PSTN. The Gateway H.323 ID of 79 will need to be added manually to the
GateKeeper | Services screen under "GK Defined Services."
The Gatekeeper Name can be customized for your needs. "MVP_IGK" is the default value.

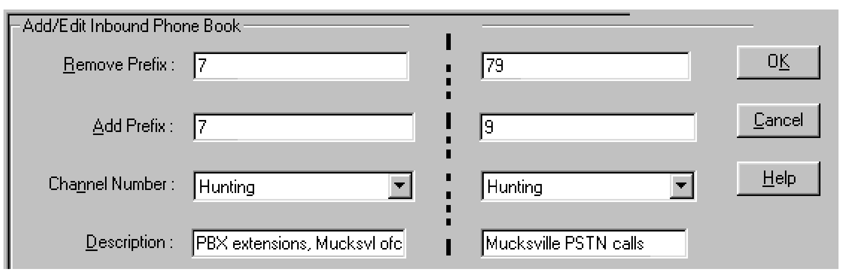
16. **MVP2410**. The Inbound Phonebook of the MVP2410 requires two entries, one for access to Mucksville PBX extensions, another for access to the Mucksville area PSTN.

Mucksville MVP2410 MultiVOIP



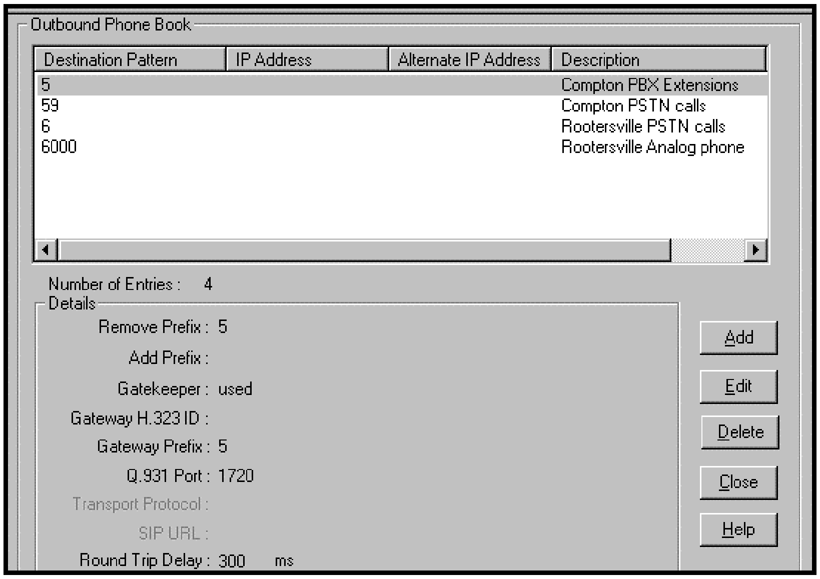
To create each of these entries, you must click on "Add" at the Inbound PhoneBook screen and enter the details for each entry in a separate Add/Edit Inbound PhoneBook screen, as shown below.

Mucksville MVP2410 MultiVOIP: Adding Inbound Phonebook Entries
giving remote users access to local PBX **… and …** *to the local area PSTN*



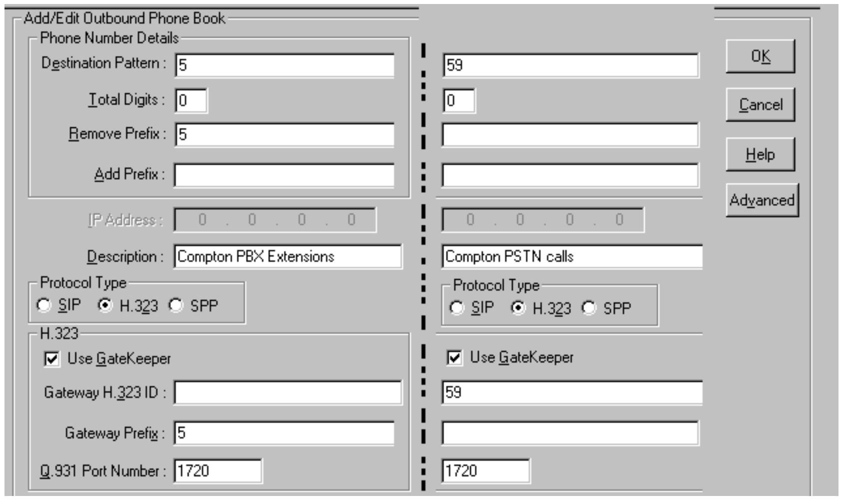
17. **MVP2410**. The Outbound Phonebook of the MVP2410 requires four entries.

Mucksville MVP2410 MultiVOIP



Two outbound phonebook entries are to gain access to Compton's PBX extensions and its local PSTN. To create each of these entries, you must click on "Add" at the Outbound PhoneBook screen and enter the details for each entry in a separate Add/Edit Outbound PhoneBook screen.

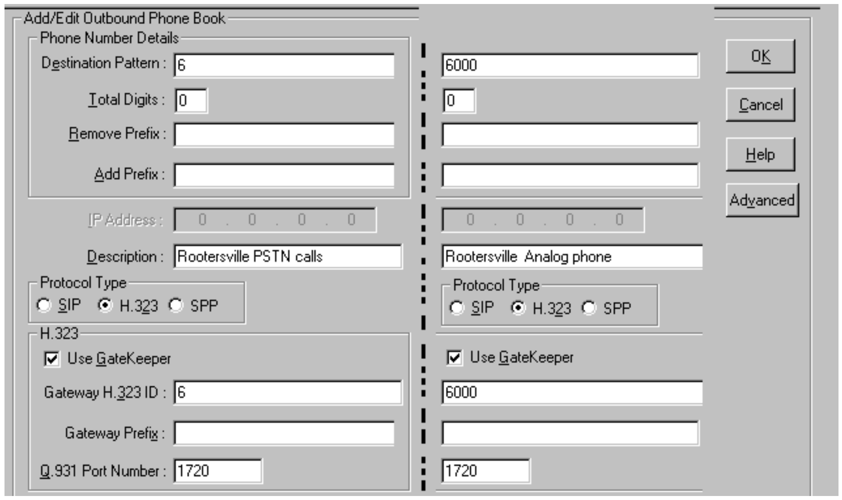
Mucksville MVP2410G MultiVOIP: Adding Outbound Phonebook Entries
gaining access to a remote site PBX **… and …** *to a remote area PSTN*



Another two outbound phonebook entries are for Rootersville, one describing access to its local PSTN and the other describing access to its office phone.

Mucksville MVP2410 MultiVOIP: Adding Outbound Phonebook Entries

gaining access to a remote area PSTN **… and …** *to a remote office phone*

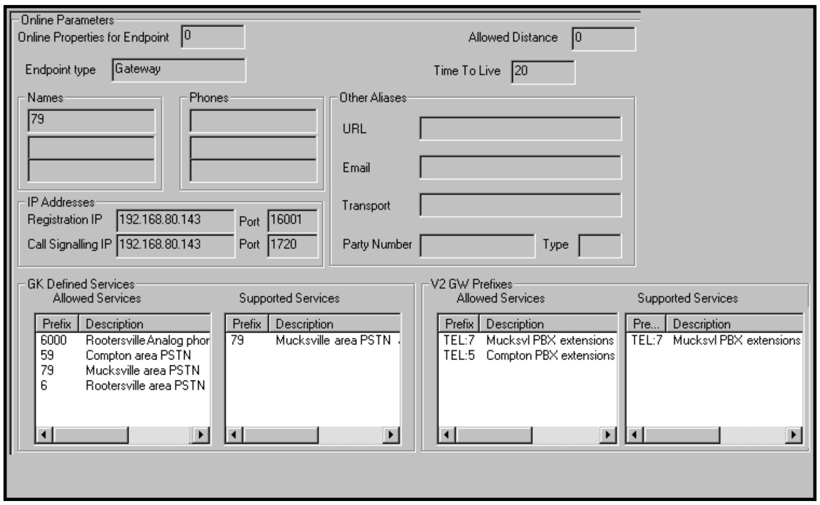


18. **MVP2410**. Save the MVP2410 PhoneBook Configuration (the Save Setup command is in the sidebar menu).

19. **MVP410G**. The gatekeeper **Online Parameters** screen (go to **Gatekeeper | Endpoints** and click the "Online Parameters" button) for the Mucksville MVP2410 shows a useful summary of system capabilities and denotes those that have been enabled for the MVP2410 in particular.

Mucksville MVP2410 MultiVOIP: its gatekeeper Online Parameters (as seen in the Compton MVP410G's MultiVOIP software display)

"allowed" services are system-wide … **whereas …** "*supported" services are those that are active in that particular voip endpoint*



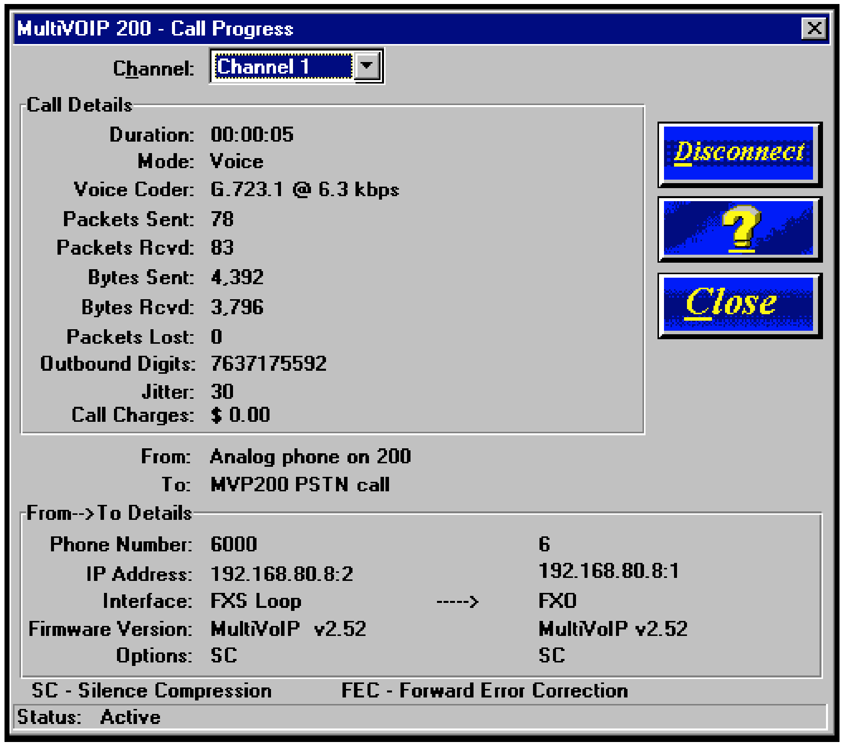
The gatekeeper will route calls to an endpoint only if the service (dialing pattern) is supported by that endpoint. (Services may be "allowed" in the system but not "supported" by an endpoint.)

"GK Allowed Services" are the set of all services (roughly the equivalent of destination patterns in phonebooks) used in the voip system that the embedded gatekeeper is overseeing. "GK Supported Services" are all services (destination patterns) that direct calls to the MVP2410 gateway.

20. **Calls**. We will now consider examples of different types of voip calls that can be made within the system. We dial a sequence, complete the call, and then look at the **Call Progress** screen of the voip unit at which the call is completed.

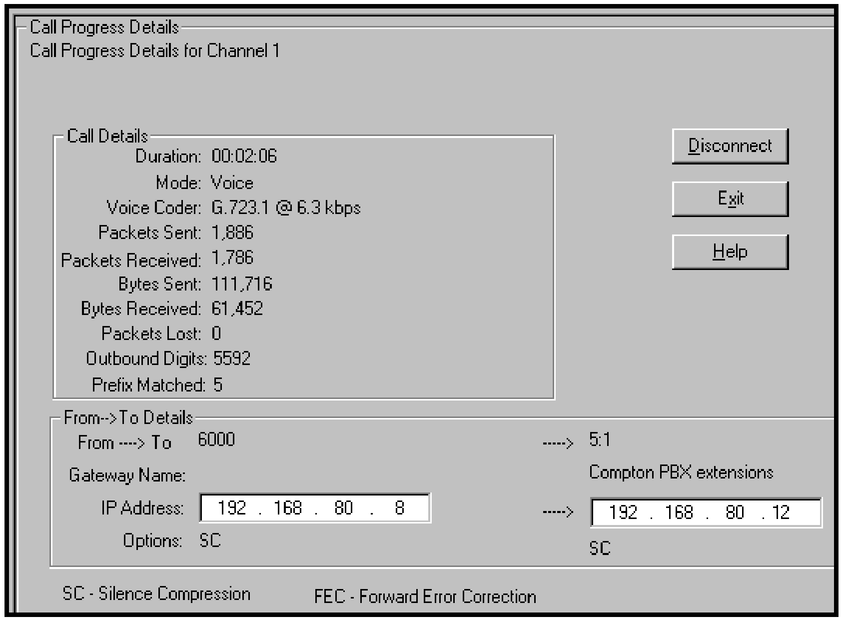
21. **MVP200**. A call from the Rootersville office to its local PSTN can be dialed 67637175592.

Rootersville MVP200 MultiVOIP



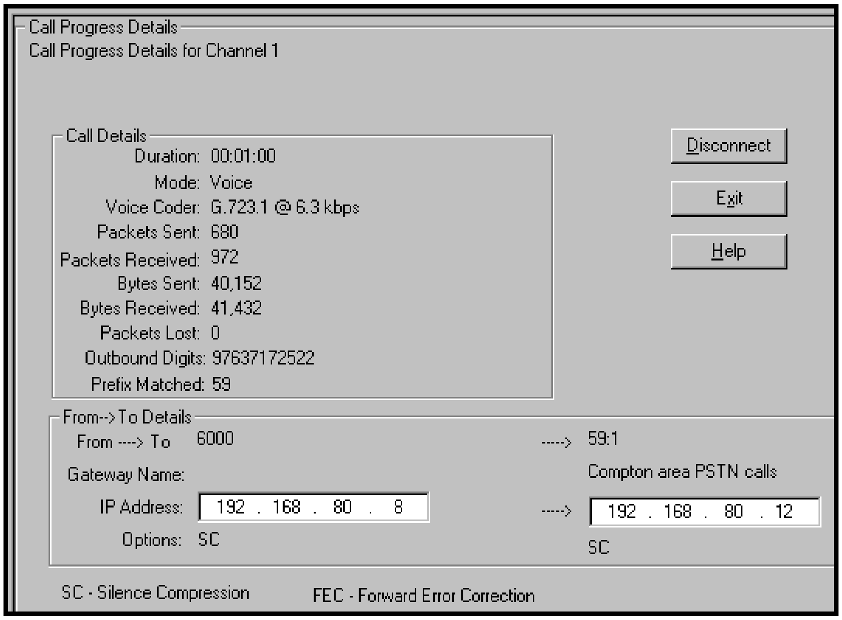
22. **MVP410G**. A call from the Rootersville analog phone to a PBX extension at the Compton office can be dialed 5592.

Compton MVP410G MultiVOIP



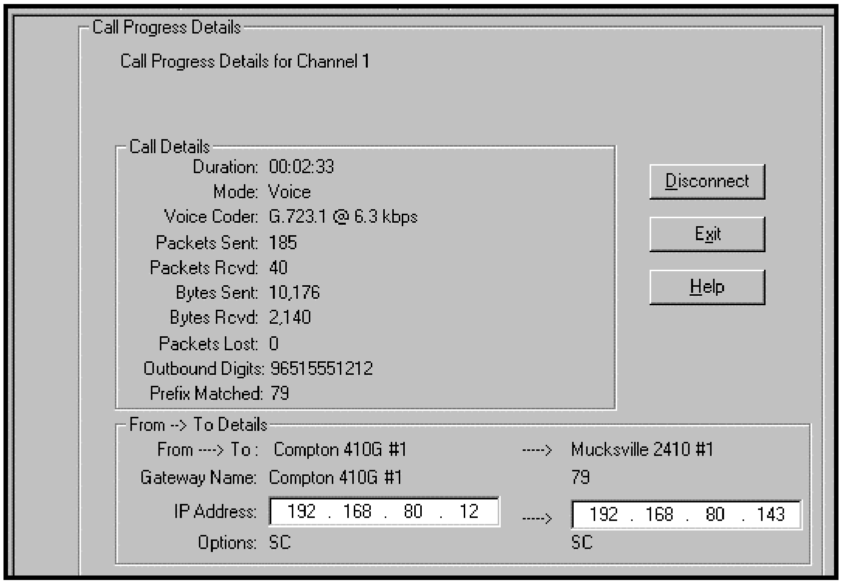
23. **MVP410G**. A call from the Rootersville analog phone to a Compton area PSTN number can be dialed 59 7637172522.

Compton MVP410G MultiVOIP



24. **MVP2410**. A call from a Compton PBX user to a Mucksville area PSTN number can be dialed 796515551212.

Mucksville MVP2410 MultiVOIP



End of Example.

Gatekeeper Basics

Introduction

Gatekeepers are optional within H.323 networks. However, when they are present, gateways (voip units) and other network endpoint devices (like terminals and Multipoint Control Units used in conferences) must use gatekeeper services. There are four functions that H.323 gatekeepers must provide to the network and many other functions, both standard and proprietary, that the gatekeeper *may* offer to network participants.

Mandatory Gatekeeper Functions

The mandatory gatekeeper functions are address translation, admission control, bandwidth control, and zone management.

Address Translation

The gatekeeper supports aliases, such as conventional E.164 phone numbers, for each endpoint registered within the zone. Users call each other within a zone by simply dialing a number or string of characters instead of an IP address. This function is particularly important when a phone on the circuit-switched network tries to call a phone connected to a gateway on an IP network.

Admission Control

The gatekeeper determines which network participants can and cannot make calls, according to established network permissions and rules. The gatekeeper controls admission using H.225 "RAS" messages (Registration, Admission, Status).

Bandwidth Control

With the MultiVOIP Gatekeeper, the network administrator can specify bandwidth limitations within a gatekeeper's zone and can specify a bandwidth limit for gateway endpoints. The gatekeeper controls bandwidth using H.225 RAS messages. A gatekeeper may determine there is no bandwidth available for a call or no additional bandwidth available for an ongoing call requesting an increase. Dynamic (situation-dependent) changes in bandwidth allocation are typically called "bandwidth management," which is considered an optional gatekeeper function.

Zone Management

Note. Zone Management and neighboring gatekeeper functionality are not included in the current software release. The discussion of this paragraph pertains primarily to the general theory of gatekeeper functionality. These functions are included in plans for subsequent software releases.

The gatekeeper allows or disallows call traffic between neighboring zones, depending upon established permissions. The zones themselves might be defined geographically (a company may have facilities in different cities, each being a separate network zone), by physical network connections (a range of IP addresses may comprise a zone, as may a subnet on a particular floor of a building), or by an organizational criterion (e.g., a large company might define separate network zones for engineering, manufacturing, marketing, and administration).

Optional Gatekeeper Functions

The MultiVOIP Gatekeeper supports the four main optional gatekeeper functions: call control signaling, call authorization, bandwidth management, and call management.

Call Control Signaling

The gatekeeper can, in “routed” mode, act as an intermediary for H.225 call-control signals between two endpoints participating in a call. In “direct” mode, this function is turned off and the endpoints exchange H.225 call-control messages directly.

Call Authorization

The gatekeeper can be programmed to restrict access (admission and registration) according to criteria set by the user.

Bandwidth Management

This is essentially dynamic bandwidth control (see “Bandwidth Control” section above).

Call Management

Note. Call Management functionality for re-routing calls is not included in the current software release. The discussion of this paragraph pertains primarily to the general theory of gatekeeper functionality. This function is included in plans for subsequent software releases.

The gatekeeper can keep a list of ongoing H.323 calls. This information allows the gatekeeper to re-route calls (where possible) to balance the traffic load on the networks.

Features

Ease of Use. The MultiVOIP Gatekeeper manages a zone, which is a collection of MultiVOIP gateways or other H.323 devices. Multiple gatekeepers can be configured to support several zones. For ease of use, the MultiVOIP Gatekeeper employs an intuitive graphical user interface. End-users can communicate using aliases (phone numbers). There's no need to remember complicated network addresses. Simple prefixes are used to access gatekeeper services such as call forwarding and out-of-zone dialing.

Capacities & Capabilities by Model. Within each zone, the MultiVOIP Gatekeeper supports a certain number of concurrent calls and registered endpoints. The capacities and capabilities of the various embedded gatekeeper voip units are described in the table below.

Model	Number of Simultaneous Calls Supported	Number of Registered Endpoints Supported	Protocols Supported
MVP210G	10	250	H.323 v4
MVP410G	20	250	H.323 v4
MVP810G	20 or 30	250	H.323 v4

•Ease of Control

With the MultiVOIP Gatekeeper, the network manager can determine the following settings:

•Network parameters
Maximum number of calls or registrations; maximum total bandwidth; upper bandwidth used per call; and frequency of sending information request (IRR) "keep alive" messages.

•Gatekeeper parameters
Gatekeeper registration policies; routing options; alias resolution policies; and endpoint permissions.

•**Gatekeeper services**
Built-in services such as call forward, zones and exit zone; and custom services.

The Gatekeeper Protocols

H.323 is an umbrella standard that consists of many subordinate protocols. Three protocols, Q.931, H.225, and H.245, are particularly relevant to gatekeepers.

The Q.931 protocol pertains to the setup and teardown of call connections between network endpoints.

The H.225 Call Signaling Protocol pertains to Registration, Admission, and Status (RAS). (Note that RAS in H.323 has nothing to do with the Remote Access Service that is used in ordinary TCP/IP networks.) H.323 RAS messages are concerned with general participation on the network (registration), specific involvement in particular calls between endpoints within and perhaps outside of the network zone (admission), and the status of endpoints (e.g., are they still "alive" or participating?).

H.245 is the conference control protocol. It pertains to negotiation between endpoints to establish a compatible set of media capabilities.

Because many user-settable parameters of the MultiTech gatekeeper software refer directly or indirectly to the H.225 protocol, we present a summary of common H.225 messages below.

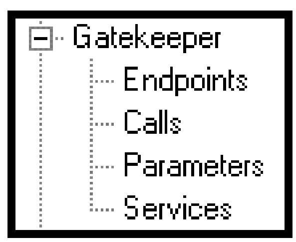
Summary of H.323 RAS* Messages (Registration, Admission, & Status) of the H.225 Call Signaling Protocol	
In a gatekeeper-controlled H.323 network, when call is made, the RAS channel between gatekeeper and endpoint is the first logical channel opened.	
Admission Control Messages	With an ARQ, an endpoint asks to participate in a phone call. The gatekeeper can either grant the request (by sending an ACF message) or deny the request (by sending an ARJ message). When admission is granted, the endpoints participating in the call can exchange (H.225) call signaling messages directly between themselves. When the call is done, each endpoint, in turn, requests disengagement (DRQ) and is granted disengagement (DCF) by the gatekeeper.
ARQ	Admission Request.
ACF	Admission Confirmation.
ARJ	Admission Rejection.
DRQ	Disengagement Request.
DCF	Disengagement Confirmation.
Bandwidth Control Messages	With a BRQ, an endpoint requests a certain amount of digital bandwidth for a call. If the gatekeeper grants the request, it returns a BCF message. If the gatekeeper denies the request, it returns a BRJ message, typically because all allocated data channels are in use. If a bandwidth request is rejected, it is possible for a call to be conducted
BRQ	Bandwidth Request
BCF	Bandwidth Confirmation
BRJ	Bandwidth Rejection
* RAS in H.323 has nothing to do with the Remote Access Service that is used in ordinary TCP/IP networks.	

Summary of H.225 RAS Messages (cont'd)	
Address Translation Messages for Out-of-Zone Calling	An LRQ is a request message between two H.323 gatekeepers to find the address of an H.323 endpoint. One gatekeeper is requesting the address translation services of the other. If the request is granted, an LCF message is returned. If the request is denied, an LRJ message is returned.
LRQ	Location Request.
LCF	Location Confirmation.
LRJ	Location Request Rejection.
Registration Control Messages	With an RRQ, an endpoint asks to be a participant in the network zone controlled by the gatekeeper. The gatekeeper can either grant the request (by sending an RCF message) or deny the request (by sending an RRJ message). If an endpoint's registration with the gatekeeper is temporary, its duration is specified in a TimeToLive field in the RCF message sent by the gatekeeper. After the registration duration has elapsed, the gatekeeper will send two IRQ messages (see "IRQ Interval" field in the **Network Parameters** screen) to see if the endpoint is still "alive." If the endpoint responds with an IRR, the registration will be extended. If not, the gatekeeper will send a URQ message to terminate the endpoint's registration. Thereafter, the endpoint must re-register with a full RRQ.
RRQ	Registration Request.
RCF	Registration Confirmation.
RRJ	Registration Rejection.

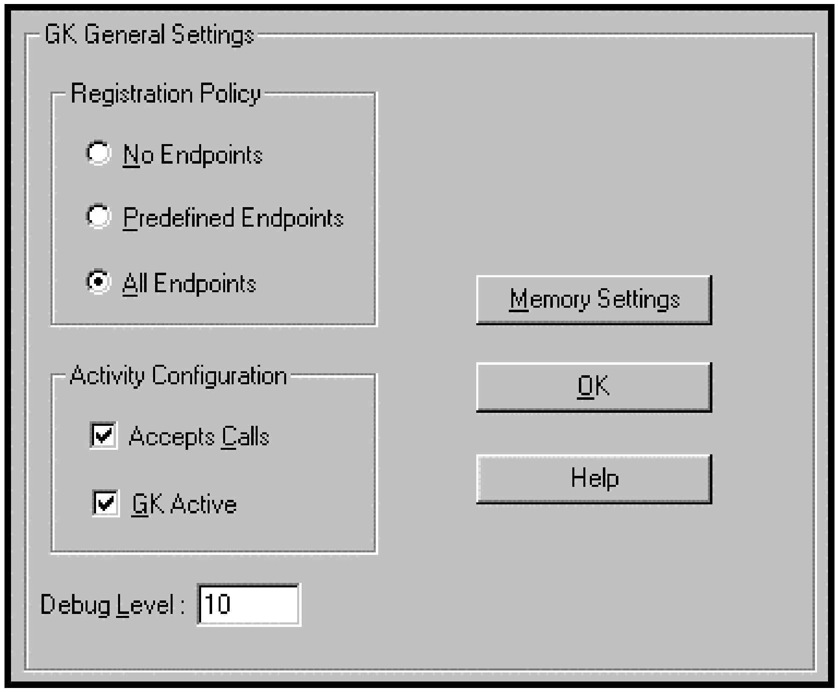
Summary of H.225 RAS Messages (cont'd)	
IRQ	Information Request
IRR	Extend Registration Request. (aka "keep-alive" request)
URQ	Unregister Request.
App URQ	When registration has timed out, the user application must decide how to respond.

MultiVOIP Gatekeeper Software Screens

Use the sidebar menu to access gatekeeper screens.

Accessing "Gatekeeper" Functions	
Pulldown	**Icon**
Sidebar	**Sidebar with Submenus**
	

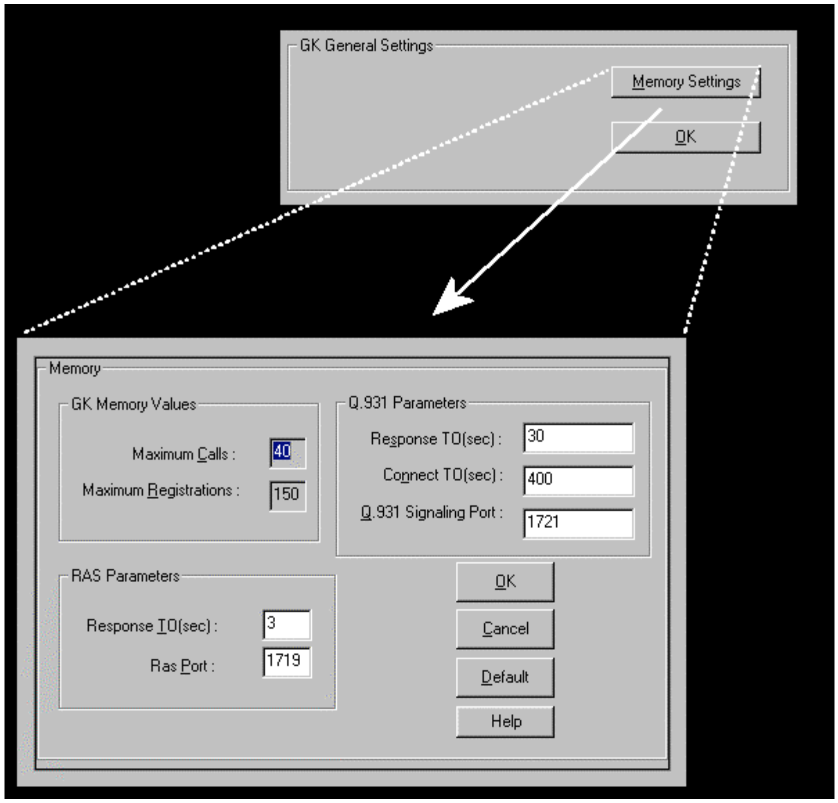
The fields in the main gatekeeper screen, the **GK General Settings** screen, are described in the table below.



GK General Settings Definitions		
Field Name	**Values**	**Description**
Registration Policy		
No Endpoints	Y/N	When selected, sets a policy whereby the Gatekeeper accepts no registrations.
Predefined Endpoints	Y/N	When selected, sets a strict zone policy, in which the Gatekeeper accepts only registrations that arrive from predefined endpoints. A strict zone policy controls network resources and services more tightly than an open zone policy.
All Endpoints	Y/N	When selected, sets an open zone policy, in which the Gatekeeper accepts any legal registration. Under this policy, the Gatekeeper can operate in “plug-and-play” mode.

GK General Settings Definitions (cont'd)		
Field Name	**Values**	**Description**
Activity Configuration		
Accepts Calls	Y/N	When checked, the voip unit will accept calls.
GK Active	Y/N	When checked, the voip unit's gatekeeper function is active.
Debug Level	0-100	The higher the value, the greater the details in Syslog or Console reports.
Buttons		
Memory Settings		Launches secondary screen on **Memory** issues. (See next table.)

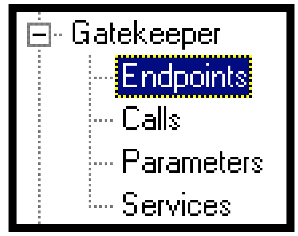
Click on the **Memory Setting** button to access the **Memory** screen.

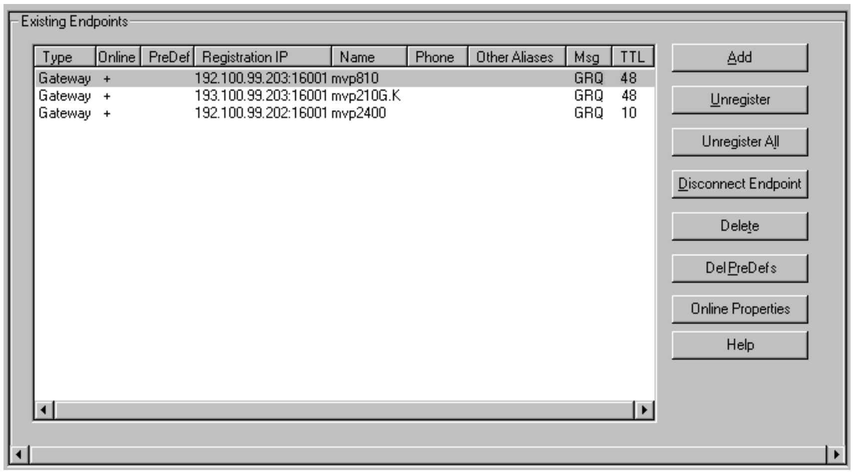


GK General Settings Definitions (cont'd)		
Field Name	**Values**	**Description**
GK Memory Values		
Maximum Calls	10, 20, 30	The maximum number of concurrent calls. MVP210G support 10 calls; MVP410G supports 20 calls; MVP810G supports 30 calls.
Maximum Registrations	2 - 250	Maximum number of endpoints that can be registered on the gatekeeper-controlled network.

GK General Settings Definitions (cont'd)		
Field Name	**Values**	**Description**
RAS Parameters		In H.323, RAS parameters pertain to Registration, Admission, and Status in the H.225 Call Signaling Protocol.
Response TO		The timeout (in seconds) before re-transmission of a RAS message that had previously fetched no response.
RAS Port		The RAS port for gatekeeper communication with endpoints. Default value = 1719
Q.931 Parameters		In H.323, Q.931 parameters are those that pertain to the set-up and tear- down of connections between H.323 endpoints.
Response TO (sec)		The timeout (in seconds) waiting for the TCP reply.
Connect TO (sec)		The timeout (in seconds) waiting for the Connect message of a call.
Q.931 Signaling Port		Logical port through which Q.931 protocol messages are handled. Default value = 1721
Buttons		
Default		Invokes default values for all parameters on the GK General Settings screen.

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



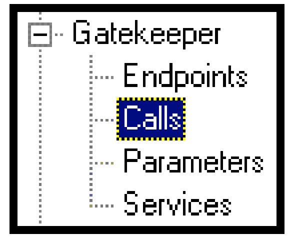


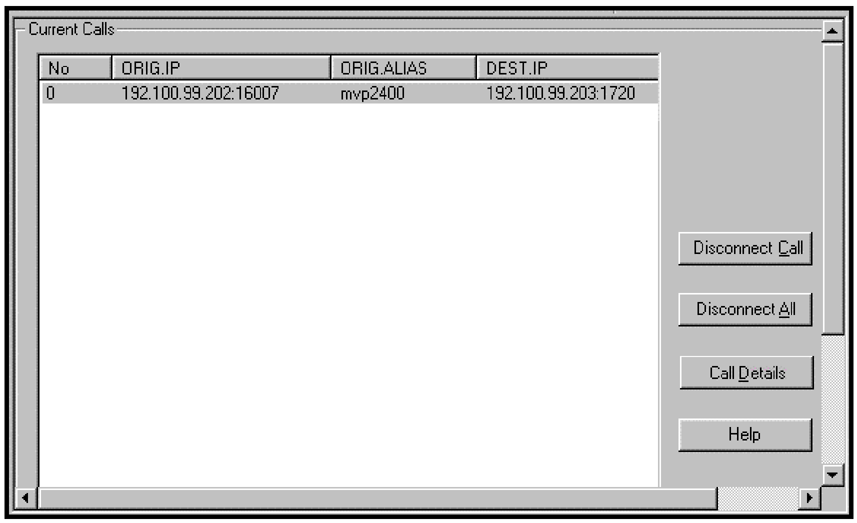
About Registration. When an endpoint registers with the gatekeeper, the endpoint is activated. That is, it becomes an acknowledged participant on the network (or on a particular zone of a network). Registration tells the gatekeeper that the endpoint is active and ready to receive calls. An endpoint's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

Existing Endpoints Parameter Definitions		
Field Name	**Values**	**Description**
Type	Gatekeeper, Gateway, MCU, Terminal, or Undefined.	The endpoint type . When an endpoint attempts to register with the Gatekeeper, the Gatekeeper compares the endpoint type with the predefined value. If the Gatekeeper detects a discrepancy, the registration is not accepted. If you are not sure of the endpoint type, select **Undefined**, which allows any endpoint of any type to register with the Gatekeeper. (Multipoint Control Units, MCUs, are used to facilitate conference calls.)
Online	+ or [blank]	When "+" appears, the endpoint's registration is dynamic or "online."
PreDef	+ or [blank]	When "+" appears, the endpoint's registration is static or "predefined."
Registration IP	n.n.n.n 0-255	The RAS address and RAS port of the endpoint.
Name		The H.323 ID alias of the endpoint.
Phone		The e164 alias number (conventional PSTN phone number)of the endpoint.
Other Aliases		Additional aliases for the endpoint: URL, e-mail address, transport address, party.address, or private network number (per ISO/IEC 11571). Alias addresses must be unique within a zone. Gatekeepers themselves cannot have aliases.
Msg	LRQ, RRQ, URQ, or AppURQ	The type of message sent by the endpoint when the mode for processing registration is manual. This can be an LRQ, RRQ, URQ, or AppURQ (which is a URQ sent by the Gatekeeper).).).
TTL	seconds	The time remaining in seconds before the TimeToLive timer expires. If the endpoint fails to reregister within this time, the endpoint is unregistered.

Existing Endpoints Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Command Buttons		
Add	--	Opens an empty Predefined Properties dialog box where you can predefine a new registration.
Unregister	--	Sends a URQ message to the selected endpoint, deleting the online (or dynamic) registration properties and unregistering the endpoint.
Unregister All	--	Sends a URQ to all the online endpoints in order to unregister them.
Disconnect Endpoint	--	Disconnects all calls with which the endpoint is involved.
Delete	--	Deletes the endpoint from the Gatekeeper database. A URQ will not be sent to the endpoint.
Del Pre-def	--	Deletes the predefined (static) properties of the endpoint.
Online Properties	--	Opens the Online properties screen or the selected endpoint whereupon are shown details of that endpoint's configuration.

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

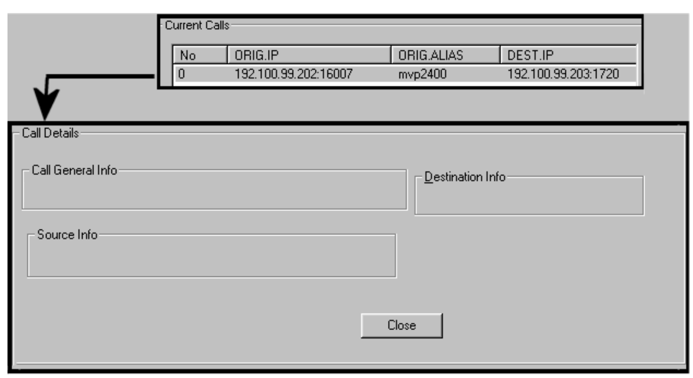




The **Calls** window displays a list of all the calls currently taking place and the basic details of the calls:

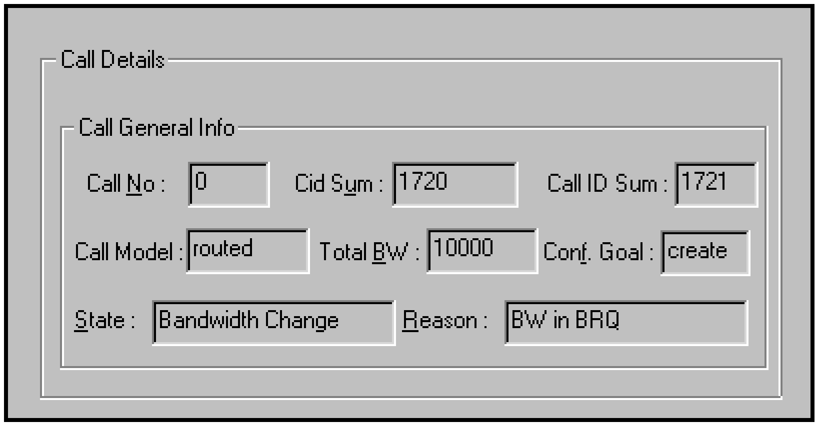
Current Calls Field Definitions		
Field Name	**Values**	**Description**
No	numeric	Number. A sequential number for identification in the list.
ORIG IP	n.n.n.n 0-255	Originating IP Address. IP Address of endpoint originating the call.
ORIG ALIAS	???	Originating Alias. The first alias given by the call's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

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



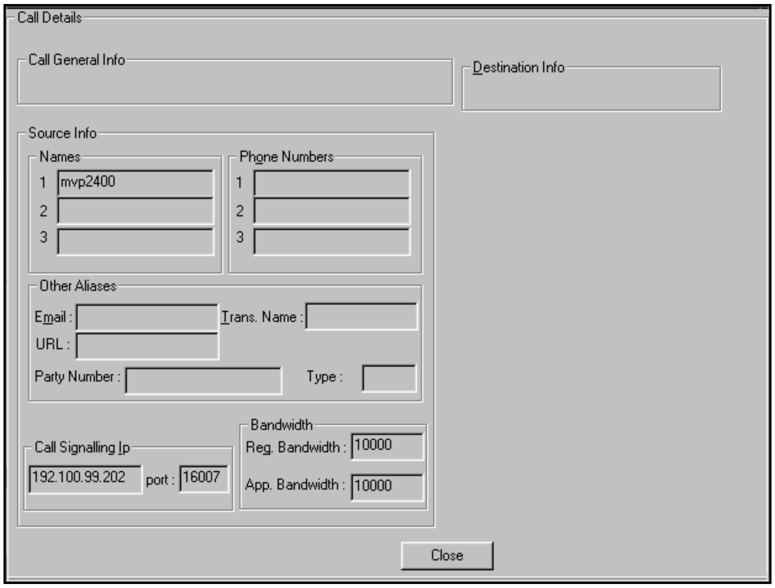
Clicking on an in-progress call, or using the "Call Details" button, yields full details about the call

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

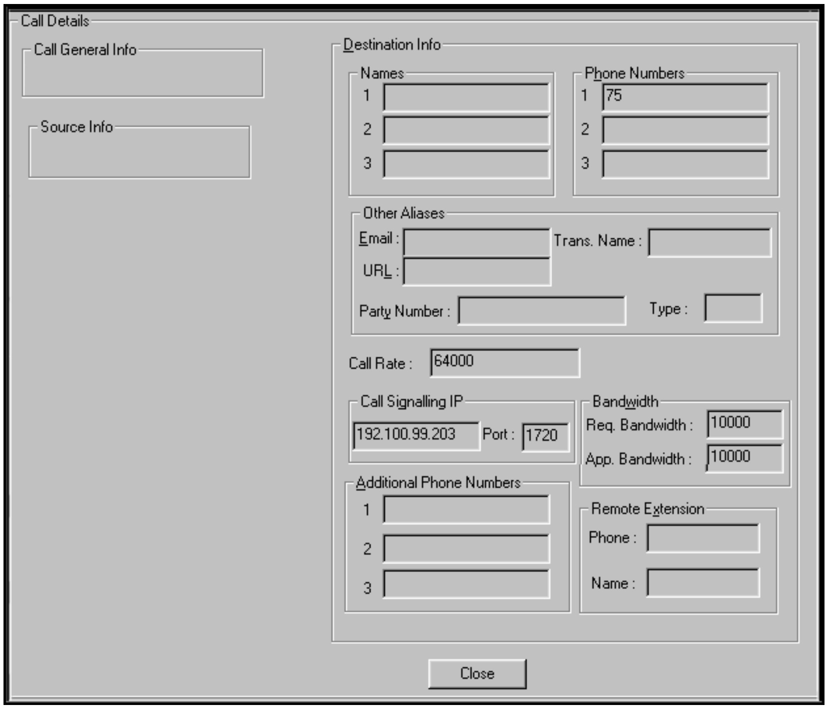


Call Details Field Definitions		
Field Name	**Values**	**Description**
Call General Info		
Call No.		Call Number. Accession number identifying a call in progress.
Cid Sum		The conference ID number (CID) is a unique non-zero value created by the calling endpoint and passed in various H.225.0 messages.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

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



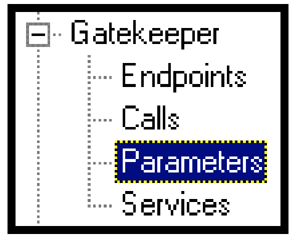
Call Details Field Definitions		
Field Name	**Values**	**Description**
Source Info fields		
Names		The H.323 alias name(s) for the originating endpoint.
Phone Numbers		The e164 alias phone number(s) of the originating endpoint.
Other Aliases: Email		An e-mail address of the originating endpoint.
OtherAliases: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

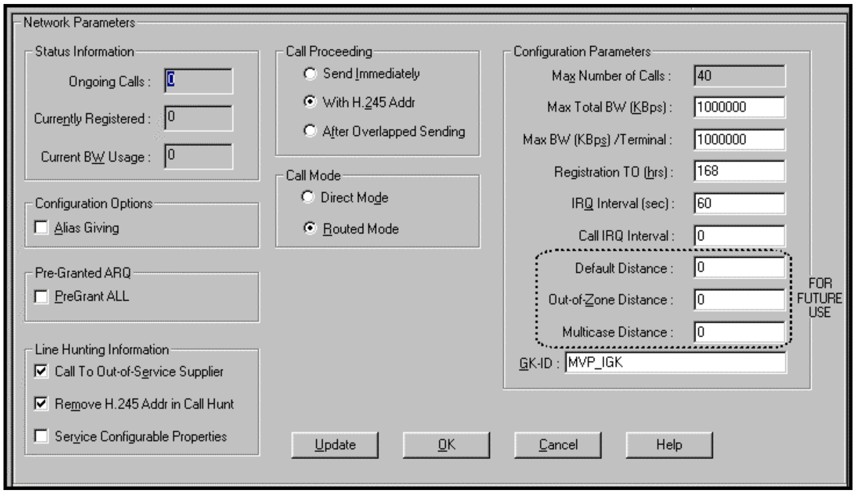


Call Details Field Definitions		
Field Name	**Values**	**Description**
Destination Info fields		
Names		The H.323 alias name used to make the call.
Phone Numbers		The e164 alias phone number used to make the call.
Other Aliases: Email		An e-mail address used to make the call.
OtherAliases: Trans. Name		A transport name alias used to make the call, consisting of an IP address and port number.
Other Aliases: URL		A URL alias used to make the call.
Call Signaling IP		The call signaling transport address of the called endpoint.

Call Details Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Destination Info fields		
Reg. Bandwidth		Requested Bandwidth. The bandwidth the called endpoint requested for the call, as it appears in the ARQ/BRQ messages.
App. Bandwidth		Approved Bandwidth. The bandwidth the Gatekeeper made available to the called endpoint for the call.
Additional Phone Numbers		These allow calling with more than one B-channel.
Remote Extension Phone		This is the phone number of the called endpoint on the remote LAN. It is used for calls between multiple gateways.
Remote Extension Name		This is the identifier (name) of the called endpoint on the remote LAN. It is used for calls between multiple gateways.

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





Network Parameter Definitions		
Field Name	**Values**	**Description**
Status Information		Use **Update** button to refresh the Status Information fields.
Ongoing Calls	number	The number of current calls with the Gatekeeper.
Currently Registered	number	The number of endpoints registered with the Gatekeeper.
Current BW Usage	number	The current bandwidth usage of the ongoing calls in Kbps.

Network Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Configuration Options		
Alias Giving	Y/N	When an endpoint sends an RRQ message, the Gatekeeper uses the additional aliases that were predefined for the endpoint as online aliases. This enables the Gatekeeper to assign terminal alias names through which the terminal can be accessed by others. The following are two examples of how this option can be used: **• Example of Alias Giving for a Terminal.** To make a terminal accessible by dialing 100, add the alias **100** to the terminal's predefined information, and select the **Alias Giving** option. When the terminal sends an RRQ message, the 100 alias becomes a dynamic (online) alias, and all calls to 100 will be directed to the terminal. **• Example of Alias Giving for Gateways.** To make all Gateways supply Service 80, add Service 80 to the Service Table, add the 80 alias as predefined information to all registered gateways, and select the **Alias Giving** option. When the gateways register, they will support Service 80.
Pre-Granted ARQ		
PreGrant ALL	Y/N	Select to cause the Gatekeeper to send a pregrantedARQ permission in the RCF message for each endpoint that wishes to register. The pregranted ARQ permission is given to both makeCall and answerCall with routed mode. When an endpoint re-ceives the permission, it may start the call with a Setup message or directly answer the call with a Connect message.

Network Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Line Hunting Information		
Call to Out-of-Service Supplier	Y/N	"Y" enables the sending of RAI messages. In a normal scenario, the gatekeeper will hunt among all the available endpoints that have been registered using the same tech-prefix. Each endpoint can inform the gatekeeper about its resource availability using an RAI (Resource Available Indication) message. Upon receiving an RAI message from an endpoint, the gatekeeper would consider that endpoint as an Out-of-Service Supplier. The 'Almost Out of Resources' configuration would allow the gatekeeper to hunt such Out-of-Service Supplier endpoints for routing the calls.
Remove H.245 Addr in Call Hunt	Y/N	When selected, the gatekeeper will not convey in its outgoing setup message the H.245 address received in an incoming setup message. This prevents H.323 terminals from establishing a channel for a call only to refuse the call later.
Service Configurable Properties	Y/N	When "Y" is selected, the gatekeeper will perform a Priority Based Line Hunting among those destinations registered using the same tech-prefix.

Network Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Proceeding		This parameter group pertains to the gatekeeper's handling of Q.931 "call-proceeding" messages.
Send Immediately	Y/N	Immediate return of call-proceeding message to originating endpoint. When selected, the gatekeeper will send the Q.931 call –proceeding message to the originating endpoint immediately after receiving that endpoint's call setup request.
With H.245 Addr	Y/N	When enabled, gatekeeper supplementary services will remove the H.245 address from the outgoing setup in order to prevent early H.245 establishment to the call's destination. This destination can be changed during Forward on Busy or during Forward on No Response (CFNR).
After Overlapped Sending	Y/N	Delayed return of call-proceeding message to originating endpoint. When selected (in routed mode), the gatekeeper will send a Q.931 call-proceeding message to the originating endpoint after it receives a return call-proceeding message back from the destination endpoint.

Network Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Mode		
Direct Mode		Sets the call mode to direct. In this mode, terminals send ARQ messages to the Gatekeeper, but pass the call signaling and media control signaling directly between them.
Routed Mode		Sets the call mode to routed. In this mode, terminals pass admission requests and call signaling through the Gatekeeper. Media control information is sent directly between the terminals. **Note**: Though direct calls consume fewer Gatekeeper resources, call control is better for indirect (or routed) calls.
Configuration Parameters		
Max Number of Calls		The maximum number of concurrent calls allowed in the zone. This number can be increased up to 100, in increments of 20, by purchasing additional concurrent call licenses.
Max Total BW (KBps)		The amount of bandwidth in Kbps that call traffic can consume at any given time.

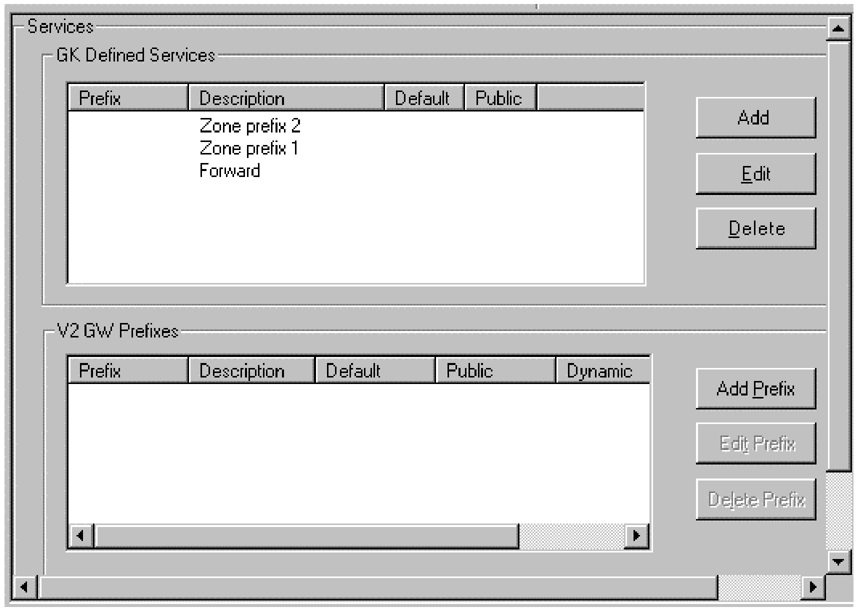
Network Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Configuration Parameters		
Registration TO (hrs)		Registration Timeout. Sets the number of hours of inactivity after which the dynamic registration of a terminal expires. Only the dynamic (online) properties will be unregistered. If the endpoint is also static (predefined), the static properties remain valid.
IRQ Interval (sec)		The interval, in seconds, between IRQ messages sent by the Gatekeeper. IRQ messages are sent to all online endpoints registered as dynamic in order to verify that the endpoints are online. The number you set determines the delay between two IRQ messages to the same endpoint. Choosing the desired delay should take into account the following factors: • IRQ messages add to the traffic already present over the network, and the shorter the delay, the more IRQ messages are sent. However, the longer the delay, the longer it takes for the Gatekeeper to detect dynamic registrations that have ceased to be online. • The delay parameter relates to the interval between two IRQ messages per one endpoint, so the actual number of the IRQ messages the Gatekeeper creates during this interval should be multiplied by the number of endpoints registered dynamically. • To disable the IRQ polling, set this value to zero. • The effective IRQ interval cannot fall below three times the RAS timeout. • IRQ messages will not be sent at a rate exceeding 20 per second.

Network Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Configuration Parameters		
Call IRQ Interval		The interval, in seconds, between IRQ messages sent by the Gatekeeper to query the status of calls. IRQ messages are sent to all online endpoints registered as dynamic and having ongoing calls in order to verify that the calls are still ongoing. The number you set determines the delay between two IRQ messages to the same endpoint regarding the same call. Choosing the desired delay should take into account the following factors: IRQ messages add to the traffic already present over the network, and the shorter the delay, the more IRQ messages are sent. However, the longer the delay, the longer it takes for the Gatekeeper to detect calls that are stale. The delay parameter relates to the interval between two IRQ messages per one call, so the actual number of the IRQ messages the Gatekeeper creates during this interval should be multiplied by the number of ongoing calls registered dynamically. To disable the IRQ polling, set this value to zero. The effective IRQ interval cannot fall below three times the RAS timeout. IRQ messages will not be sent at a rate exceeding 20 per second.

Network Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Configuration Parameters		
Default Distance		The "distance" (number device-to-device hops that a call must traverse between endpoints) allowed for endpoints which are only dynamically registered, such as an endpoint with no predefined values. This distance is compared to the distances of the neighbor gatekeepers and to the multicast distance in order to determine if an LRQ can be sent on behalf of the requesting endpoint. **NOTE**: The neighboring gatekeeper feature is not supported in the current software version.
Out-of-Zone Distance		The "distance" (number device-to-device hops that a call must traverse between endpoints) allowed for an out-of-zone endpoint that is making a call through the Gatekeeper. This distance is compared to the distances of the neighbor gatekeepers and to the multicast distance in order to see if an LRQ can be sent on behalf of the requesting endpoint. **NOTE**: The neighboring gatekeeper feature is not supported in the current software version.

Network Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Configuration Parameters		
Multicast Distance		The "distance" (number device-to-device hops that a call must traverse between endpoints) associated with sending an LRQ by multicast. **NOTE**: The neighboring gatekeeper feature is not supported in the current software version.
GK-ID		The name of the Gatekeeper. The terminals identify the Gatekeeper by this name during the discovery process. The Gatekeeper responds only to Discovery requests that either contain a matching Gatekeeper identifier or have no Gatekeeper identifier.
Update (button)	--	Click to update information in the "Status Information" fields of the Network Parameters screen.

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



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

Services Screen Definitions (cont'd)		
Field Name	**Values**	**Description**
GK Defined Services		
Public		For any GK-defined service being used, the user must select either "Default" or "Public." When Public is selected, the service is accessible to all endpoints that are not part of the zone.
V2 GW Prefixes		H.323 Version 2 enables the gateway to specify prefixes that the user should dial before the WAN number in order to make a call using a certain medium. E.g., the user could dial the prefix 3 for voice calls or 77 for H.320 video calls. The prefixes are defined in the RRQ message at registration. Prefix can be any H.323 alias, including an H.323 ID & mail address. When a terminal places a LAN to WAN call, it should add one of the prefixes to the dialed number. The Gatekeeper identifies the prefix & routes the call to the appropriate gateway. If more than one gateway supplies the same prefix, line hunting is possible between the gateways.
Prefix		Identifies the service. The prefix can be a numeric code, alphanumeric string, name, or phone number that the user dials. Per H.323 Vers. 2, prefixes can also be of URL and e-mail type. Also for H.323 Vers. 2, the type must precede the prefix. For example, TEL: 3 or NAME: John.
Description		A description of the service that is accessible by dialing the prefix.
Default		Select to make the service accessible to all endpoints that are not predefined in the zone.
Public		Select to make the service accessible to all endpoints that are not part of the zone.

Services Screen Definitions (cont'd)		
Field Name	**Values**	**Description**
V2 GW Prefixes		
Dynamic	Y/N	Indicates whether the service is static (essentially permanent) or timed & conditional (dynamic). This field indicates whether the service has been added manually (non-dynamically; field value =N) or dynamically (field value = Y) as part of registration from endpoints.
Buttons		These buttons allow you add, edit, or delete a selected service or prefix.

GK Defined Service Types

You can either define your own Gatekeeper services, or use any of the built-in services, which are predefined internally and supported by the Gatekeeper.

Example of a Gatekeeper Service

You can define a service named TECHSUPP and register five different terminals that provide technical support. Any call directed to TECHSUPP can connect to one of the five terminals.

To do so:

1. Add a service with a prefix TECHSUPP.
2. Make sure the terminals register with the additional alias TECHSUPP.
3. When a call for TECHSUPP arrives, the Gatekeeper automatically routes the call to one of terminals that provides the TECHSUPP Service.

Endpoints must be registered with the service name to receive calls for the service. This is achieved using one of the following methods:

- The endpoint is pre-configured using its own configuration. Then, using RAS messages, the endpoint is registered with a name or a phone number identical to the service prefix.
- The service prefix is predefined for the endpoint, using the configuration application of the Gatekeeper as an ID or phone number, and the **Alias Giving** option is activated. See the description of the **Alias Giving** option in the Network Parameters window section.

Built-in Gatekeeper-Defined Services

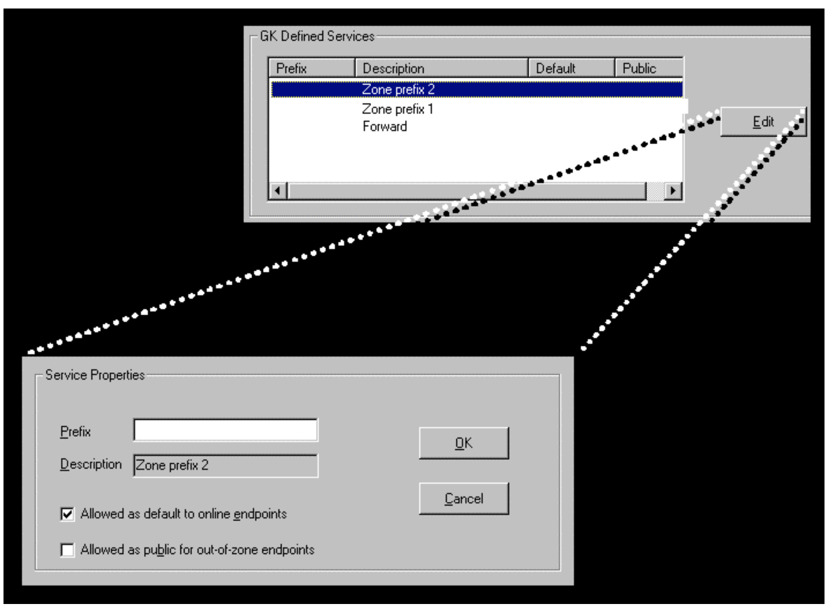
The current version of the Gatekeeper software supports the following services:

- Zone Prefix 1
- Zone Prefix 2
- Forward

Service Types: Zone Prefixes (1 and 2)

Note: This feature is for future use. Zone Prefix functionality is implemented in the current software release but it operates only in a context of neighboring gatekeeper functionality, which is not implemented in the current release. The discussion of this section pertains to a context in which neighboring gatekeeper functionality is implemented. Such functionality is included in plans for subsequent software releases.

MultiVOIP gatekeeper can operate in multiple zones. You can define one or two prefixes for a zone by entering the prefix for the services. The zone prefix functions in the same way as a telephone area code.



When one of the zone prefixes is defined, no calls from other zones can reach this zone, unless preceded by the prefix. If an endpoint in a zone dials a zone prefix before its number, and the Gatekeeper cannot resolve it in its zone, the Gatekeeper attempts to locate and route the call to a Neighbor Gatekeeper with the same prefix. For such calls, the Gatekeeper strips the zone prefix and then applies the destination location mechanism to route the call to its final destination.

You can use the zone prefix to devise a dialing plan in a multi-zone environment. If zone prefixes are not defined, the zone accepts the following calls:

- Calls prefixed to a service defined in the zone and allowed as default.
- Calls to on-line terminals in the zone.
- Calls to terminals marked as Forward in the zone.

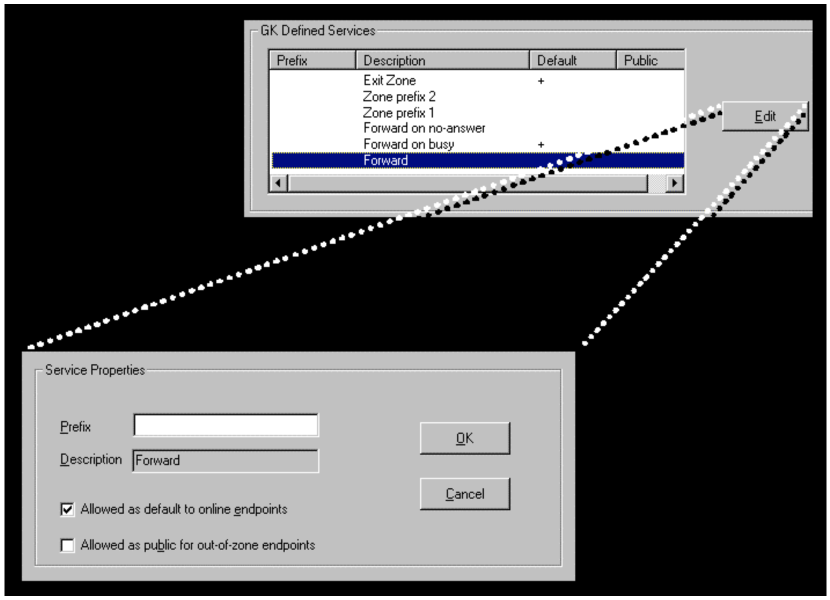
Example of comparing Zone prefix use when using Zone prefixes

- Zone A has a 01 prefix. In this zone, the phone number of user A1 is 123 and the phone number of user A2 is 456. The Gateway service has a prefix of 8.
- Zone B has a 02 prefix. In this zone, the phone number of user B1 is 123 and the phone number of user B2 is 456. The Gateway number is 555444 and the Gateway service has a prefix of 9.
- A1 calls A2 by dialing 456.
- A1 calls using zone A Gateway 8555444.
- A1 calls B1 by dialing 02123.

Note: The call is completed only if the Gateway service is allowed as default in Zone B.

Service Types: Forward

This call-forwarding feature is non-contingent, i.e., it forwards all calls for a selected station to another destination.



Gatekeeper Log Data Data Files

The embedded gatekeeper does not create files for its log data. For debugging or other purposes, such log data can be viewed/printed using a SysLog application program or HyperTerminal.

Gatekeeper Software User License Agreement

The MultiVOIP Gatekeeper software is licensed by Multi-Tech Systems, Inc., to the original end-user purchaser of the product, hereafter referred to as “Licensee.” The License includes the distribution disc, other accompanying programs, and the documentation. The MultiVOIP Gatekeeper software, hereafter referred to as “Software,” consists of the computer program files included on the original distribution disc.

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Chapter 12 Warranty, Service, and Tech Support

Limited Warranty

Multi-Tech Systems, Inc. ("MTS") warrants that its products will be free from defects in material or workmanship for a period of two years from the date of purchase, or if proof of purchase is not provided, two years from date of shipment. MTS MAKES NO OTHER WARRANTY, EXPRESSED OR IMPLIED, AND ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE HEREBY DISCLAIMED. This warranty does not apply to any products which have been damaged by lightning storms, water, or power surges or which have been neglected, altered, abused, used for a purpose other than the one for which they were manufactured, repaired by the customer or any party without MTS's written authorization, or used in any manner inconsistent with MTS's instructions.

MTS's entire obligation under this warranty shall be limited (at MTS's option) to repair or replacement of any products which prove to be defective within the warranty period, or, at MTS's option, issuance of a refund of the purchase price. Defective products must be returned by Customer to MTS's factory—transportation prepaid.

MTS WILL NOT BE LIABLE FOR CONSEQUENTIAL DAMAGES AND UNDER NO CIRCUMSTANCES WILL ITS LIABILITY EXCEED THE PURCHASE PRICE FOR DEFECTIVE PRODUCTS.

Repair Procedures for U.S. and Canadian Customers

In the event that service is required, products may be shipped, freight prepaid, to our Mounds View, Minnesota factory:

Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, MN 55112
Attn: Repairs, Serial # ________________

A Returned Materials Authorization (RMA) is not required. Return shipping charges (surface) will be paid by MTS.

Please include, inside the shipping box, a description of the problem, a return shipping address (it must be a street address, not a P.O. Box number), your telephone number, and if the product is out of warranty, a check or purchase order for repair charges.

For out-of-warranty repair charges, go to www.
multitech.com/documents/warranties

Extended two-year overnight replacement service agreements are available for selected products. Please call MTS at (888) 288-5470, extension 5308, or visit our web site at www.multitech.com/programs/orc
for details on rates and coverages.

Please direct your questions regarding technical matters, product configuration, verification that the product is defective, etc., to our Technical Support department at (800) 972-2439 or email
tsupport@multitech.com. Please direct your questions regarding repair expediting, receiving, shipping, billing, etc., to our Repair Accounting department at (800) 328-9717 or (763) 717-5631, or email
mtsrepair@multitech.com.

Repairs for damages caused by lightning storms, water, power surges, incorrect installation, physical abuse, or used-caused damages are billed on a time-plus-materials basis.

Technical Support

Multi-Tech Systems has an excellent staff of technical support personnel available to help you get the most out of your Multi-Tech product. If you have any questions about the operation of this unit, or experience difficulty during installation you can contact Tech Support via the following:

Contacting Technical Support

Country	By E-mail	By telephone
France	support@multitech.fr	(33) 1-64 61 09 81
India	support@ multitechindia.com	(91) 124-340778
U.K.	support@ multitech.co.uk	(44) 118 959 7774
U.S. & Canada	tsupport@ multitech.com	(800) 972-2439
Rest of World	support@ multitech.com	(763) 785-3500

Internet: http://www.multitech.com/ **_forms/email_tech_support.htm**

Please have your product information available, including model and serial number.

Chapter 13: 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 89/336/EEC of 3 May 1989 on the approximation of the laws of Member States relating to electromagnetic compatibility,
and
Council Directive 73/23/EEC of 19 February 1973 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 Declaration

NOTE: 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 instructions, 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.

This device complies with Part 15 of the FCC rules.

Operation is subject to the following two conditions:

(1) This device may not cause harmful interference.

(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.

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.

FCC Part 68 Telecom

1. This equipment complies with part 68 of the Federal Communications Commission Rules. On the outside surface of this equipment is a label that contains, among other information, the FCC registration number. This information must be provided to the telephone company.
2. As indicated below, the suitable jack (Universal Service Order Code connecting arrangement) for this equipment is shown. If applicable, the facility interface codes (FIC) and service order codes (SOC) are shown.
3. An FCC compliant telephone cord and modular plug is provided with this equipment. This equipment is designed to be connected to the telephone network or premises wiring using a compatible modular jack that is Part 68 compliant. See installation instructions for details.
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. If advance notice is not practical, the telephone company will notify the customer as soon as possible.
5. The telephone company may make changes in its facilities, equipment, operation, or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice to allow you to make necessary modifications to maintain uninterrupted service.
6. If trouble is experienced with this equipment (the model of which is indicated below), please contact Multi-Tech Systems, Inc. at the address shown below for details of how to have repairs made. If the equipment is causing harm to the network, the telephone company may request you to remove the equipment form t network until the problem is resolved.
7. 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.
8. Manufacturer: Multi-Tech Systems, Inc.
   Trade name: MultiVOIP
   Model number: MVP2400
   FCC registration number: US: AU7DDNAN46050

Modular jack (USOC):	RJ-48C
Service center in USA:	Multi-Tech Systems, Inc. 2205 Woodale Drive Mounds View, MN 55112 Tel: (763) 785-3500 FAX: (763) 785-9874

Canadian Limitations Notice

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

Appendix A: Expansion Card Installation (MVP24-48 & MVP30-60)

Installation

Both the MVP2410 and the MVP3010 use the same mechanical chassis. This chassis accommodates a second MultiVOIP circuit card or motherboard module. The add-on module for the MVP2410 is the MVP24-48 product; the add-on module for the MVP3010 is the MVP30-60 product. The MVP2410G will not accept an expansion card because its second card slot is occupied by gatekeeper circuitry.

To install an expansion card into an MVP2410 or MVP3010, you must:

1. Power down and unplug the MVP2410/3010 unit.

2. Using a Phillips or star-bit screwdriver, remove the blank plate at the rear of the MVP2410/3010 chassis (see Figure A-1). Save the screw.

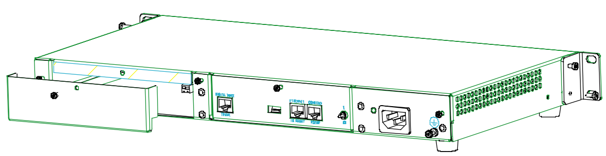


Figure A-1: Remove Plate Covering Expansion Slot

3. A power cable for the expansion card (+5V) is already present within the MVP2410/3010 unit. This power cable has a two-pin "molex" connector. When the rear cover plate has been removed, the cable is accessible from the rear at the right side of the expansion slot. Locate this connector within the MVP2410/3010. See Figure A-2.

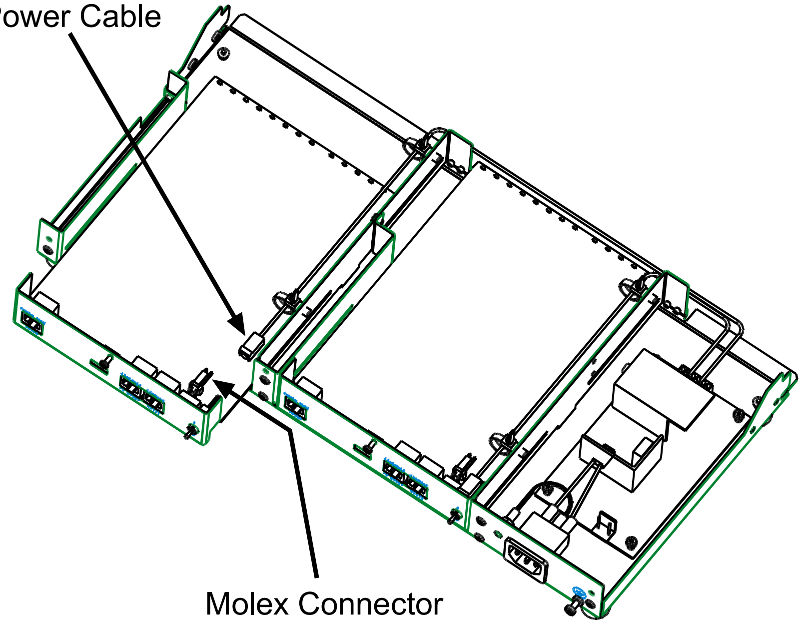


Figure A-2: MVP2410/3010 Chassis (top/rear view)

4. While keeping the power cable out of the way, fit the MVP24-48 or MVP30-60 card into the grooves of the expansion slot. Push it in far enough to allow connection of the power cable to the receptacle on the vertical plate of the expansion card. (See Figure A-2.) Connect the power cable.

5. Push the expansion card fully into the chassis. See Figure A-3.

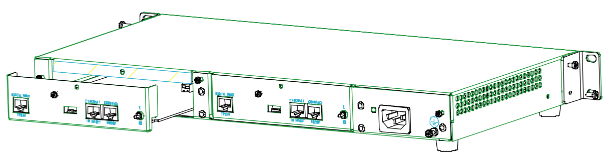


Figure A-3: Sliding Expansion Card into Chassis

Secure the vertical plate of the expansion card to the chassis with a screw.

Operation

The MVP2410/3010 front panel has two sets of identical LEDs. In the MVP2410/3010 without an expansion card, only the left-hand set of LEDs is functional. However, when the MultiVOIP unit has been upgraded with an MVP24-48 or MVP30-60 expansion card, the right-hand set of LEDs will also become active.

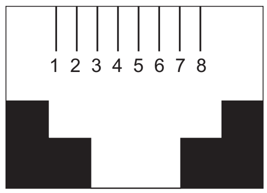
Remember that the expansion card must be configured as though it were simply another complete MultiVOIP unit: it requires its own T1/E1 line; it requires its own connection to a computer running the MultiVOIP configuration software. All of the procedures and operations that apply to the original motherboard of the MVP2410/3010 will also apply to the expansion card. See applicable User Guide chapters for details.

Appendix B: Cable Pinouts

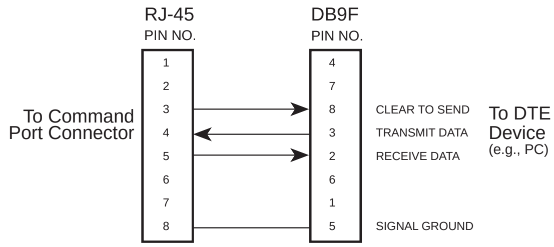
Appendix B: Cable Pinouts

Command Cable

RJ-45 Connector



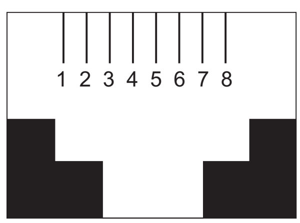
End-to-End Pin Info



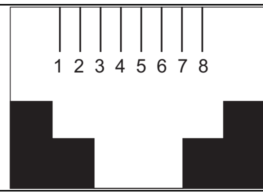
RJ-45 connector plugs into Command Port of MultiVOIP.
DB-9 connector plugs into serial port of command PC (which runs MultiVOIP configuration software).

Ethernet Connector

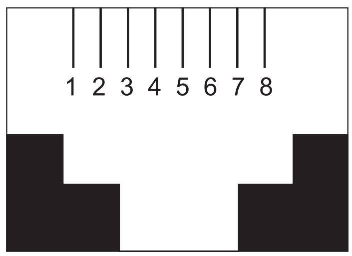
The functions of the individual conductors of the MultiVOIP's Ethernet port are shown on a pin-by-pin basis below.

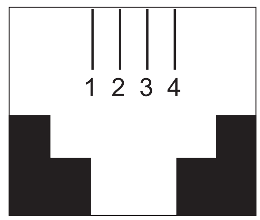
RJ-45 Ethernet Connector	Pin	Circuit Signal Name
	1	TD+ Data Transmit Positive
	2	TD- Data Transmit Negative
	3	RD+ Data Receive Positive
	6	RD- Data Receive Negative

T1/E1 Connector

T1/E1 Connector	
	1 2 } Receive Pair (from line) 4 5 } Transmit Pair (to line)

Voice/Fax Channel Connectors





Pin Functions (E&M Interface)		
Pin	**Descr**	**Function**
1	M	Input
2	E	Output
3	T1	4-Wire Output
4	R	4-Wire Input, 2-Wire Input
5	T	4-Wire Input, 2-Wire Input
6	R1	4-Wire Output
7	SG	Signal Ground (Output)
8	SB	Signal Battery (Output)

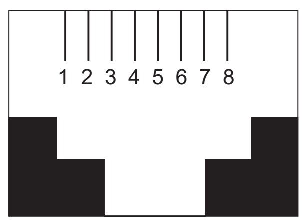
Pin Functions (FXS/FXO Interface)			
FXS Pin	**Description**	**FXO Pin**	**Description**
2	N/C	2	N/C
3	Ring	3	Tip
4	Tip	4	Ring
5	N/C	5	N/C

ISDN BRI RJ-45 Pinout Information

The S/T interface uses an 8-conductor modular cable terminated with an 8-pin RJ-45 plug. An 8-pin RJ-45 jack located on the terminal is used to connect the terminal to the DSL (Digital Subscriber Loops) using this modular cable.

The table below shows the Pin Number, Terminal Pin Signal Name and Network Pin Signal name for the S/T interface.

Pin	TE Signal		NT Signal	Pin
1	Not used		Not used	1
2	Not used		Not used	2
3	Tx+		Rx+	3
4	Rx-		Tx-	4
5	Rx+		Tx+	5
6	Tx-		Rx-	6
7	Not used		Not used	7
8	Not used		Not used	8



TE=Terminal Equipment

NT=Network

ISDN Interfaces: “ST” and “U”

The MVP410ST and MVP810ST are ISDN-BRI voip units that use an S/T outlet interface. You will need an NT1 device to connect these units to any network equipment that has the “U” ISDN interface. In the UK, and in many European countries, the telco supplies an NT1 device for ISDN-BRI service.

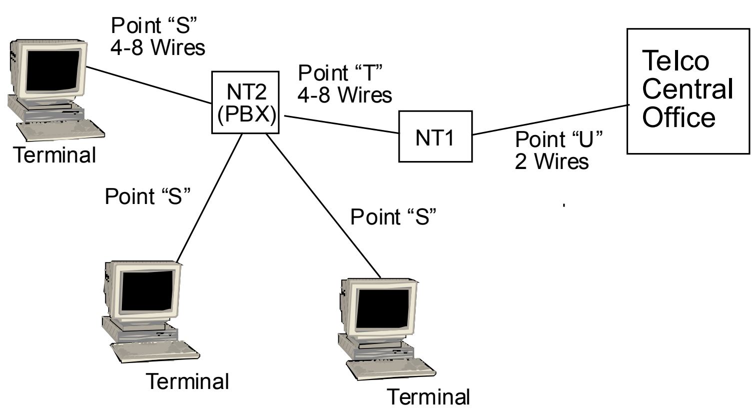
An ISDN Basic Rate (BRI) U-Loop consists of two conductors from the telco central office to the customer premises. The equipment on both sides of the U-loop accommodates the extensive length of the U-loop and the noisy environment in which it may operate. At the customer premises, the U-loop is terminated by an NT1 (network termination 1) device. An NT1 device makes an end-user’s 4-wire terminal equipment compatible with the telco’s 2-wire twisted pair ISDN-BRI line.

The NT1 drives an S/T bus. The S/T bus is usually made up of 4 wires, but in some cases may be 6 or 8 wires.

“S” and “T” refer to connection points in the ISDN specification.

When a PBX is present, *S* refers to the connection between the PBX and the terminal. (“Terminal” can mean any sort of end-user ISDN device: data terminals, telephones, FAX machines, voip units, etc.)

Point *T* refers to the connection between the NT1 device and customer supplied equipment. Terminals can connect directly to the NT1 device at point *T*, or there may be a PBX (private branch exchange, i.e., a customer-owned telephone exchange). The figure below shows “S” and “T” connection points in an ISDN network.



Appendix C: 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

Well-Known Port Numbers

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

Appendix D: Installation Instructions for MVP428 Upgrade Card

Installation Instructions for MVP428 Upgrade Card

In this procedure, you will install an additional circuit board into the MVP410, converting it from a 4-channel voip to an 8-channel voip.

Summary: **(A) Attach four standoffs to main circuit card.**
(B) Mate the 60-pin connectors (male connector on main circuit card; female on upgrade card).
(C) Attach upgrade card to main circuit card (4 screws).

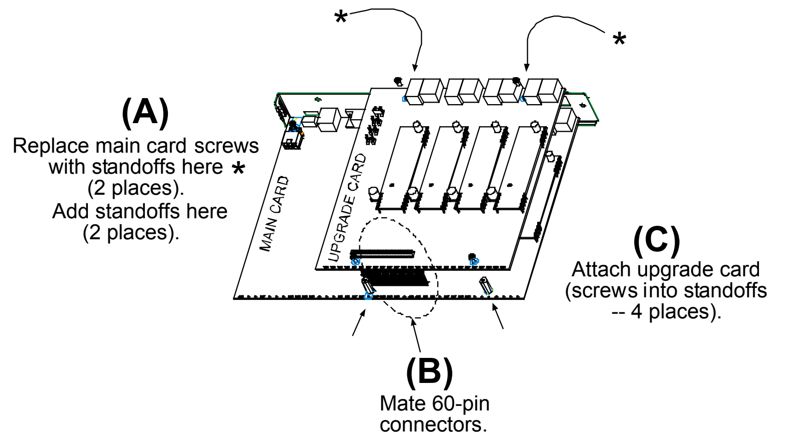


Figure D-1. Installation Summary

Procedure in Detail

1. Power down and unplug the MVP410 unit.

2. Using a Phillips driver, remove the blank cover plate at the rear of the MVP410 chassis. Save the screws.

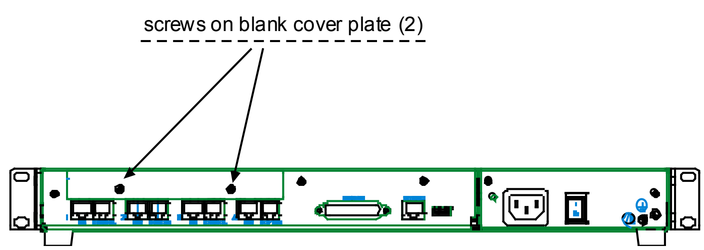


Figure D-2: Removing screws from blank cover plate

3. Using a Phillips driver, remove the three screws that secure the main circuit board and back panel assembly to the chassis.

NOTE:
Follow standard ESD precautions to protect the circuit board from static electricity damage.

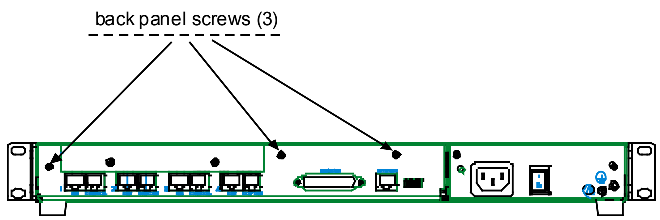


Figure D-3: Removing screws from back panel

4. Slide the main circuit board out of the chassis far enough to unplug the power connector.

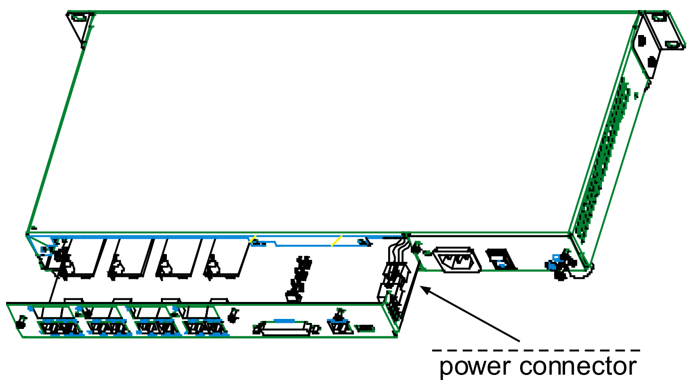


Figure D-4: Accessing power connector

5. Unplug the power connector from the main circuit board.
6. Slide the main circuit board completely out of the chassis and place on a non-conductive, static-safe tabletop surface.
7. Remove mounting hardware (2 screws, 2 nuts, and 4 standoffs) from its package.

8. On the phone-jack side of the circuit card, three screws attach the circuit card to the back panel. Two of these screws are adjacent to the four phone-jack pairs. Remove these two screws.

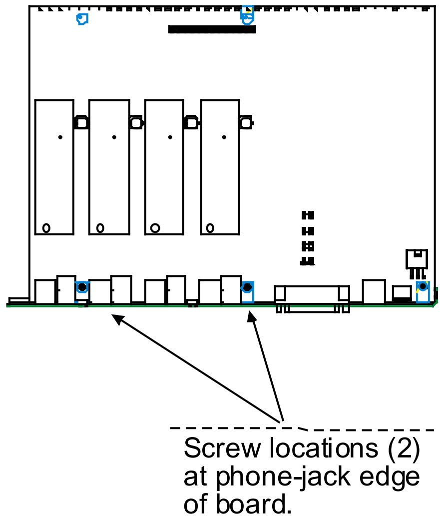


Figure D-5: Screws to be removed and replaced with standoffs (phone-jack edge of board; top view)

9. Replace these two screws with standoffs.
10. There are two copper-plated holes at the LED edge of the circuit card. Place a nut beneath each hole (lockwasher side should be in contact with board) and attach a standoff to each location).

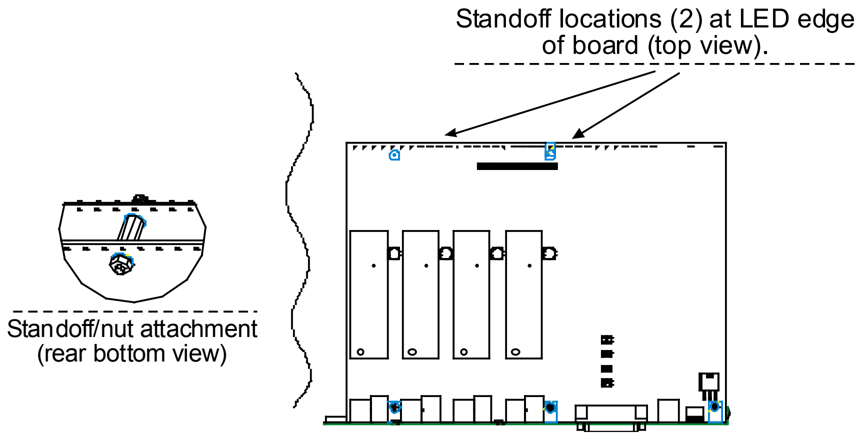


Figure D-6: Standoffs at LED edge of board (top view)

11. Locate the male 60-pin vertical connector near the LED edge of the main circuit card. Check that pins are straight and evenly spaced. If not, then correct for straightness and spacing. Locate the 60-pin female connector on the upgrade circuit card.
12. Set the upgrade circuit card on top of the main circuit card. Align the upgrade card's 4 pairs of phone-jacks with the 4 pairs of holes in the backplane of the main card. Slide the phone jacks into the holes.
13. Mate the upgrade card's 60-pin female connector with the main card's 60-pin male connector.

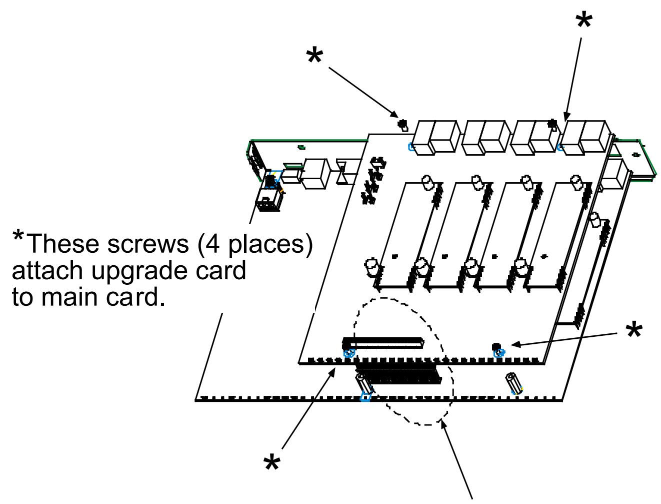


Figure D-7. Attaching upgrade card to main circuit card (secure 4 Phillips screws; mate 60-pin connectors)

14. There are four copper-plated attachment holes, two each at the front and rear edges of the upgrade card. Attach the upgrade card to the main card using 4 Phillips screws. The upgrade card should now be firmly attached to the main card.
15. Slide the main circuit card back into the chassis far enough to allow re-connection of power cable.
16. Re-connect power cable.
17. Slide the main circuit card fully into the chassis.
18. Re-attach the backplane of the main circuit card to the chassis with 3 screws.

Appendix E: Call States & Reasons for Embedded Gatekeepers

Call States and Call Reasons

MultiVOIP units with embedded gatekeeper functionality track call states and the reasons for those states. We present here a complete listing of these call states and call reasons. These relate to the **Call Details** screen, which is a secondary screen that can be launched from the **Calls** ("Current Calls") screen of the embedded gatekeeper software.

Possible Call States of which the Embedded Gatekeeper Software can be notified

No	State	Description
1	Wait Orig Admission	Needs application approval for sending an ACF to the origin.
2	Wait NW Setup	Waits for the Setup message to arrive after sending an ACF back to the origin.
3	Wait Dest Admission	Needs application approval for sending an ACF to the Destination.
4	Wait NW Connect	Waits for the Connect message to arrive after sending an ACF back to the destination.
5	Wait Dest Connect	Needs application approval for Connecting the destination to the origin.
6	Connect Sent To Orig	The Gatekeeper passed the Connect message of the destination back to the origin.
7	Setup Arrived	A Setup message is received from the network.
8	Wait Orig Offering	Needs application approval before sending a Setup message from the originator of the call to the destination.
9	Wait LRQ	Needs application approval to do an LRQ for the call.
10	Sending LRQ	A notification is given for each outgoing LRQ.

		Call States Listing (cont'd)
No	**State**	**Description**
11	LRQ Sent	An LRQ was sent on the network. Waiting for a reply.
12	received LCF	An LCF was received. The application should decide whether or not to accept it.
13	Setup Sent To Dest	The Gatekeeper sends the Setup message to the Destination.
14	Call To Forward Service	A call is to the forward service and hence will be disconnected.
15	Dial Tone	A Setup message was sent.
16	Proceeding	Waiting for the end user's phone to ring.
17	Setup Ack	A Notification given on a SetupAck message arrived from the destination of a call.
18	Dest Alert	The end user's phone is ringing.
19	Disconnected	A connected call was disconnected.
20	Call Cannot Complete	The destination did not connect. Waiting for application instruction whether to disconnect or perform address translation again after the application sets new addresses.
21	Connected	The call connected.
22	Address Resolution	The application may replace call addresses. The various reasons for this state are mentioned in the reason table. Each time the address is changed by the Gatekeeper (such as stripping a zone prefix or translating an alias to IP address), the application is notified with the suitable reason.
23	Address Resolution Done	The application may review the final destination. This can be sent with two reasons: 1)AddressFound 2)NeedLRQ. The application needs to approve the final result or reroute the call
24	Admission Reject	Lets the application know about the reject.
25	Setup Reject	Lets the application know about the reject.
26	Orig Admission Reject	Lets the application know about the reject.
27	Dest Admission Reject	Lets the application know about the reject.

		Call States Listing (cont'd)
No	**State**	**Description**
28	GK Disconnected Call	Lets the application know about a call that the Gatekeeper disconnected.
29	Wait Line Hunting	Line Hunting failed on one line. Line Hunting can still continue after application approval.
30	DRJ Sent	Lets the application know when sending a DRJ.
31	DCF Sent	Lets the application know when sending a DCF.
32	ARJ Sent	Lets the application know when sending an ARJ.
33	GK Initiated DRQ	Lets the application know when the Gatekeeper initiated a DRQ.
34	Bandwidth Change	Notification of a change of the call bandwidth.
35	Idle	The call was terminated. Waiting for the application to release the handle.
36	Unknown	State unknown.

Call Reasons sent to Embedded Gatekeeper Software with respect to a Call State.

No	State	Description
1	Undefined	No reason.
2	Resource Unavailable	The call was rejected because of a lack of Gatekeeper resources.
3	Invalid Endpoint	The ARQ/DRQ was rejected because no valid endpoint was identified.
4	Route Call To GK	The destination ARQ was rejected because no Setup message preceded it.
5	Lines Busy	The call cannot be completed because Line Hunting failed.
6	Destination Out Of Service	The call cannot be completed because the destination cannot be reached.
7	Destination Busy	The call cannot be completed because destination is busy.
8	No Answer at Destination	The call cannot be completed because the user at the destination did not answer in the given time.
9	Destination Rejected the Call	The call cannot be completed because the party at the destination rejected the call.
10	Origin Disconnected	A connected call was disconnected because of the origin. The reason for state Disconnected.
11	Destination Disconnected	A connected call was disconnected because of the destination. The reason for state Disconnected.
12	New Admission from Origin	The reason for address resolution because of a new admission.
13	New Setup from Origin	The reason for address resolution because of a new Setup.
14	Origin Setup	The reason for wait offering when the Setup is not the first message in call. (An ARQ was received.)
15	Destination Info In LRQ	An LCF arrived with no CallSignal Address but with a new destinationInfo alias. The Gatekeeper sent an Address Resolution state with this reason in order to translate the new found alias to a valid IP address.

		Call Reasons Listing (cont'd)
No	**State**	**Description**
16	No Change. Service Prohibited	The reason for address resolution. The required service is not allowed for the endpoint.
17	Zone Prefix Removed	The reason for address resolution after the zone prefix was removed.
18	Exit Zone Prefix Removed	The reason for address resolution after the exit zone prefix was removed.
19	Ip Address Set	The reason for address resolution after the IP address was found from the aliases.
20	Address Forwarded	The reason for address resolution after finding that the call should be forwarded.
21	Address Found	The reason for state AddressResolutionDone.
22	Need to Send LRQ	The reason for state AddressResolutionDone.
23	Failure in App. Event Handler	The call cannot be completed because of a failure in the application event handler. (For example, the return value < 0.)
24	Internal Failure	The call cannot be completed because of an internal error.
25	Service Not Allowed	The call cannot be completed because a required service is not allowed.
26	Exit Zone Not Allowed	The call cannot be completed because it was dialed without an exit zone prefix, or the exiting zone is not allowed for call.
27	No Destination in Call	The call cannot be completed because it was dialed without a destination.
28	Cannot Send LRQ	The call cannot be completed because an LRQ cannot be sent.
29	Address Not Found after LRQ	The call cannot be completed because an LCF was not accepted for the LRQ.
30	Call Not Register	The reason for sending a DRJ.
31	Origin Connected First	The reason for a Connect message that arrives without first asking the application. This happens when the origin is already connected when the destination connects, which is an error.
32	DCF to Origin	A DCF was sent to the origin.
33	DCF to Dest	A DCF was sent to the destination.
34	App. Disconnected Destination	An application initiated disconnect of the destination (associated with the Call Cannot Complete state or with GK Disconnect Call state.)

		Call Reasons Listing (cont'd)
No	**State**	**Description**
35	App. Timeout	The call was disconnected because of a timeout on waiting for an application reply.
36	call cannot complete-missing line hunting addresses	The call cannot be completed because no application Line Hunting addresses were supplied when the application Line Hunting mode was on.
37	Additional Address Complete	The Additional Address information exchange has been completed.
38	Additional Address	The Additional Address procedure (digit collection) is in progress.
39	GK Connect Call	The Gatekeeper has connected to the call as the destination, forming a one-legged call. This reason accompanies the Wait Dest Connect state when the application replies to Setup Arrived with the Send Connect To Orig reply.
40	GK Initiated Call	This reason accompanies the Address Resolution and Connected states to indicate a one-legged call initiated from the Gatekeeper by the application.
41	Unknown	Reason unknown

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