MultiVOIP®

Voice over IP gateways

User Guide

Digital Models: MVP-2410/3010

AnalogModels: MVP-130/130FXS
& MVP-210/410/810

BRI Models: MVP-410ST/810ST

Multi Tech Systems®

User Guide
S000249K

Analog MultiVOIP Units (Models MVP130,MVP130FXS, MVP210, MVP410, MVP810)
ISDN-BRI MultiVOIP Units (Models MVP410ST, and MVP810ST)
Digital MultiVOIP Units (Models MVP2410, & MVP3010)
Upgrade Units (MVP24-48 and MVP30-60)

Multi-Tech Systems, Inc. makes no representations or warranties with respect to the contents hereof and specifically disclaims any implied warranties of merchantability or fitness for any particular purpose. Furthermore, Multi-Tech Systems, Inc. reserves the right to revise this publication and to make changes from time to time in the content hereof without obligation of Multi-Tech Systems, Inc. to notify any person or organization of such revisions or changes.

Record of Revisions

Revision	Description
A	Initial Release. (05/10/02)
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)
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)
I	Updated for x.06 software release. (04/29/04)
J	Updated ISDN-BRI to software release 5.06.AK (08/27/04)
K	Updated for x.07 software release. MVP130FXS added. Added link to website for warranty (05/12/09)

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.

Trademark

Trademark of Multi-Tech Systems, Inc. is the Multi-Tech logo. Windows and NetMeeting are registered trademarks of Microsoft.

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

For warranty information, please visit: http://www.multitech.com

CONTENTS

CHAPTER 1: OVERVIEW 7

ABOUT THIS MANUAL 8
INTRODUCTION TO TI MULTIVOIPS (MVP2410 & MVP24-48) 11
T1 Front Panel LEDs *16*
INTRODUCTION TO EI MULTIVOIPS (MVP3010 & MVP30-60) 18
E1 Front Panel LEDs *23*
E1 LED Descriptions *24*
INTRODUCTION TO ANALOG MULTIVOIPS (MVP-130/130FXS, MVP-210/410/810 & MVP428) 25
Analog MultiVOIP Front Panel LEDs *30*
INTRODUCTION TO ISDN-BRI MULTIVOIPS (MVP410ST & MVP810ST) 34
ISDN BRI MultiVOIP Front Panel LEDs *38*
ISDN-BRI MultiVOIP LED Descriptions *39*
COMPUTER REQUIREMENTS 40
SPECIFICATIONS 41
Specs for Digital T1 MultiVOIP Units *41*
Specs for Digital E1 MultiVOIP Units *42*
Specs for Analog/BRI MultiVOIP Units *43*
INSTALLATION AT A GLANCE 44
RELATED DOCUMENTATION 44

CHAPTER 2: QUICK START INSTRUCTIONS 45

INTRODUCTION 46
MULTIVOIP STARTUP TASKS 46
*Phone/IP Details *Absolutely Needed* Before Starting the Installation* *47*
Gather IP Information 47
Gather Telephone Information (T1) 47
Gather Telephone Information (E1) 48
Gather Telephone Information (Analog) 48
Gather Telephone Information (ISDN BRI) 49
Obtain Email Address for VOIP (for email call log reporting) 50
Identify Remote VOIP Site to Call 50
Identify VOIP Protocol to be Used 50
Placement *51*
The Command/Control Computer (Specs & Settings) *51*
Quick Hookups *52*
Load MultiVOIP Control Software onto PC *58*
Phone/IP Starter Configuration *59*
Phonebook Starter Configuration (with remote voip) *63*
Outbound Phonebook 63
Inbound Phonebook 67
Phonebook Tips *70*
Phonebook Example *73*
Connectivity Test *78*
Troubleshooting *82*

CHAPTER 3: MECHANICAL INSTALLATION AND CABLING 84

INTRODUCTION 85
SAFETY WARNINGS 85
Lithium Battery Caution *85*
Safety Warnings Telecom *85*
UNPACKING YOUR MULTIVOIP 86
Unpacking the MVP2410/3010 .... *86*
Unpacking the MVP-410/810 .... *87*
Unpacking the MVP210 .... *88*
Unpacking the MVP-130/130FXS *89*
RACK MOUNTING INSTRUCTIONS FOR MVP-2410/3010 & MVP-410/810 90
Safety Recommendations for Rack Installations *91*
19-Inch Rack Enclosure Mounting Procedure *92*
CABLING 93
Cabling Procedure for MVP2410/3010 .... *93*
Cabling Procedure for MVP-410/810 .... *95*
Cabling Procedure for MVP-410ST/810ST *100*
Cabling Procedure for MVP210 .... *104*
Cabling Procedure for MVP-130/130FXS *108*

CHAPTER 4: SOFTWARE INSTALLATION 110

INTRODUCTION 111
LOADING MULTIVOIP SOFTWARE ONTO THE PC 111
UN-INSTALLING THE MULTIVOIP CONFIGURATION SOFTWARE 118

CHAPTER 5: TECHNICAL CONFIGURATION FOR DIGITAL T1/E1 MULTIVOIPS (MVP2410, MVP3010) 121

CONFIGURING THE DIGITAL T1/E1 MULTIVOIP 122
LOCAL CONFIGURATION 124
Pre-Requisites *124*
IP Parameters 124
T1 Telephony Parameters (for MVP2410) 125
E1 Telephony Parameters (for MVP3010) 126
SMTP Parameters (for email call log reporting) 127
Local Configuration Procedure (Summary) *128*
Local Configuration Procedure (Detailed) *129*
Modem Relay *150*

CHAPTER 6: TECHNICAL CONFIGURATION FOR ANALOG/BRI MULTIVOIPS (MVP-130/130FXS, MVP-210, MVP-410, MVP-810 & MVP-410ST/810ST) 206

CONFIGURING THE ANALOG/BRI MULTIVOIP 207
LOCAL CONFIGURATION 210
Pre-Requisites *210*
IP Parameters 210
Analog Telephony Interface Parameters (for MVP-130/130FXS/210/410/810) 211
ISDN-BRI Telephony Parameters (for MVP-410ST/810ST) 212
SMTP Parameters (for email call log reporting) 213
Local Configuration Procedure (Summary) *214*
Local Configuration Procedure (Detailed) *215*

Modem Relay *238*

CHAPTER 7: T1 PHONEBOOK CONFIGURATION 307

CONFIGURING THE MVP2410 MULTIVOIP PHONEBOOKS 308
T1 PHONEBOOK EXAMPLES 336
3 Sites, All-T1 Example *336*
Configuring Mixed Digital/Analog VOIP Systems *342*
Call Completion Summaries *351*
Variations in PBX Characteristics *354*

CHAPTER 8: E1 PHONEBOOK CONFIGURATION 355

MVP3010 INBOUND AND OUTBOUND MULTIVOIP PHONEBOOKS 356
Free Calls: One VOIP Site to Another *357*
Local Rate Calls: Within Local Calling Area of Remote VOIP *358*
National Rate Calls: Within Nation of Remote VOIP Site *360*
Inbound versus Outbound Phonebooks *361*
PHONEBOOK CONFIGURATION PROCEDURE 364
E1 PHONEBOOK EXAMPLES 387
3 Sites, All-E1 Example *387*
Configuring Digital & Analog VOIPs in Same System *394*
Call Completion Summaries 402
Variations in PBX Characteristics *405*
International Telephony Numbering Plan Resources *406*

CHAPTER 9: ANALOG/BRI PHONEBOOK CONFIGURATION 408

CHAPTER 10: OPERATION AND MAINTENANCE 410

OPERATION AND MAINTENANCE 411
System Information screen *411*
Statistics Screens *414*
About Call Progress *414*
About Logs *420*
About IP Statistics *425*
About Link Management *429*
About Packetization Time *432*
About T1/E1 and BRI Statistics *435*
About Registered Gateway Details *447*
MULTIVOIP PROGRAM MENU ITEMS 449
Configuration Option *451*
Configuration Port Setup *451*
Date and Time Setup *452*
Obtaining Updated Firmware *452*
Implementing a Software Upgrade *456*
Identifying Current Firmware Version 456
Downloading Firmware 457
Downloading Factory Defaults 460
Downloading IFM Firmware (Analog Voips only) *462*
Setting and Downloading User Defaults *466*
Setting a Password (Windows GUI) *468*

Setting a Password (Web Browser GUI) *472*
Un-Installing the MultiVOIP Software *473*
Upgrading Software *475*
FTP SERVER FILE TRANSFERS ("DOWNLOADS") 476
WEB BROWSER INTERFACE 486
SYSLOG SERVER FUNCTIONS 491

CHAPTER 11: REGULATORY INFORMATION 494

EMC, Safety, and R&TTE Directive Compliance *495*
FCC DECLARATION 495
Industry Canada *496*
FCC Part 68 Telecom *496*
Canadian Limitations Notice *497*

APPENDIX A: EXPANSION CARD INSTALLATION (MVP24-48 & MVP30-60) 498

INSTALLATION 499
OPERATION 501

APPENDIX B: CABLE PINOUTS 502

APPENDIX B: CABLE PINOUTS 503
Command Cable *503*
Ethernet Connector *503*
T1/E1 Connector *504*
Voice/Fax Channel Connectors *504*
ISDN BRI RJ-45 Pinout Information *506*
ISDN Interfaces: "ST" and "U" *507*

APPENDIX C: TCP/UDP PORT ASSIGNMENTS 508

WELL KNOWN PORT NUMBERS 509
PORT NUMBER ASSIGNMENT LIST 509

APPENDIX D: INSTALLATION INSTRUCTIONS FOR MVP428 UPGRADE CARD 510

INSTALLATION INSTRUCTIONS FOR MVP428 UPGRADE CARD 511

INDEX 515

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 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 & MVP130FXS.
4. ISDN-BRI MultiVOIP units, models MVP410ST & MVP810ST.

These MultiVOIP units can inter-operate with the earlier generation of MultiVOIP products (the MVP200, MVP400, MVP800, MVP120, etc.)

In this manual, an “x” suffix to a model number means the assertion applies to all suffix types of that model number. E.g., “MVP410x” refers collectively to MVP410 and MVP410ST.

The table below describes the vital characteristics of the various models described in this manual.

MultiVOIP Product Family					
Description / Model		**MVP-2410**	**MVP 24-48**	**MVP 3010**	**MVP 30-60**
Function		T1 digital VOIP unit	T1 digital VOIP add-on card	E1 digital VOIP unit	E1 digital VOIP add-on card
Capacity		24 channels	24 added channels	30 channels	30 added channels
Chassis/ Mounting		19" 1U rack mount	circuit card only	19" 1U rack mount	circuit card only
Description / Model	**MVP 810**	**MVP 428**	**MVP 410**	**MVP 210**	**MVP-130/ 130FXS**
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. "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.

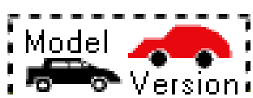


Figure 1-1: Variable Model/Version Icon

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 (MVP2410 & MVP24-48)

We proudly present MultiTech's T1 Digital Multi-VOIP products. 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. These voice-over-IP products have fax capabilities. 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. They can also accommodate T1 lines of the ISDN Primary Rate Interface type (ISDN-PRI).

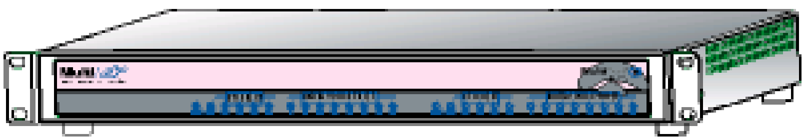


Figure 1-2. MultiVOIP MVP2410 LEDs

Scale-ability. The MVP2410 is 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 MVP2410 accepts its outbound traffic from a T1 trunk that's connected to either a PBX or to a telco/carrier. The MVP2410 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 MVP2410 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 MVP2410 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 MVP2410 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 MultiVOIP MVP2410 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 MVP2410 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.

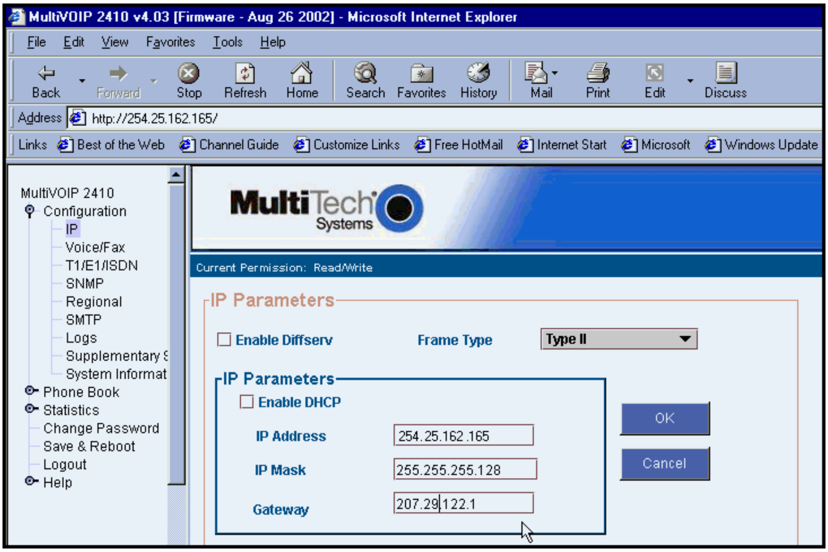
Ports. The MVP2410 has one 10/100 Mbps Ethernet LAN interface and one Command port for configuration. An MVP2410 upgraded with the MVP24-48 kit will have two Ethernet LAN interfaces and two Command ports.

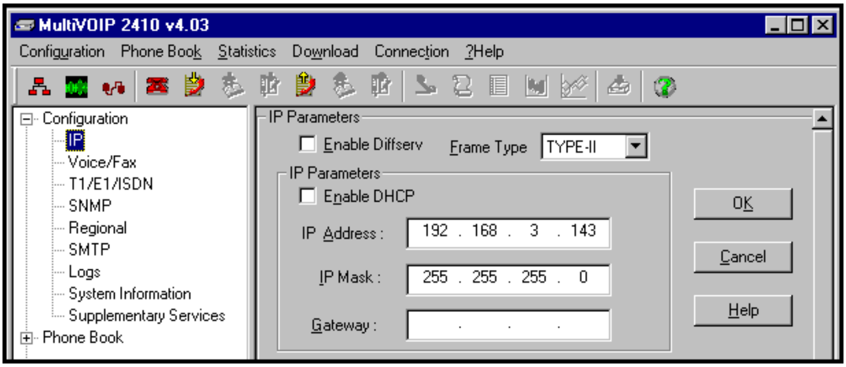
PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

Gatekeeper. T1 voip systems can have gatekeeper functionality by adding, as an endpoint, a Multi-Tech 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 stand-alone gatekeeper software performs all of the standard gatekeepers functions (address translation, admission control, and bandwidth control) and also supports many valuable optional functions (call control signaling, call authorization, bandwidth management, and call management).

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).

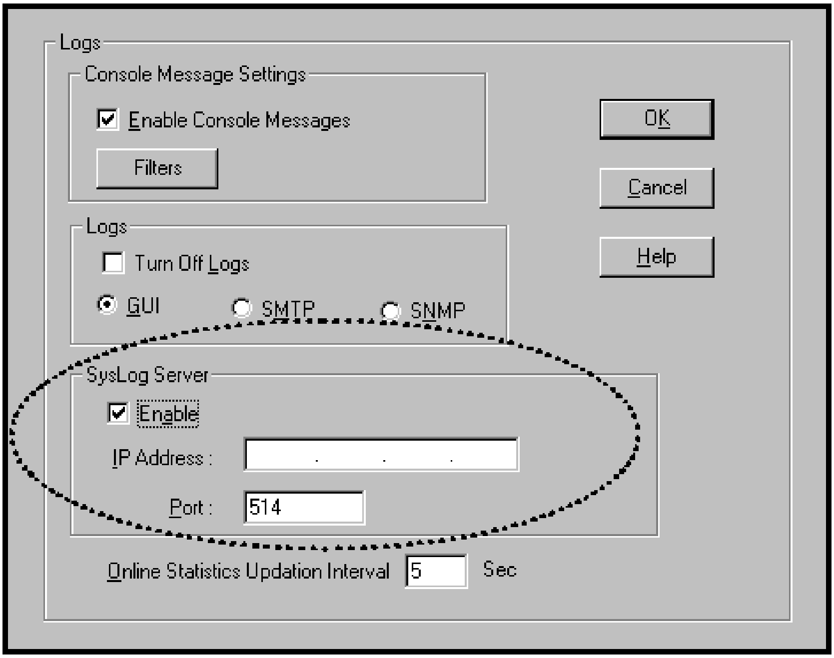




The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

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

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.

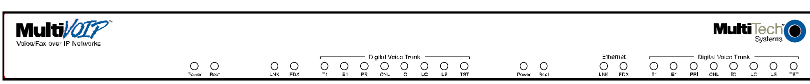


Figure 1-3: MVP2410 LEDs

T1 LED Descriptions. The descriptions below apply to the 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.

MVP2410 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 MVP2410 is booting.
FDX	Full-Duplex & Collision LED. This LED indicates whether the Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions.
LNK	Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
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 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.

Introduction to EI 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).

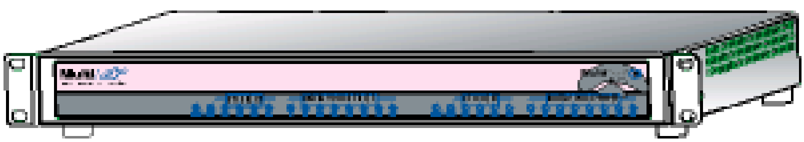


Figure 1-4. MultiVOIP MVP3010 Chassis

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.

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 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.

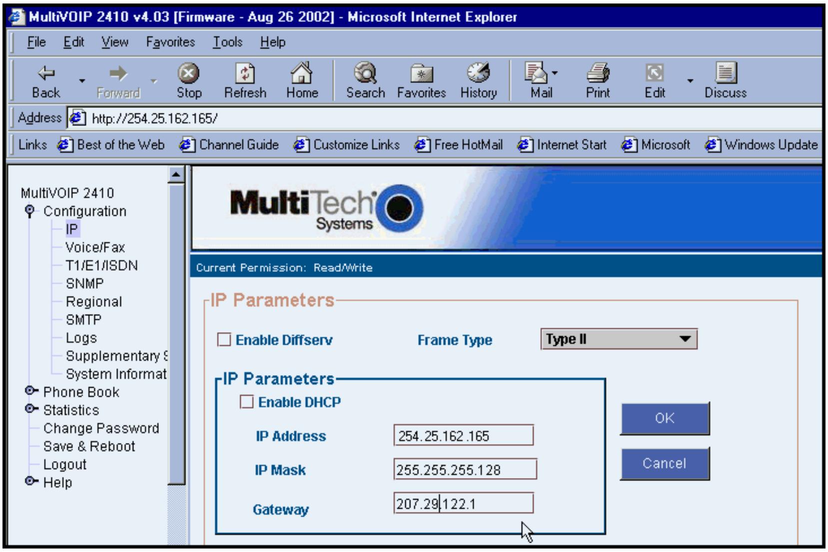
Ports. The MVP3010 also has a 10/100 Mbps Ethernet LAN interface, and a Command port for configuration. An MVP3010 upgraded with the MVP30-60 kit will have two Ethernet LAN interfaces and two Command ports.

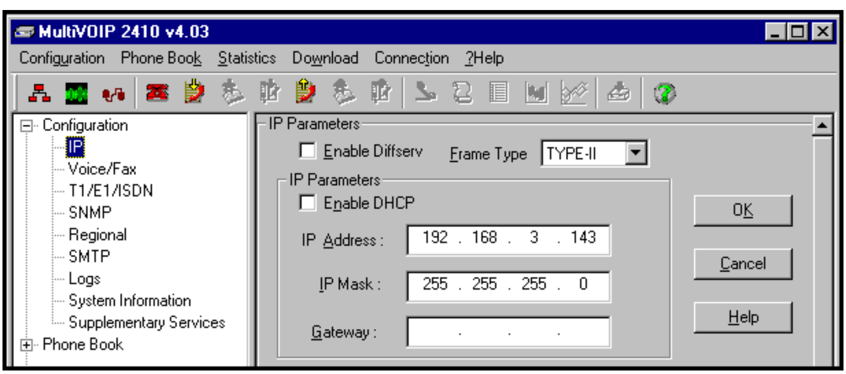
PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

Gatekeeper. E1 voip systems can have gatekeeper functionality by adding, as an endpoint, a Multi-Tech 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 stand-alone gatekeeper software performs all of the standard gatekeepers functions (address translation, admission control, bandwidth control, and zone management) and also supports many valuable optional functions (call control signaling, call authorization, and bandwidth management).

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).

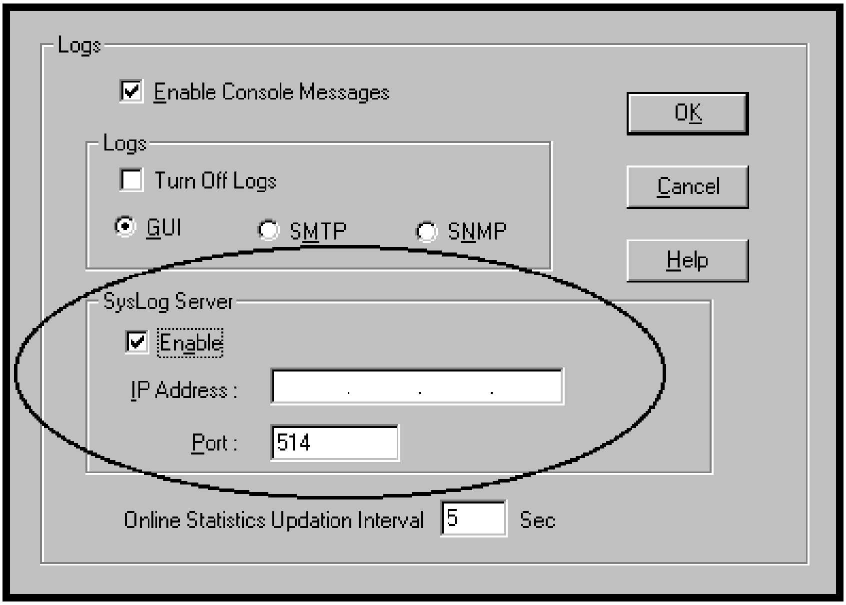




The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

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.

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.

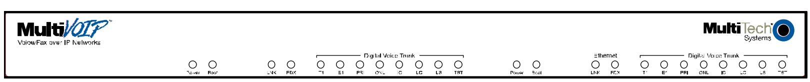


Figure 1-5: MVP3010 LEDs

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.
FDX	Full-Duplex & Collision LED. This LED indicates whether the Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions.
LNK	Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
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.

Introduction to Analog MultiVOIPs (MVP-130/130FXS, MVP-210/410/810 & MVP428)

VOIP: The Free Ride. We proudly present Multi-Tech's MVP-130/130FXS and MVP-210/410/810 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 analog MultiVOIPs inter-operate readily with T1 or E1 MultiVOIP units.

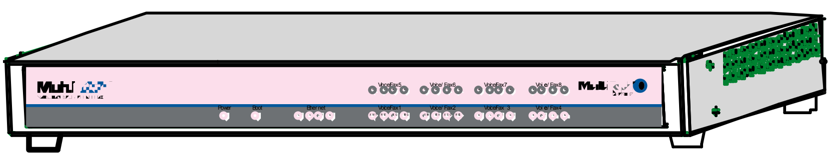


Figure 1-6: MVP-410/810 Chassis

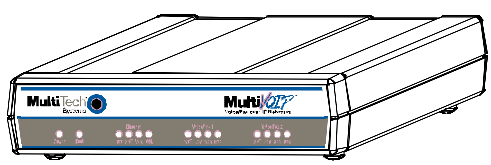


Figure 1-7: MVP-210 Chassis

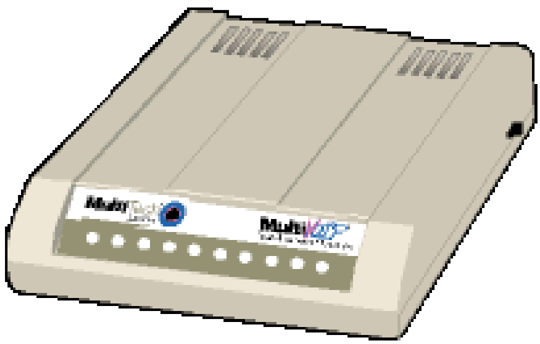


Figure 1-8: MultiVOIP MVP-130/130FXS Chassis

Capacity. MultiVOIP model MVP810 is an eight-channel unit, the model MVP410 is a four-channel, the model MVP210 is a two-channel units, the MV130 is a single-channel unit and the MVP130FXS is a single-channel unit that supports the FXS telephony interface only. 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, MVP-130/130FXS 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.

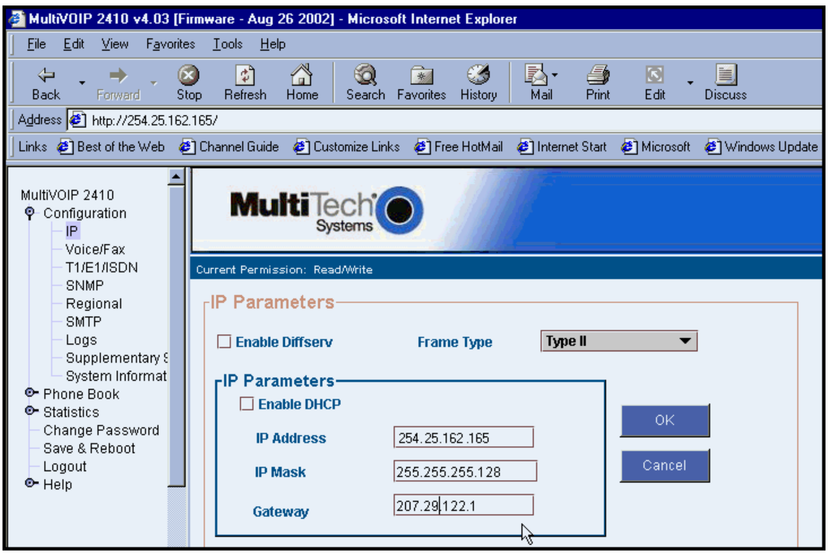
Data Compression & Quality of Service. The analog MultiVOIP unit comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

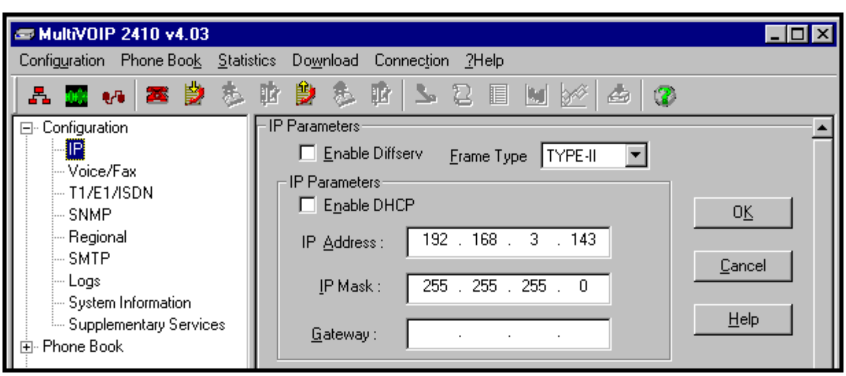
PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

Gatekeepers. For voip systems built with MultiTech's analog gateway units, users can have 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 stand-alone gatekeeper software performs all of the standard gatekeepers functions (address translation, admission control, and bandwidth control) and also supports many valuable optional functions (call control signaling, call authorization, bandwidth management, and call management).

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).

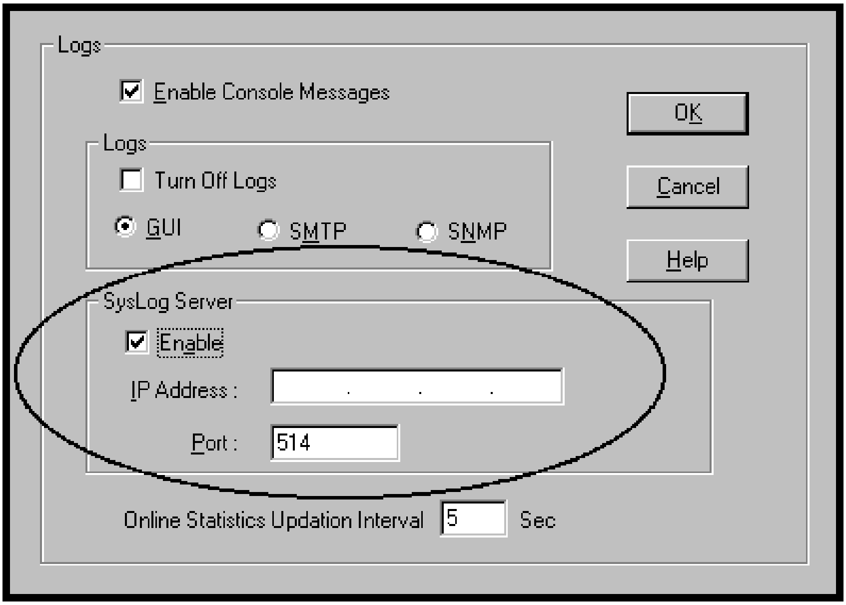




The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

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.

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.

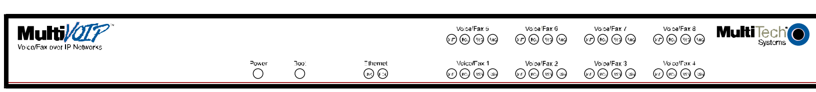


Figure 1-9. MVP410/810 LEDs

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

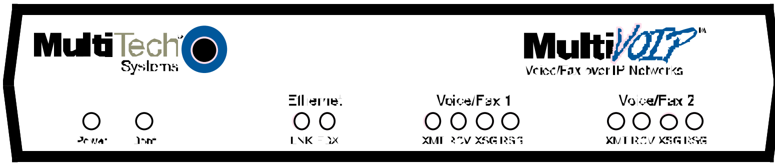


Figure 1-10. MVP210 LEDs

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

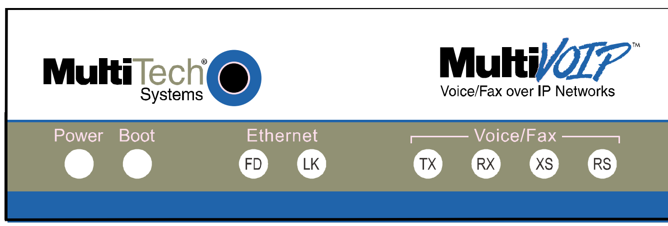


Figure 1-11. MVP-130/130FXS LEDs

Analog MultiVOIP LED Descriptions

MVP-210/410/810 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	**FDX.** LED indicates whether Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions. **LNK.** Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
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.

MVP-130/130FXS 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	**FDX.** LED indicates whether Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions. **LNK.** Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
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 (MVP130 only) is receiving a ring from the Telco or PBX.
RS	**Receive Signal**. This indicator lights when the FXS-configured channel is ringing or the FXO-configured channel (MVP130 only) has taken the line off-hook.

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).

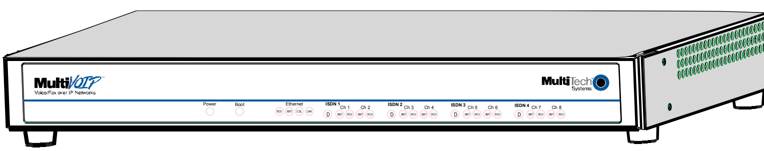


Figure 1-12: MVP-410ST/810ST Chassis

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.

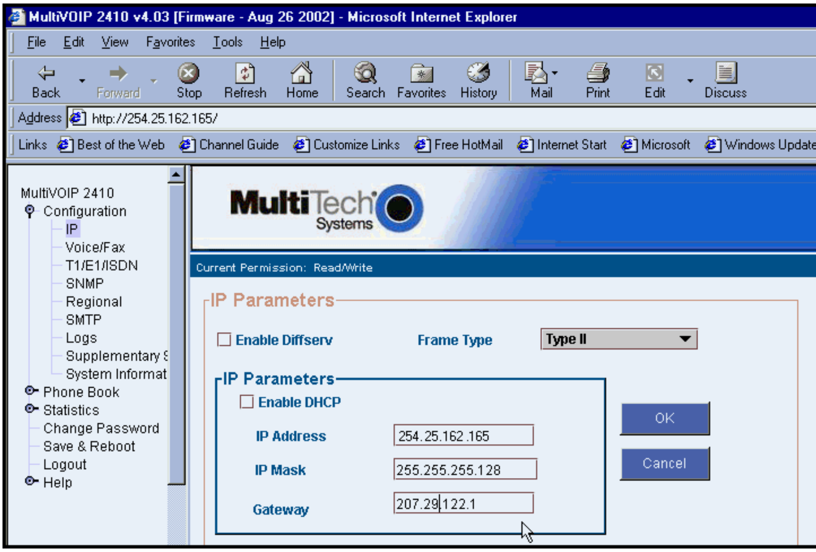
Data Compression & Quality of Service. The BRI MultiVOIP unit comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

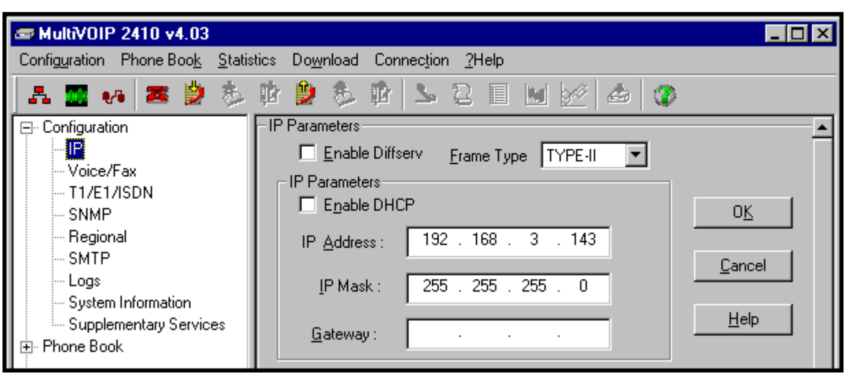
PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

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. 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).

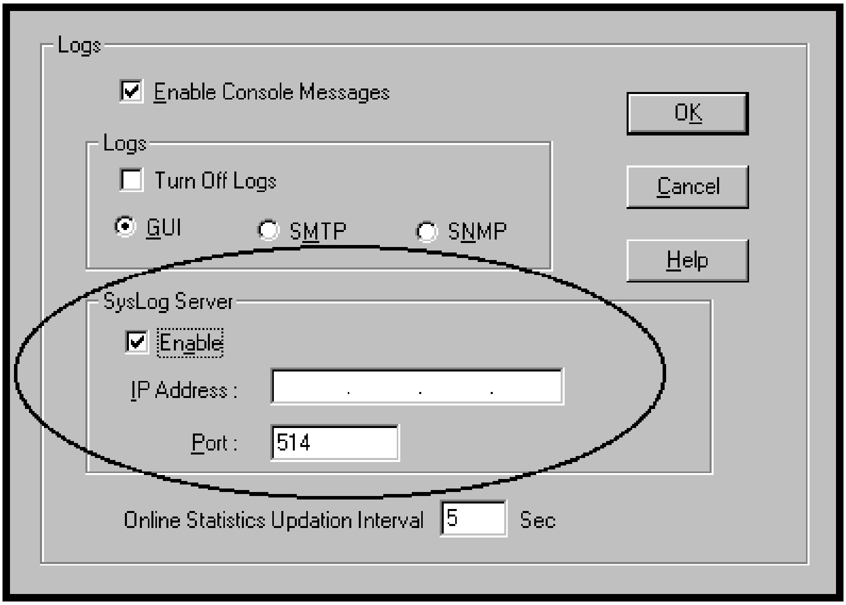




The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

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.

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).

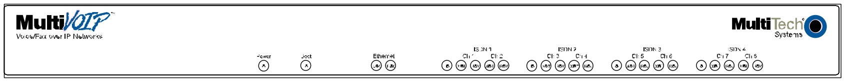


Figure 1-13. MVP-410ST/810ST LEDs

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	**FDX.** LED indicates whether Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions. **LNK.** Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
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-2410**	**MVP-2410 w/ MVP24-48 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 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	MVP410	MVP810 or MVP410 + 428
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**	**MVP810ST**	**MVP-130/130FXS**
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

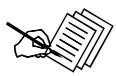
This chapter gets the MultiVOIP up and running quickly. The details we've skipped to make this brief can be found elsewhere in the manual (see Table of Contents and Index).

MultiVOIP Startup Tasks

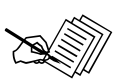
Task	Summary
● **Collecting Phone/IP Details (vital!)**	The MultiVOIP must be configured to interface with your particular phone system and IP network. To do so, certain details must be known about those phone and IP systems.
● **Placement**	Decide where you'll mount the voip.
● **Command/Control Computer 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

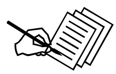
✓	*Ask your computer network administrator.*	*Info needed to operate:* **all MultiVOIP models.**
	 IP Network Parameters: Record for each VOIP Site in System	
	• IP Address	
	• IP Mask	
	• Gateway	
	• Domain Name Server (DNS) Info *(not implemented; for future use)*	

Gather Telephone Information (T1)

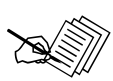
✓	**T1 Phone Parameters** *Ask phone company or PBX maintainer.*	*Info needed to operate:* **MVP2410**
	 T1 Telephony Parameters: Record for this VOIP Site	
	• Which frame format is used? ESF___ or D4___	
	• Which CAS or PRI protocol is used? _______________	
	• Clocking: Does the PBX or telco switch 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)

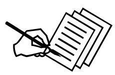
Gather Telephone Information (E1)

✓	**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)

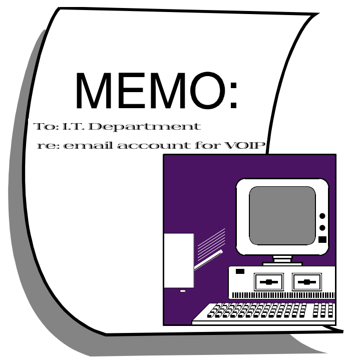
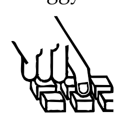
✓	**Analog Phone Parameters** *Ask phone company or telecom manager.*	*Needed for:* **MVP810** **MVP410** **MVP210** **MVP130** **MVP130FXS**
	 Analog Telephony Interface Parameters: Record for this VOIP Site	
	• Which interface type (or "signaling") is used? E&M_____ FXS/FXO_____	
	• If FXS, determine 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)

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.	 voip-unit2@biggytech.com 

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)

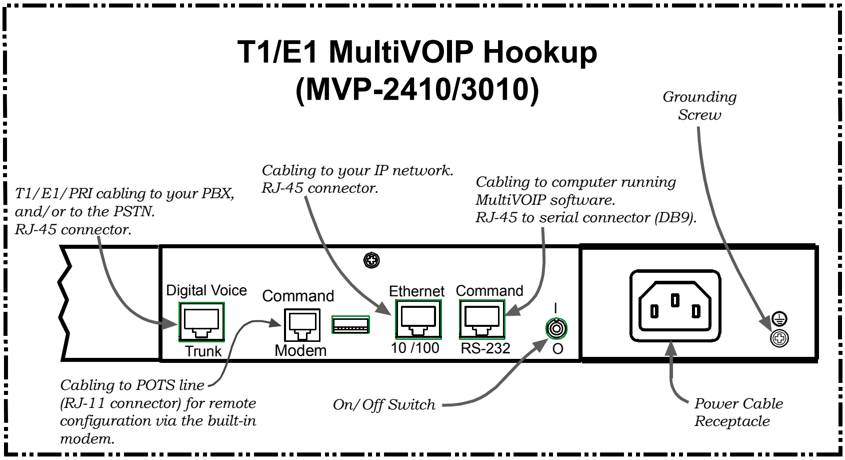
The computer used for command and control of the MultiVOIP
(a) must be an IBM-compatible PC,
(b) must use a Microsoft operating system,
(c) must be connected to your local network (Ethernet) system, and
(d) must have an available serial COM port.

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.

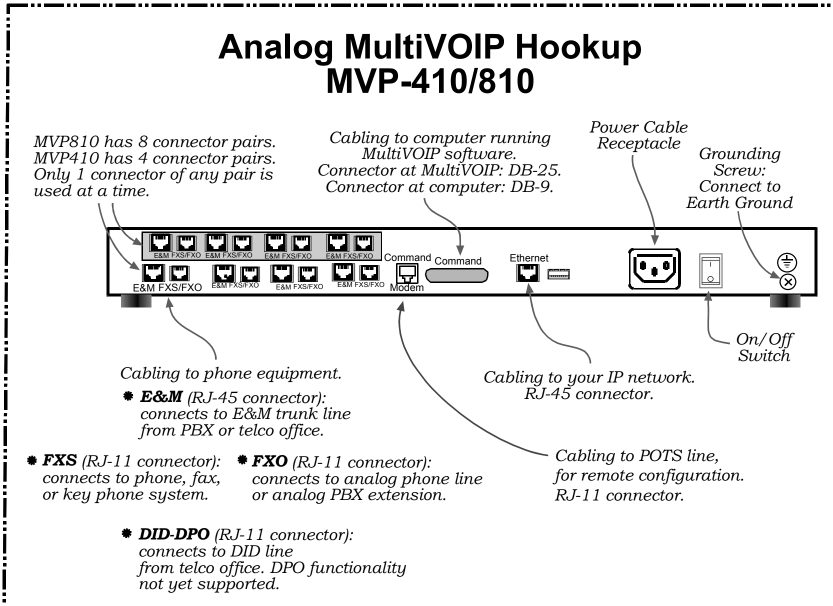
COM port on controller PC. You'll need an available COM port on the controller PC. You'll need to know which COM port is available for use with the MultiVOIP (COM1, COM2, etc.).

Quick Hookups

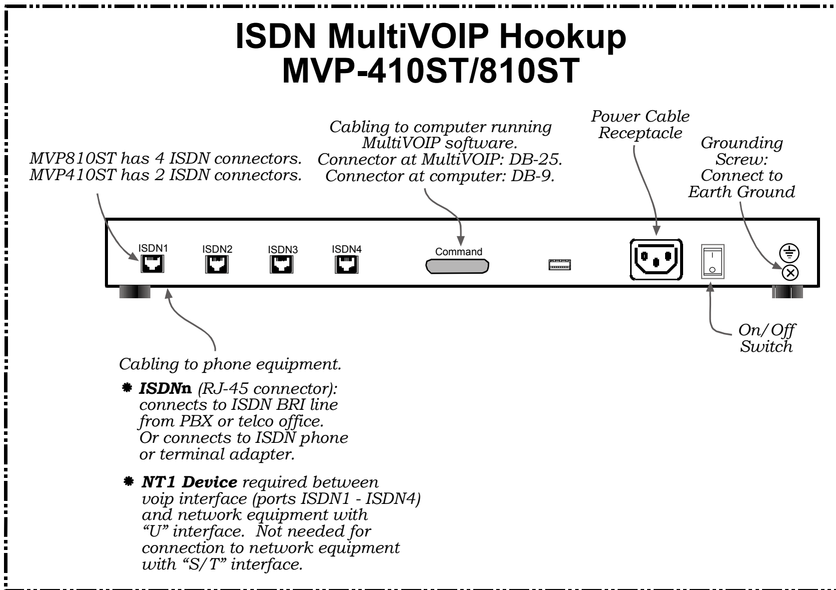
Hookup for MVP2410 & MVP3010



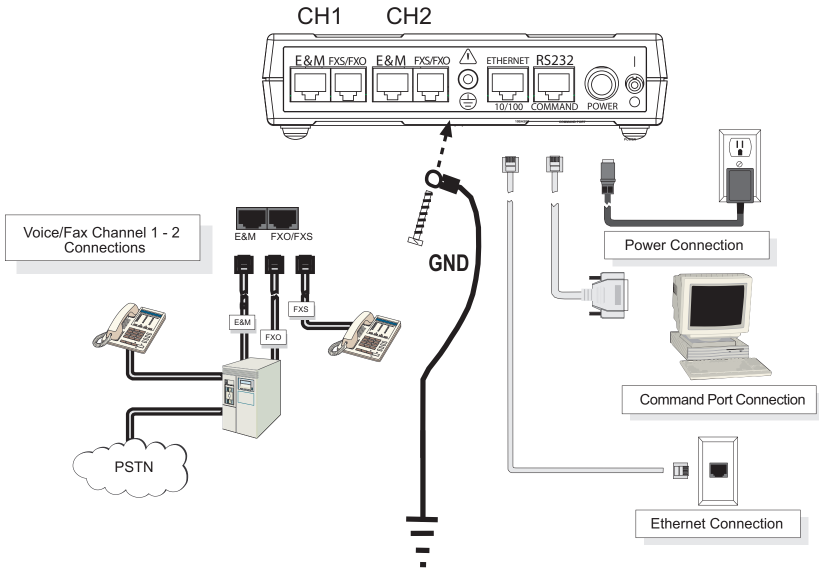
Hookup for MVP410 & MVP810



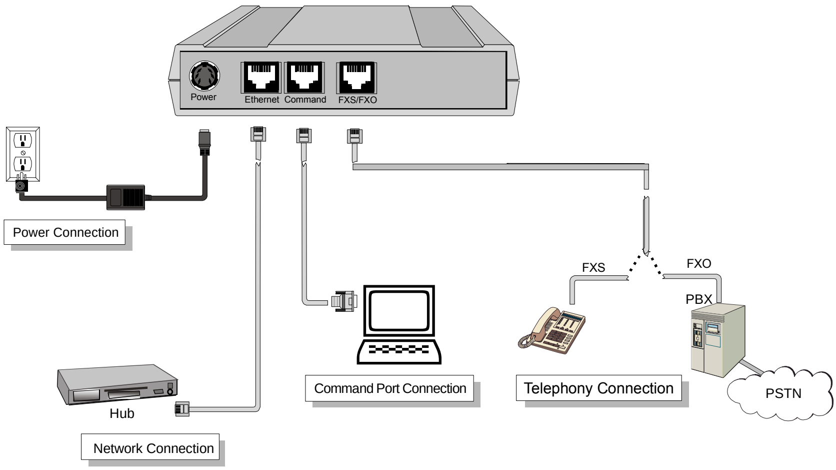
Hookup for MVP410ST & MVP810ST



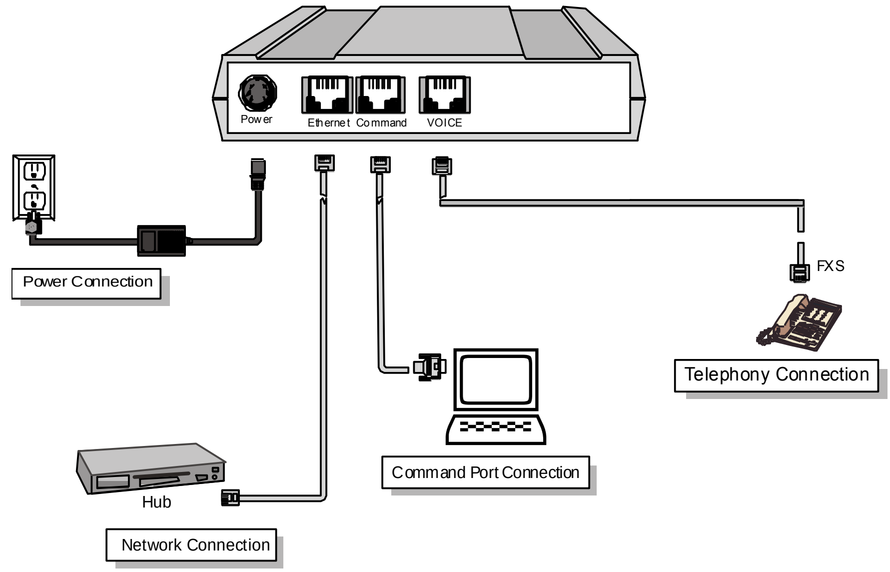
Hookup for MVP210



Hookup for MVP130



Hookup for MVP130FXS



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:

MVP2410 MVP3010	*Chapter 5: Technical Configuration for Digital T1/E1 MultiVOIPs* in User Guide.
MVP130 MVP130FXS MVP210 MVP410 MVP810	*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 1.4.2_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
MVP-130/130FXS,
MVP-210/410/810

Go to
Configuration | Interface.
Enter parameters obtained from phone company or PBX administrator.

Digital MultiVOIPs
MVP-2410/3010

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.

8. Go to **Configuration | Regional Parameters**. In the **Country Selection for Built-In Modem** field (drop-down list), select the country that best fits your situation. (This may not be the same as your selection for the **Country/Region** field. The selections in the **Country Selection for Built-In Modem** field entail more detailed

groupings of telephony parameters than do the **Country/Region** values.)

Click **OK** to exit from the **Regional Parameters** dialog box.

9. 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 11.
If YES, continue with step 10.

10. 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.

11. 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)

12. Enable premium (H.450) telephony features.

    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.

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).

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 71) 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 Pattern 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 Pattern 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, SIP, or SPP).

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.

U.K. Country Code

Phonebook Entry → 0044207

International Access Code

Leading Zero Dropped from Area Code

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)

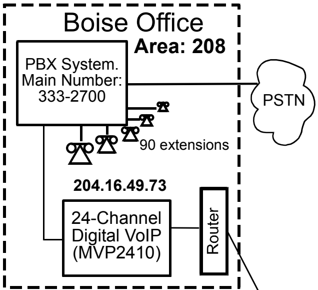
5. **Ease of Use**. The phonebook setup determines how easy the voip system is to use. Generally, you'll want to make it so dialing a voip call is very similar to dialing any other number (on the PSTN or through the PBX).

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

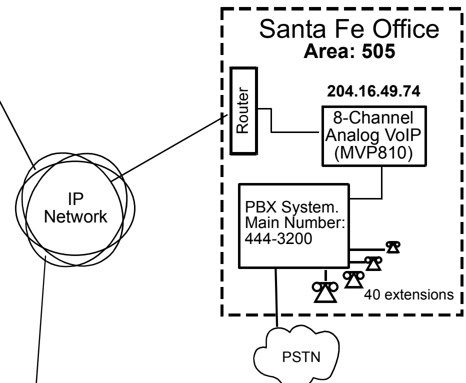
Voip Example. This company has offices in three different cities. The PBX units all operate alike. Notably, they all give access to outside lines using "9." They all are 'smart' enough to identify voip calls without using a special access digit ("8" is used in some systems). Finally, the system operates 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.

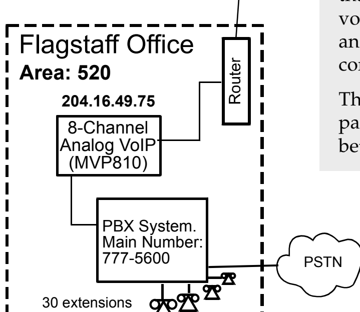
The second **Inbound** Phonebook entry (1 digit) is for receiving calls from company employees in the other two cities. The out-of-town employee simply dials 3 digits. The first of the three digits is uniquely used at each site and so acts as a destination pattern (Boise extensions are 7xx, Santa Fe extensions 2xx, Flagstaff extensions 6xx).

The local voip sees the pattern in its inbound phone book and notes the first digit (here either 2, 5, or 6). To make the match, this first digit, 2, 5, or 6 is put in the "Remove Prefix" field. This first digit must then be added back once again so that the voip will send all three digits to the PBX. The PBX can then dial the specific extension identified by the three-digit number.

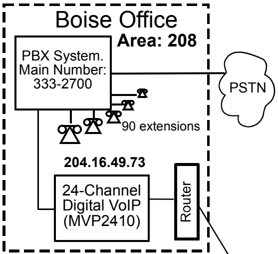


Each **Outbound Phonebook** contains two pairs of entries, two entries for each remote site. Whenever an out-of-town employee dials a 12-digit number beginning with the listed 5-digit destination pattern (9+1+area code) of another company location, the PBX hands the call to the voip system. The local 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.

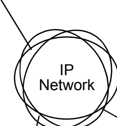
The one-digit **Outbound** destination patterns pertain to 3-digit calling between company employees.

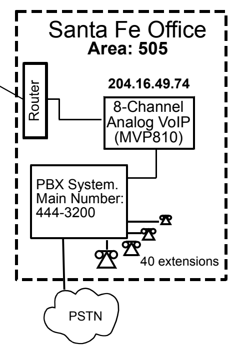


Voip Sites with Phonebooks

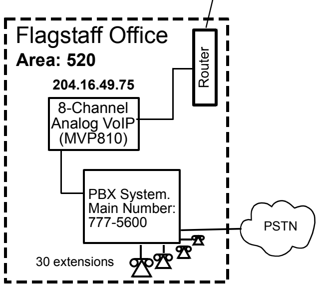


Boise Voip			Boise Voip					
Inbound Phonebook			Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
91208	9,	Incoming calls to PSTN, Boise Area	91505	12	none	none	204.16.49.74	Outgoing calls to Santa Fe area
7	7	i ncoming calls to extensions of company's PBX system in Boise	2	3	none	none	204.16.49.74	3-digit calls to Santa Fe employees (extensions 200 to 240)
			91520	12	none	none	204.16.49.75	Outgoing calls to Flagstaff area
			6	3	none	none	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
91505	9,	Incoming calls to PSTN, Santa Fe local calls	91208	12	none	none	204.16.49.73	Outgoing calls to Boise area
2	2	Incoming calls to extensions of company's PBX system in Santa Fe	7	3	none	none	204.16.49.73	3-digit calls to Boise employees (extensions 700-790)
			91520	12	none	none	204.16.49.75	Outgoing calls to Flagstaff area
			6	3	none	none	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
91520	9	Incoming calls to PSTN, Flagstaff local calls	91505	12	none	none	204.16.49.74	Outgoing calls to Santa Fe area
6	6	Incoming calls to extensions of company's PBX system in Flagstaff	2	3	none	none	204.16.49.74	3-digit calls to Santa Fe employees (extensions 200-240)
			91208	12	none	none	204.16.49.73	Outgoing calls to Boise area
			7	3	none	none	204.16.49.73	3-digit calls to Boise employees (extensions 700-790)

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
91208	9,	Incoming calls to PSTN, Boise Area	91505	12	none	none	204. 16.49. 74	Outgoing calls to Santa Fe area
7	7	i ncoming calls to extensions of company's PBX system in Boise	2	3	none	none	204. 16.49. 74	3-digit calls to Santa Fe employees (extensions 200 to 240)
			91520	12	none	none	204. 16.49. 75	Outgoing calls to Flagstaff area
			6	3	none	none	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
91505	9,	Incoming calls to PSTN, Santa Fe local calls	91208	12	none	none	204. 16.49. 73	Outgoing calls to Boise area
2	2	Incoming calls to extensions of company's PBX system in Santa Fe	7	3	none	none	204. 16.49. 73	3-digit calls to Boise employees (extensions 700-790)
			91520	12	none	none	204. 16.49. 75	Outgoing calls to Flagstaff area
			6	3	none	none	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
91520	9	Incoming calls to PSTN, Flagstaff local calls	91505	12	none	none	204.16 .49.74	Outgoing calls to Santa Fe area
6	6	Incoming calls to extensions of company's PBX system in Flagstaff	2	3	none	none	204.16 .49.74	3-digit calls to Santa Fe employees (extensions 200-240)
			91208	12	none	none	204.16 .49.73	Outgoing calls to Boise area
			7	3	none	none	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:

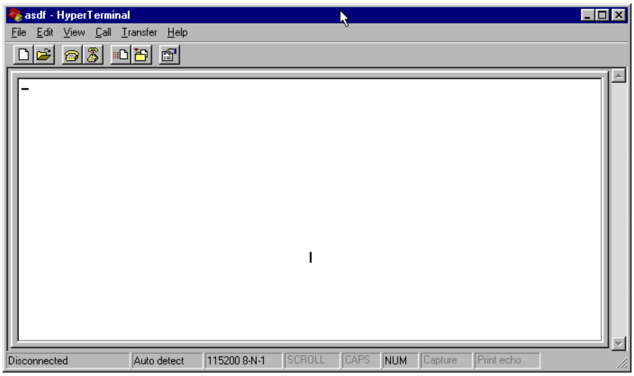
Connections	
for digital MultiVOIPs (MVP-2410/3010)	**for analog MultiVOIPs (MVP-130/130FXS & MVP-210/410/810)**
MultiVOIP to local PBX	MultiVOIP to local phone station –OR-- MultiVOIP to extension of key phone system
MultiVOIP to command PC	MultiVOIP to command PC
MultiVOIP to Internet	MultiVOIP to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

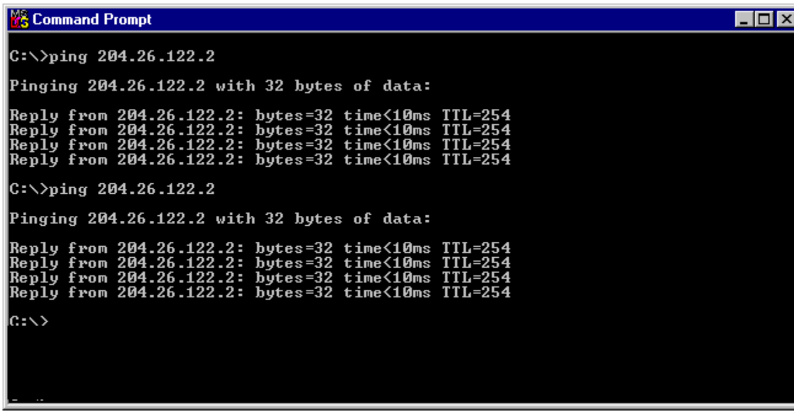
   where x is the number of the voip channel carrying the call

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.



```
Command Prompt
C:\>ping 204.26.122.2

Pinging 204.26.122.2 with 32 bytes of data:

Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254

C:\>ping 204.26.122.2

Pinging 204.26.122.2 with 32 bytes of data:

Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254
Reply from 204.26.122.2: bytes=32 time<10ms TTL=254

C:\>
```

2. Verify the telephone connections.

A. **For 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 MVP-130/130FXS, 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, MVP130FXS and MVP210 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

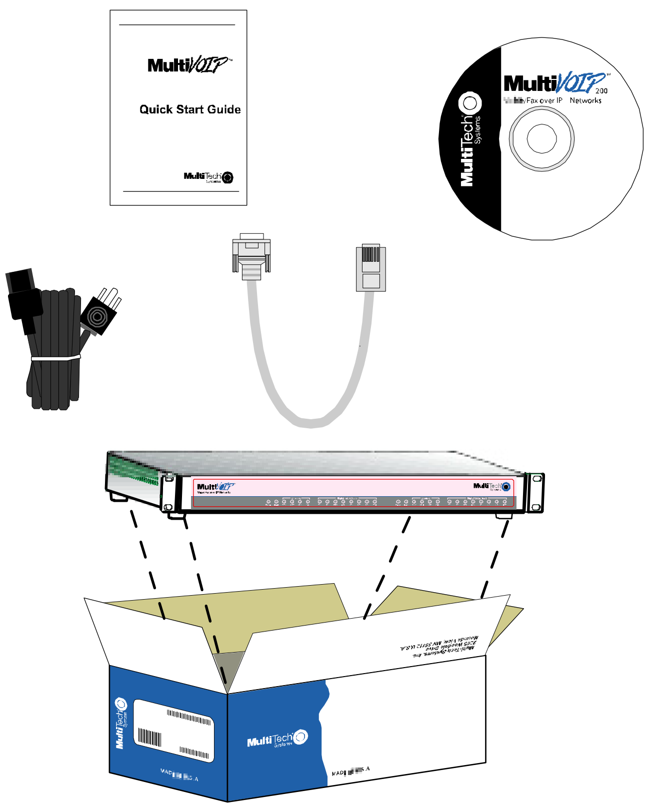


Figure 3-1: Unpacking the MVP2410/3010

Unpacking the MVP-410/810

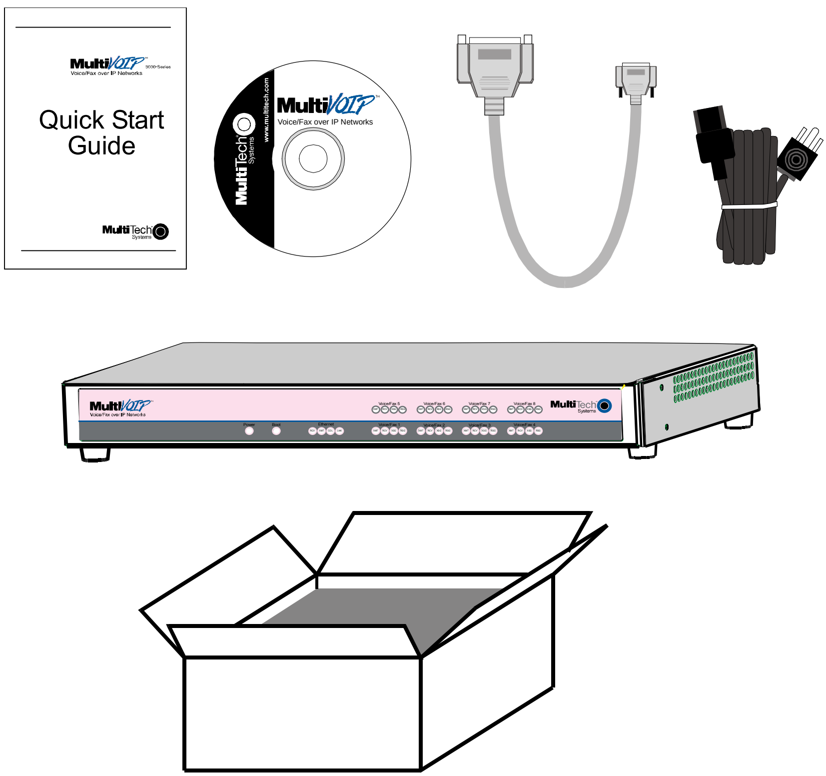


Figure 3-2: Unpacking the MVP-410/810

Unpacking the MVP210

Figure 3-3: Unpacking the MVP210

Unpacking the MVP-130/130FXS

Figure 3-4: Unpacking the MVP-130/130FXS

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

The MultiVOIPs can be mounted in an industry-standard EIA 19-inch rack enclosure, as shown in Figure 3-5.

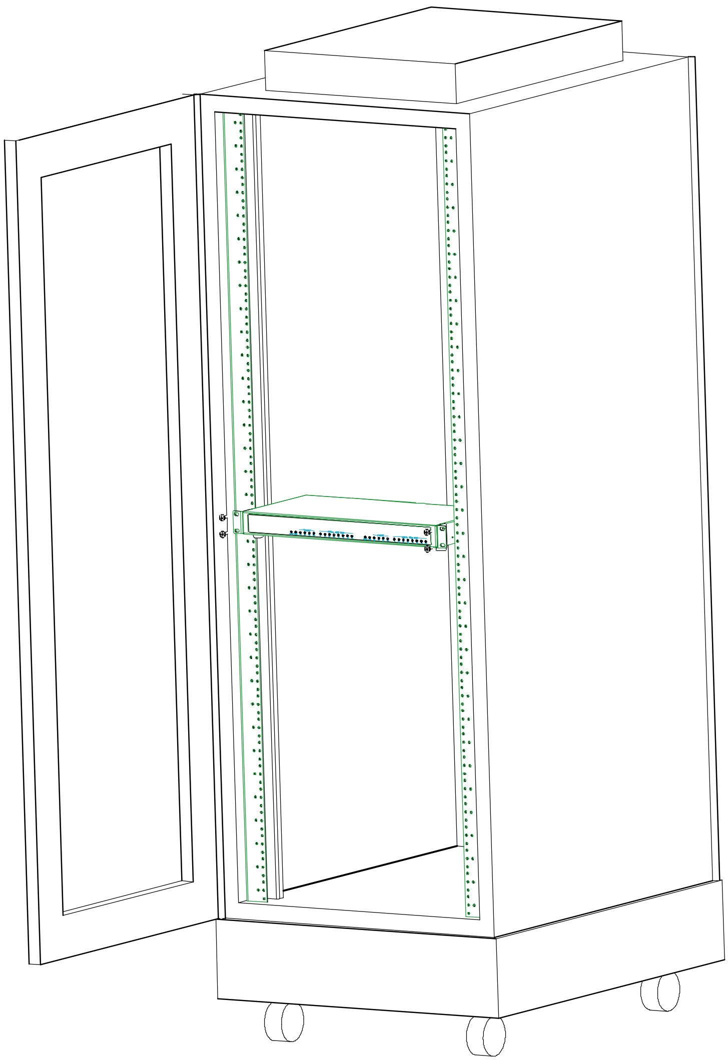


Figure 3-5: Rack-Mounting (MVP2410/3010 or MVP410/810)

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 60 degrees Celsius (140 degrees Fahrenheit) at 20-90% non-condensing relative humidity. 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 3-6, and then secure unit to rack rails by the brackets, as shown in Figure 3-7. 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.

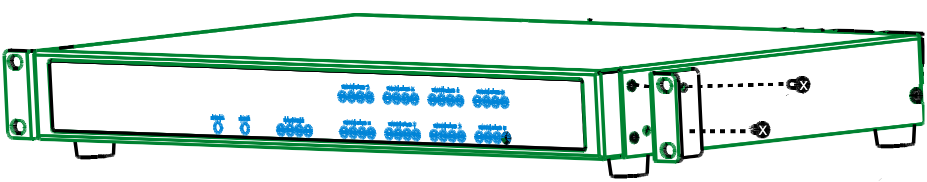


Figure 3-6: Bracket Attachment for Rack Mounting (MVP-2410/3010 & MVP-410/810)

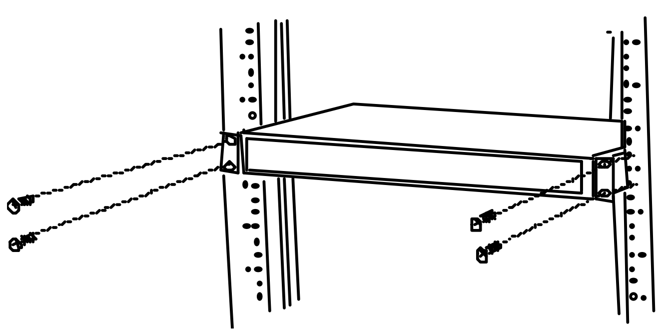


Figure 3-7: Attaching MultiVOIP to Rack Rail (MVP-2410/3010 & MVP-410/810)

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 3-8 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 3-8.

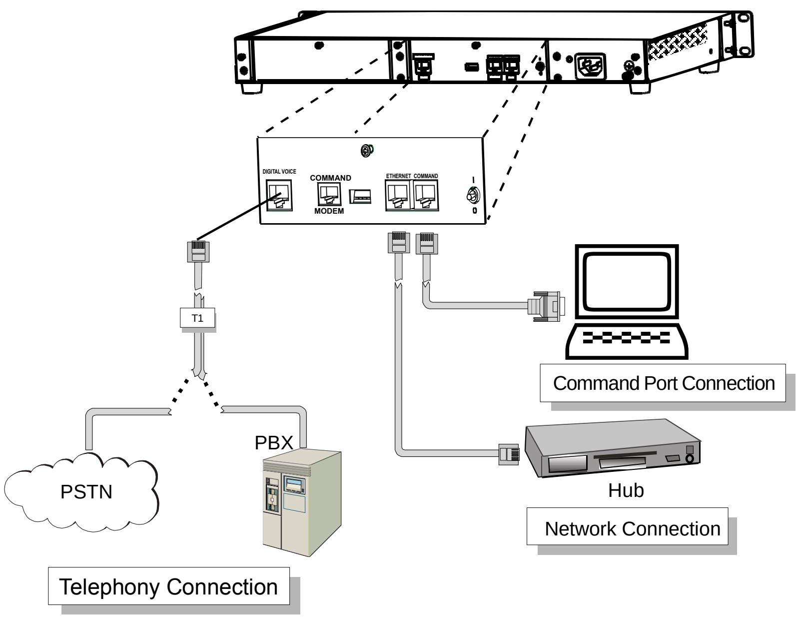


Figure 3-8. Cabling for MVP2410/3010

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 3-8.

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. If you intend to configure the MultiVOIP remotely using the MultiVOIP Windows GUI, connect an RJ-11 phone cable between the Command Modem connector (at the rear of the MultiVOIP) and a receptacle served by a telco POTS line. See Figure 3-9.

   The Command Modem is built into the MultiVOIP unit. To configure the MultiVOIP remotely using its Windows GUI, you must call into the MultiVOIP's Command Modem. Once a connection is made, the configuration process is identical to local configuration with the Windows GUI.

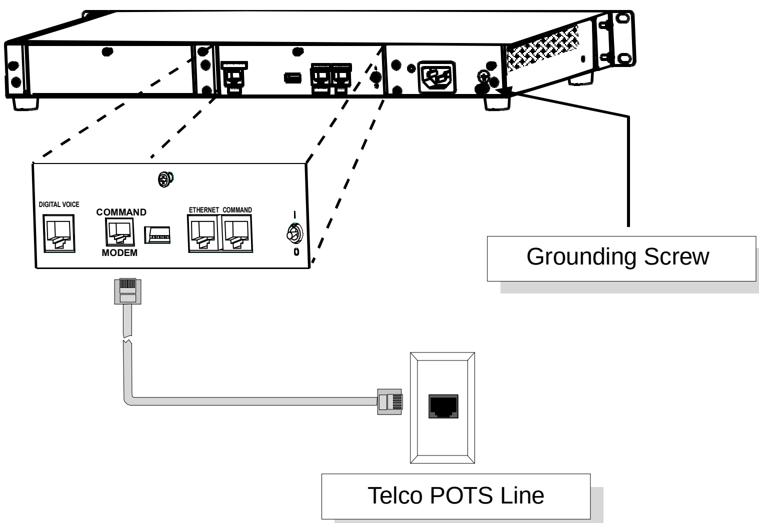


Figure 3-9. MVP-2410/3010 Voip Connections for GND & Remote Config Modem

5. 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 grounding screw (see Figure 3-9) and a metallic object that will provide an electrical ground.

6. 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-410/810

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

1. **For DID channels only**. If all channels of your MultiVOIP will be using either FXS, FXO, or E&M telephony interfaces, skip to step 2.

   For any channel on which you are using the DID interface type, you must change the jumper on the MultiVOIP circuit card.

   a. Disconnect power. Unplug the AC power cord from the wall outlet or from the receptacle on the MultiVOIP unit.

   b. Using a #1 Phillips driver, remove the three screws (at back of unit) that attach the main circuit card to the chassis of the MultiVOIP.

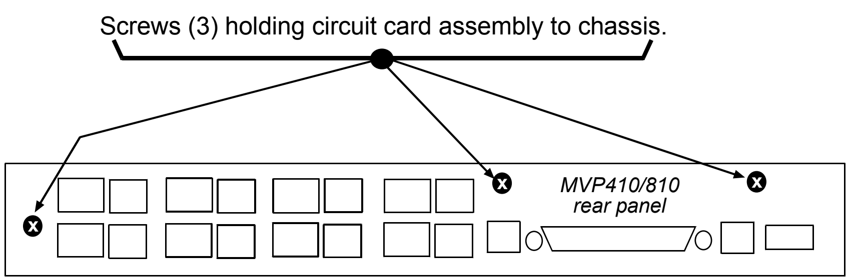


Figure 3-10. MVP-410/810 Rear Screw Locations

   c. Pull the main circuit card out about 5 inches (the power connection to the board prevents it from being removed entirely from the chassis).

d. Identify the channels on which the DID interface will be used.

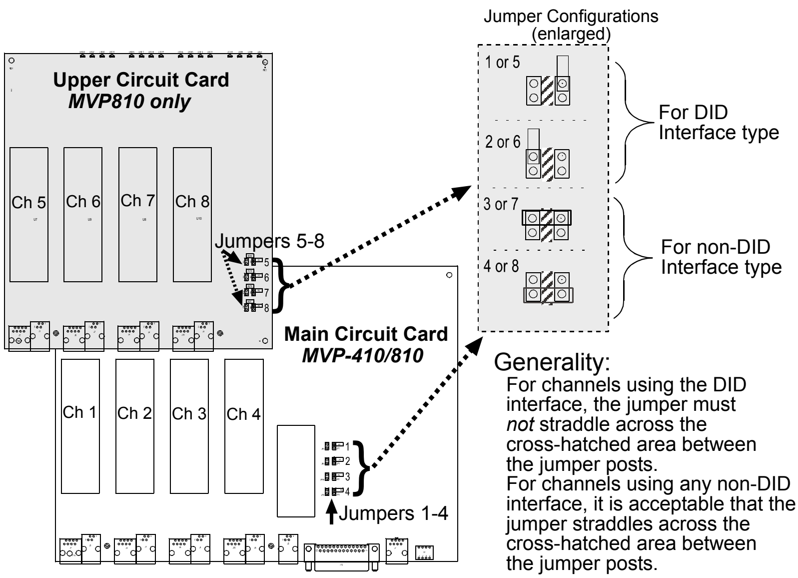


Figure 3-11. MVP-410/810 Channel Jumper Settings

e. Position the jumper for each DID channel so that it does not connect the two jumper posts. For DID operation of a voip channel, the MultiVOIP will work properly if you simply remove the jumper altogether, but that is inadviseable because the jumper might be needed later if a different telephony interface is used for that voip channel.

f. Slide the main circuit card back into the MultiVOIP chassis and replace the three screws.

2. 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.

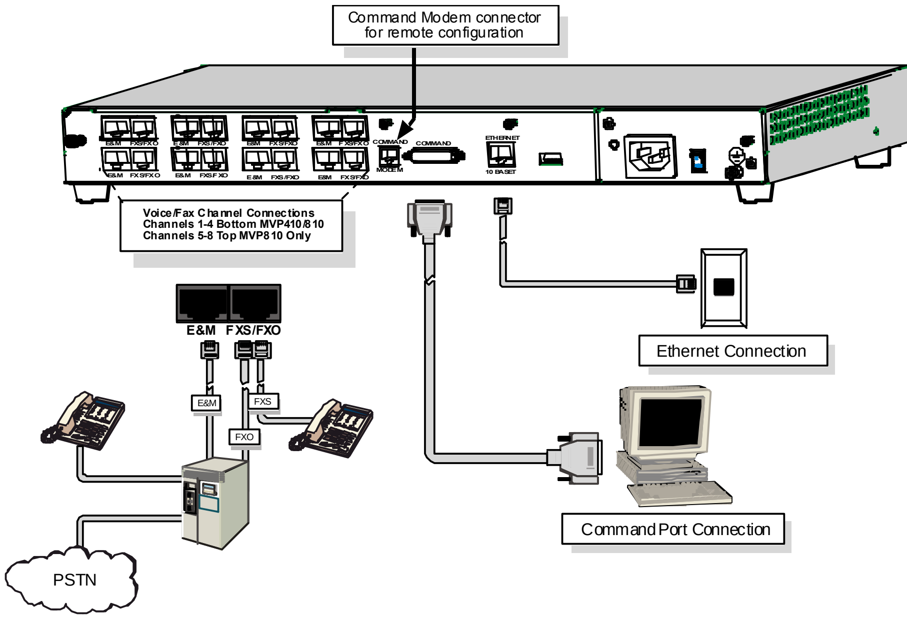


Figure 3-12: Cabling for MVP-410/810

3. 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.

4. 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.

5. **For an FXS or FXO connection**.
(*FXS Examples: analog phone, fax machine, Key Telephone System*.)
(*FXO Examples: PBX extension, POTS line from telco central office*.)
Connect one end of an RJ-11 phone cord to the Channel **1 FXS/FXO** connector on the back of the MultiVOIP.
Connect the other end to the device or phone jack.

   For an E&M connection.
(*E&M Example: trunk line from telephone switch*.)
Connect one end of an RJ-45 phone cord to the Channel **1 E&M** connector on the back of the MultiVOIP.

Connect the other end to the trunk line.
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 supported by the telephone switch. See Appendix B for an E&M cabling pinout.

For a DID connection.
(*DID Example: DID fax system or DID voice phone lines*.)
Connect one end of an RJ-11 phone cord to the Channel **1 FXS/FXO** connector on the back of the MultiVOIP.
Connect the other end to the DID jack.

> **NOTE**: DID lines are polarity sensitive. If, during testing, the DID line rings busy consistently, you will need to reverse the polarity of one end of the connector (swap the connections of the wires to the two middle pins of one RJ-11 connector).

6. Repeat step 5 to connect the remaining telephone equipment to each channel on your MultiVOIP. Although a MultiVOIP's channels are often all configured identically, each channel is individually configurable. So, for example, some channels of a MultiVOIP might use the FXO interface and others the FXS; some might use the DID interface and others E&M, etc.

7. If you intend to configure the MultiVOIP remotely using the MultiVOIP Windows GUI, connect an RJ-11 phone cable between the Command Modem connector (at the rear of the MultiVOIP) and a receptacle served by a telco POTS line. See Figure 3-13.

   The Command Modem is built into the MultiVOIP unit. To configure the MultiVOIP remotely using its Windows GUI, you must call into the MultiVOIP's Command Modem. Once a connection is made, the configuration process is identical to local configuration with the Windows GUI.

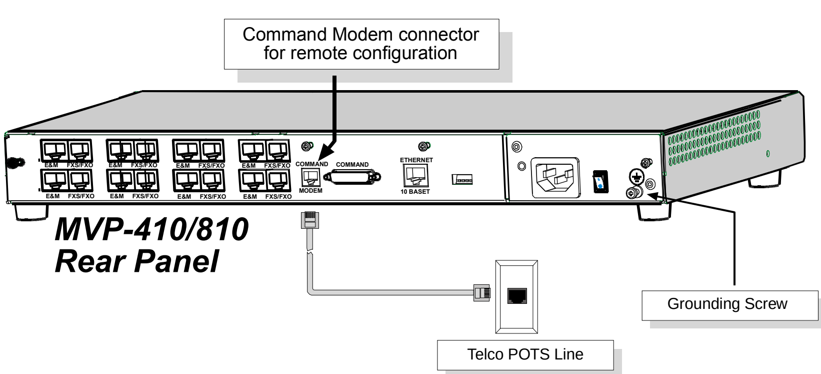


Figure 3-13. MVP-410/810 Voip Connections for GND & Remote Config Modem

8. 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 grounding screw (see Figure 3-13) and a metallic object that will provide an electrical ground.

9. 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 3-14.

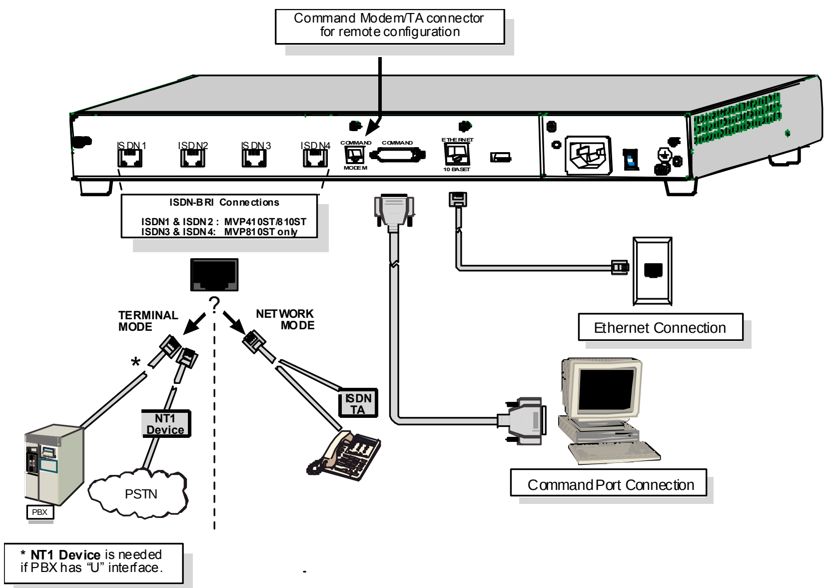


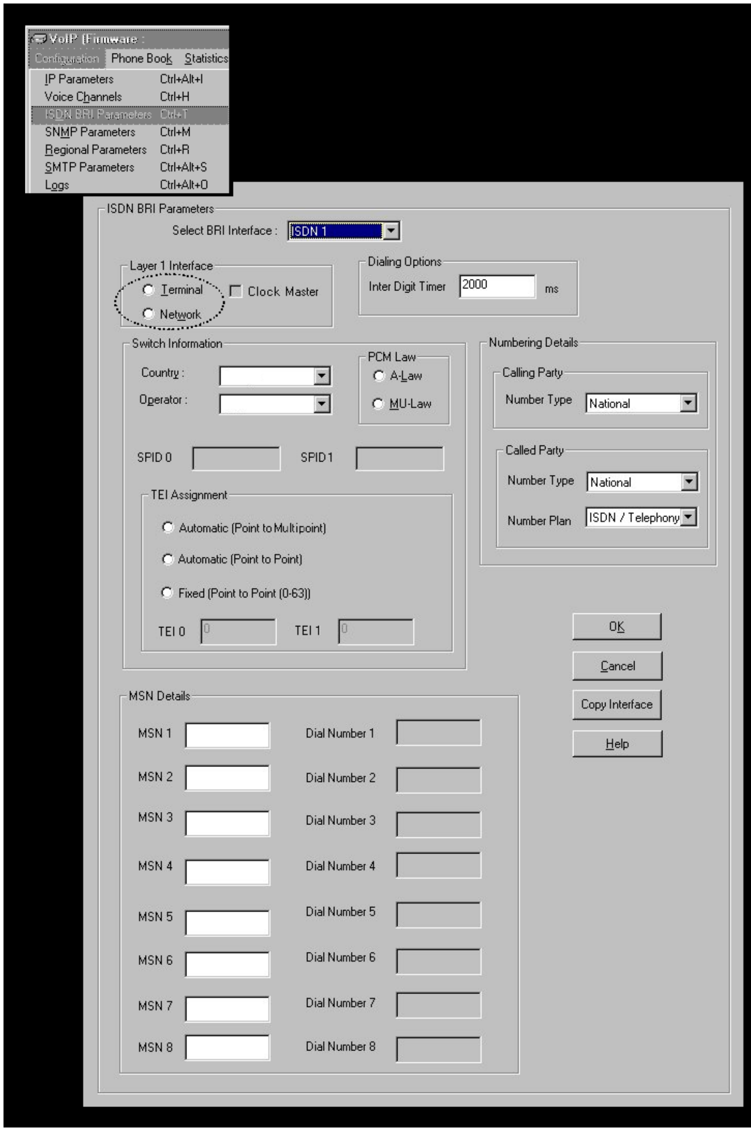
Figure 3-14: Cabling for MVP-410ST/810ST

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 3-14.

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.

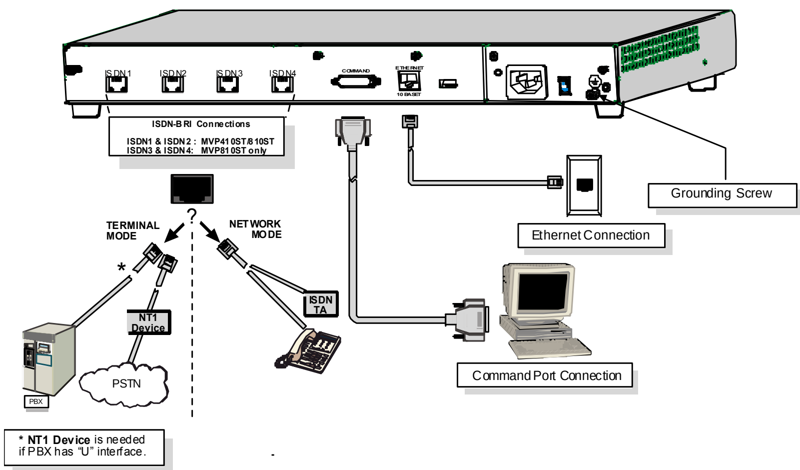


Figure 3-15: ISDN/BRI Voip Connections for GND & Remote Configuration Modem

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 grounding screw (see Figure 3-15) 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 MVP210

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

1. **For DID channels only**. If both channels of your MVP210 MultiVOIP will be using either FXS, FXO, or E&M telephony interfaces, skip to step 2.

   For any channel on which you are using the DID interface type, you must change the jumper on the MultiVOIP circuit card.

   a. Disconnect power. Unplug the AC power cord from the wall outlet or from the receptacle on the MultiVOIP unit.

   b. Using a #1 Phillips driver, remove the screw (at bottom of unit, near the back-cover end) that attaches the main circuit card to the chassis of the MVP210.

   c. Pull the main circuit card out about half way.

d. Identify the channels on which the DID interface will be used.

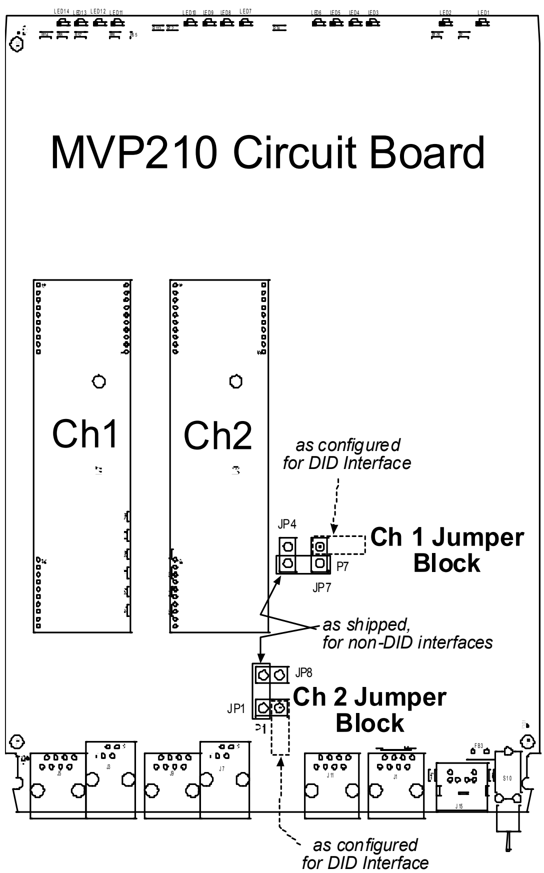


Figure 3-16. MVP210 Channel Jumper Settings

e. Position the jumper for each DID channel so that it does not connect the two jumper posts. For DID operation of a voip channel, the MultiVOIP will work properly if you simply remove the jumper altogether, but that is inadviseable because the jumper might be needed later if a different telephony interface is used for that voip channel.

f. Slide the main circuit card back into the MultiVOIP chassis and replace the screw at the bottom of the unit.

2. 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 3-17.

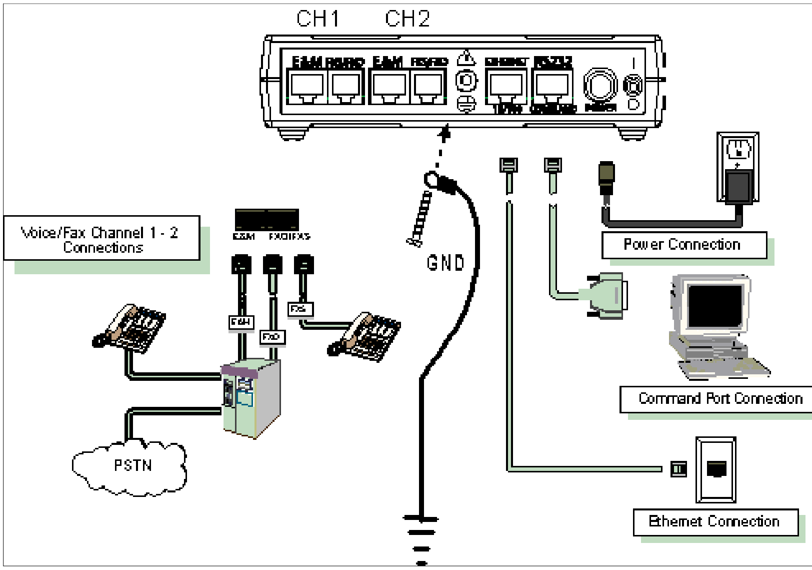


Figure 3-17: Cabling for MVP210

3. 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 3-17.

4. 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.

5. **For an FXS or FXO connection**.
   (*FXS Examples: analog phone, fax machine, Key Telephone System*.)
   (*FXO Examples: PBX extension, POTS line from telco central office*.)
   Connect one end of an RJ-11 phone cord to the Channel **1 FXS/FXO** connector on the back of the MultiVOIP.
   Connect the other end to the device or phone jack.

   For an E&M connection.
   (*E&M Example: trunk line from telephone switch*.)
   Connect one end of an RJ-45 phone cord to the Channel **1 E&M** connector on the back of the MultiVOIP.
   Connect the other end to the trunk line.
   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 supported by the telephone switch. See Appendix B for an E&M cabling pinout.

   For a DID connection.
   (*DID Example: DID fax system or DID voice phone lines*.)

Connect one end of an RJ-11 phone cord to the Channel **1 FXS/FXO** connector on the back of the MultiVOIP.
Connect the other end to the DID jack.

> **NOTE**: DID lines are polarity sensitive. If, during testing, the DID line rings busy consistently, you will need to reverse the polarity of one end of the connector (swap the connections of the wires to the two middle pins of one RJ-11 connector).

6. Repeat the above step to connect the remaining telephone equipment to the second channel on your MultiVOIP.
7. 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.
8. 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 the *Software Installation* chapter to load the MultiVOIP software.

Cabling Procedure for MVP-130/130FXS

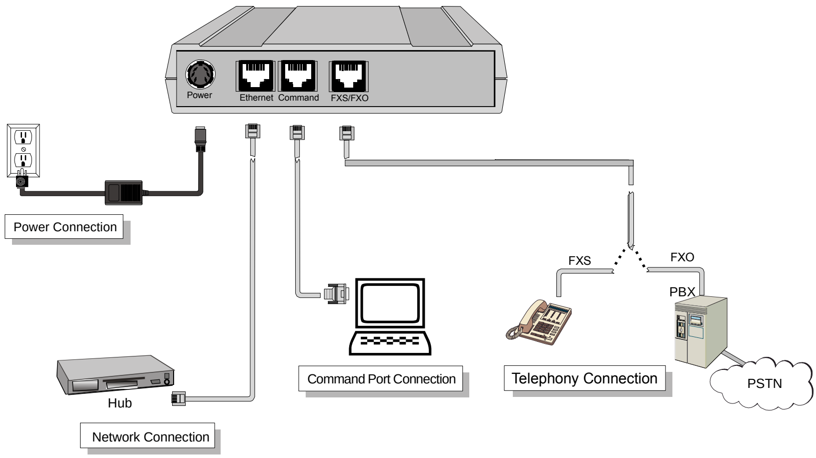


Figure 3-18: Cabling for MVP-130/130FXS

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 3-18.
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 3-18.
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. Since the MVP130FXS supports the FXS interface only, its connection options differ from that of the MVP130, which supports both FXS and FXO.

A. **For MVP130**. To connect 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, attach 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.

B. **For MVP130FXS**. To connect a station device such as an analog telephone or fax machine to your MultiVOIP, attach one end of an RJ-11 phone cord to the **VOICE** 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. The MVP130FXS supports only the FXS interface.

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).

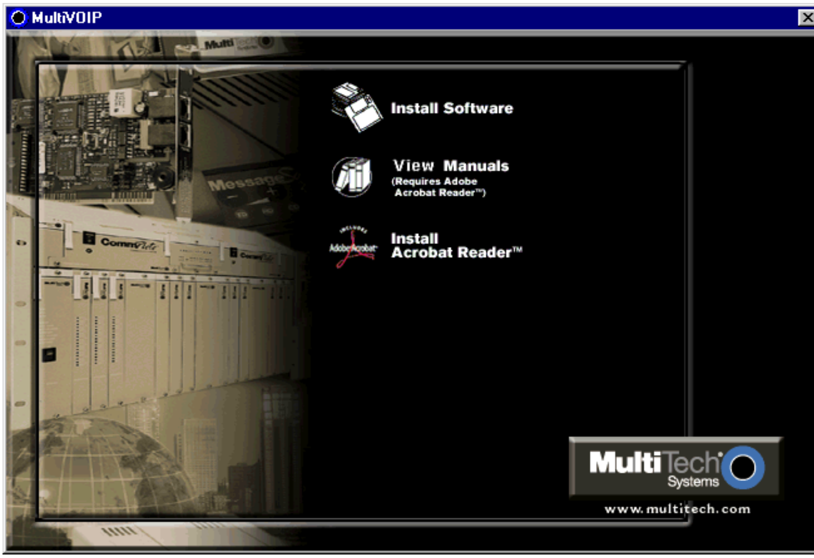
Loading MultiVOIP Software onto the PC

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.

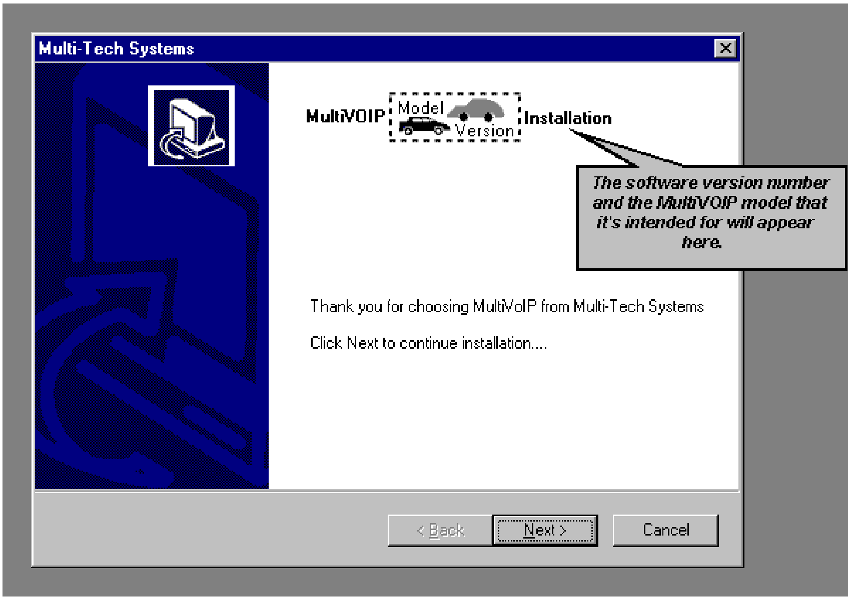
2. Insert the MultiVOIP CD into your CD-ROM drive. The CD should start automatically. It may take 10 to 20 seconds for the Multi-Tech CD installation window to display.



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.

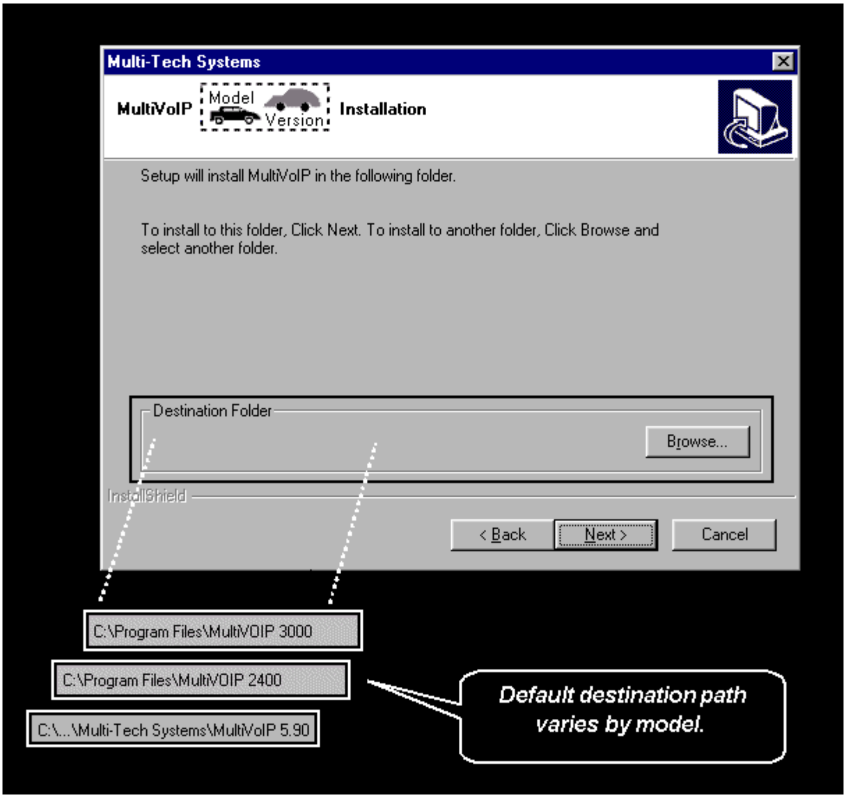
3. When the Multi-Tech Installation CD dialog box appears, click the **Install Software** icon.

4. A 'welcome' screen appears.



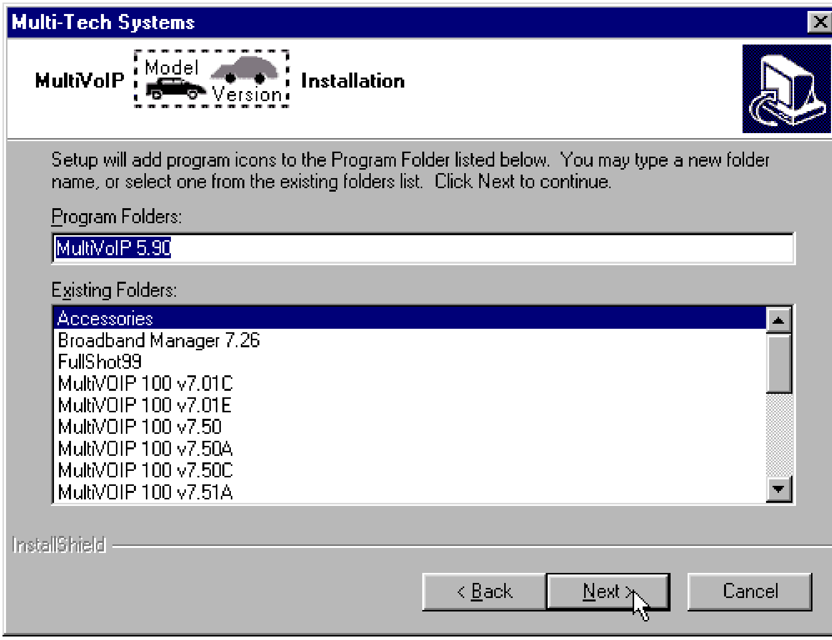
Press **Enter** or click **Next** to continue.

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.



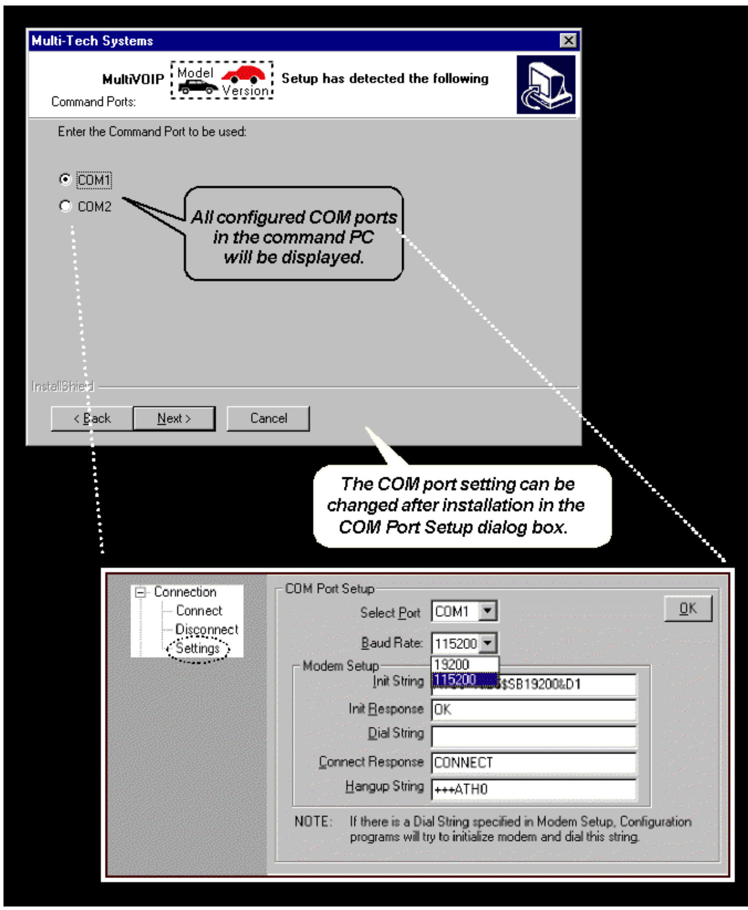
Choose a location and click **Next.**

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



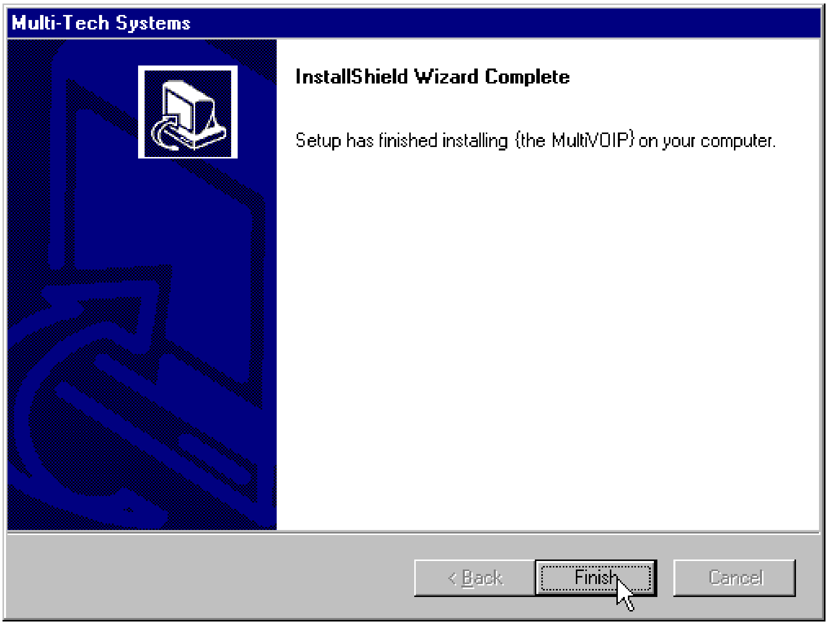
Click **Next**. Transient progress screens will appear while files are being copied.

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**.

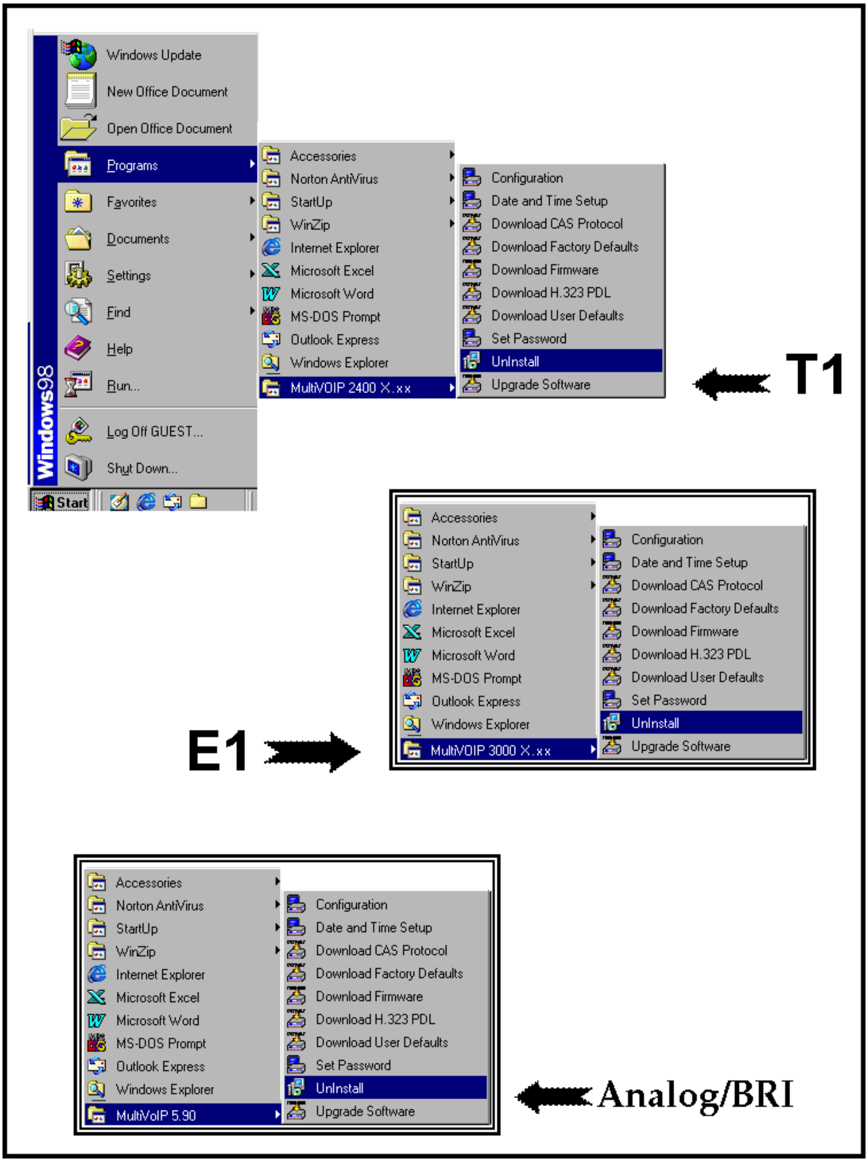
9. When setup of the MultiVOIP software is complete, you will be prompted to run the MultiVOIP software to configure the VOIP.

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

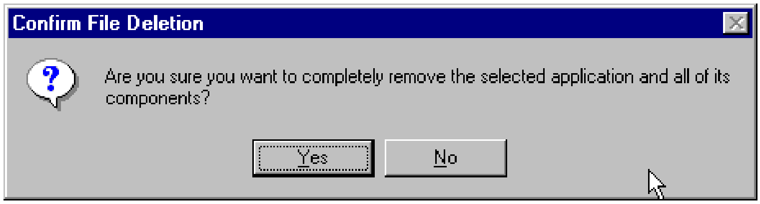
Technical Configuration instructions are in the next two chapters of this manual: Chapter 5 for T1/E1 MultiVOIP units and Chapter 6 for Analog MultiVOIP units.

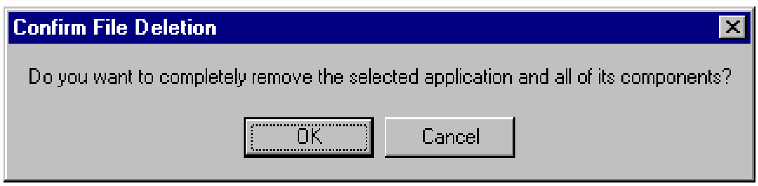
Un-Installing the MultiVOIP Configuration Software

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

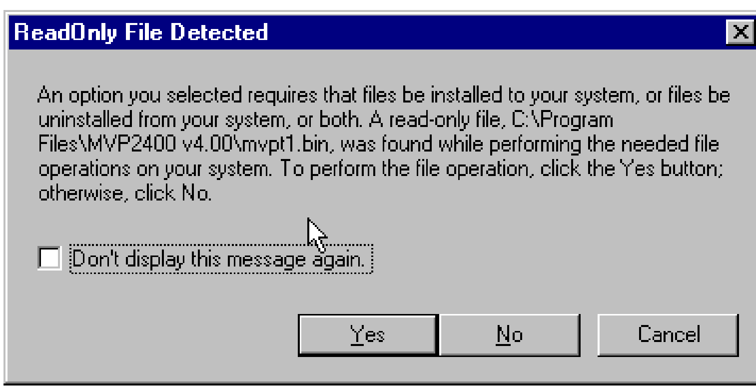


2. Two confirmation screens will appear. Click **Yes** and **OK** when you are certain you want to continue with the uninstallation process.

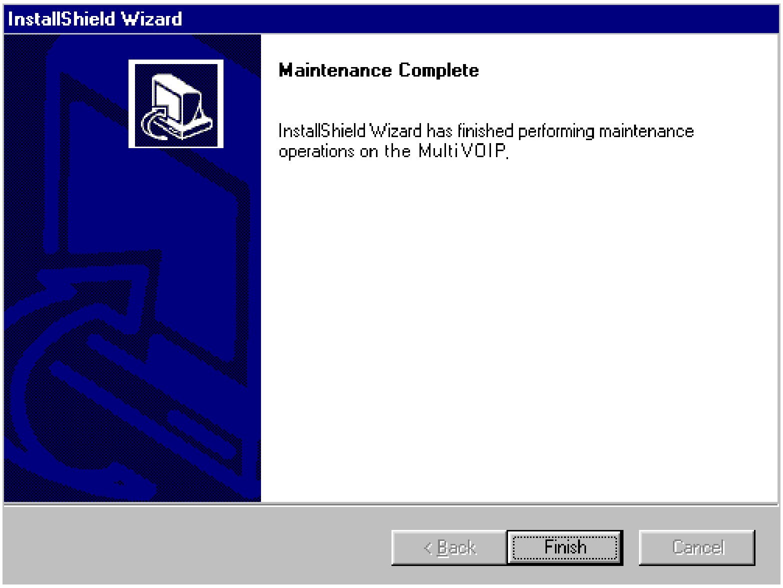




3. A special warning message similar to that shown below may appear concerning the MultiVOIP software's ".bin" file. Click **Yes**.



4. A completion screen will appear.



Click **Finish**.

Chapter 5: Technical Configuration for Digital T1/E1 MultiVOIPs (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.

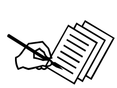
Pre-Requisites

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

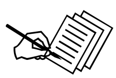
The following parameters must be known about the network (LAN, WAN, Internet, etc.) to which the MultiVOIP will connect:

✓	*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 MVP2410)

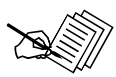
The following parameters must be known about the PBX or telco central office equipment to which the T1 MultiVOIP will connect:

✓	**T1 Phone Parameters** *Ask phone company or PBX maintainer.*	*Info needed to operate:* **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:

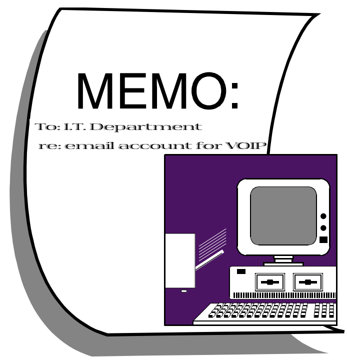
✓	**E1 Phone Parameters** *Ask phone company or PBX maintainer.*	*Info needed to operate:* **MVP3010**
	 E1 Telephony Parameters: Record for this VOIP Site	
	• Which frame format is used? Double Frame_____ MultiFrame w/ CRC4_____ MultiFrame w/ CRC4 modified_____	
	• Which CAS or PRI protocol is used? ______________	
	• 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)	

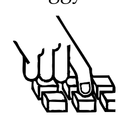
Write down the values for these E1 parameters. You will need to enter these values in the “T1/E1 Parameters” screen in the Configuration section of the MultiVOIP software.

SMTP Parameters (for email call log reporting)

<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>

voip-unit2@biggytech.com
</td></tr>
</table>

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 & Cadences and setup for built-in Remote Configuration/Command Modem).

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).

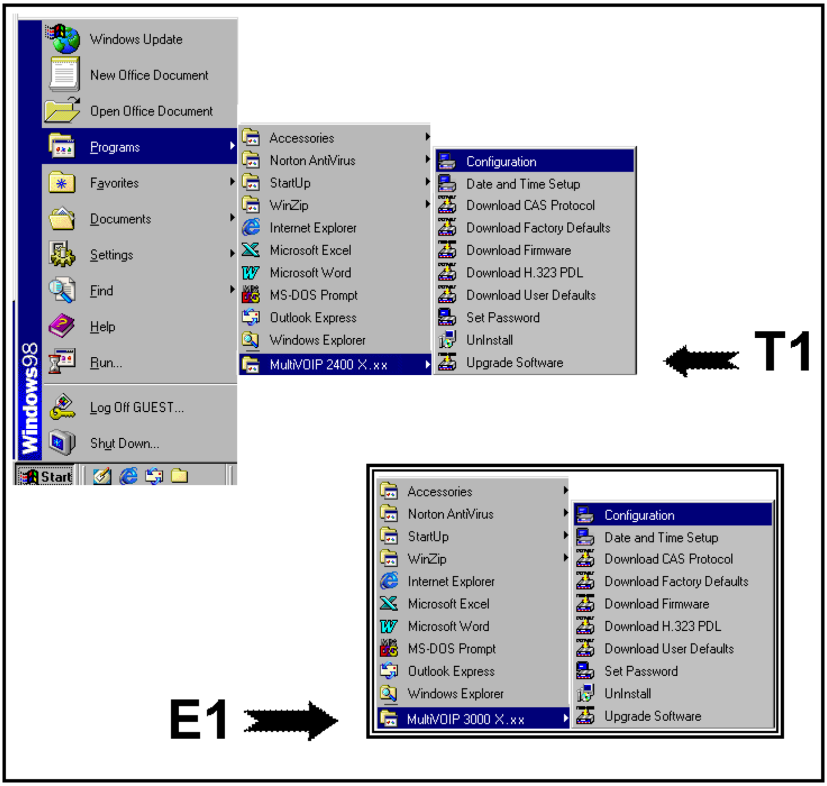
Local Configuration Procedure (Detailed)

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.

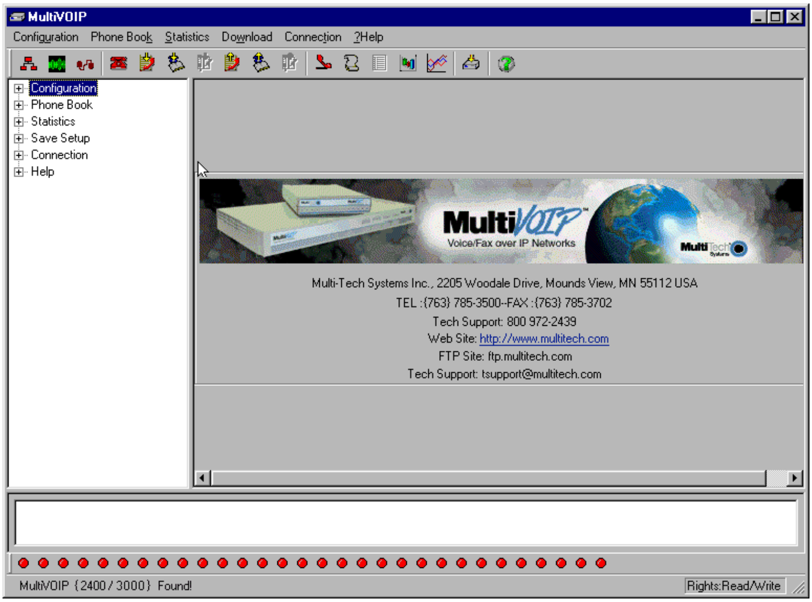
1. **Check Power and Cabling**. Be sure the MultiVOIP is turned on and connected to the computer via the MultiVOIP's Command Port (DB9 connector at computer's COM port; RJ45 connector at MultiVOIP).

   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.

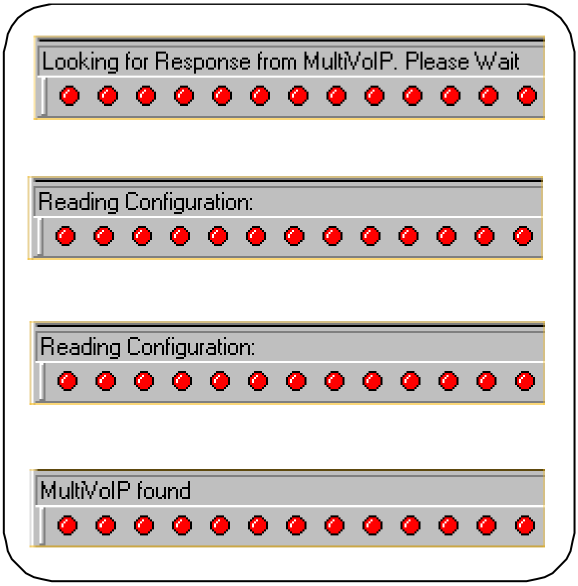
2. **Start MultiVOIP Configuration Program**. Launch the MultiVOIP program from the Windows **Start** menu (from the folder location determined during installation).



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.

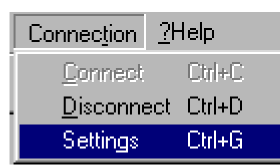


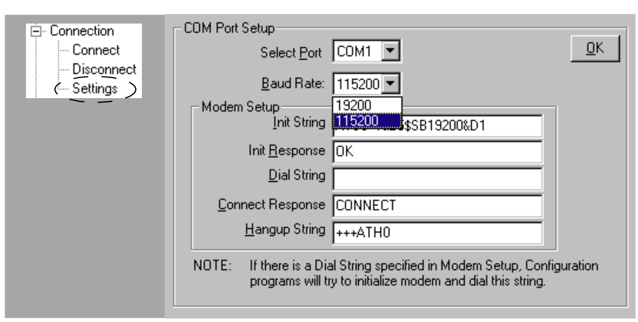
4. **Solving Common Connection Problems**.

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.

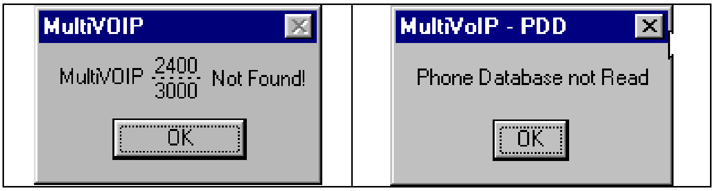
To change the COM port setting, use the **COM Port Setup** dialog box, which is accessible via the keyboard shortcut **Ctrl + G** or by going to the **Connection** pull-down menu and choosing "Settings." In the "Select Port" field, select a COM port that is available on the PC. (If no COM ports are currently 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").

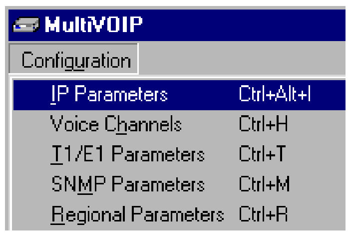
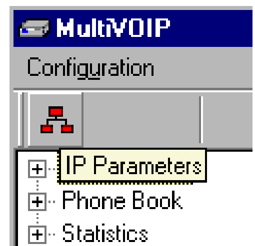
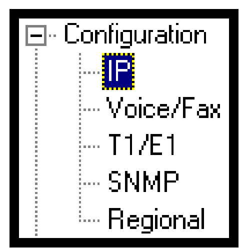


In this case, the MultiVOIP is simply disconnected from the network. For instructions on MultiVOIP cable connections, see the "Cabling" section of Chapter 3.

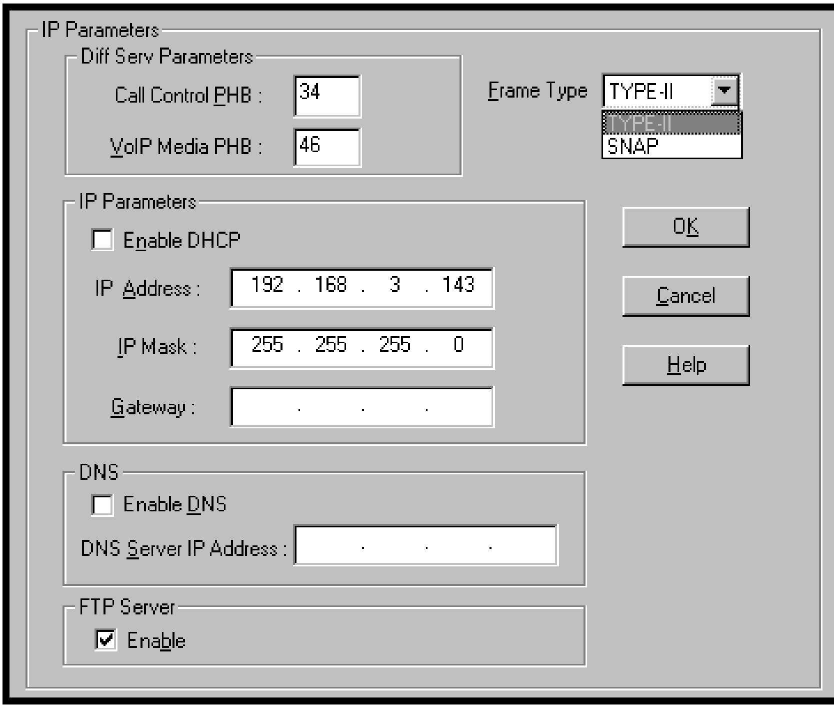
5. **Configuration Parameter Groups: Getting Familiar, Learning About Access**. The first part of configuration concerns IP parameters, Voice/FAX parameters, T1/E1 parameters, SNMP parameters, Regional parameters, SMTP parameters, Supplementary Services parameters, Logs, and System Information. In the 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**
DiffServ Parameter fields	DiffServ PHB (Per Hop Behavior) values pertain to a differential prioritizing system for IP packets as handled by DiffServ-compatible routers. There are 64 values, each with an elaborate technical description. These descriptions are found in TCP/IP standards RFC2474, RFC2597, and, for present purposes, in RFC3246, which describes the value 34 (34 decimal; 22 hex) for Assured Forwarding behavior (default for Call Control PHB) and the value 46 (46 decimal; 2E hexadecimal) for Expedited Forwarding behavior (default for Voip Media PHB). Before using values other than these default values of 34 and 46, consult these standards documents and/or a qualified IP telecommunications engineer. To disable DiffServ, configure both fields to 0 decimal. The next page explains DiffServ in the context of the IP datagram.	
Call Control PHB	0 – 63 default = 34	Value is used to prioritize call setup IP packets.
Voip Media PHB	0 – 63 default = 46	Value is used to prioritize the RTP/RTCP audio IP packets.
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.

The IP Datagram with Header, Its Type-of-Service field, & DiffServ

bits =>

<table>
<tr><td>0</td><td>4</td><td>8</td><td>16</td><td>19</td><td>24</td><td>31</td></tr>
<tr><td>VERS</td><td>HLEN</td><td>TYPE OF SERVICE</td><td colspan="4">TOTAL LENGTH</td></tr>
<tr><td colspan="3">IDENTIFICATION</td><td>FLAGS</td><td colspan="3">FRAGMENT OFFSET</td></tr>
<tr><td colspan="2">TIME TO LIVE</td><td>PROTOCOL</td><td colspan="4">HEADER CHECKSUM</td></tr>
<tr><td colspan="7">SOURCE IP ADDRESS</td></tr>
<tr><td colspan="7">DESTINATION IP ADDRESS</td></tr>
<tr><td colspan="5">IP OPTIONS (if any)</td><td colspan="2">PADDING …
end of header</td></tr>
<tr><td colspan="7">DATA</td></tr>
<tr><td colspan="7">…</td></tr>
</table>

The TOS field consists of eight bits, of which only the first six are used. These six bits are called the “Differentiated Service Codepoint” or DSCP bits.

The Type of Service or “TOS” field

<table>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td></tr>
<tr><td colspan="3">PRECEDENCE</td><td>D</td><td>T</td><td>R</td><td colspan="2">*unused*</td></tr>
</table>

The three precedence have eight values, 0-7, ranging from “normal” precedence (value of 0) to “network control” (value of 7). When set, the *D* bit requests low delay, the *T* bit requests high throughput, and the *R* bit requests high reliability.

Routers that support DiffServ can examine the six DSCP bits and prioritize the packet based on the DSCP value. The DiffServ Parameters fields in the MultiVOIP IP Parameters screen allow you to configure the DSCP bits to values supported by the router. Specifically, the Voip Media PHB field relates to the prioritizing of audio packets (RTP and RTCP packets) and the Call Control PHB field relates to the prioritzing of non-audio packets (packets concerning call set-up and tear-down, gatekeeper registration, etc.).

The MultiVOIP Call Control PHB parameter defaults to 34 decimal (22 hex; 100010 binary – consider vis-à-vis TOS field above) for Assured Forwarding behavior. The MultiVOIP Voip Media PHB parameter defaults to the value 46 decimal (2E hex; 101110 binary – consider vis-à-vis TOS field above). To disable DiffServ, configure both fields to 0 decimal.

IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
IP Parameter fields		
Enable DHCP	Y/N disabled by default	Dynamic Host Configuration Protocol is a method for assigning IP address and other IP parameters to computers on the IP network in a single message with great flexibility. IP addresses can be static or temporary depending on the needs of the computer.
IP Address	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.
DNS Parameter fields		
Enable DNS	Y/N	Enables Domain Name Space/System function where computer names are resolved using a worldwide distributed database.
DNS Server IP Address	4-places, 0-255.	IP address of specific DNS server to be used to resolve Internet computer names.
FTP Parameter fields		
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.

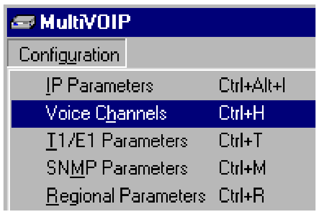
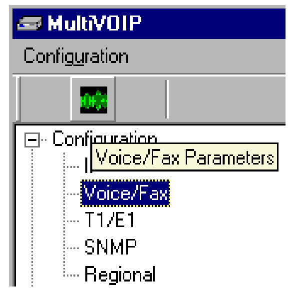
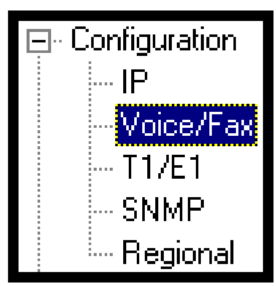
F. Browse to IP address of MultiVOIP unit.

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

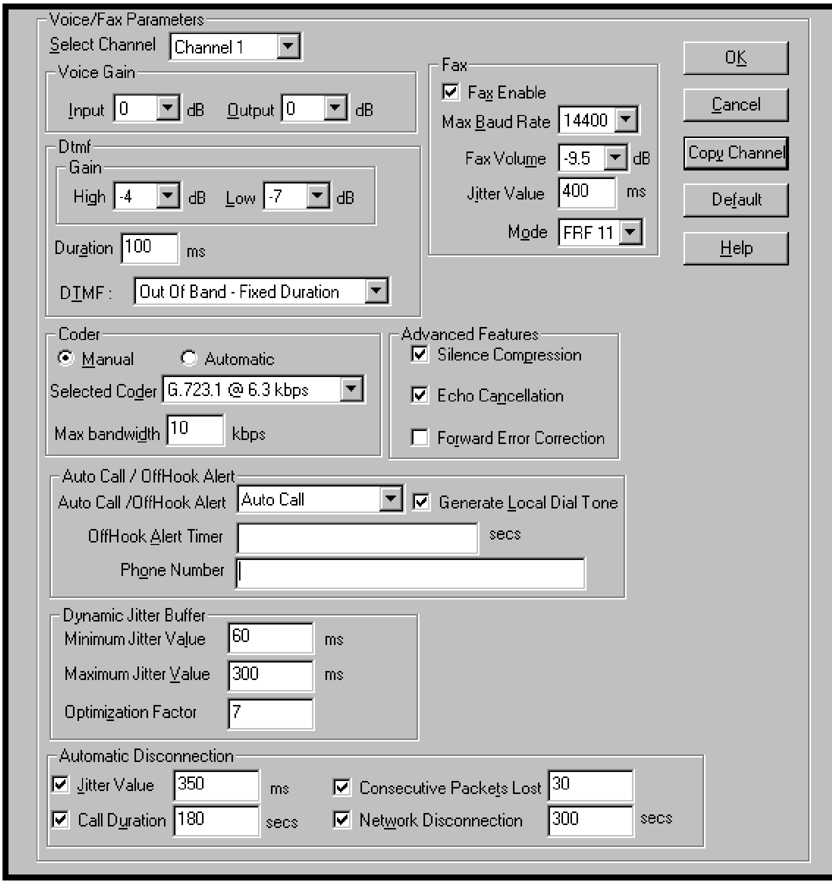
H. Use web browser GUI to configure or operate MultiVOIP unit. The configuration screens in the web browser GUI will have the same content as their counterparts in the Windows GUI; only the graphic presentation will be different.

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

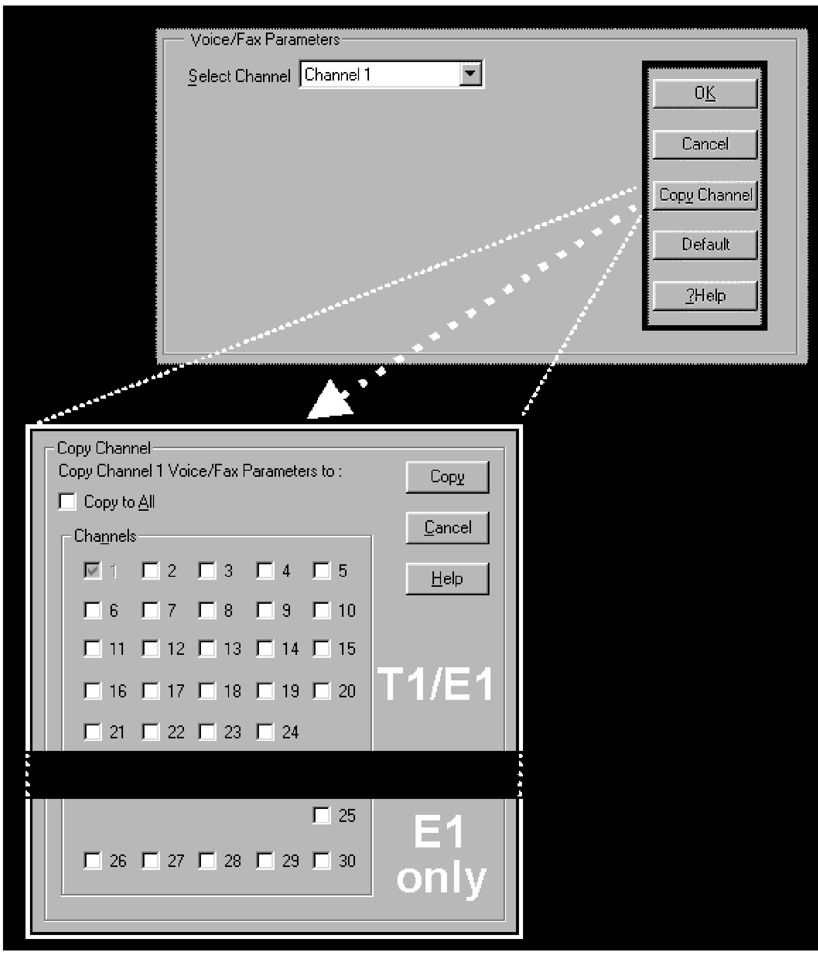
8. **Set Voice/FAX Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

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

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall, Offhook Alert	The **AutoCall** option enables the local MultiVOIP to call a remote MultiVOIP without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote MultiVOIP identified in the **Phone Number** box of this option. This is essentially a hotline function that gives an immediate connection between two points. The **Offhook Alert** option is not supported in T1/E1 MultiVOIP units. **AutoCall** applies on a channel-by-channel basis. It would not be appropriate for this function to be applied to a channel that serves in a pool of available channels for general phone traffic. **AutoCall** requires an entry in the Outgoing phonebook of the local MultiVOIP and a matched setting in the Inbound Phonebook of the remote voip. Also, especially for the AutoCall function,

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Offhook Alert Timer	N/A	Not supported in T1/E1 MultiVOIP units.
Phone Number	N/A	Not supported in T1/E1 MultiVOIP units.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dynamic Jitter Buffer Parameters		
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

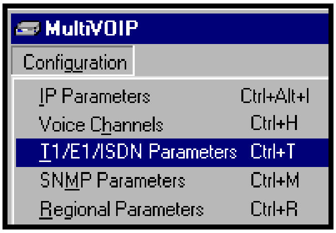
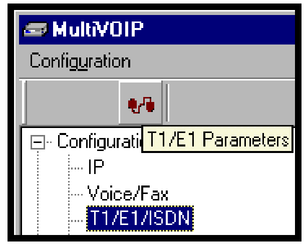
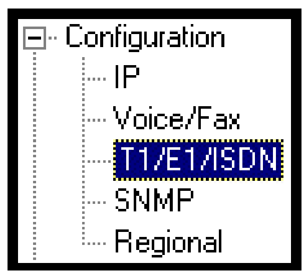
Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dynamic Jitter Buffer Parameters		
Maximum Jitter Value	60 to 400 ms	The default maximum dynamic jitter buffer of **300** milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 msec
Optimizat-ion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

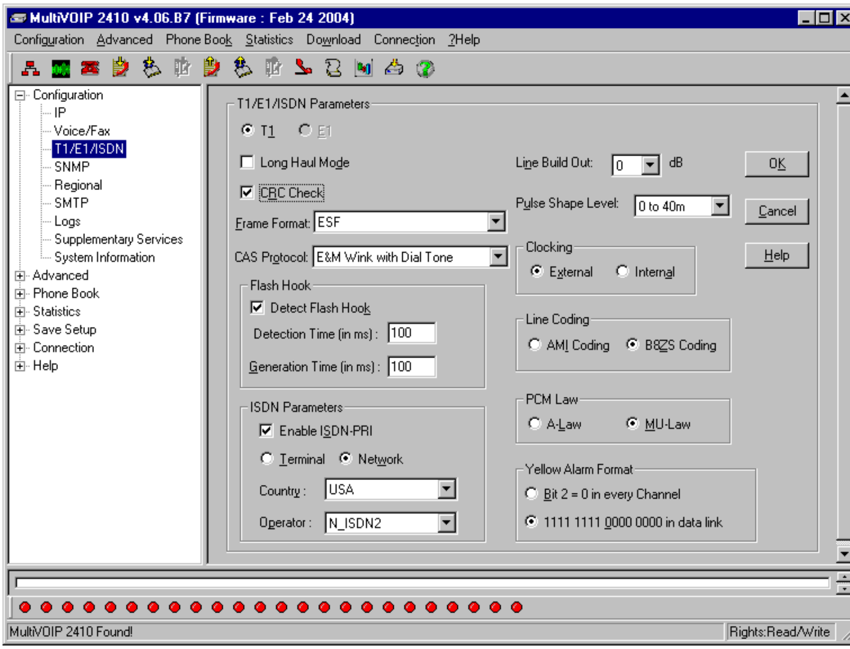
To place modem traffic onto the voip network (an application called “modem relay”), use Coder G.711 mu-law at 64kbps.

Voice/Fax Parameter Definitions (cont'd))		
Field Name	**Values**	**Description**
Auto Disconnect Parameters		The **Automatic Disconnection** group has 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.

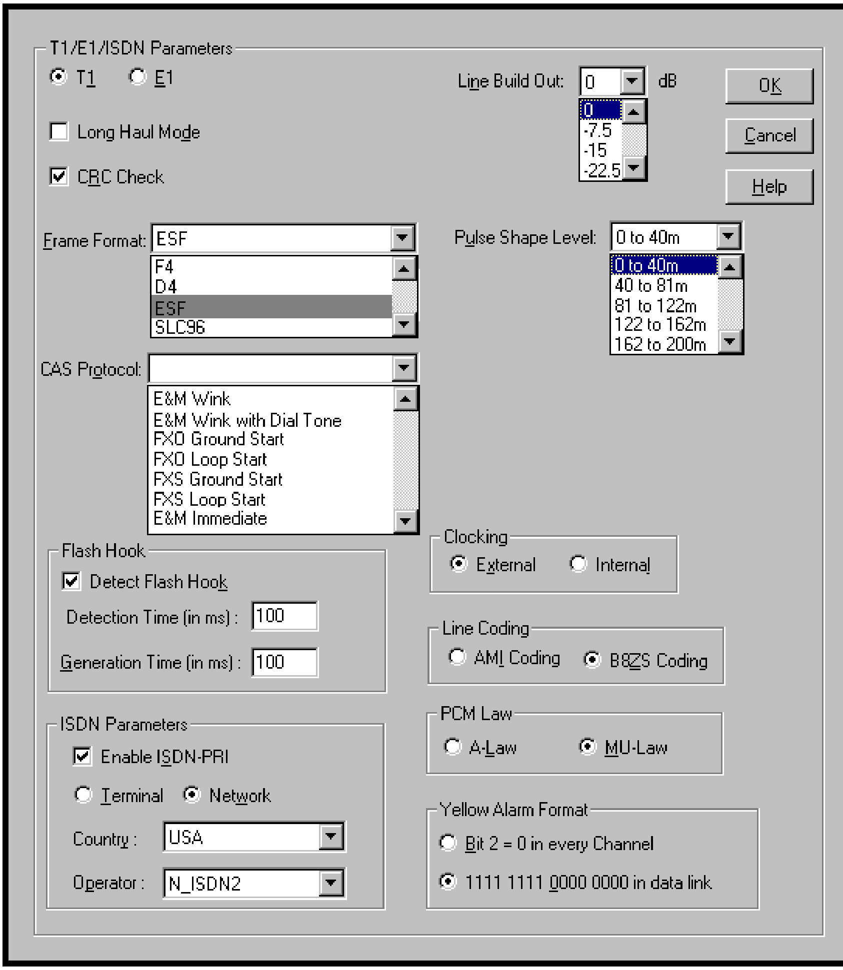
9. **Set T1/E1/ISDN Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

Accessing "T1/E1/ISDN Parameters"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
Ctrl + T	

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.

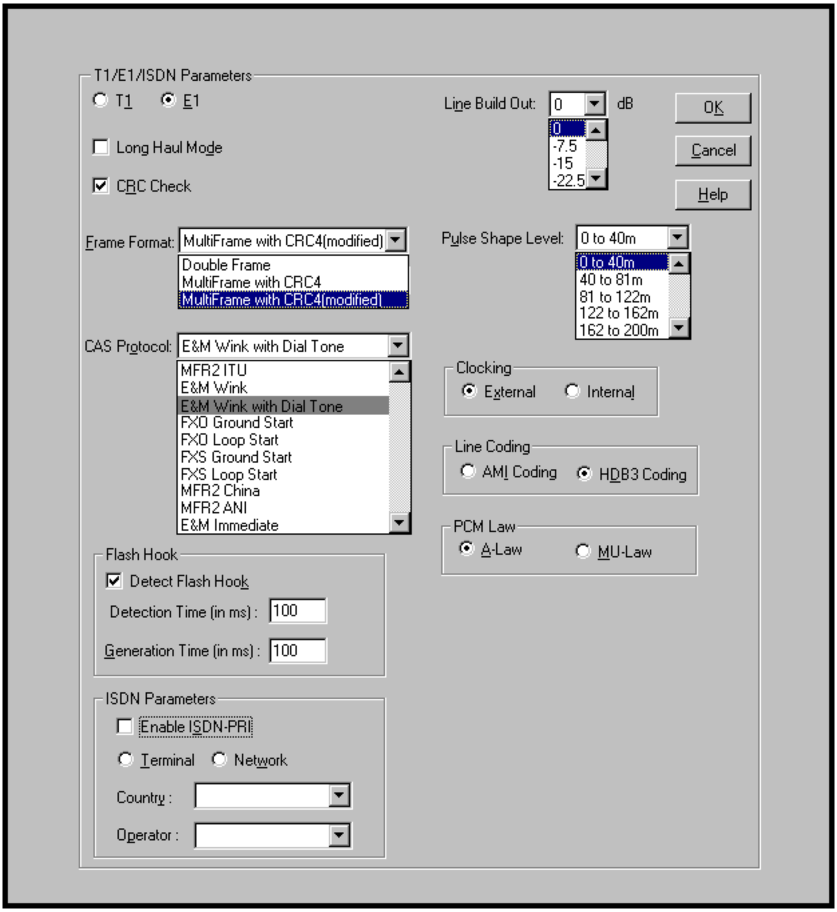
T1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
CAS Protocol	E&M Immed Strt E&M Wink Start E&M Wink with dial tone FXO Ground Strt FXO Loop Start FXS Ground Strt FXS Loop Start	Channel Associated Signaling (CAS) is a method of incorporating telephony signaling info into a T1 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)		
Field Name	**Values**	**Description**
Detect Flash Hook	Y/N	This setting determines whether or not the MultiVOIP responds to hook-flash signals.
Detection Time	100 – 1500 milliseconds	Minimum hook-flash time that will be interpreted as a valid flash by the MultiVOIP.
Generation Time	100 – 1500 milliseconds	In some systems, a MultiVOIP might receive a hook-flash signal from an upstream device (a PBX, voip or other device) and must replicate it to a downstream device. This parameter determines the duration of the hook-flash signal that is passed to a downstream device.

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.

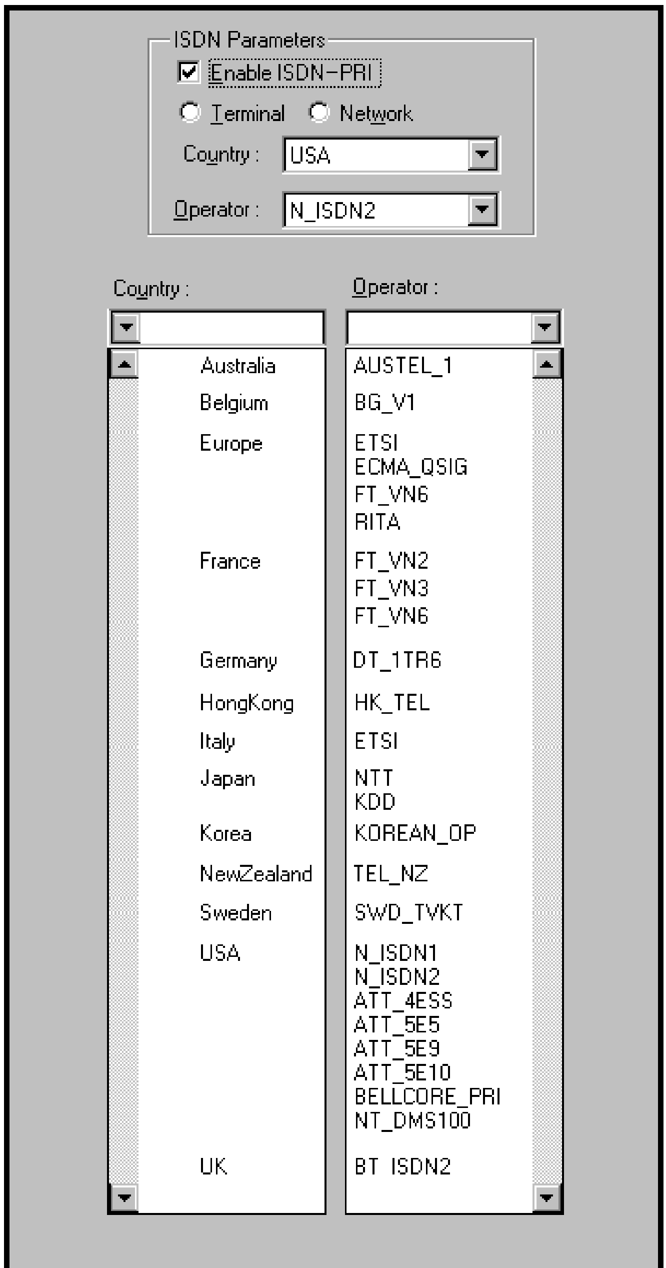
E1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
CAS Protocol	E&M Immed Strt E&M Wink Start E&M Wink with dial tone FXO Ground Strt FXO Loop Start FXS Ground Strt FXS Loop Start MFR2ITU MFR2 China MFR2 ANI	Channel Associated Signaling (CAS) is a method of incorporating telephony 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)		
Field Name	**Values**	**Description**
Detect Flash Hook	Y/N	This setting determines whether or not the MultiVOIP responds to hook-flash signals.
Detection Time	100 – 1500 milliseconds	Minimum hook-flash time that will be interpreted as a valid flash by the MultiVOIP.
Generation Time	100 – 1500 milliseconds	In some systems, a MultiVOIP might receive a hook-flash signal from an upstream device (a PBX, voip or other device) and must replicate it to a downstream device. This parameter determines the duration of the hook-flash signal that is passed to a downstream device.

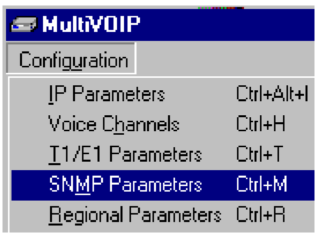
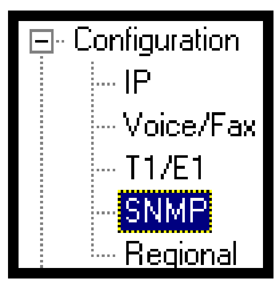
E1 Parameter Definitions (cont'd)		
ISDN Parameters		
Field Name	**Values**	**Description**
Enable ISDN-PRI	Y/N	If digital connection is ISDN-PRI type, this box should be checked. When ISDN is enabled, the "CAS Protocols" field is grayed out (ISDN has its own signaling method).
Terminal/ Network	either "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.

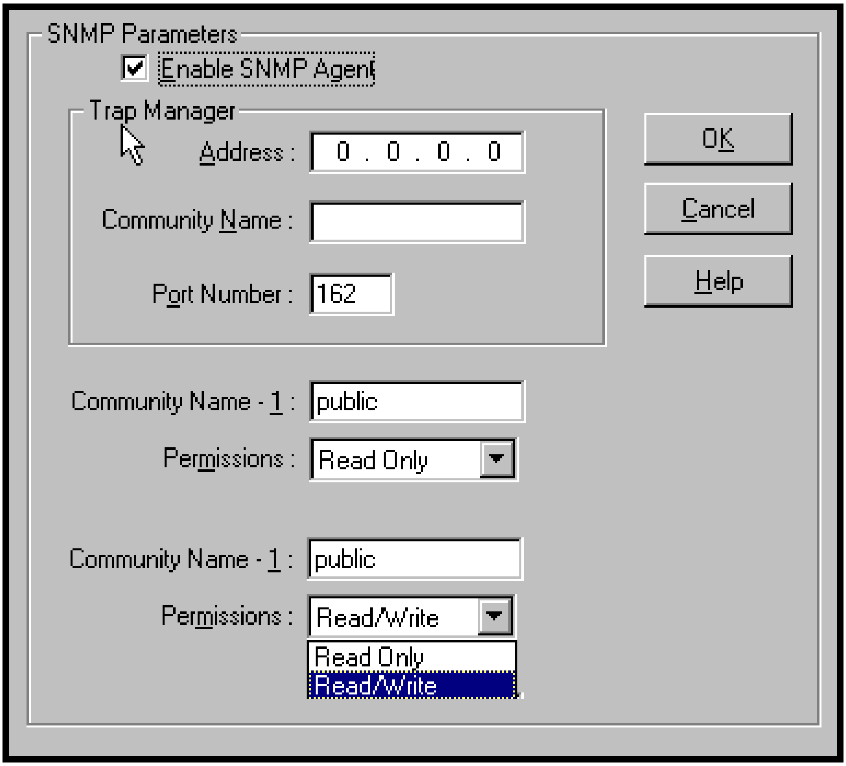
10. **Set ISDN Parameters** (if applicable). These parameters are accessible in the **T1/E1/ISDN Parameters** screen. If your T1 or E1 phone line is a Primary Rate Interface ISDN line, enable ISDN-PRI and set it for the particular implementation of ISDN that your telco uses. The ISDN types supported 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>`Ctrl + M`</td><td>
</td></tr>
</table>

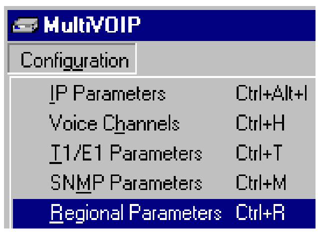
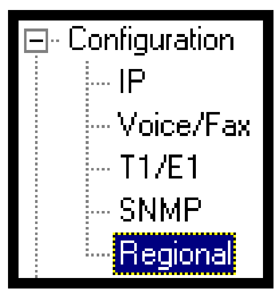
In each field, enter the values that fit your particular system.



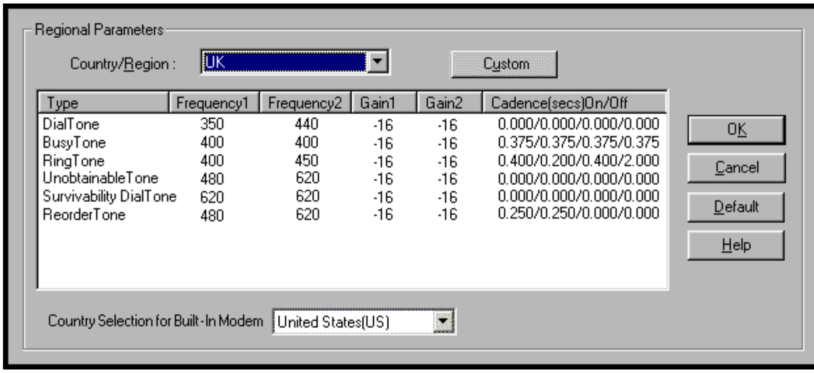
The SNMP Parameter fields are described in the table below.

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.

12. **Set Regional Parameters** (Phone Signaling Tones & Cadences and setup for built-in Remote Configuration/Command Modem)**.** 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.



Remote Configuration/Command Modem. Each MVP2410 or MVP3010 unit contains a built-in modem. This modem allows the MultiVOIP to be configured remotely when a standard POTS line is connected to the "Command Modem" connector on the back panel of the MultiVOIP. In the **Country Selection for Built-In Modem** field (drop-down list), select the country that best fits your situation. This may not be the same as your selection for the **Country/Region** field. The selections in the **Country Selection for Built-In Modem** field entail more detailed groupings of telephony parameters than do the **Country/Region** values.

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

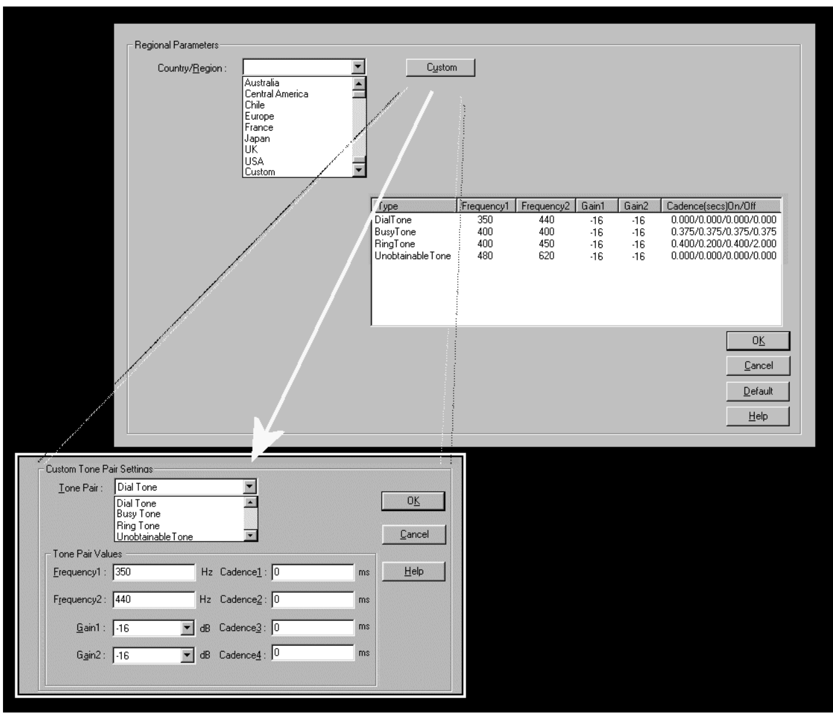
The **Regional Parameters** fields are described in the table below.

"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), **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), re-order tone.	Type of telephony tone-pair for which frequency, gain, and cadence are being presented.

"Regional Parameter" Definitions		
Field Name	**Values**	**Description**
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.
Country Selection for Built-In Modem	country name	MultiVOIP units operating with the X.06 software release (and above) include a built-in modem. The administrator can dial into this modem to configure the MultiVOIP unit remotely. The country name values in this field set telephony parameters that allow the modem to work in the listed country. This value may be different than the Country/Region value. For example, a user may need to choose “Europe” as the Country/Region value but “Denmark” as the Country-Selection-for-Built-In-Modem value.

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.

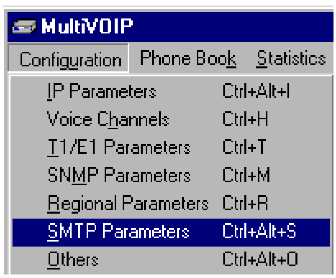
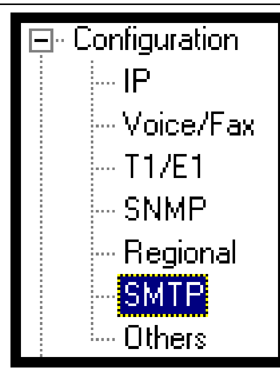


The **Custom Tone-Pair Settings** fields are described in the table below.

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**

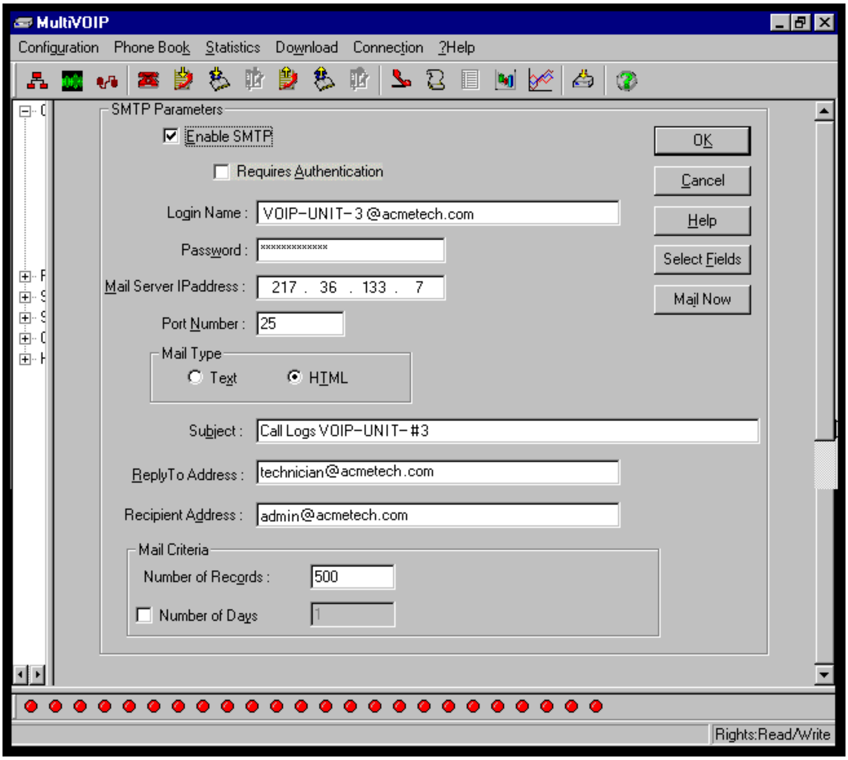
Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Cadence 1	integer time value in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable tone (fast busy), dial tone (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.

Accessing "SMTP Parameters"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + Alt + S`	

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.

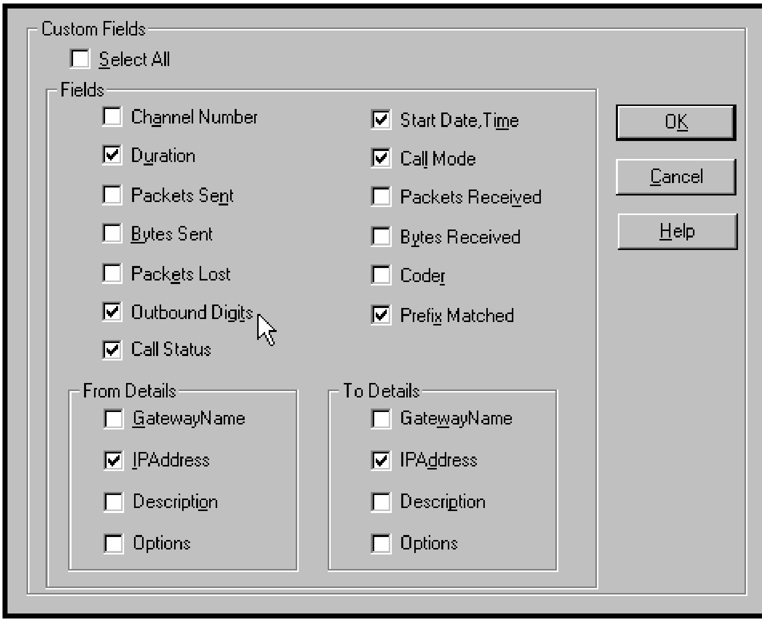


"SMTP Parameters" Definitions		
Field Name	**Values**	**Description**
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select "SMTP" in the **Logs** screen.
Requires Authentication	Y/N	If this checkbox is checked, the MultiVOIP will send Authentication information to the SMTP server. The authentication information indicates whether or not the email sender has permission to use the SMTP server.
Login Name	alpha-numeric, 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.

......

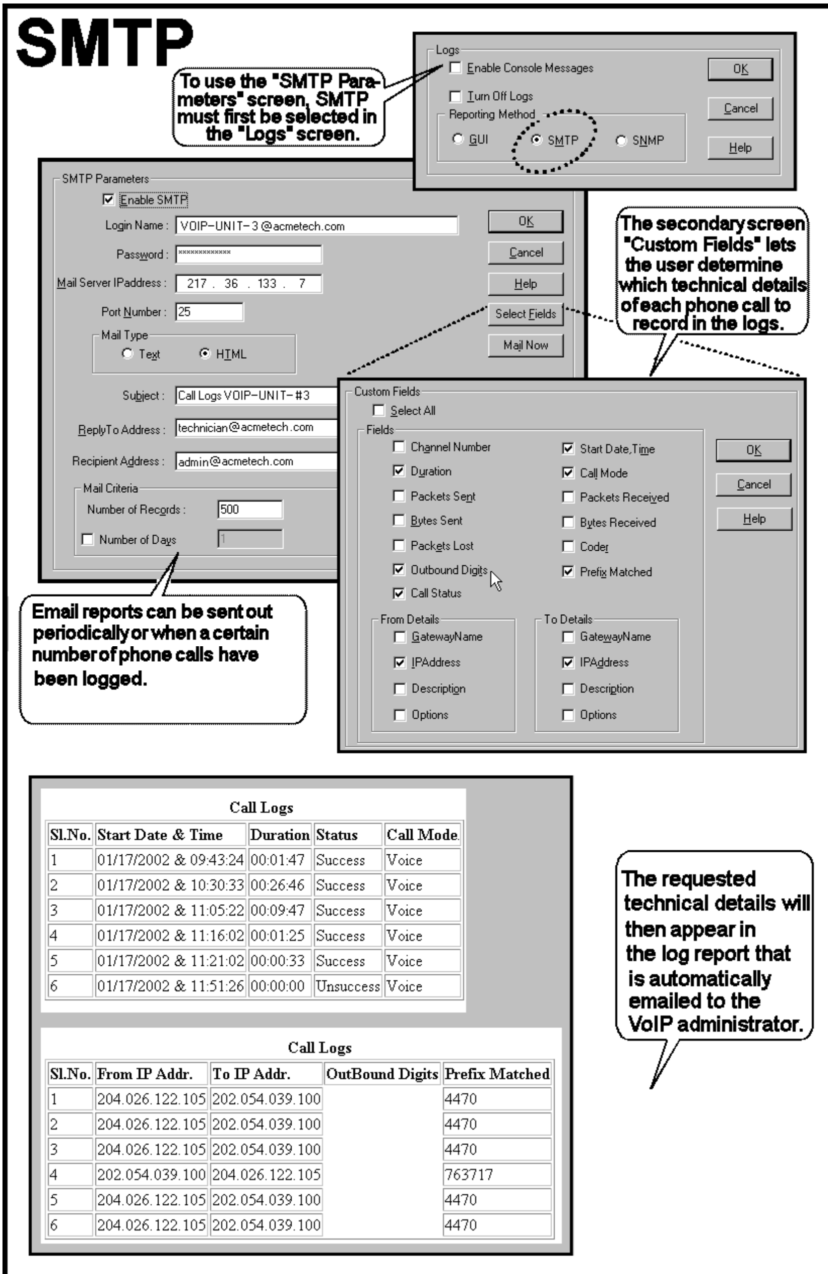
“SMTP Parameters” Definitions (cont’d)		
Field Name	**Values**	**Description**
Mail Type	text or html	Mail type in which log reports will be sent.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP unit.
Reply-To Address	email address	User specified. 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

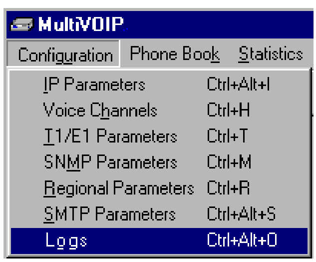
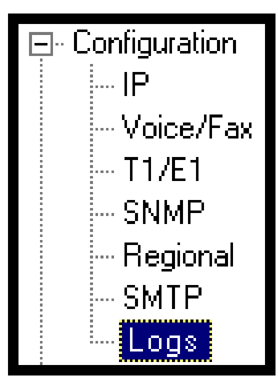
Sl.No.	From IP Addr.	To IP Addr.	OutBound Digits	Prefix Matched
1	204.026.122.105	202.054.039.100		4470
2	204.026.122.105	202.054.039.100		4470
3	204.026.122.105	202.054.039.100		4470
4	202.054.039.100	204.026.122.105		763717
5	204.026.122.105	202.054.039.100		4470
6	204.026.122.105	202.054.039.100	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

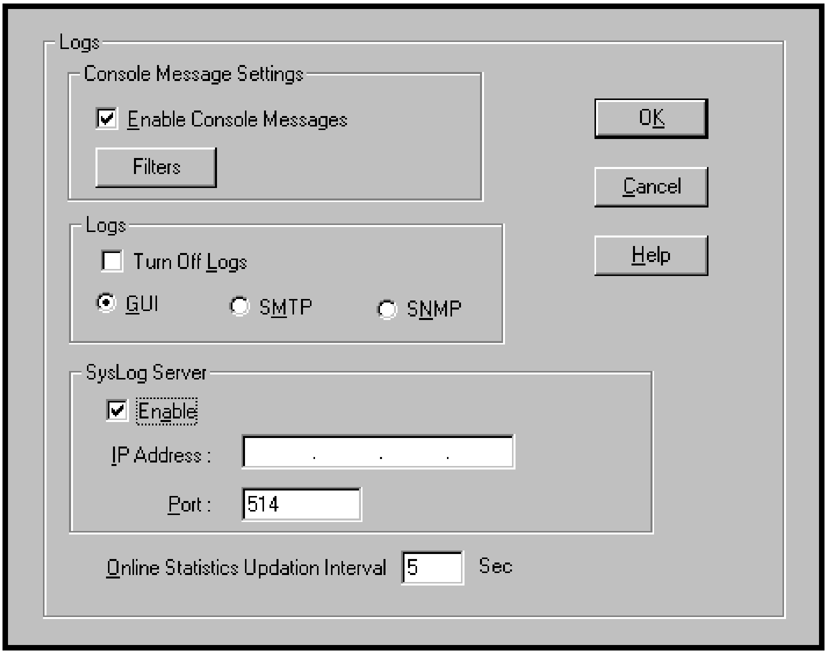
A. in the MultiVOIP program (GUI),

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`	

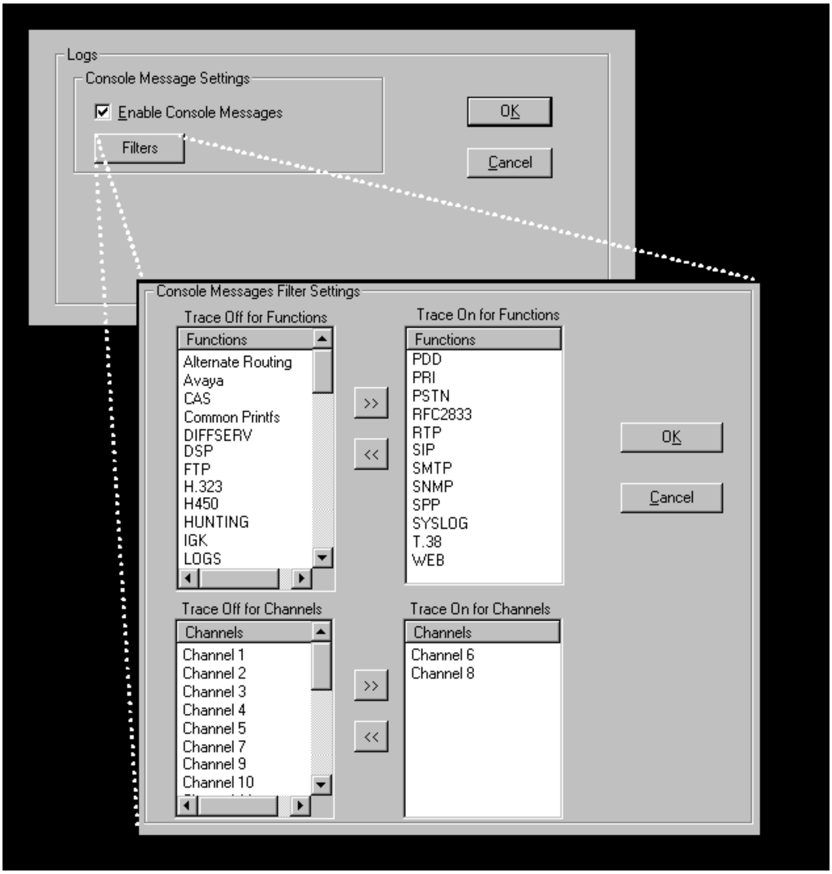
If you enable console messages, you can customize the types of messages to be included/excluded in log reports by clicking on the "Filters" button and using the **Console Messages Filter Settings** screen (see subsequent page). If you use the logging function, select the logging option that applies to 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).



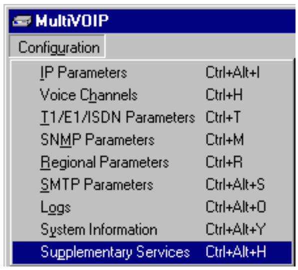
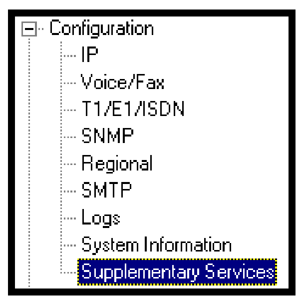
“Logs” Screen Definitions		
Field Name	**Values**	**Description**
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic tele-communications program like HyperTerminal ™ or similar application. Normally, this should be disabled because it consumers MultiVOIP pro-cessing 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.

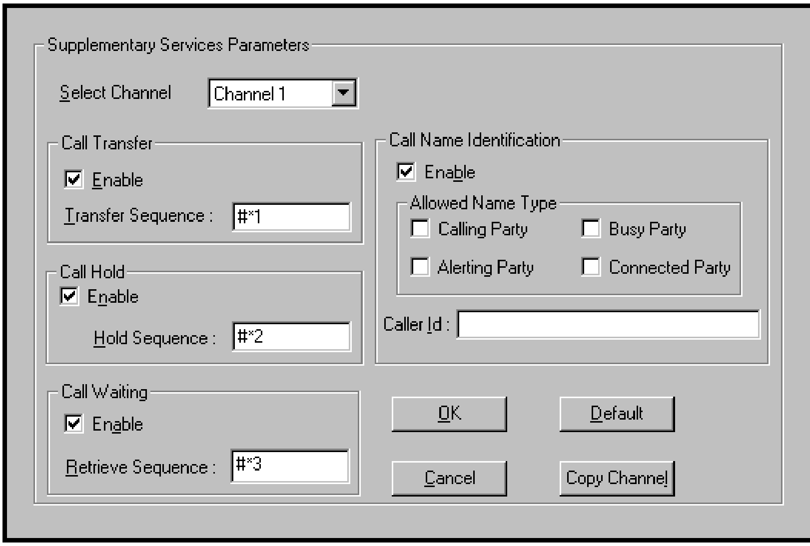


16. **Set Supplementary Services Parameters.** This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “Supplementary Services 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 +H</code></td><td>
</td></tr>
</table>

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**.

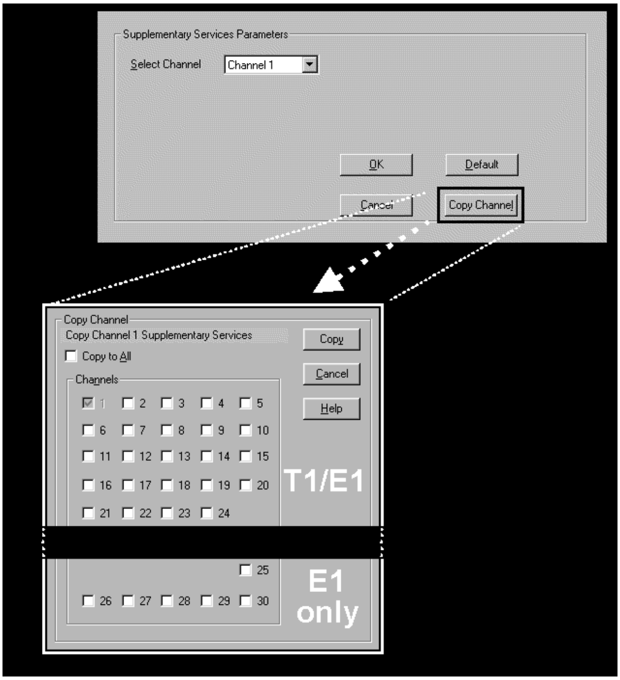
Call Transfer. Call Transfer allows one party to re-connect the party with whom they have been speaking to a third party. The first party is disconnected when the third party becomes connected. Feature is invoked by a programmable phone keypad sequence (for example, #7).

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 (MVP-130/ 130FXS 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. A brief musical jingle is played for the caller on hold.
Transfer Sequence	any phone keypad character	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call transfer. The call-transfer sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). The sequences for call transfer, call hold, and call waiting can be from 1 to 4 digits in length consisting of any combination of digits 1234567890*#.

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.

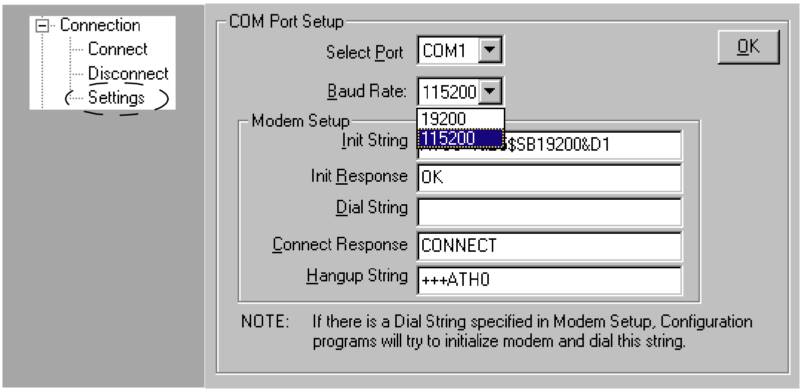
Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Alerting Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving the call and **Alerting Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit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

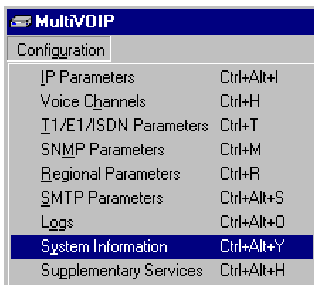
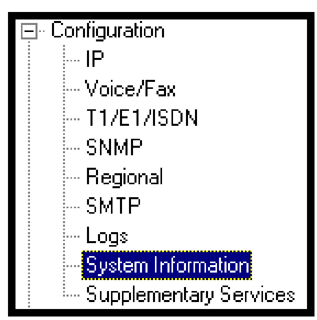


First, it is important to note that the default COM port established by the MultiVOIP program is COM1. ***Do not accept the default value until you have checked the COM port allocation on your PC***. To do this, check for COM port assignments in the system resource dialog box(es) of your Windows operating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

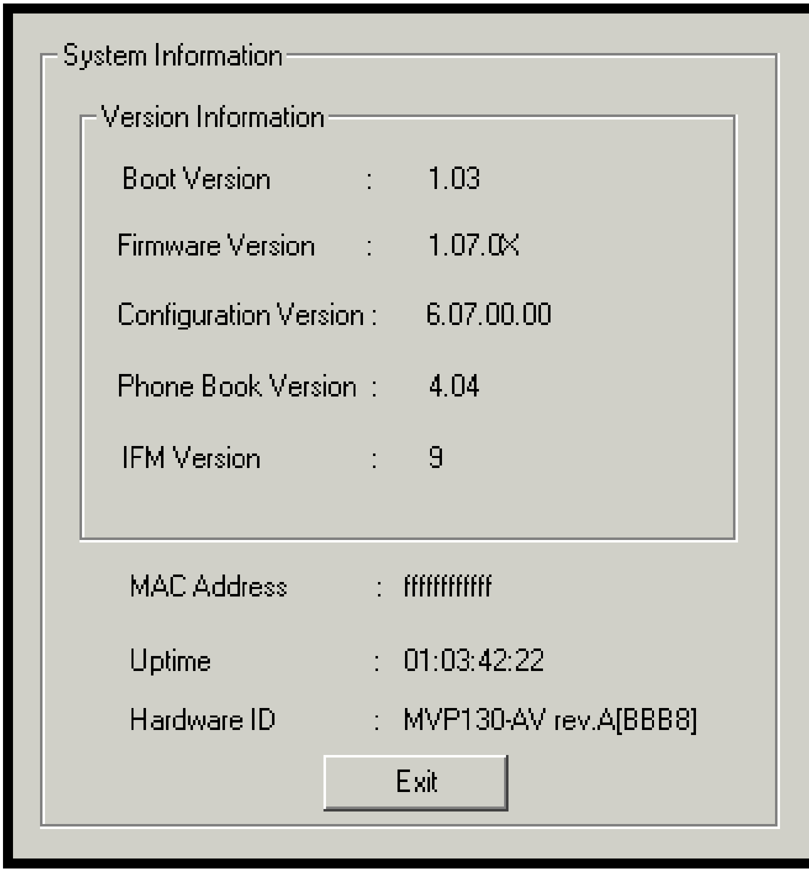
The default baud rate is 115,200 bps.

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><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 +Y</code></td><td>
</td></tr>
</table>

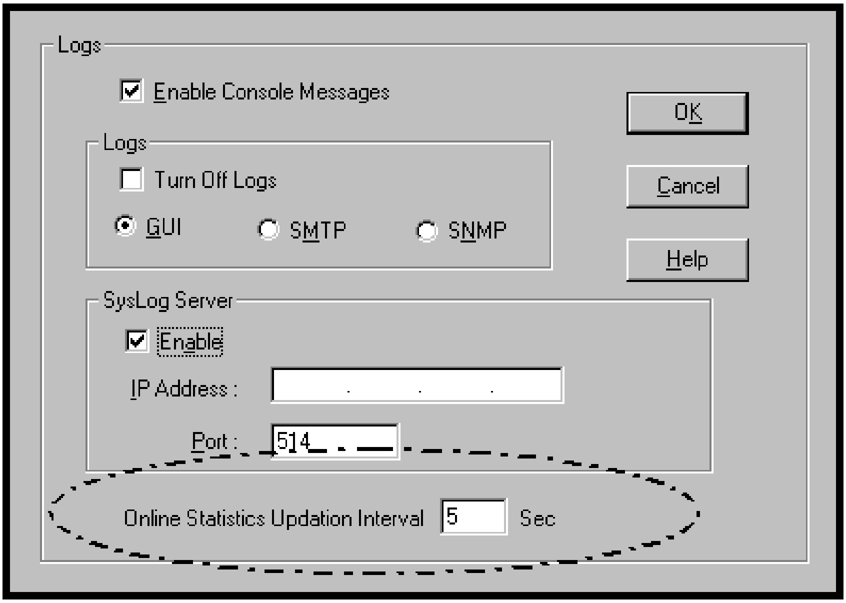
This screen presents vital system information at a glance. Its primary use is in troubleshooting.



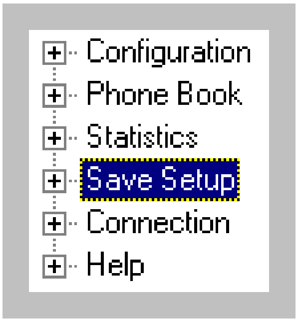
System Information Parameter Definitions		
Field Name	**Values**	**Description**
Boot Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Firmware Version	alpha-numeric	Indicates version of MultiVOIP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

System Information Parameter Definitions		
Field Name	**Values**	**Description**
Configur-ation Version	nn.nn.nn.nn alpha-numeric	Indicates version of MultiVOIP Configuration software (which includes screens for IP Parameters, SNMP Parameters, SMTP Parameters, Regional Parameters, etc.
Phone Book Version	numeric	Indicates the version of the inbound and outbound phonebook portion of the MultiVOIP software.
IFM Version	numeric	Indicates the version of the firmware running on the MultiVOIP's Interface Module, which is its analog telephony hardware.
Mac Address	alpha-numeric	Denotes the number assigned as the voip unit's unique node identifier.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.
Hardware ID	alpha-numeric	Indicates the version of the MultiVOIP unit's circuit board and components.

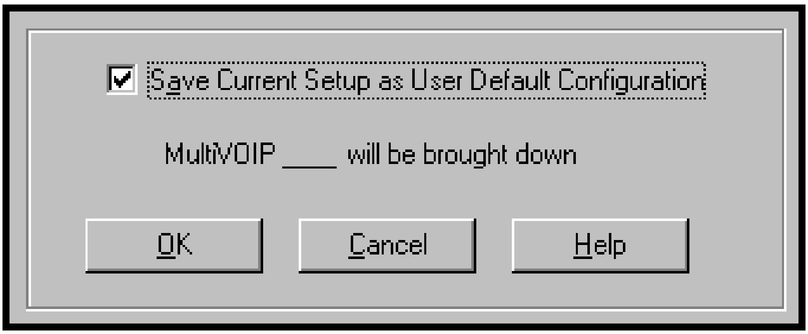
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



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 (MVP-130/130FXS, MVP-210, MVP-410, MVP-810 & 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.

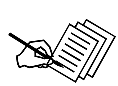
Pre-Requisites

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:

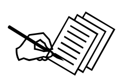
✓	*Ask your computer network administrator.*	*Info needed to operate:* **all MultiVOIP models.**
	 IP Network Parameters: Record for each VOIP Site in System	
	• IP Address	
	• IP Mask	
	• Gateway	
	• Domain Name Server (DNS) Info *(not implemented; for future use)*	

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 MVP-130/130FXS/210/410/810)

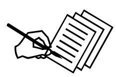
The following parameters must be known about the PBX or telco central office equipment to which the analog MultiVOIP will connect:

<table>
<tr><td>✓</td><td>Analog Phone Parameters

Ask phone company or telecom manager.</td><td>Needed for:
MVP810
MVP410
MVP210
MVP130
MVP130FXS</td></tr>
<tr><td></td><td colspan="2"> Analog Telephony Interface Parameters:
Record for this VOIP Site</td></tr>
<tr><td></td><td colspan="2">• Which interface type (or “signaling”) is used?
E&M_____ FXS/FXO_____</td></tr>
<tr><td></td><td colspan="2">• If FXS, determine whether the line will be used for a phone, fax, or KTS (key telephone system)</td></tr>
<tr><td></td><td colspan="2">• If FXO, determine if line will be an analog PBX extension or an analog line from a telco central office</td></tr>
<tr><td></td><td colspan="2">• 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?</td></tr>
</table>

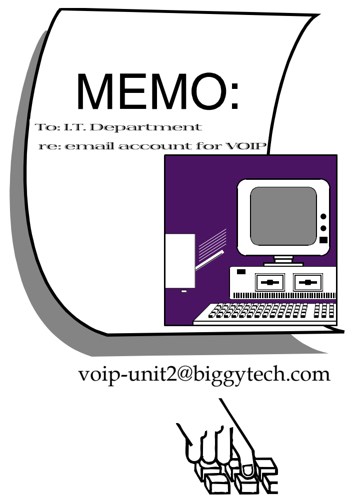
ISDN-BRI Telephony Parameters (for MVP-410ST/810ST)

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 & Cadences and setup for built-in Remote Configuration/Command Modem).

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).

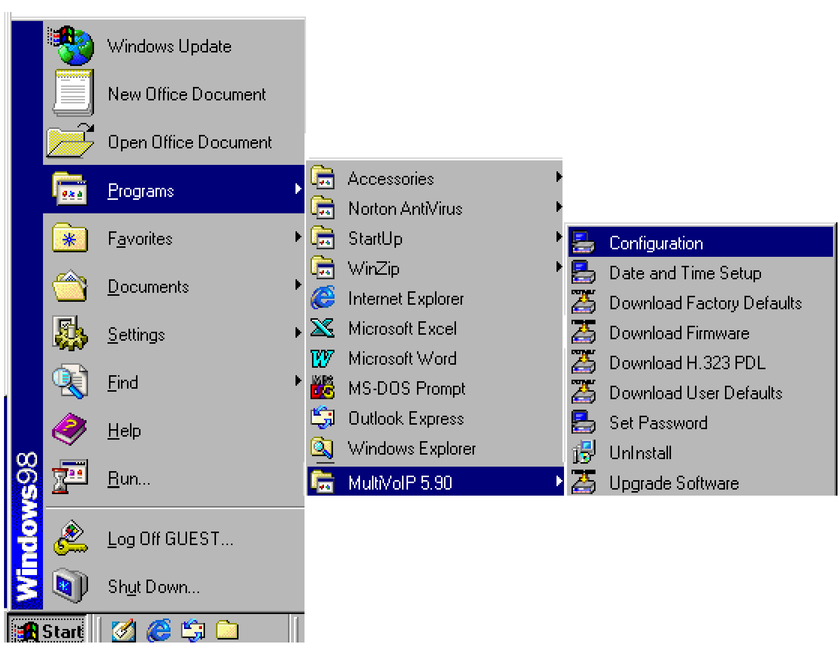
When technical configuration is complete, you will need to configure the MultiVOIP's inbound and outbound phonebooks. This manual has separate chapters describing ***T1 Phonebook Configuration*** for North-American-influenced telephony settings and ***E1 Phonebook Configuration*** for Euro-influenced telephony settings.

Local Configuration Procedure (Detailed)

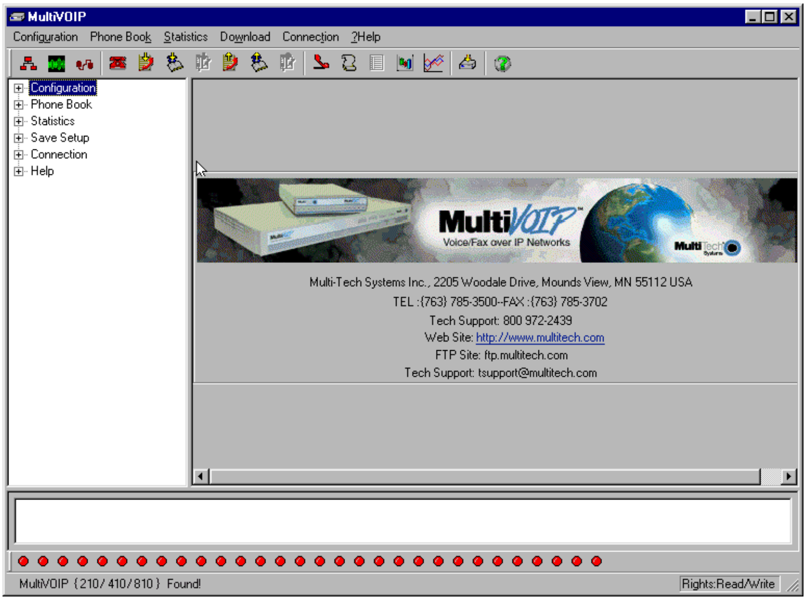
You can begin the configuration process as a continuation of the MultiVOIP software installation. You can establish your configuration or modify it at any time by launching the MultiVOIP program from the Windows **Start** menu.

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).

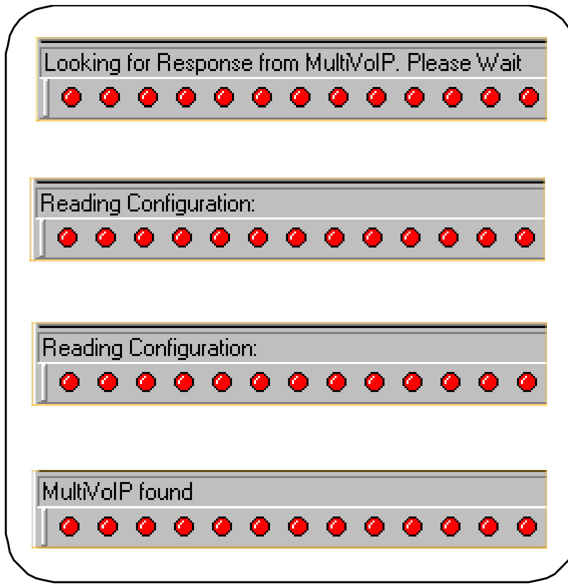
2. **Start MultiVOIP Configuration Program**. Launch the MultiVOIP program from the Windows **Start** menu (from the folder location determined during installation).



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.

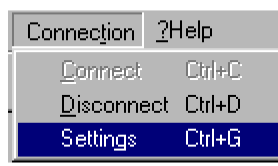


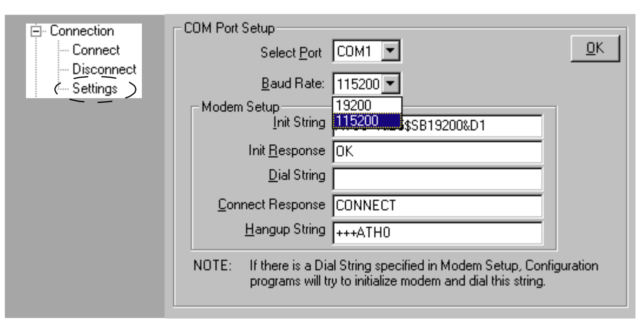
4. **Solving Common Connection Problems**.

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.

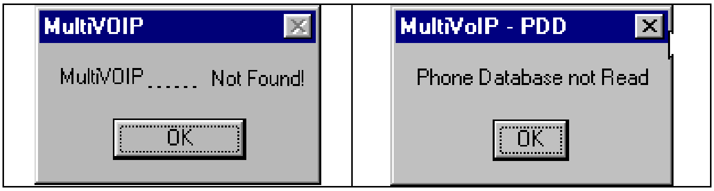
To change the COM port setting, use the **COM Port Setup** dialog box, which is accessible via the keyboard shortcut **Ctrl + G** or by going to the **Connection** pull-down menu and choosing “Settings.” In the “Select Port” field, select a COM port that is available on the PC. (If no COM ports are currently 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").

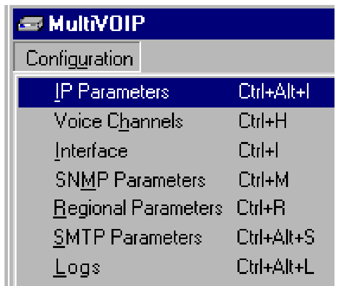
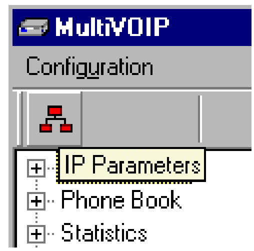
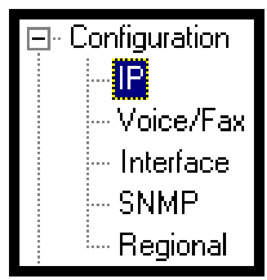


In this case, the MultiVOIP is simply disconnected from the network. For instructions on MultiVOIP cable connections, see the Cabling section of Chapter 3.

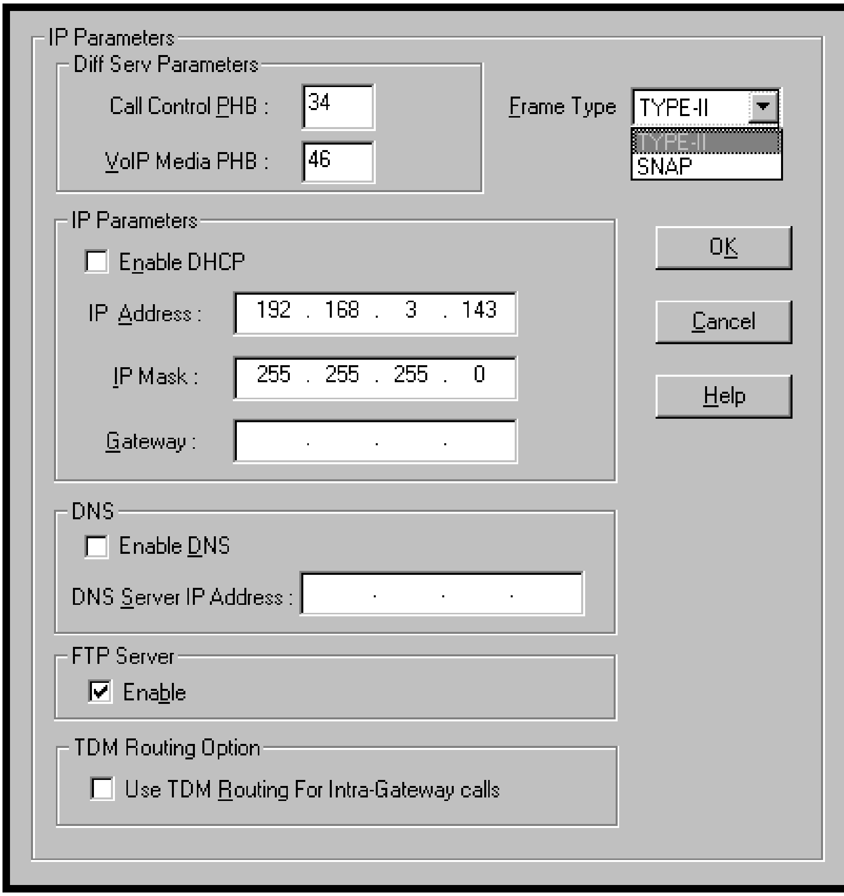
5. **Configuration Parameter Groups: Getting Familiar, Learning About Access**. The first part of configuration concerns IP parameters, Voice/FAX parameters, Telephony Interface parameters, SNMP parameters, Regional parameters, SMTP parameters, Supplementary Services parameters, Logs, and System Information. 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.

Accessing "IP Parameters"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + Alt + I`	

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**
DiffServ Parameter fields	DiffServ PHB (Per Hop Behavior) values pertain to a differential prioritizing system for IP packets as handled by DiffServ-compatible routers. There are 64 values, each with an elaborate technical description. These descriptions are found in TCP/IP standards RFC2474, RFC2597, and, for present purposes, in RFC3246, which describes the value 34 (34 decimal; 22 hex) for Assured Forwarding behavior (default for Call Control PHB) and the value 46 (46 decimal; 2E hexadecimal) for Expedited Forwarding behavior (default for Voip Media PHB). Before using values other than these default values of 34 and 46, consult these standards documents and/or a qualified IP telecommunications engineer. To disable DiffServ, configure both fields to 0 decimal. The next page explains DiffServ in the context of the IP datagram.	
Call Control PHB	0 – 63 default = 34	Value is used to prioritize call setup IP packets.
Voip Media PHB	0 – 63 default = 46	Value is used to prioritize the RTP/RTCP audio IP packets.
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.

The IP Datagram with Header, Its Type-of-Service field, & DiffServ

bits =>

<table>
<tr><td>0</td><td>4</td><td>8</td><td>16</td><td>19</td><td>24 31</td></tr>
<tr><td>VERS</td><td>HLEN</td><td>TYPE OF SERVICE</td><td colspan="3">TOTAL LENGTH</td></tr>
<tr><td colspan="3">IDENTIFICATION</td><td>FLAGS</td><td colspan="2">FRAGMENT OFFSET</td></tr>
<tr><td colspan="2">TIME TO LIVE</td><td>PROTOCOL</td><td colspan="3">HEADER CHECKSUM</td></tr>
<tr><td colspan="6">SOURCE IP ADDRESS</td></tr>
<tr><td colspan="6">DESTINATION IP ADDRESS</td></tr>
<tr><td colspan="5">IP OPTIONS (if any)</td><td>PADDING …
end of header</td></tr>
<tr><td colspan="6">DATA</td></tr>
<tr><td colspan="6">…</td></tr>
</table>

The TOS field consists of eight bits, of which only the first six are used. These six bits are called the “Differentiated Service Codepoint” or DSCP bits.

The Type of Service or “TOS” field

<table>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td></tr>
<tr><td colspan="3">PRECEDENCE</td><td>D</td><td>T</td><td>R</td><td colspan="2">*unused*</td></tr>
</table>

three precedence have eight values, 0-7, ranging from “normal” precedence (value of 0) to “network control” (value of 7). When set, the *D* bit requests low delay, the *T* bit requests high throughput, and the *R* bit requests high reliability.

Routers that support DiffServ can examine the six DSCP bits and prioritize the packet based on the DSCP value. The DiffServ Parameters fields in the MultiVOIP IP Parameters screen allow you to configure the DSCP bits to values supported by the router. Specifically, the Voip Media PHB field relates to the prioritizing of audio packets (RTP and RTCP packets) and the Call Control PHB field relates to the prioritzing of non-audio packets (packets concerning call set-up and tear-down, gatekeeper registration, etc.).

The MultiVOIP Call Control PHB parameter defaults to 34 decimal (22 hex; 100010 binary – consider vis-à-vis TOS field above) for Assured Forwarding behavior. The MultiVOIP Voip Media PHB parameter defaults to the value 46 decimal (2E hex; 101110 binary – consider vis-à-vis TOS field above). To disable DiffServ, configure both fields to 0 decimal.

IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
IP Parameter fields		
Enable DHCP	Y/N disabled by default	Dynamic Host Configuration Protocol is a method for assigning IP address and other IP parameters to computers on the IP network in a single message with great flexibility. IP addresses can be static or temporary depending on the needs of the computer.
IP Address	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 Parameter fields		
Enable DNS	Y/N Default = disabled	Enables Domain Name Space/System function where computer names are resolved using a worldwide distributed database.
DNS Server IP Address	4-places, 0-255.	IP address of specific DNS server to be used to resolve Internet computer names.

IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
FTP Parameter fields		
FTP Server Enable	Y/N Default = disabled 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.
TSM Routing Option Parameter fields		
Use TDM Routing for Intra-Gateway calls	Y/N; enabled by default	Allows calls placed between ports on the same MultiVOIP voice channel board to be routed over internal Time Division Multiplex bus without conversion to IP. TDM routing effectively eliminates the delay introduced by IP conversion. If you require all calls to be IP routed, disable the "use TDM Routing for Intra-Gateway Calls" option. Since this is not normally required, we generally recommend leaving TDM Routing enabled. *Not applicable to MVP-130/130FXS.*

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.

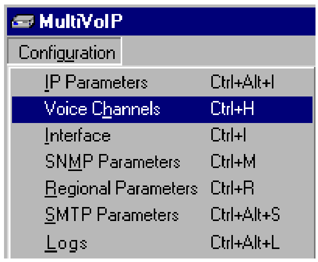
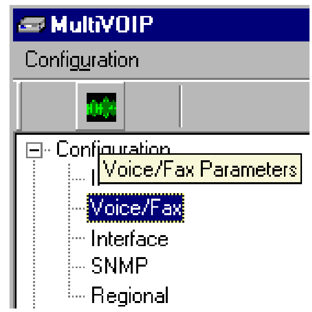
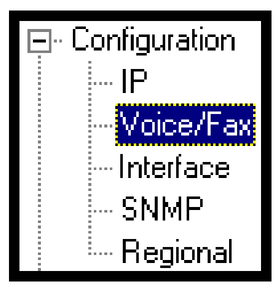
F. Browse to IP address of MultiVOIP unit.

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

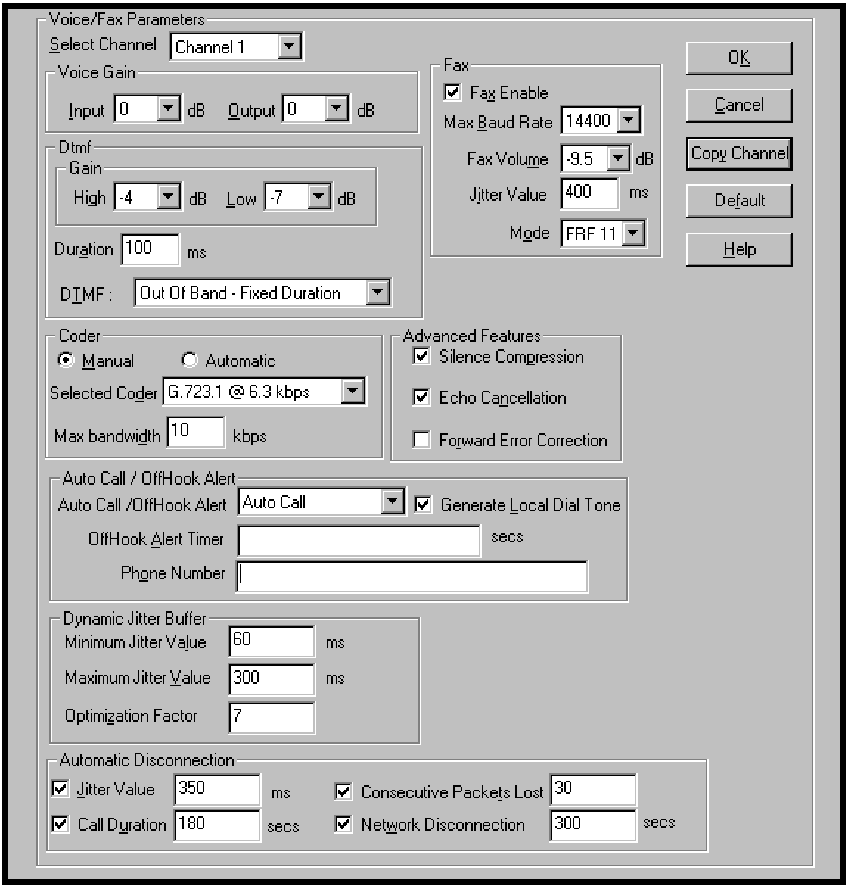
H. Use web browser GUI to configure or operate MultiVOIP unit. The configuration screens in the web browser GUI will have the same content as their counterparts in the Windows GUI; only the graphic presentation will be different.

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

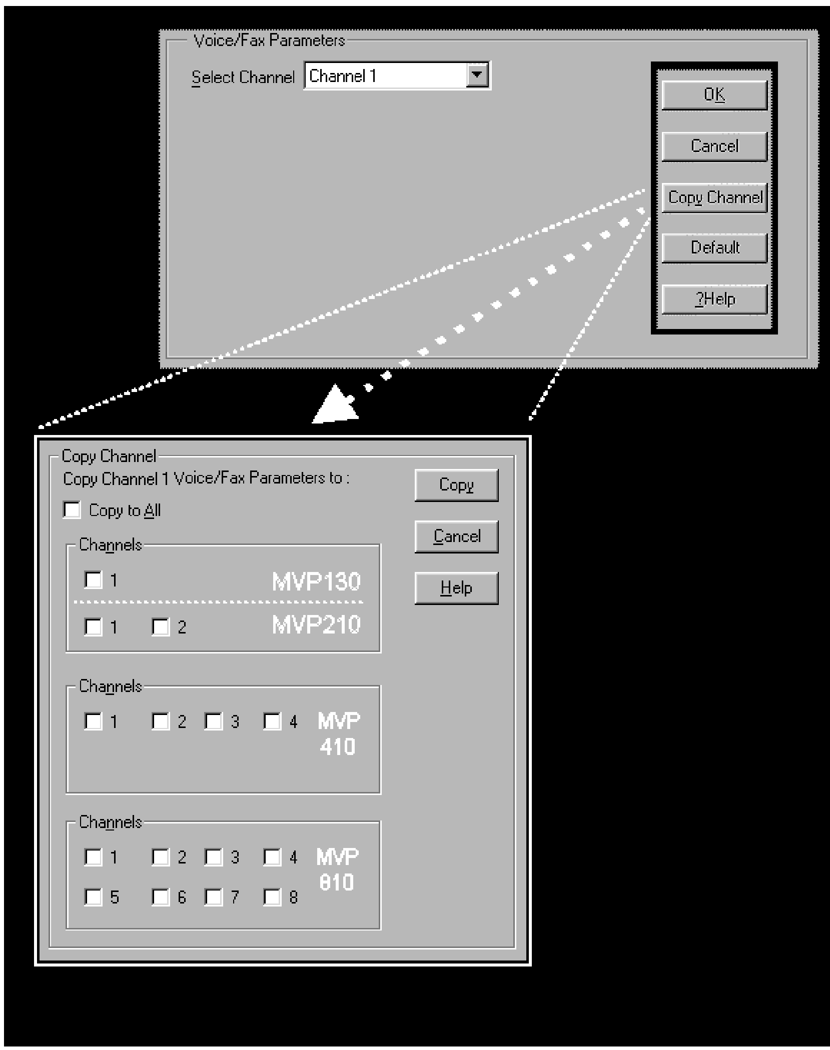
8. **Set Voice/FAX Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

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. *Not applicable to MVP130.*
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 DTMF tones sent out for Touch-Tone dialing.
DTMF Gain, High Tones	+3dB to -31dB & "mute"	Default value: **-4 dB**. Not to be changed except under supervision of MultiTech's Technical Support.
DTMF Gain, Low Tones	+3dB to -31dB & "mute"	Default value: **-7 dB**. Not to be changed except under supervision of MultiTech's Technical Support.

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

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall, Offhook Alert	The **AutoCall** option enables the local MultiVOIP to call a remote MultiVOIP without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote MultiVOIP identified in the **Phone Number** box of this option. If the "Pass Through Enable" field is checked in the Interface Parameters screen, AutoCall must be used. The Offhook Alert option applies only to FXS channels. The **Offhook Alert** option works like this: if a phone goes offhook and yet no number is dialed within a specific period of time (as set in the **Offhook Alert Timer** field), then that phone will automatically dial the Alert phone number for the voip channel. (The Alert phone number must be set in the **Voice/Fax Parameters | Phone Number** field; if the voip system is working without a gatekeeper unit, there must also be a matching phone number entry in the Outbound Phonebook.). One use of this feature would be for emergency use where a user goes off hook but does not dial, possibly indicating a crisis situation. The Offhook Alert feature uses the **Intercept Tone**, as listed in the **Regional Parameters** screen. This tone will be outputted on the phone that was taken off hook but that did not dial. The other end of the connection will hear audio from the "crisis" end as is it would during a normal phone call.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall, Offhook Alert	*(continued from previous page)* Both functions apply on a channel-by-channel basis. It would not be appropriate for either of these functions to be applied to a channel that serves in a pool of available channels for general phone traffic. Either function requires an entry in the Outgoing phonebook of the local MultiVOIP and a matched setting in the Inbound Phonebook of the remote voip.
Generate Local Dial Tone	Y/N	*Used for AutoCall only*. If selected, dial tone will be generated locally while the call is being established between gateways. The capability to generate dial tone locally would be particularly useful when there is a lengthy network delay.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Offhook Alert Timer	0 – 3000 seconds	The length of time that must elapse before the offhook alert is triggered and a call is automatically made to the phone number listed in the **Phone Number** field.
Phone Number	--	Phone number used for Auto Call function or Offhook Alert Timer function. This phone number must correspond to an entry in the Outbound Phonebook of the local MultiVOIP and in the Inbound Phonebook of the remote MultiVOIP (unless a gatekeeper unit is used in the voip system).

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

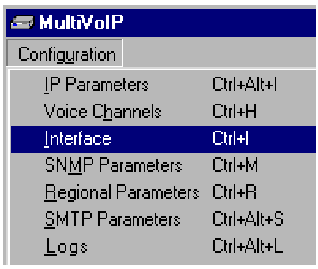
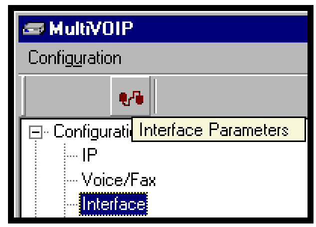
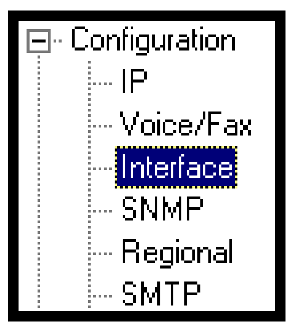
Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dynamic Jitter		
Maximum Jitter Value	60 to 400 ms	The maximum dynamic jitter buffer of **400** milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 msec
Optimizat-ion Factor	0 to 12	The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

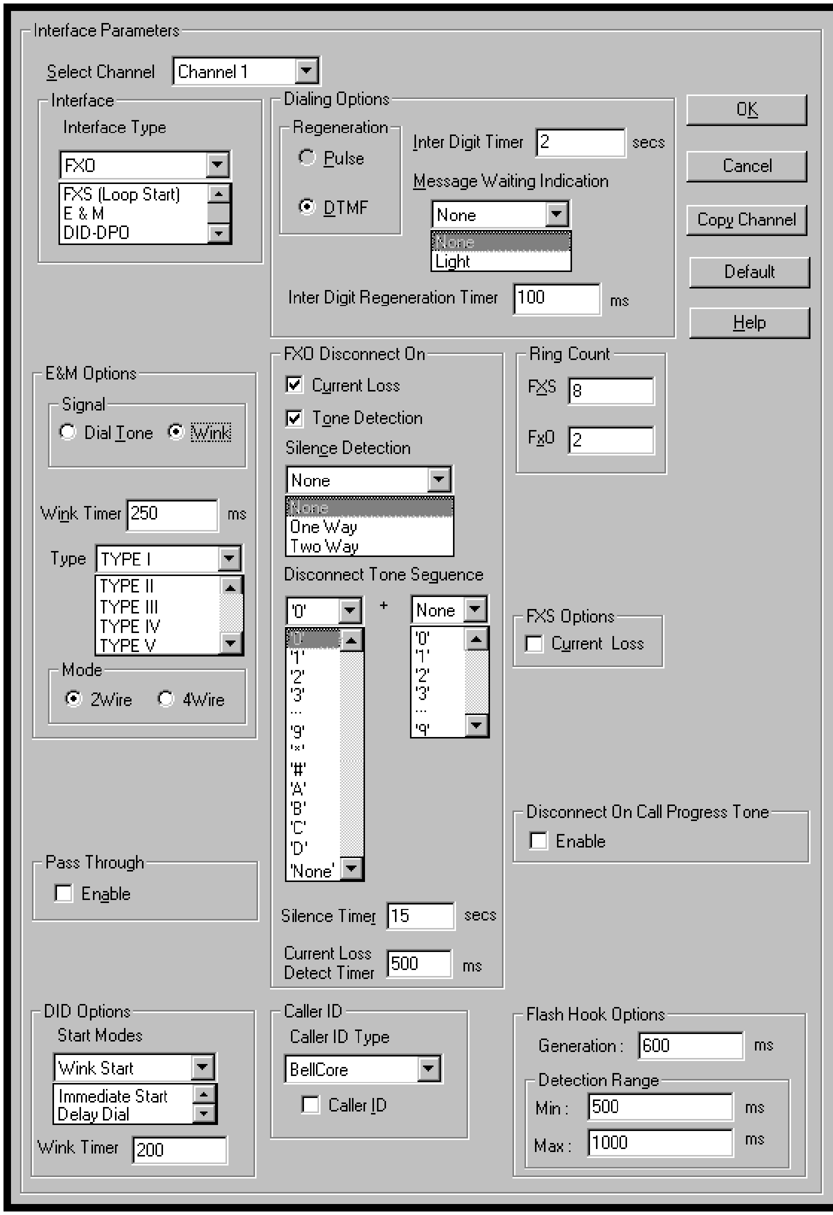
To place modem traffic onto the voip network (an application called "modem relay"), use Coder G.711 mu-law at 64kbps.

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.

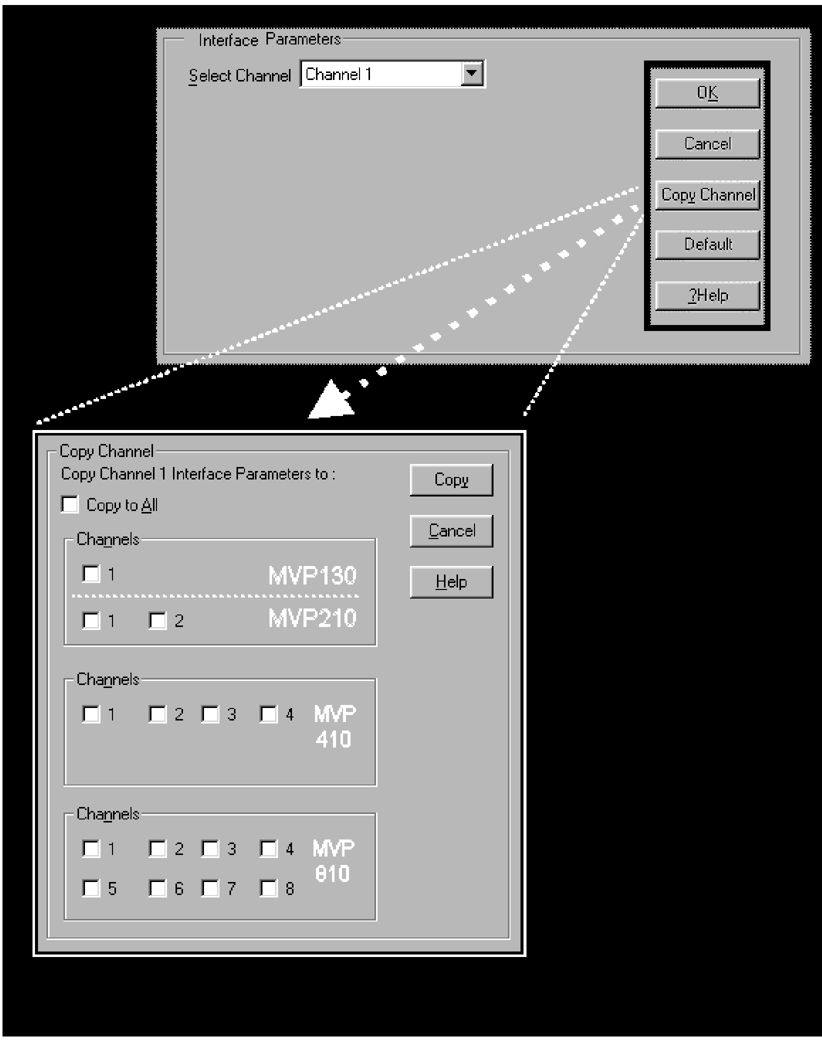
<table>
<tr><th colspan="2">Accessing Telephony Interface Parameters</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td>
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + I</td><td>
</td></tr>
</table>

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

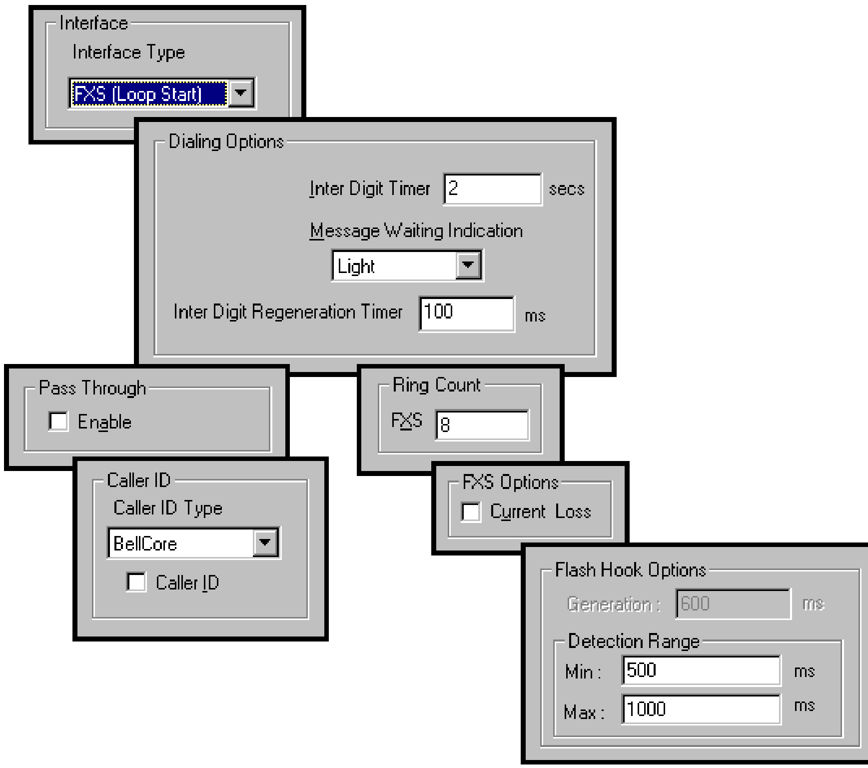


The kinds of parameters for which values must be chosen depend on the type of telephony supervisory signaling or interface used (FXO, E&M, etc.). We present here the various parameters grouped and organized by interface type.

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



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



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

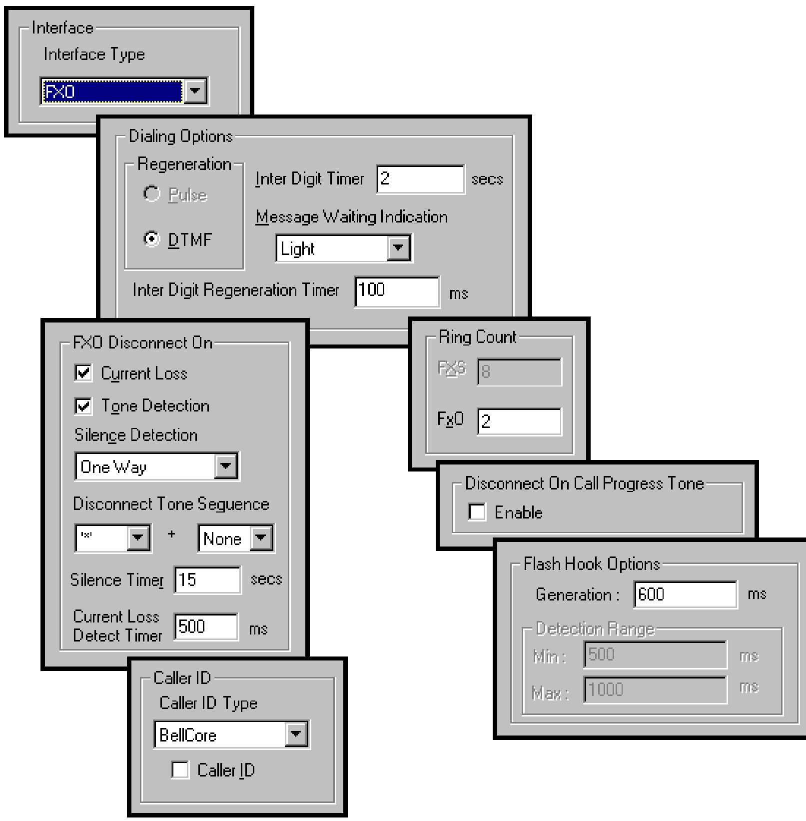
FXS Loop Start Interface: Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dialing Options fields		
Inter Digit Timer	1 - 10 seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the outbound phonebook for the number entered and place the call accordingly. Default = 2.
Message Waiting Indication	--	Not applicable to FXS Loop Start interface
Inter Digit Regeneration Time	in milliseconds	The length of time between the outputting of DTMF digits. Default = 100 ms.
Ring Count, FXS	1-99	Maximum number of rings that the MultiVOIP will issue before giving up the attempted call.
FXS Options, Current Loss	Y/N	When enabled, the MultiVOIP will interrupt loop current in the FXS circuit to initiate a disconnection. This tells the device connected to the FXS port to hang up. The Multi-VOIP cannot drop the call; the FXS device must go on hook.

FXS Loop Start Interface: Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Flash Hook Options fields		
Generation	--	*not applicable to FXS interface*
Detection Range	*for* Min. *and* Max., 50 - 1500 milliseconds	For a received flash hook to be regarded as such by the MultiVOIP, its duration must fall between the minimum and maximum values given here.
Pass Through Enable	Y/N	When enabled, this parameter creates an open audio path through the MultiVOIP. If the Pass-Through feature is enabled, the AutoCall feature must be enabled for this voip channel in the Voice/Fax Parameters screen.
Caller ID fields		
Caller ID Type	Bellcore	The MultiVOIP currently supports only one implementation of Caller ID. That implementation is Bellcore type 1 with Caller ID placed between the first and second rings of the call.
Caller ID enable	Y/N	Caller ID information is a description of the remote calling party received by the called party. The description has three parts: name of caller, phone number of caller, and time of call. The 'time-of-call' portion is always generated by the receiving MultiVOIP unit (on FXS channel) based on its date and time setup.

FXS Loop Start Interface: Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Caller ID fields		
Caller ID enable (cont'd)	Y/N	The forms of the 'Caller Name' and 'Caller Phone Number' differ depending on the IP transmission protocol used (H.323, SIP, or SPP) and upon entries in the phonebook screens of the remote (CID generating) voip unit. The CID Name and Number appearing on the phone at the terminating FXS end will come either from a central office switch (showing a PSTN phone number), or the phonebook of the remote (CID sending) voip unit.

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

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



FXO Interface: Parameter Definitions* *Not applicable to MVP130FXS		
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	1 to 10 seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the phonebook for the number entered. Default = 2.
Message Waiting Indication	--	Not applicable to FXO interface.

FXO Interface: Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dialing Options (cont'd)		
Inter Digit Regeneration Time	50 to 20,000 milliseconds	The length of time between the outputting of DTMF digits. Default = 100 ms.
FXO Disconnect On		There are three possible criteria for disconnection under FXO: current loss, tone detection, and silence detection. Disconnection can be triggered by more than one of the three criteria.
Current Loss	Y/N	Disconnection to be triggered by loss of current. That is, when Current Loss is enabled ("Y"), the MultiVOIP will hang up the call when it detects a loss of current initiated by the attached device.
Current Loss Detect Timer	integer values (in milliseconds)	The minimum time required for detecting the current loss signal on the FXO interface. In other words, this is the minimum length of time the current must be absent to validate 'current loss' as a disconnection criterion. Default = 500 ms.
Tone Detection	Y/N	Disconnection to be triggered by a tone sequence.

<table>
<tr><th colspan="3">FXO Interface: Parameter Definitions (cont'd)</th></tr>
<tr><th>Field Name</th><th>Values</th><th>Description</th></tr>
<tr><td colspan="2">FXO Disconnect On (cont'd)</td><td></td></tr>
<tr><td>Disconnect Tone Sequence</td><td>1st tone pair
+
2nd tone pair</td><td>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.</td></tr>
<tr><td></td><td colspan="2">
<table>
<tr><th colspan="6">DTMF Tone Pairs</th></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td>Low Tones</td></tr>
<tr><td></td><td>1</td><td>2</td><td>3</td><td>A</td><td>697Hz</td></tr>
<tr><td></td><td>4</td><td>5</td><td>6</td><td>B</td><td>770Hz</td></tr>
<tr><td></td><td>7</td><td>8</td><td>9</td><td>C</td><td>852Hz</td></tr>
<tr><td></td><td>*</td><td>0</td><td>#</td><td>D</td><td>941Hz</td></tr>
<tr><td>High Tones</td><td>1209Hz</td><td>1336Hz</td><td>1447Hz</td><td>1633Hz</td><td></td></tr>
</table>
</td></tr>
<tr><td>Silence Detection</td><td>One-Way or Two-Way</td><td>Disconnection to be triggered by silence in one direction only or in both directions simultaneously.</td></tr>
<tr><td>Silence Timer in seconds</td><td>integer value</td><td>Duration of silence required to trigger disconnection.</td></tr>
<tr><td></td><td></td><td></td></tr>
<tr><td>Disconnect on Call Progress Tone</td><td>Y/N</td><td>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.</td></tr>
<tr><td></td><td></td><td></td></tr>
<tr><td>Ring Count, FXO</td><td>1-99</td><td>Number of rings required before the MultiVOIP answers the incoming call.</td></tr>
</table>

FXO Interface: Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Flash Hook Options fields		
Generation	50 - 1500 milliseconds	Length of flash hook that will be generated and sent out when the remote end initiates a flash hook and it is regenerated locally. Default = 600 ms.
Detection Range	--	Not applicable to FXO.
Caller ID fields		
Caller ID Type	Bellcore	The MultiVOIP currently supports only one implementation of Caller ID. That implementation is Bellcore type 1 with caller ID placed between the first and second rings of the call.
Caller ID enable	Y/N	Caller ID information is a description of the remote calling party received by the called party. The description has three parts: name of caller, phone number of caller, and time of call. The 'time-of-call' portion is always generated by the receiving MultiVOIP unit (on FXS channel) based on its date and time setup. The forms of the 'Caller Name' and 'Caller Phone Number' differ depending on the IP transmission protocol used (H.323, SIP, or SPP) and upon entries in the phonebook screens of the remote (CID generating) voip unit. The CID Name and Number appearing on the phone at the terminating FXS end will come either from a central office switch (showing a PSTN phone number), or the phonebook of the remote (CID sending) voip unit.

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

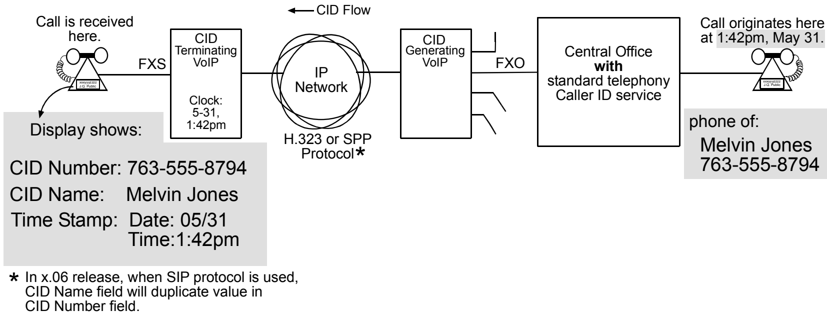


Figure 6-1: Voip Caller ID Case #1 – Call, through telco central office *with* standard CID, enters voip system

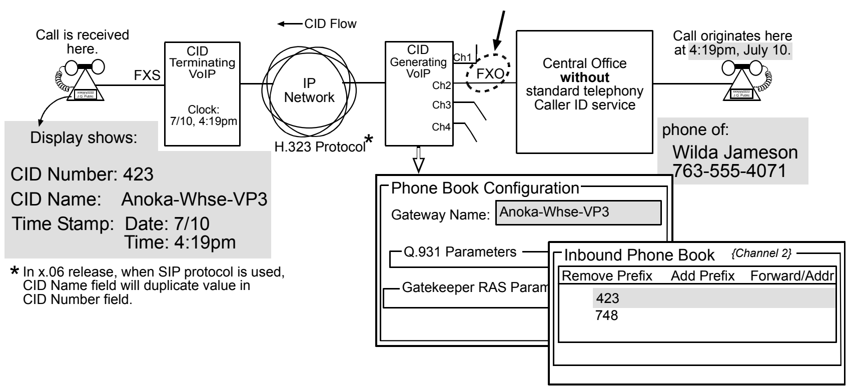


Figure 6-2: Voip Caller ID Case #2 – Call, through telco central office *without* standard CID, enters H.323 voip system

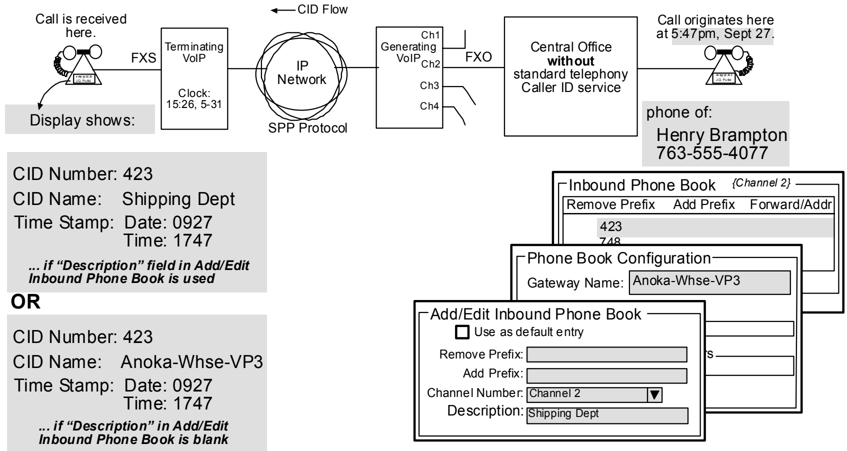


Figure 6-3: Voip Caller ID Case #3 – Call, through telco central office *without* standard CID, enters SPP voip system

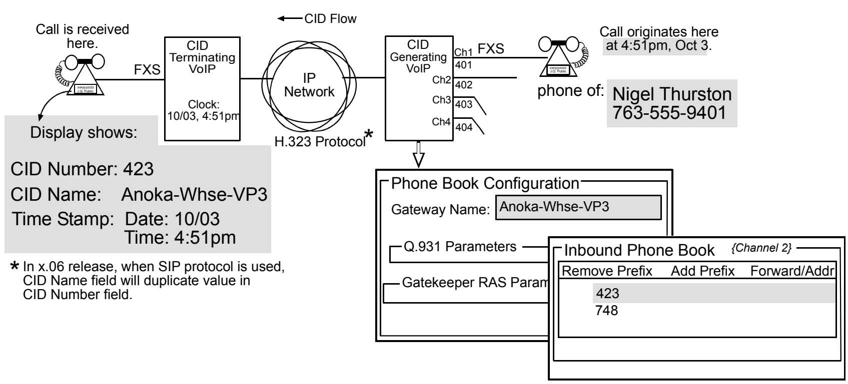


Figure 6-4: Voip Caller ID Case #4 – Remote FXS call on H.323 voip system

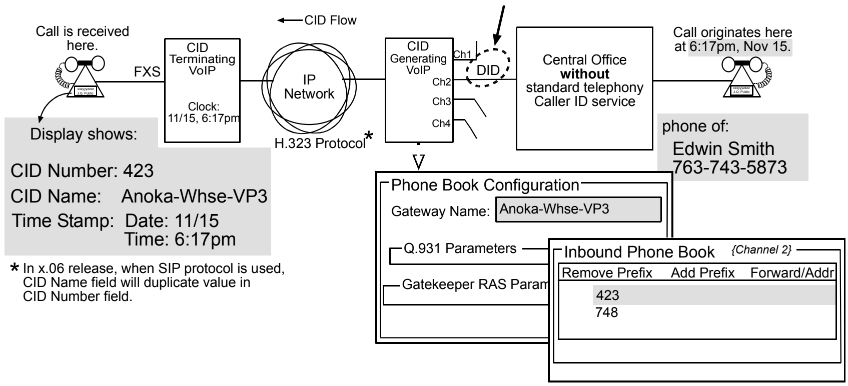
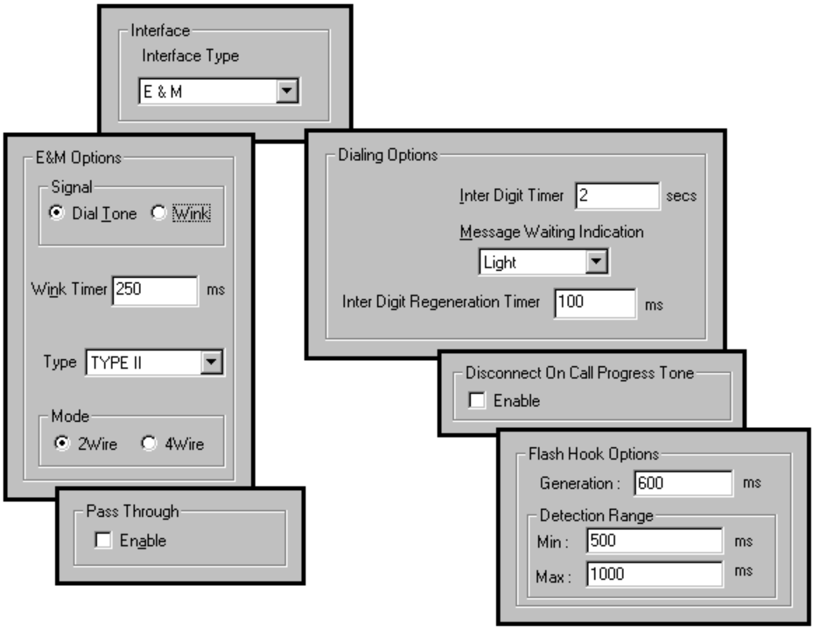


Figure 6-5: Voip Caller ID Case #5 – Call through telco central office without standard CID enters DID channel in H.323 voip system

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

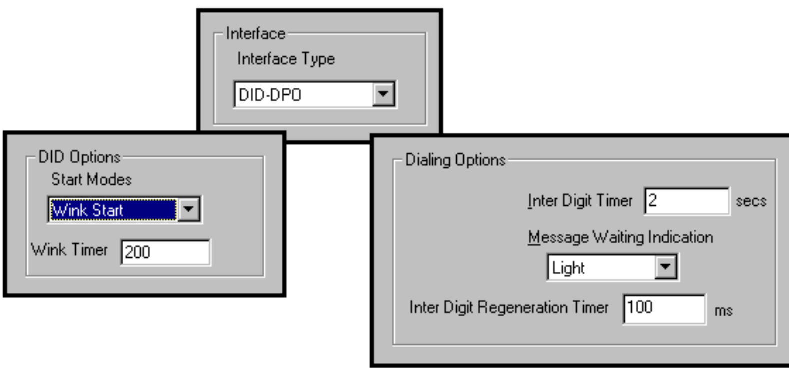


E&M Interface Parameter Definitions **Not applicable to MVP130FXS*		
Field Name	**Values**	**Description**
Interface	E&M	enables E&M functionality
Type	Types 1-5.	Refers to the type of E&M interface being used.
Mode	2-wire or 4-wire	Each E&M interface type can be either 2-wire or 4-wire audio.
Signal	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 Enable	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.

E&M Interface Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dialing Options		
Inter Digit Timer	integer values, in seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the phonebook for the number entered. Default = 2.
Message Waiting Indication	Light or None	Allows MultiVOIP to pass mode-code sequences between Avaya Magix PBXs to turn on and off the message-waiting light on a PBX extension phone. Mode codes: *53 + *PBX extension* ➔ turns message light on. #53 + *PBX extension* ➔ turns message light off. Signals to turn message-waiting lights on/off are not sent to phones connected directly to the MultiVOIP on FXS channels, not to other non-Avaya Magix PBX phone stations on the voip network.
Inter Digit Regeneration Timer	milliseconds	The length of time between the outputting of DTMF digits. Default = 100 ms.

E&M Interface Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dialing Options (cont'd)		
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.
Flash Hook Options fields		
Generation	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.
Detection Range	*for Min. and Max.*, 50 1500 milliseconds	For a received flash hook to be regarded as such by the MultiVOIP, its duration must fall between the minimum and maximum values given here.

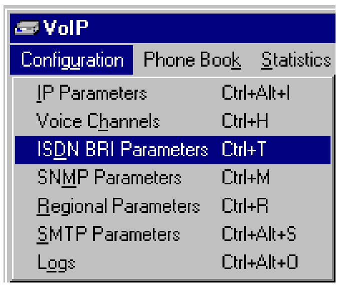
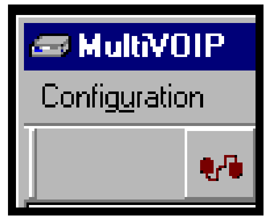
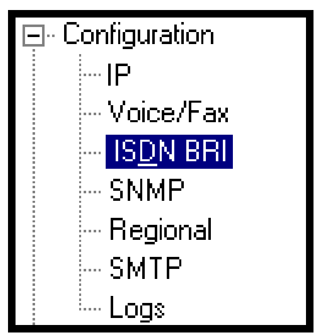
DID Parameters. The parameters applicable to the Direct Inward Dial (DID) telephony interface type are shown in the figure below and described in the table that follows. The DID interface allows one phone line to direct incoming calls to any one of several extensions without a switchboard operator. Of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



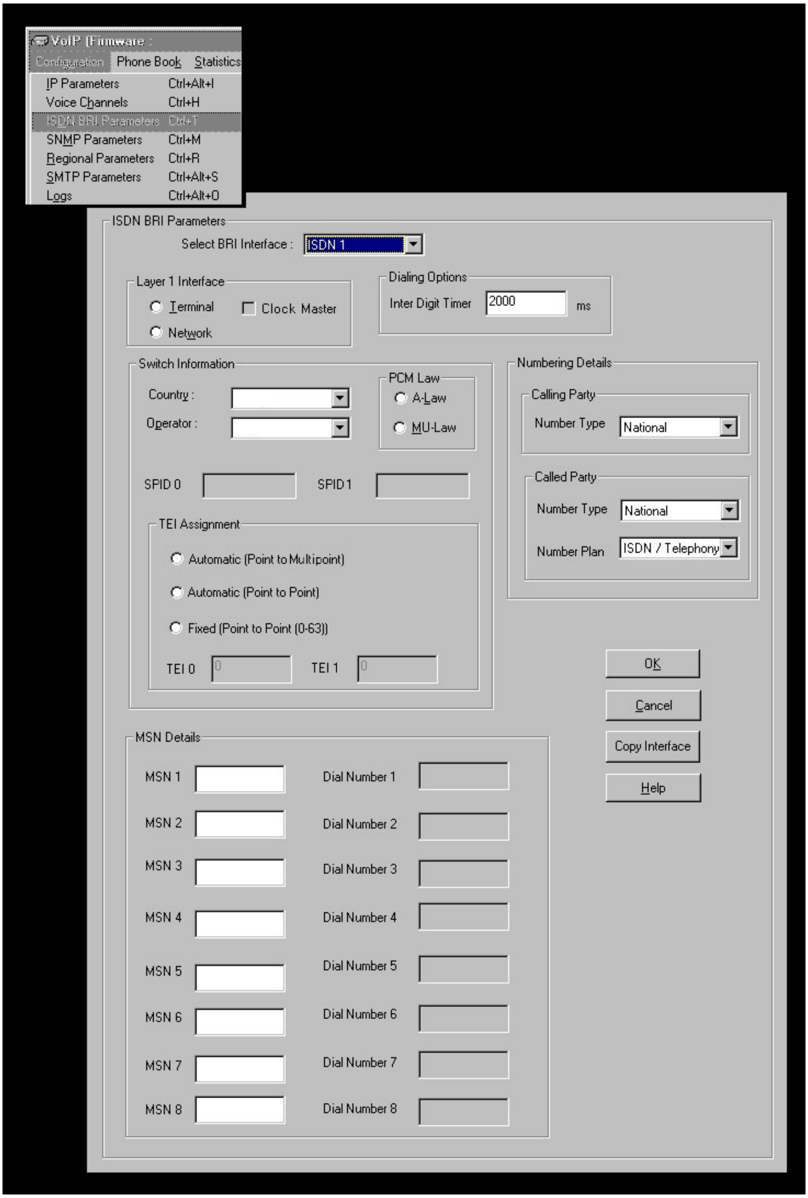
DID Interface Parameter Definitions **Not applicable to MVP130FXS*		
Field Name	**Values**	**Description**
Interface	DID-DPO	Enables the customer-premises side of DID functionality
DID Options		MultiVOIP's use of DID applies only for incoming DID calls. The Start Mode used by the MultiVOIP must match that used by the originating telephony equipment, else DID calls cannot be completed.
Start Modes	Immediate Start, Wink Start, Delay Dial	For **Immediate Start**, the voip detects the off-hook condition initiated by the telco central-office call and becomes ready to receive dial digits immediately.

DID Interface Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
DID Options (cont'd)		
Start Modes	Immediate Start, Wink Start, Delay Dial	For **Wink Start**, the voip detects the off-hook condition. Then the voip reverses battery polarity for a specified time (140-290 ms; a "wink") and then becomes ready to receive dial digits. For **Delay Dial**, the voip detects detects the off-hook condition. Then the voip reverses battery polarity for a specified time (reverse polarity duration has wider acceptable range than for Wink Start) and then becomes ready to receive dial digits.
Wink Timer (in ms)	integer values, in milliseconds	This is the length of the wink for Wink Start and Delay Dial signaling modes.. Applicable only when **Start Mode** parameter is set to "Wink Start" or "Delay Dial."
Dialing Options		
Inter Digit Timer	integer values, in seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the phonebook for the number entered. Default = 2.
Message Waiting Indication	--	Not applicable to DID-DPO interface.
Inter-Digit Regeneration Timer	integer values, in milliseconds	This parameter is applicable when digits are dialed onto a DID-DPO channel after the connection has been made. The length of time between the outputting of DTMF digits. Default = 100 ms.

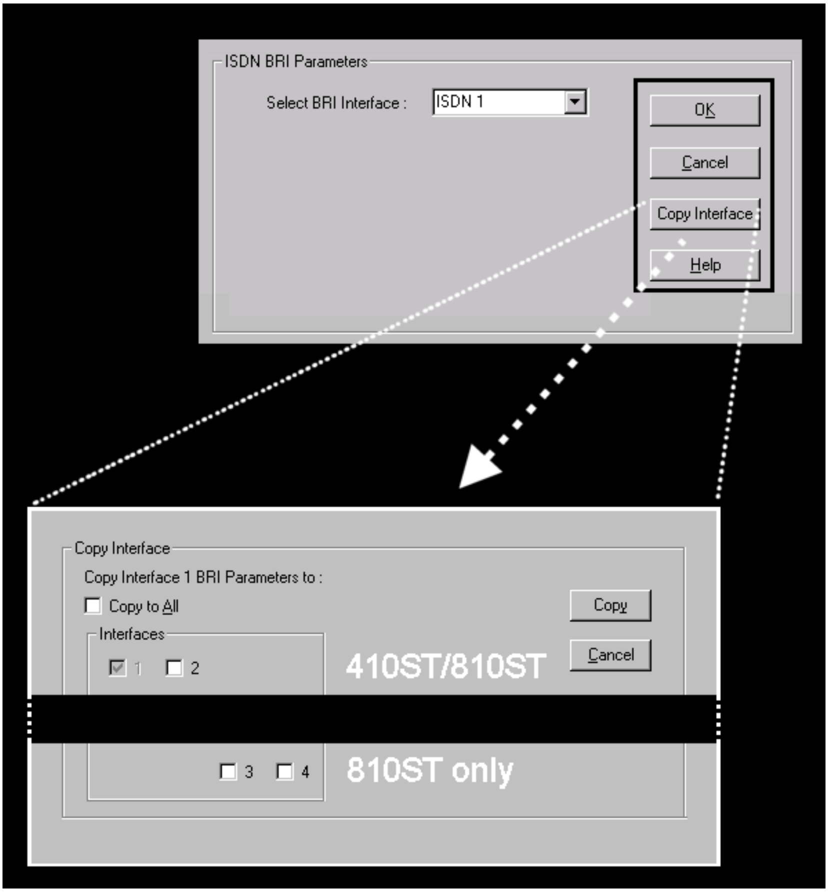
9b. (**for ISDN-BRI MultiVOIP units**). **Set ISDN Parameters**. This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

Accessing ISDN (BRI) Parameters	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + T`	

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**.



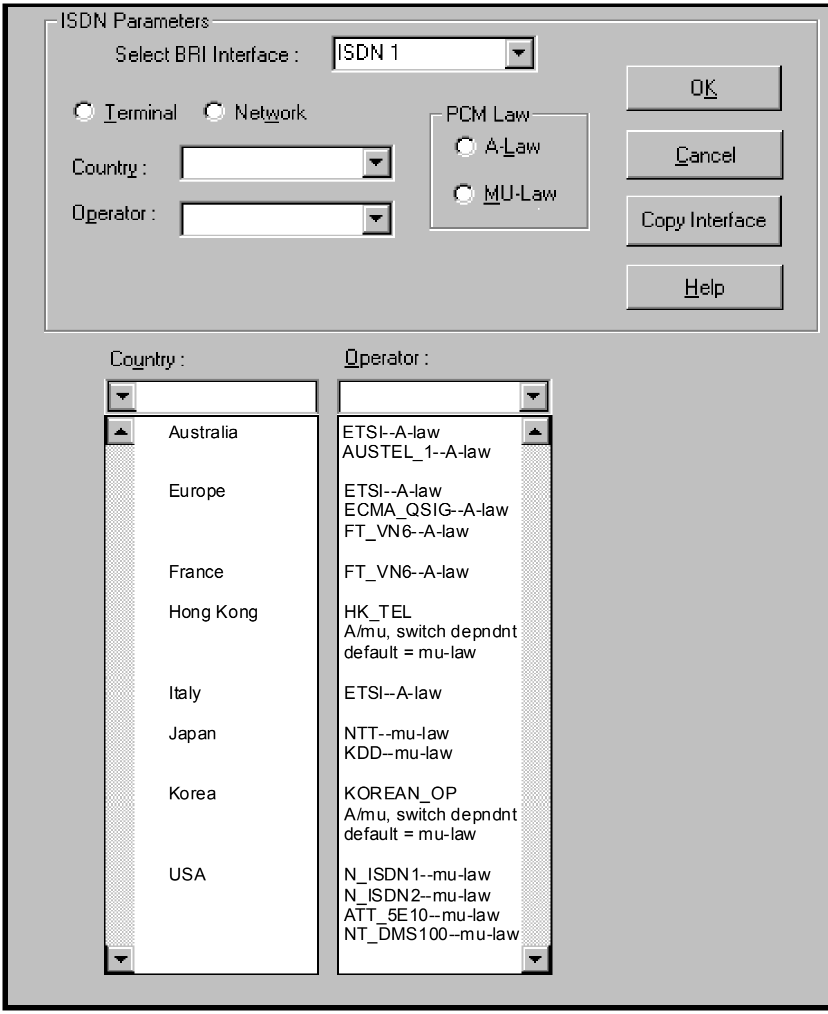
ISDN-BRI Parameter Definitions		
Field Name	**Values**	**Description**
Select BRI Interface	ISDN*n* for n= 1-2 (410ST) for n=1-4 (810ST)	In this field, you will choose which ISDN port you are configuring. The 410ST has two ISDN –BRI ports (or "interfaces"); the 810ST has four ISDN-BRI ports (or "interfaces"). Each port has two channels.
Layer 1 Interface	Terminal, Network, Clock Master	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. ISDN 1 Terminal Clock Master – The BRI VOIP uses an internal crystal as the clock master when any one of the ports is set for Network mode. If all ports are set for Terminal mode, then the first port to be activated becomes the clock master and it gets its clock from the network. The clock master can only be activated on ISDN1 interface in Terminal mode. If ISDN1 is selected and any other interface is set to Network mode, then clock master can be enabled. If enabled, ISDN1 is set as the clock master for all time, whether or not it is activated. This feature enables the BRI VOIP to have all of its interfaces synchronized to other network equipment and allows the terminal devices connected to the VOIP to be synchronized to the network equipment as well.

ISDN-BRI Parameter Definitions (continued)		
Field Name	**Values**	**Description**
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).
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 America standard. Refer to the PCM-Law defaults based on country and operator below.
TEI *n* Assignment	Terminal or Network or both Automatic (Point-to-Point) Network – Automatic (Point-to-Multipoint) Fixed (Point-to-Point (0-63) TEI 0 & TEI 1 active	TEI (Terminal Endpoint Identifier) is a number to uniquely identify each device connected to the ISDN. TEI Assignment displays the value for each TEI assigned to the BRI port. Depending on the layer 1 interface selection (Terminal or Network) and the country selection, some fields are grayed out (inactive) as they have no meaning for this configuration. The TEI range is zero to 63 for Fixed and 64 to 126 for Automatic assignment. An interface in Network mode has the added option of Point-to-Multipoint Automatic TEI. This added option should be used in cases where the interface, or BRI port, has one or more terminal devices connected to it. This option may also help resolve some problems that occur when set for Point-to-Point Automatic TEI, even with only one terminal device connected to the interface. Point-to-Point Automatic TEI is then to be used when there is one network device and one terminal device, default setting. When selecting Fixed TEI, the connection must be Point-to-Point, one network and one terminal device.

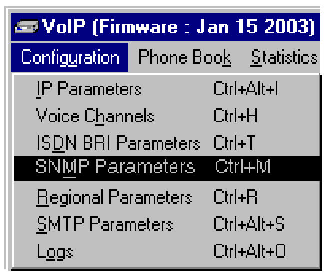
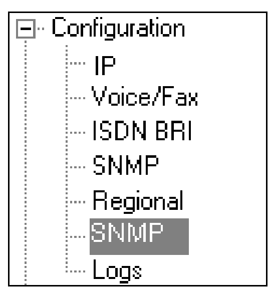
ISDN-BRI Parameter Definitions (continued)		
Field Name	**Values**	**Description**
Numbering Details		
Calling Party	**Number Type:** Unknown, International, National, Net_Spf, Subscriber, Local, Abbreviated	Support for the user to select the Calling Party Number Type. Local is valid only for AT5 operator – local (directory) number. There may be cases where the default Type of Number and/or Numbering Plan may cause conflicts with the network or terminal device to which the VOIP is attached. Should this be the case, then changing to a different Type of Number might resolve the problem.
Called Party	**Number Type:** Unknown, International, National, Net_Spf, Subscriber, Local, Abbreviated **Number Plan:** Unknown, ISDN, Telephone, Data, Telex, National, Private, Reserved	Support for the user to select the Calling Party Number Type. Local is valid only for AT5 operator – local (directory) number. There may be cases where the default Type of Number and/or Numbering Plan may cause conflicts with the network or terminal device to which the VOIP is attached. Should this be the case, then changing to a different Type of Number might resolve the problem. Support for the user to select the Called Numbering Plan and Number Type. ISDN/telephony - CCITT E.164/E.163, Telephony – not in CEPT, Data – CCITT x.121, Telex – CCITT F.69, National – Standard, Reserved for extension. There may be cases where the default Type of Number and/or Numbering Plan may cause conflicts with the network or terminal device to which the VOIP is attached. Should this be the case, then changing to a different Type of Number might resolve the problem. Typically, setting the Called Party Type of Number to “unknown” and the Number Plan to “Unknown” will resolve such issues. These problems normally occur for calls going to the ISDN from the VOIP (numbers in the Inbound Phone Book)

ISDN-BRI Parameter Definitions (continued)		
Field Name	**Values**	**Description**
MSN Details		
MSN n	Right most digits to be distinguished	Multiple Subscriber Numbering (MSN) – In Euro-ISDN, and some country specific variants, it is possible to have several ISDN numbers for the same BRI or PRI connection. This feature allows you to assign different ISDN numbers to different applications, or different physical equipment, e.g., modem, phone instrument, fax machine, etc. When specifying MSN Numbers, don't enter the whole number, but only enough of the right most digits so that the numbers allocated to the line can be distinguished. Right now, a maximum of eight MSN Numbers is supported. For all ISDN incoming calls, when the dialed digits are matching with the configured MSN number, then, if the Dial Number corresponding to the matched MSN number is present, then using that Dial Number an IP outgoing call is made. Otherwise, for the matched MSN number, if no Dial Number is present, dial tone is provided to the user to dial the actual digits with which to make the IP outgoing call.
"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. MSN details are not copied due to the nature of MSN assignment.

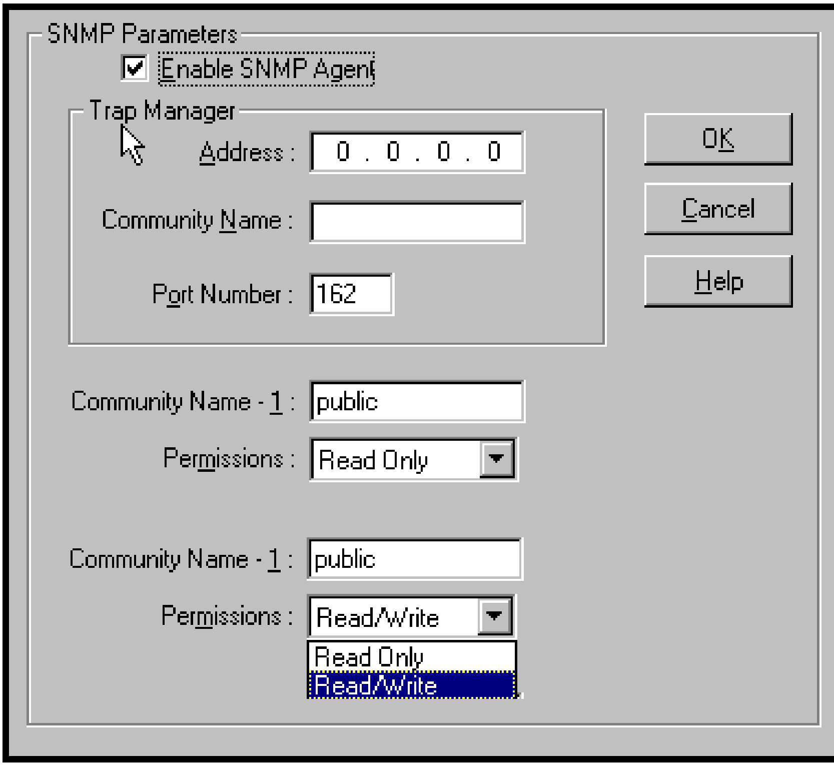
Country and Operator options for the MVP-410ST/810ST voip units are listed below.



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><code>Ctrl + M</code></td><td>
</td></tr>
</table>

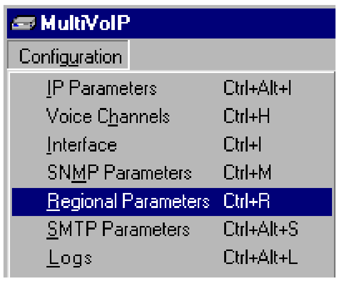
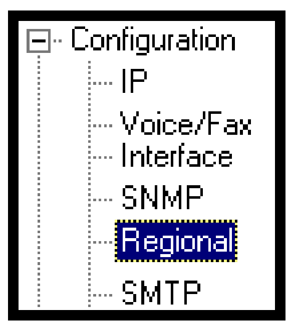
In each field, enter the values that fit your particular system.



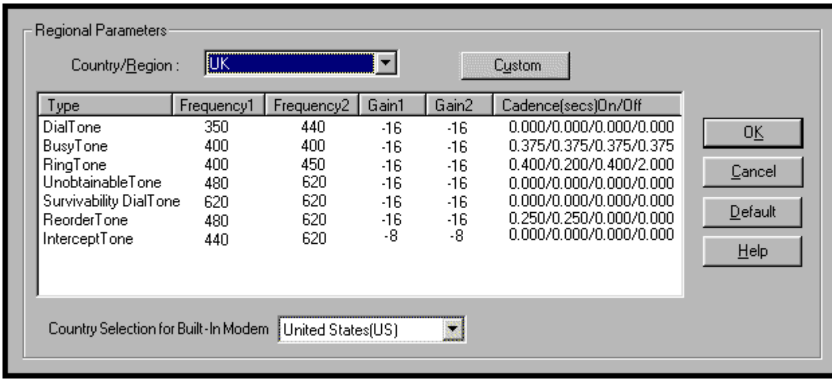
The SNMP Parameter fields are described in the table below.

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.

11. **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.



Remote Configuration/Command Modem. Each MVP410 and MVP810 MultiVOIP unit contains a built-in modem. This modem allows the MultiVOIP to be configured remotely when a standard POTS line is connected to the "Command Modem" connector on the back panel of the MultiVOIP. In the **Country Selection for Built-In Modem** field (drop-down list), select the country that best fits your situation. This may not be the same as your selection for the **Country/Region** field. The selections in the **Country Selection for Built-In Modem** field entail more detailed groupings of telephony parameters than do the **Country/Region** values.

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

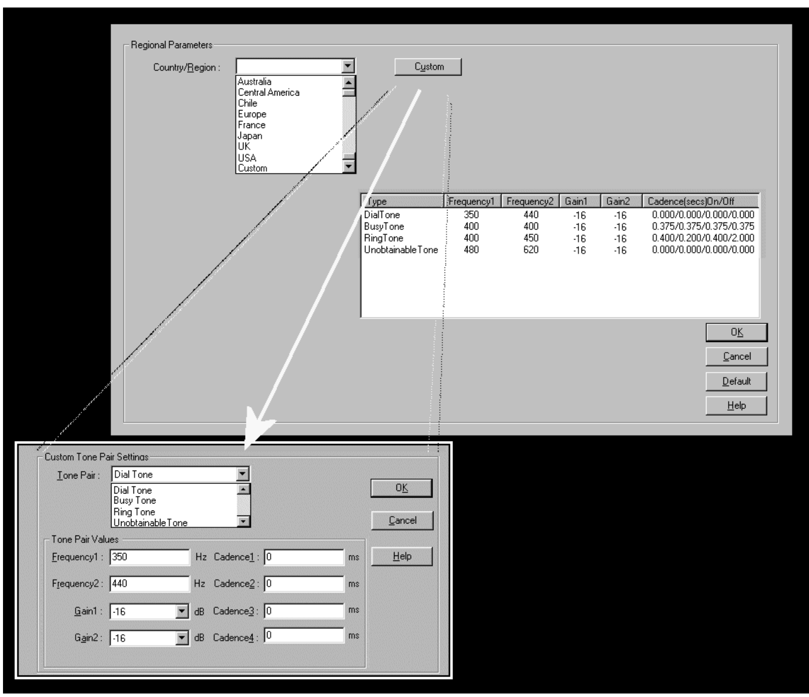
The **Regional Parameters** fields are described in the table below.

"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), **survivability tone** (tone heard briefly, 2 seconds, after going offhook denoting survivable mode of VOIP unit), **re-order tone** (a tone pattern indicating the need for the user to hang up the phone), and **intercept tone** (a tone that warns an a party that has gone off hook but has not begun dialing, within a prescribed time, that an automatic emergency or attendant number will be called; the automatic call can be used to direct an attendant's attention to a disabled or distressed caller, allowing an appropriate response to be made). In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The "Custom" option (button) assures that any tone-pairing scheme worldwide can be accommodated. **Note**: **Intercept tone** is applicable only when the FXS telephony interface has been chosen in the **Interface** screen and when the AutoCall / OffHook Alert field is set to OffHook Alert in the **Voice/Fax Parameters** screen. The time allowed for dialing before the automatic calling process begins is set in the Offhook Alert Timer field of the **Voice/Fax Parameters** screen.

"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.
Country Selection for Built-In Modem (*not applicable to MVP-130/130FXS MVP210, MVP410ST, or MVP810ST*)	country name	MultiVOIP units operating with the X.06 software release (and above) include a built-in modem. The administrator can dial into this modem to configure the MultiVOIP unit remotely. The country name values in this field set telephony parameters that allow the modem to work in the listed country. This value may be different than the Country/Region value. For example, a user may need to choose "Europe" as the Country/Region value but "Denmark" as the Country-Selection-for-Built-In-Modem value.

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) 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.)

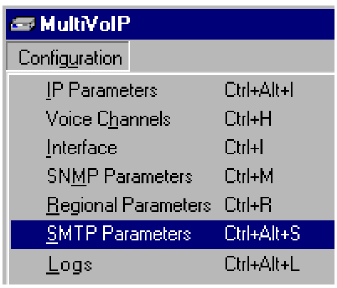
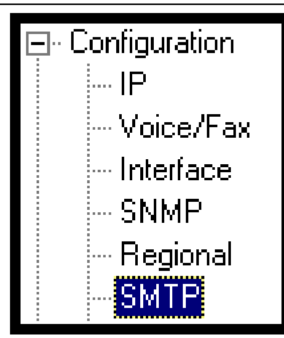


The **Custom Tone-Pair Settings** fields are described in the table below.

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**

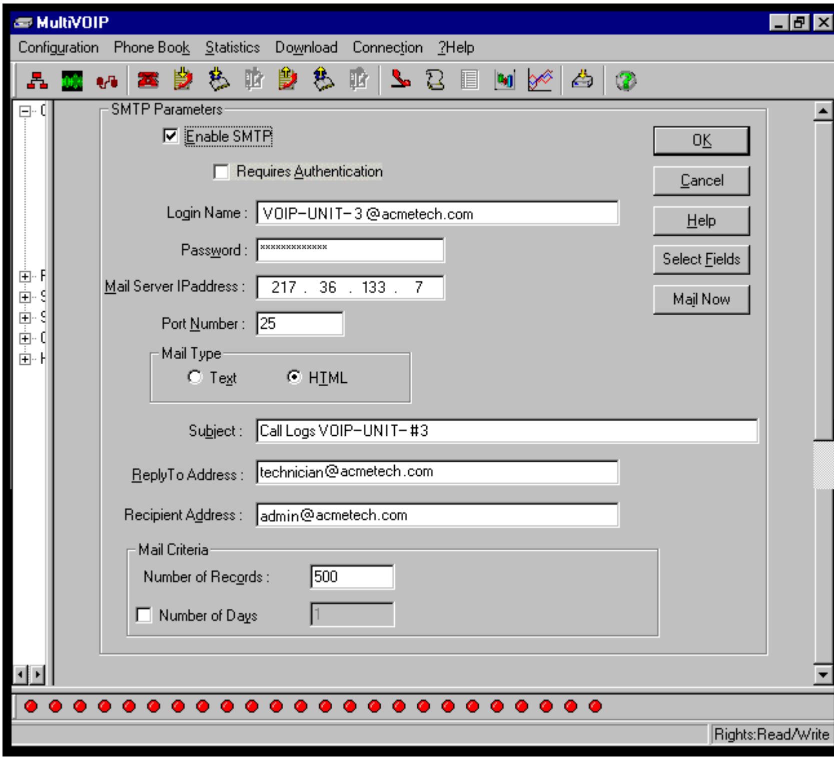
Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Cadence 1	integer time value in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, dial tone ("0" indicates continuous tone) survivability 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.

Accessing "SMTP Parameters"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + Alt + S`	

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

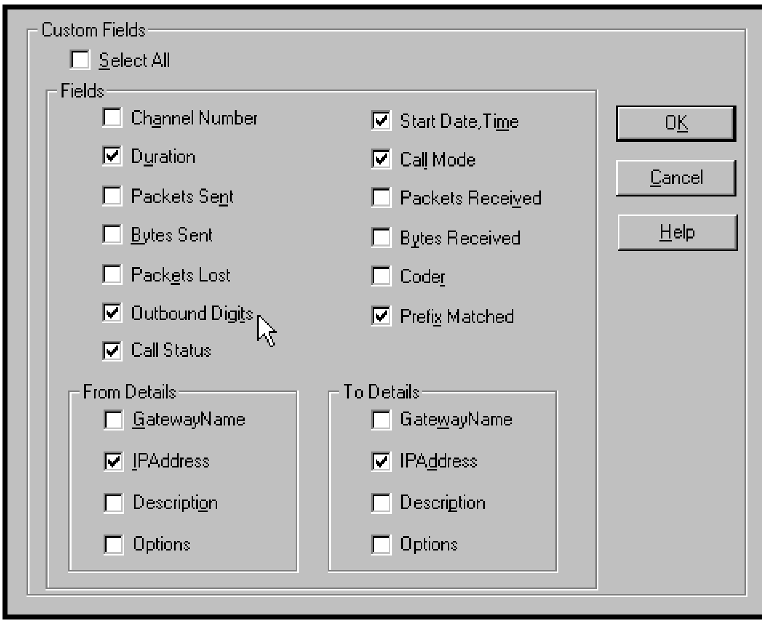


"SMTP Parameters" Definitions		
Field Name	**Values**	**Description**
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select "SMTP" in the **Logs** screen.
Requires Authentication	Y/N	If this checkbox is checked, the MultiVOIP will send Authentication information to the SMTP server. The authentication information indicates whether or not the email sender has permission to use the SMTP server.
Login Name	alpha-numeric, per email domain	This is the User Name for the MultiVOIP unit's email account.

.

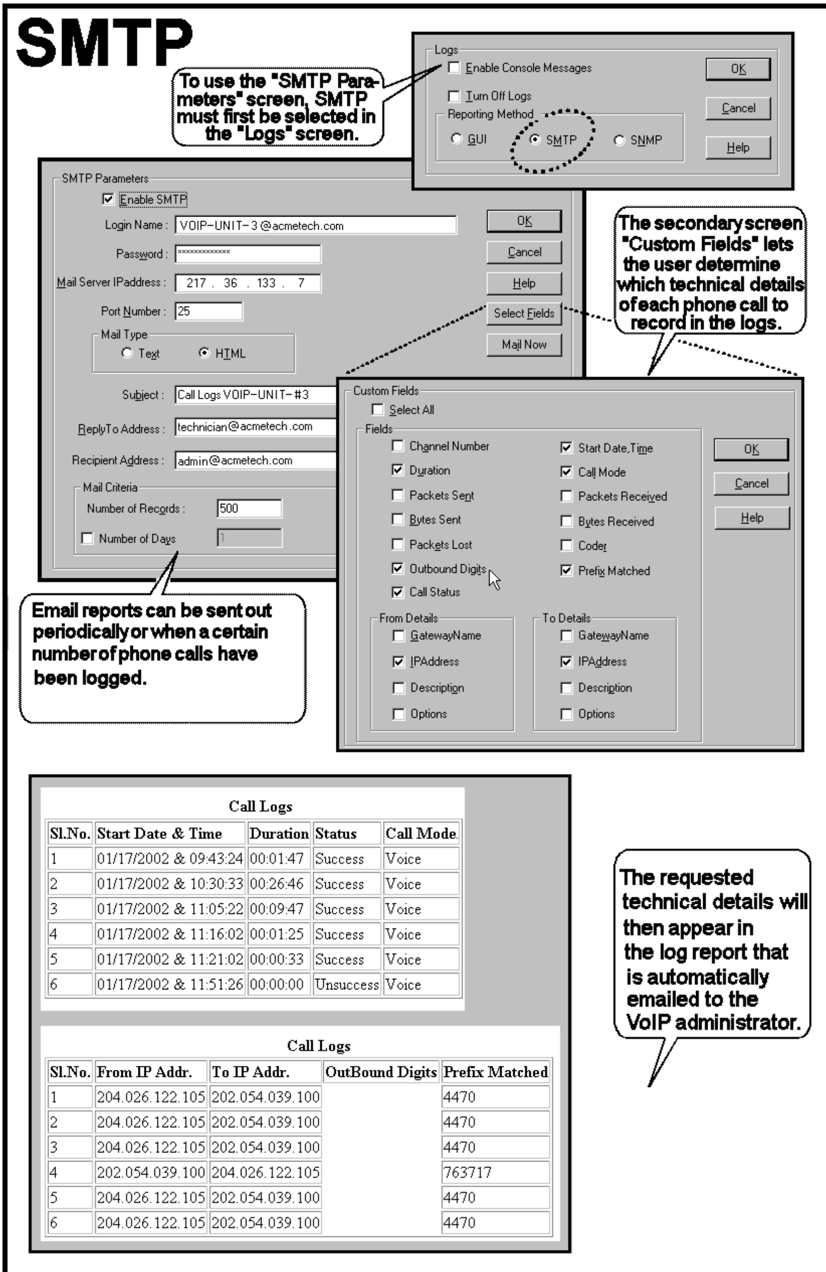
"SMTP Parameters" Definitions (cont'd)		
Field Name	**Values**	**Description**
Password	alpha-numeric	Login password for MultiVOIP unit's email account.
Mail Server IP Address	n.n.n.n for n= 0 to 255	This is the mail server's IP address. This mail server must be accessible on the IP network to which the MultiVOIP is connected.
Port Number	25	25 is a standard port number for SMTP.
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

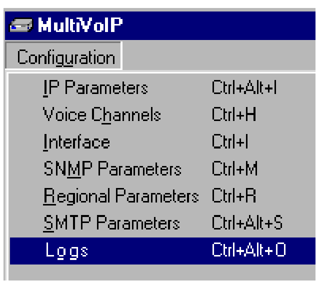
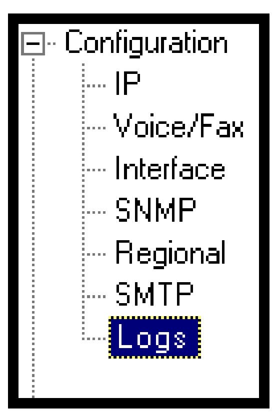
Sl.No.	From IP Addr.	To IP Addr.	OutBound Digits	Prefix Matched
1	204.026.122.105	202.054.039.100		4470
2	204.026.122.105	202.054.039.100		4470
3	204.026.122.105	202.054.039.100		4470
4	202.054.039.100	204.026.122.105		763717
5	204.026.122.105	202.054.039.100		4470
6	204.026.122.105	202.054.039.100	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

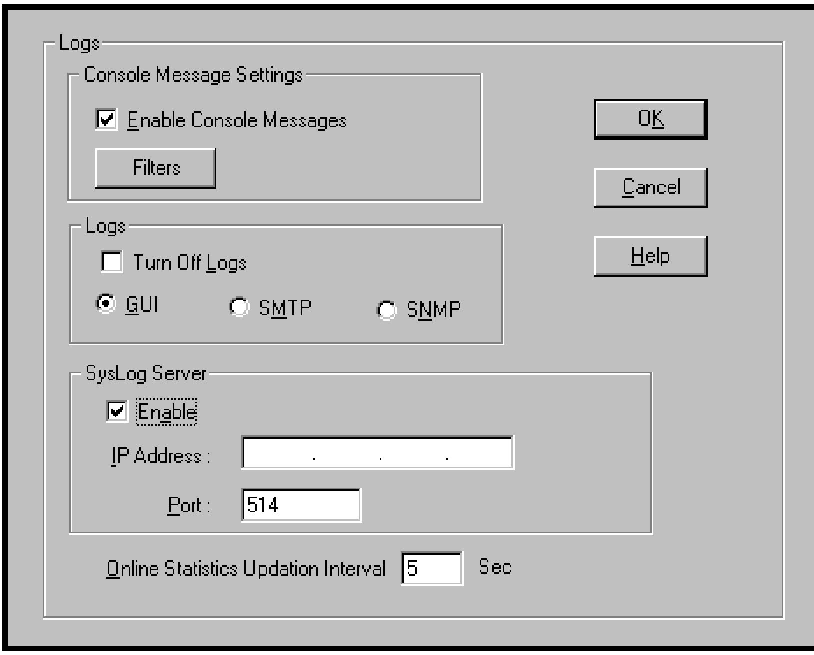
    A. in the MultiVOIP program (GUI),

    B. via email (SMTP), or

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

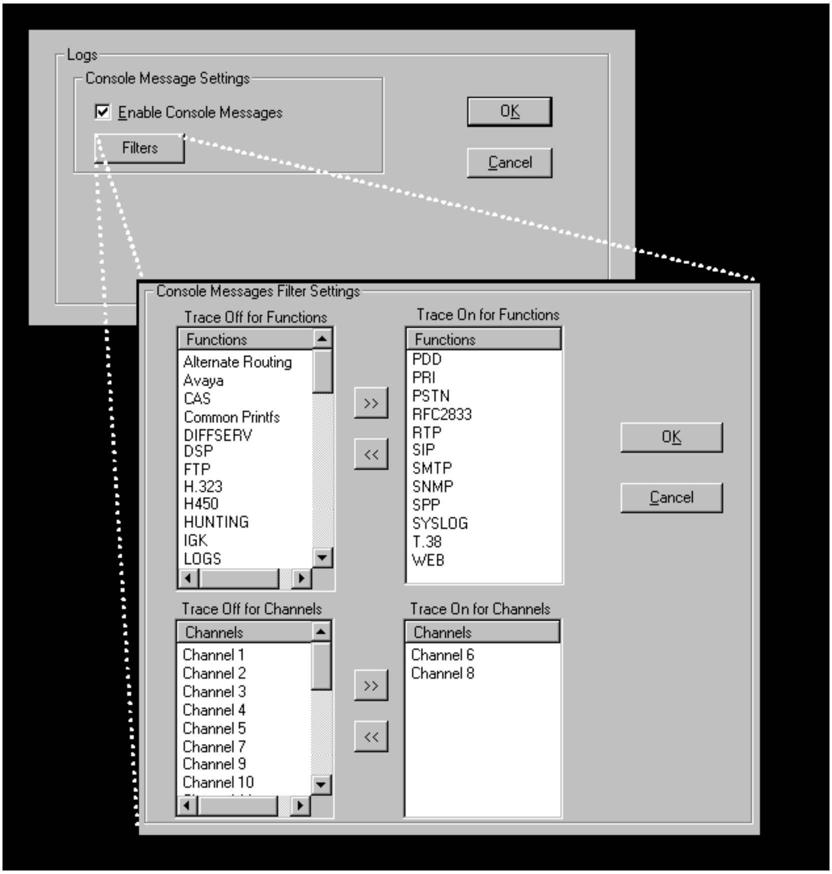
Accessing "Logs" Screen	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + Alt + O`	

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).

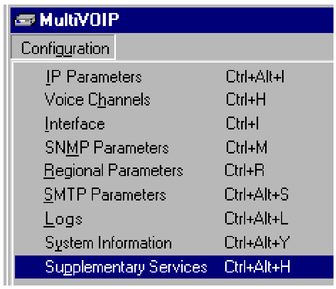
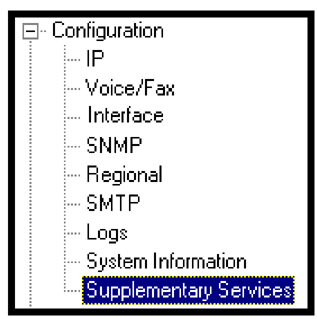


“Logs” Screen Definitions		
Field Name	**Values**	**Description**
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic terminal program like HyperTerminal ™ or equivalent. Normally, this should be disabled because it uses MultiVOIP processing resources. Console messages 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.)
Turn Off Logs	Y/N	Check to disable log-reporting function.
Logs Buttons		Only one of these three log reporting methods, GUI, SMTP, or SNMP, may be chosen.
GUI	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.

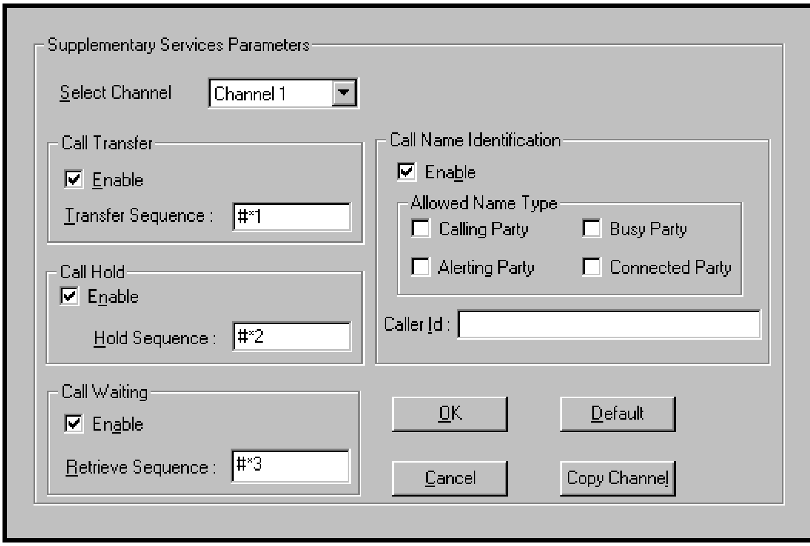


15. **Set Supplementary Services Parameters.** This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

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**.

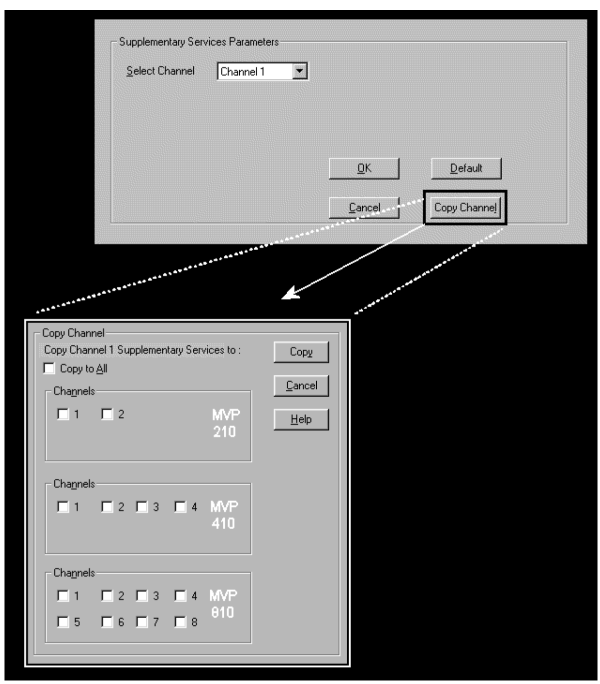
Call Transfer. Call Transfer allows one party to re-connect the party with whom they have been speaking to a third party. The first party is disconnected when the third party becomes connected. Feature is invoked by a programmable phone keypad sequence (for example, #7).

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 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		
Field Name	**Values**	**Description**
Select Channel	1 (MVP-130/ 130FXS) 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. A brief musical jingle is played for the caller on hold.
Transfer Sequence	any phone keypad character	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call transfer. The call-transfer sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). The sequences for call transfer, call hold, and call waiting can be from 1 to 4 digits in length consisting of any combination of digits 1234567890*#.

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.

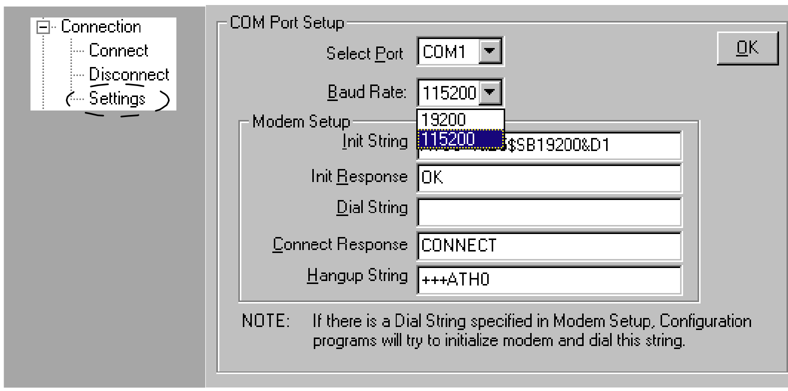
Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Alerting Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving the call and **Alerting Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit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

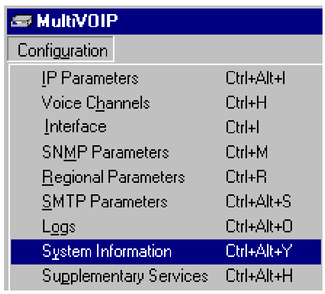
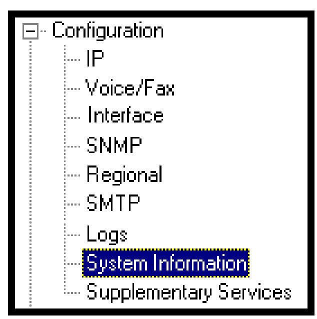


First, it is important to note that the default COM port established by the MultiVOIP program is COM1. ***Do not accept the default value until you have checked the COM port allocation on your PC***. To do this, check for COM port assignments in the system resource dialog box(es) of your Windows operating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

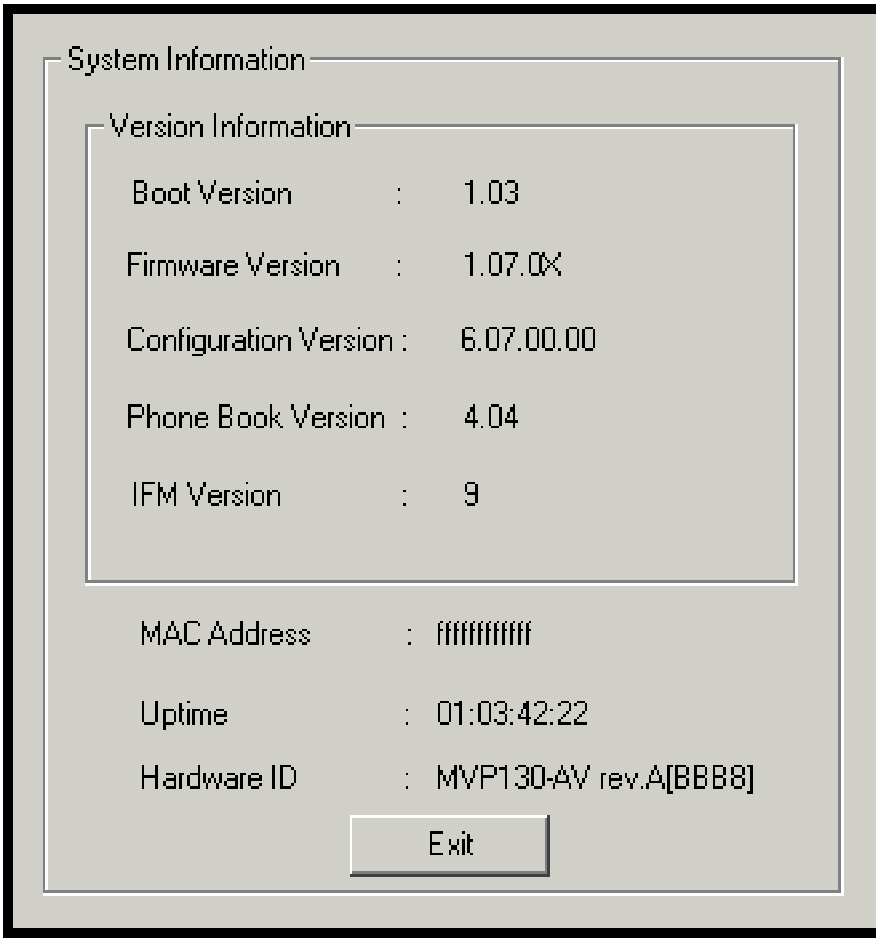
The default baud rate is 115,200 bps.

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

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

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

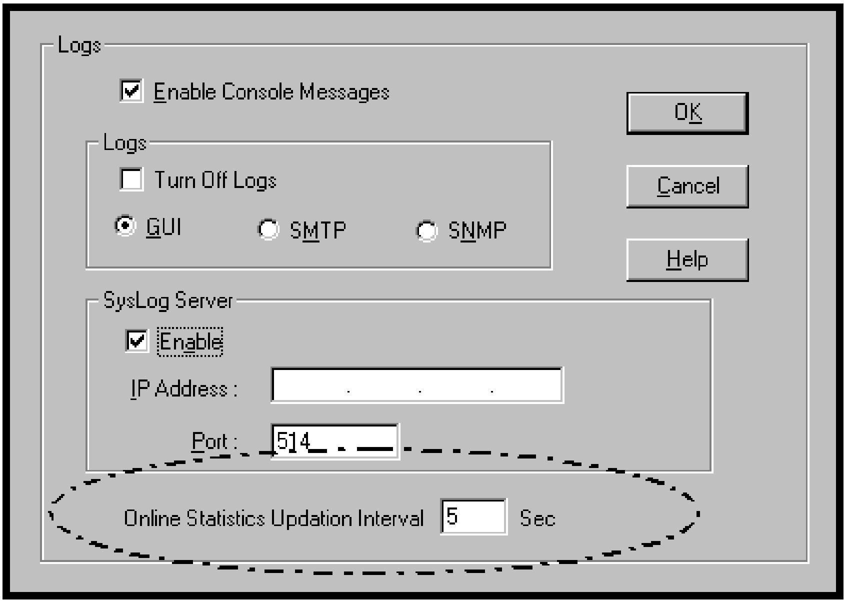
This screen presents vital system information at a glance. Its primary use is in troubleshooting.



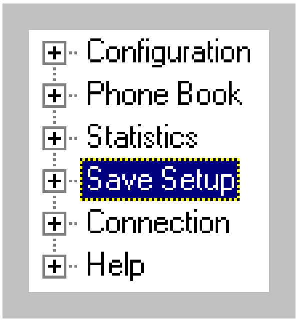
System Information Parameter Definitions		
Field Name	**Values**	**Description**
Boot Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Firmware Version	alpha-numeric	Indicates version of MultiVOIP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

System Information Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Configur-ation Version	nn.nn.nn.nn alpha-numeric	Indicates version of MultiVOIP Configuration software (which includes screens for IP Parameters, SNMP Parameters, SMTP Parameters, Regional Parameters, etc.
Phone Book Version	numeric	Indicates the version of the inbound and outbound phonebook portion of the MultiVOIP software.
IFM Version	numeric	Indicates the version of the firmware running on the MultiVOIP's Interface Module, which is its analog telephony hardware.
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.
Hardware ID	alpha-numeric	Indicates the version of the MultiVOIP unit's circuit board and components.

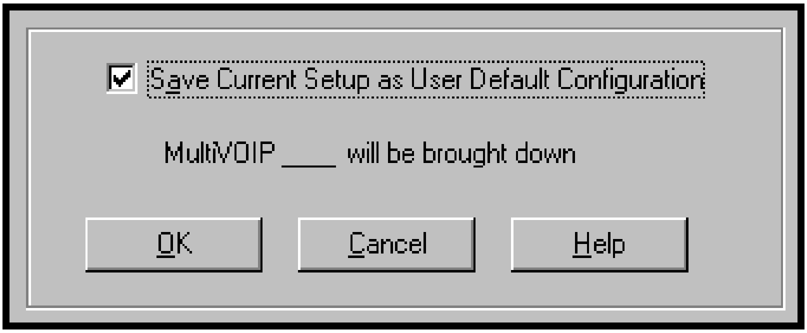
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



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 MVP2410 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" phone-books of each VoIP in the system.

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

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

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

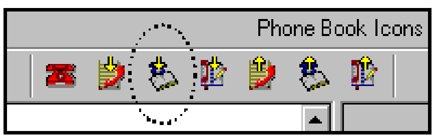
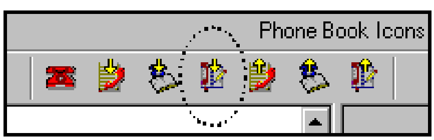
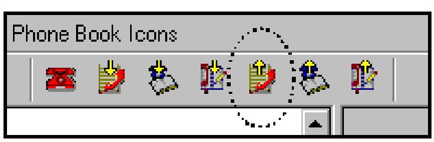
Consider two types of calls in the three-city system described above: (1) calls originating from the Miami office and terminating in the New York (Manhattan) office, and (2) calls originating from the Miami office and terminating in New York City but off the company's premises in an adjacent area code, 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.

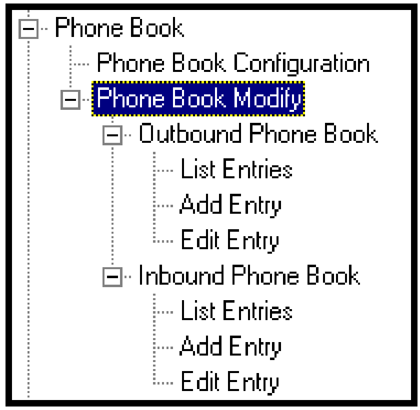
The second type of call similarly requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound Phonebook of the New York VOIP. However, these entries will be longer and more complicated. Any Miami call to New York City local numbers will be sent through the VOIP 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.

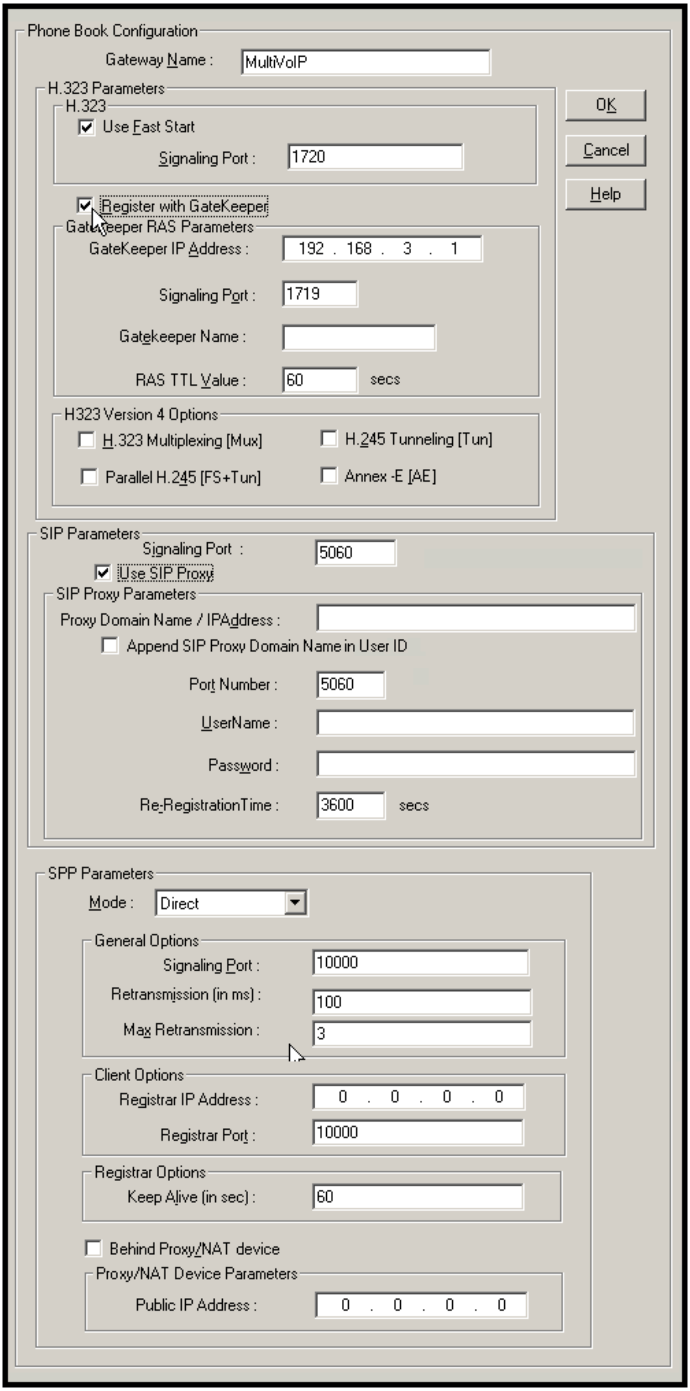
Phonebook configuration screens can be accessed using icons or the sidebar menu.

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 determine which protocol you will use (H.323, SIP, or SPP). Then fill in the IP address, signaling port, and other parameters, as needed. (The parameters needed for each protocol are different.)

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.
H.323 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.
Signaling Port	port number	Default: 1720 (H.323)
GateKeeper RAS Parameters		
Gatekeeper / IP Address	n.n.n.n, for n = 0 - 255	IP address of the GateKeeper.
Signaling Port		Well-known port number for GateKeepers. Must match port number of GateKeeper, 1719.
Gatekeeper Name	*alpha-numeric string*	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register.

.

PhoneBook Configuration Parameter Definitions (cont'd)		
GateKeeper RAS Parameters		
Field Name	**Values**	**Description**
RAS TTL Value	*in seconds*	The H.323 Gatekeeper "Time to Live" value. As soon as a MultiVOIP gateway registers with a gatekeeper (allowing the gatekeeper to control its call traffic) a countdown timer begins. The RAS TTL Value is the interval of the countdown timer. Before the TTL countdown expires, the MultiVOIP gateway needs to register with the gatekeeper in order to maintain the connection. If the MultiVOIP does not register before the TTL interval expires, the MultiVOIP gateway's registration with the gatekeeper will expire and the gatekeeper will no longer permit call traffic to or from that gateway. Calls in progress will continue to function even if the gateway becomes de-registered.

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
H.323 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**
SIP Proxy Parameters		
Signaling Port		Port number on which the MultiVOIP UserAgent software module will be waiting for any incoming SIP requests.
Use SIP Proxy	Y/N	Allows the MultiVOIP to work in conjunction with a proxy server.
Proxy Domain Name / IP Address	n.n.n.n where n=0-255	Network address of the proxy server that the voip is using.
Append SIP Proxy Domain Name in User ID	Y/N	When checked, the domain name of the SIP Proxy serving the MultiVOIP gateway will be included as part of the User ID for that gateway. If unchecked, the SIP Proxy's IP address will be included as part of the User ID instead of the SIP Proxy's domain name.
Port Number		Logical port number for proxy communications.
User Name	Values: alphanumeric 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.	

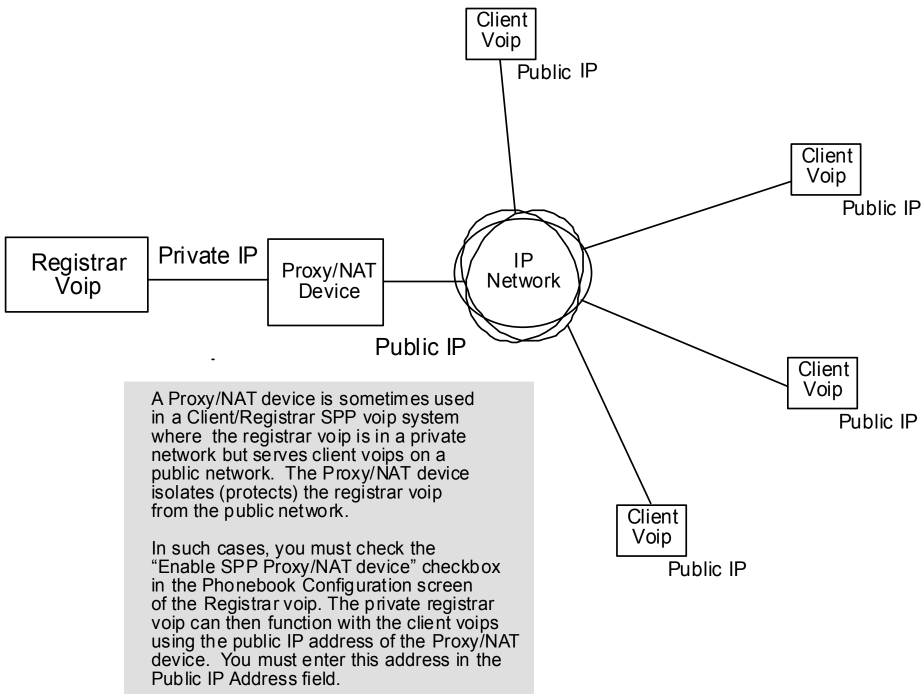
PhoneBook Configuration Parameter Definitions (cont'd)	
Field Name	**Values & Description**
SIP Proxy Parameters	
Password	Values: alphanumeric Description: Password for proxy server function. See "User Name" description above.
Re-Registration Time	Values: numeric (in seconds) Description: This is the timeout interval for registration of the MultiVOIP with a SIP proxy server. The time interval begins the moment the MultiVOIP gateway registers with the SIP proxy server and ends at the time specified by the user in the Re-Registration Time field (this field). When/if registration lapses, call traffic routed to/from the MultiVOIP through the SIP proxy server will cease. However, calls in progress will continue to function until they end.

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)

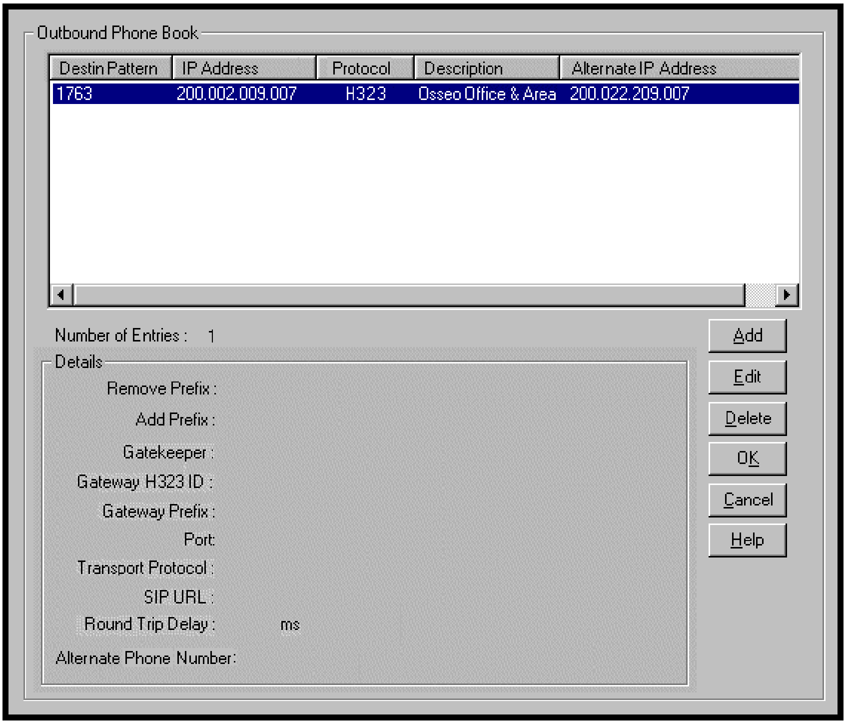
PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP) [continued]		
Client Options		Client Option fields are active only in registrar/client mode and only for client voip units.
Registrar IP Address		This is the IP address of the registrar 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.
Proxy/NAT Device Parameters		
Behind Proxy/NAT device	Y/N	Enables MultiVOIP (running in SPP Registrar mode) to operate 'behind' a proxy/NAT device (NAT = Network Address Translation).
Proxy/NAT Device Parameters – Public IP Address	n.n.n.n where n=0-255	The public IP address of the proxy/NAT device which the MultiVOIP is behind.

About SPP Proxy/NAT Device Parameters

SPP Client/Registrar System

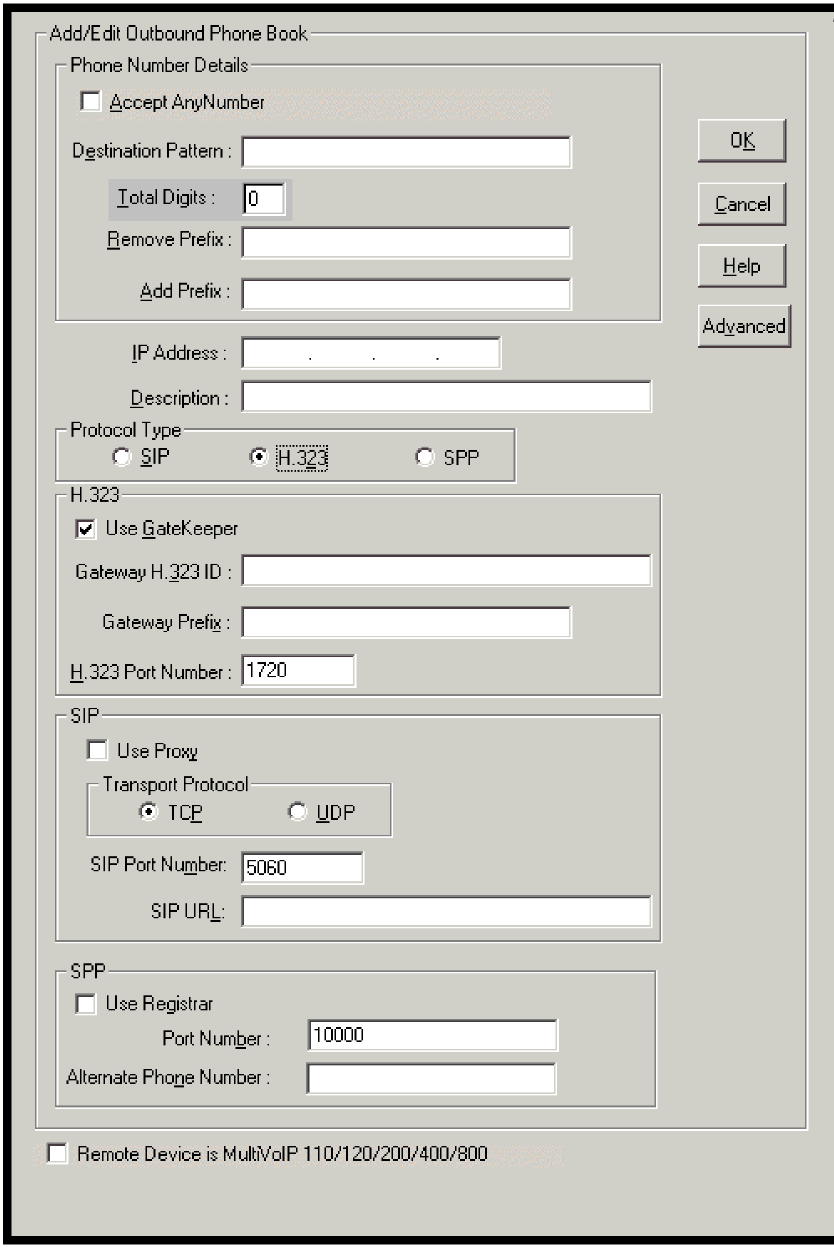


2. Select **PhoneBook Modify** and then select
Outbound Phone Book/List Entries.
Fields in the "Details" section will differ depending on the protocol (H.323, SIP, or SPP) of the selected list entry to which the details pertain.



Click **Add**.

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



Enter Outbound PhoneBook data for your MVP2410. 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**
Accept Any Number	Y/N	When checked, “Any Number” appears as the value in the Destination Pattern field. The **Any Number** feature works differently depending on whether or not an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the **IP Address** in the Add/Edit Outbound Phone Book screen. “Any Number” can be used in addition to one or more Destination Patterns. **When external routing device is used**. If Any Number is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the external routing device used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). The IP Address of the external routing device must be set in the Phone Book Configuration screen.

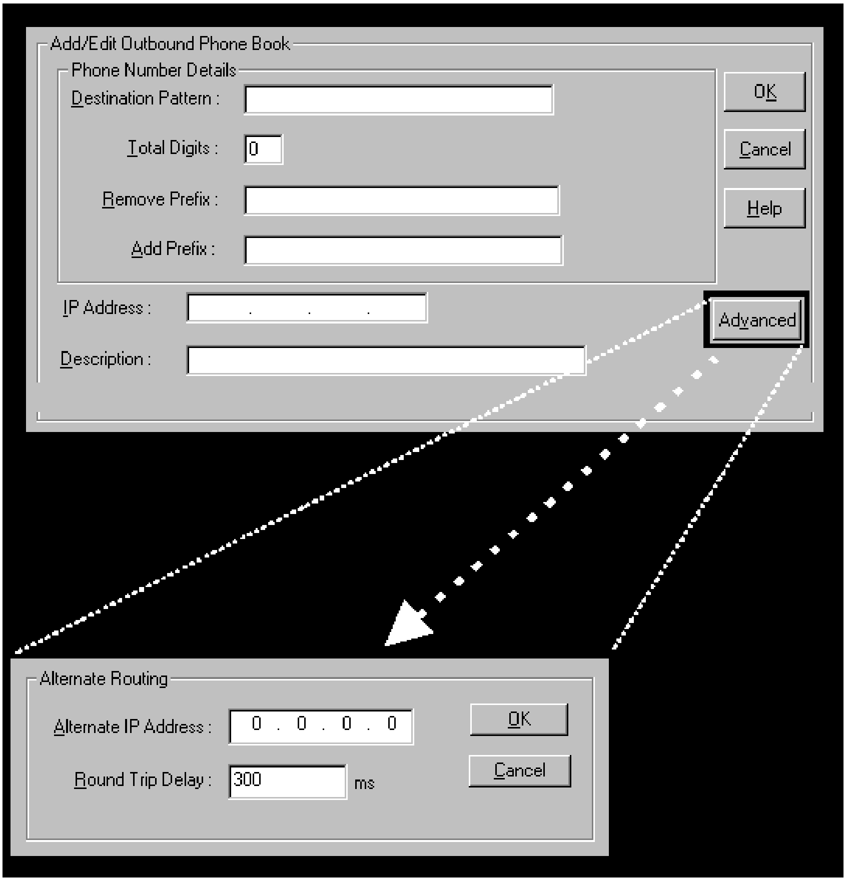
Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	*This field currently disabled.* 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.
Gateway H.323 ID	alpha-numeric	The H.323 ID assigned to the destination MultiVOIP. Only valid if "Use Gatekeeper" is enabled for this entry.
Gateway Prefix	numeric	This number becomes registered with the GateKeeper. Call requests sent to the gatekeeper and preceded by this prefix will be routed to the VOIP gateway.
H.323 Port Number	1720	This parameter pertains to Q.931, which is the H.323 call signaling protocol for setup and termination of calls (aka ITU-T Recommendation I.451). H.323 employs only one "well-known" port (1720) for Q.931 signaling. If Q.931 message-oriented signaling protocol is used, 1720 must be chosen as the H.323 Port Number.

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.

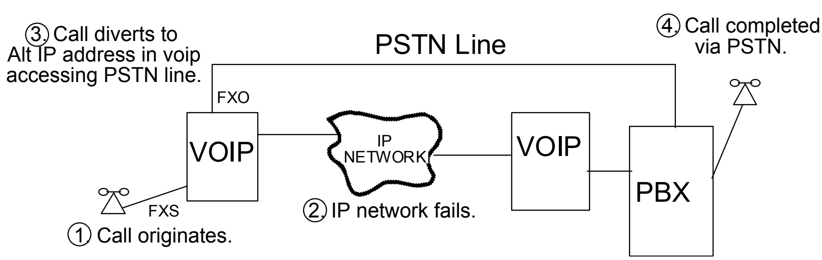
Add/Edit Outbound Phone Book: Field Def'ns (cont'd)		
Field Name	**Values**	**Description**
SPP Fields		
Use Registrar	**Values:** Y/N **Description:** Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set 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.
Remote Device is [legacy voip]	Y/N	When checked, this MultiVOIP can operate with 'first-generation' MultiVOIP units in the same IP network. These include MVP-110/120/200/400/800.
Advanced button	**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.



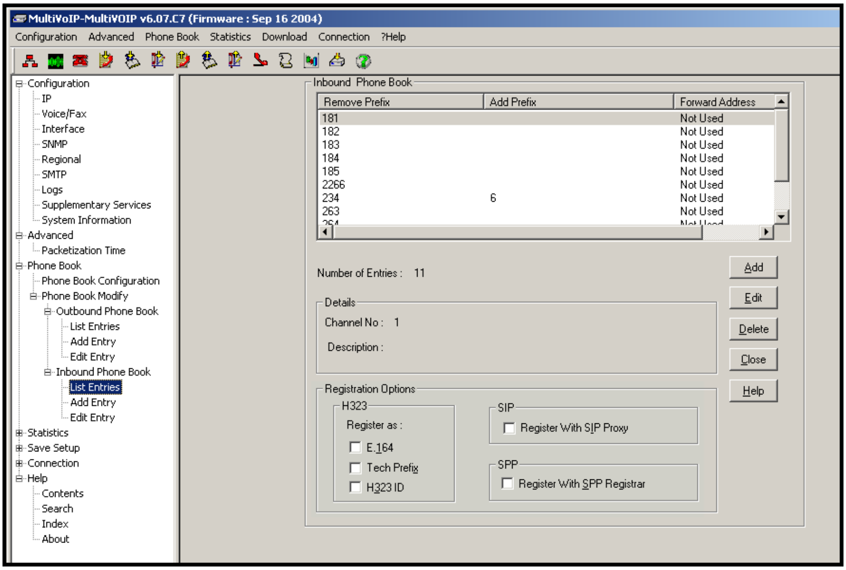
Alternate Routing Field Definitions		
Field Name	**Values**	**Description**
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when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

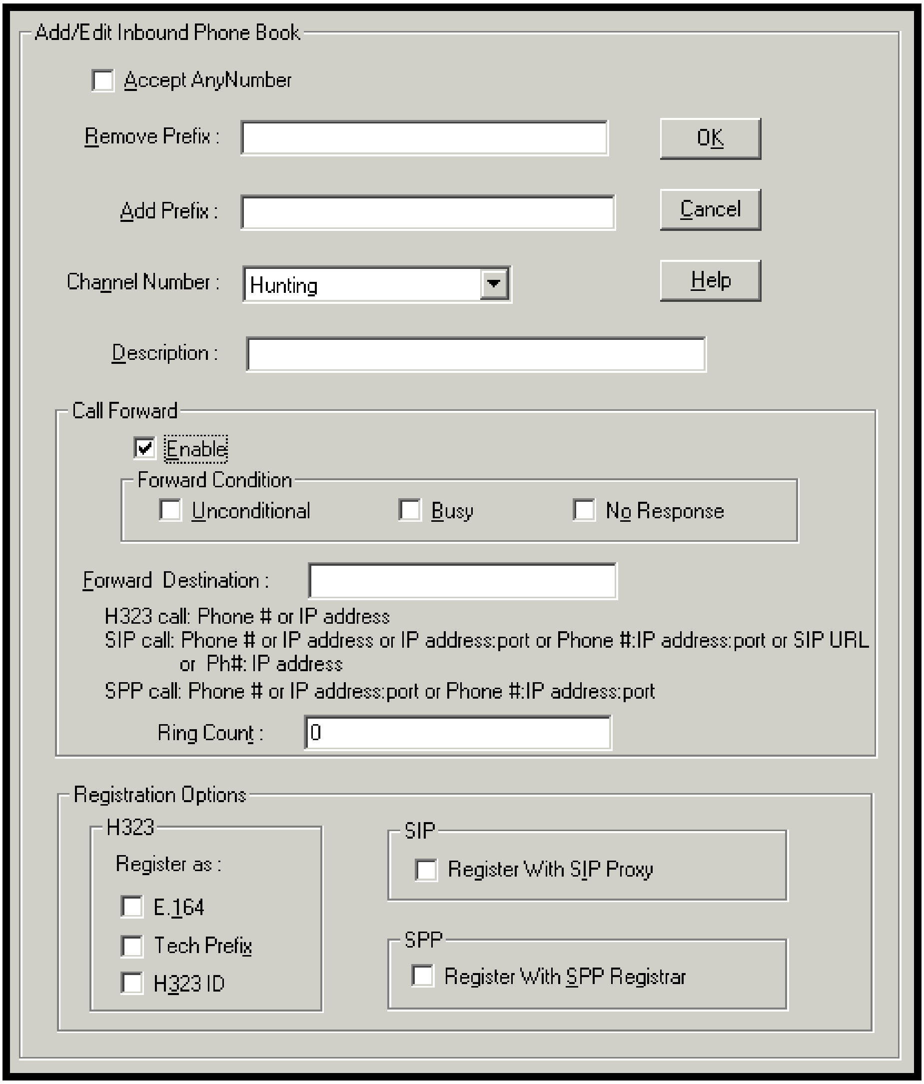


PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

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**
Accept Any Number	**Values**: Y/N **Description:** When checked, "Any Number" appears as the value in the Remove Prefix field. The **Any Number** feature of the Inbound Phone Book does not work when an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the voip on the channel listed in the **Channel Number** field. "Any Number" can be used in addition to one or more Prefixes.	
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)
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.
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.

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Forward Parameters		
Forward Condition	Condit.; 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. Forwarding can be conditioned on both "Busy" and "No Response."
Forward Destination IP address, phone number, port number, etc.	Phone number or IP address to which calls will be directed. For H.323 calls, the Forward Destination can be either a Phone Number or an IP Address. For SIP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address, (c) IP address: port number, (d) phone number:IP addr: port number, (e) SIP URL, or (f) phone #: IP address. For SPP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address: port, or (c) phone number: IP address: port.	

Add/Edit Inbound Phone Book: Field Definitions (cont'd)	
Field Name	**Values and Description**
Ring Count	0, 1, 2, 3, etc. When "No Response" is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.
Registration Option Parameters	In an H.323 voip system, gateways can register with the system using one of these identifiers: (a) an E.164 identifier, (b) a Tech Prefix identifier, or (c) an H.323 ID identifier. In a SIP voip system, gateways can register with the SIP Proxy. In an SPP voip system, gateways can register with the SPP Registrar voip unit.

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.

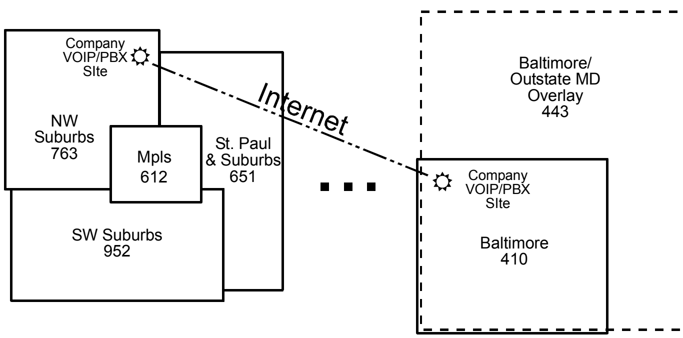
Remember that the initial MVP2410 setup must be done locally or via the built-in Remote Configuration/Command Modem using the MultiVOIP program. After the initial configuration is complete, all of the MVP2410 units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVOIP web GUI software program or the MultiVOIP program (in conjunction with the built-in modem).

T1 Phonebook Examples

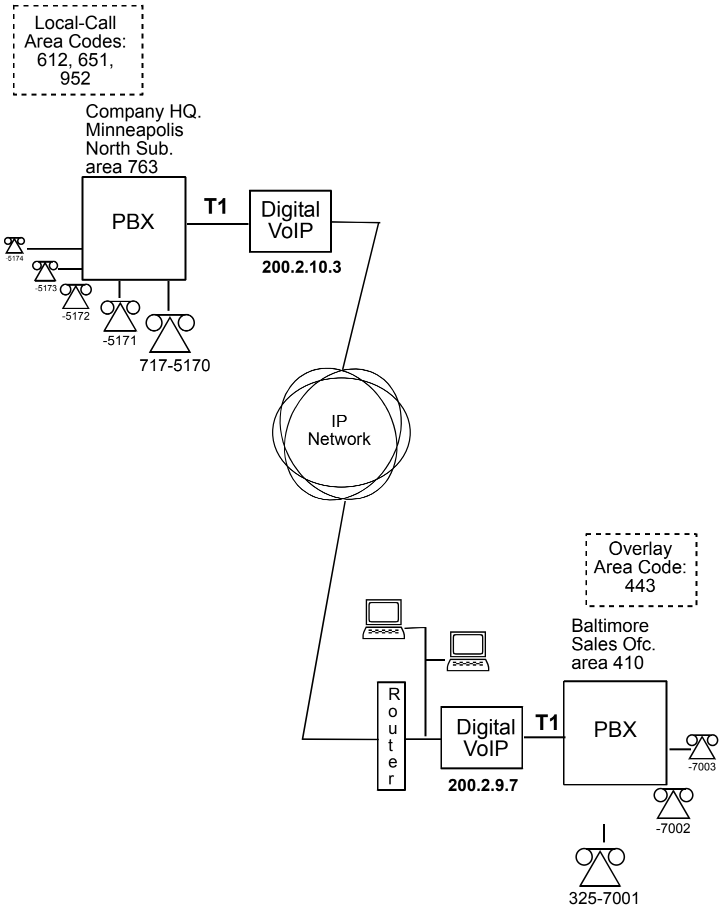
The following example demonstrates how Outbound and Inbound PhoneBook entries work in a situation of multiple area codes. Consider a company with offices in Minneapolis and Baltimore.

3 Sites, All-T1 Example

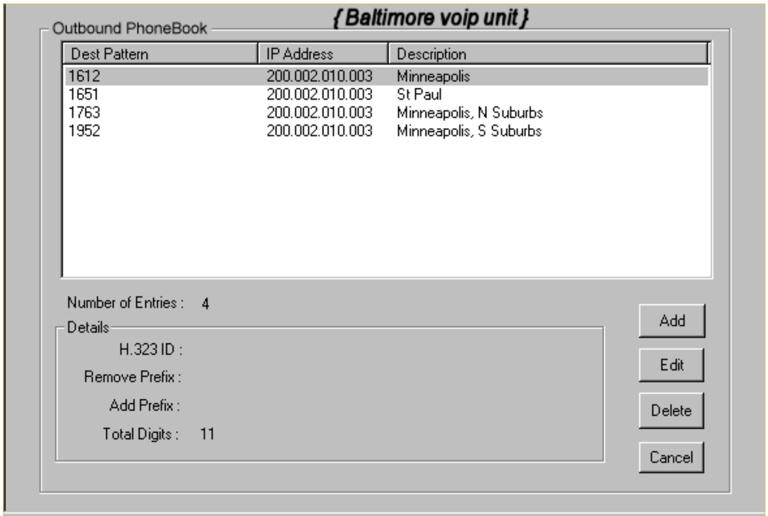
Notice first the area code situation in those two cities: Minneapolis's local calling area consists of multiple adjacent area codes; Baltimore's local calling area consists of a base area code plus an overlay area code.



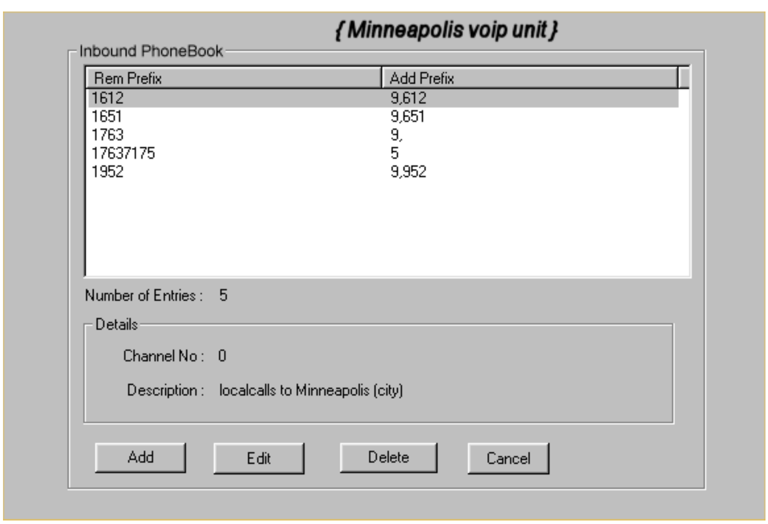
An outline of the equipment setup in both offices is shown below.



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.)

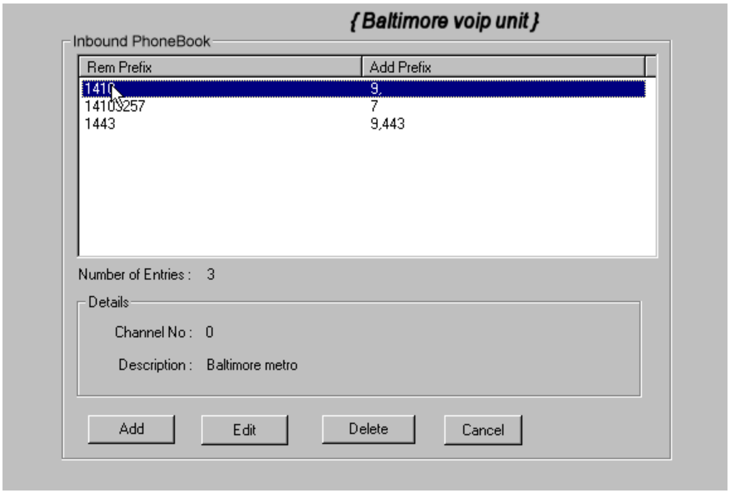
If a Baltimore employee dials any phone number in the 612 area code, the call will automatically be handled by the company's voip system. Upon receiving such a call, the Minneapolis voip will remove the digits "1612". But before the suburban-Minneapolis voip can complete the call to the PSTN of the Minneapolis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

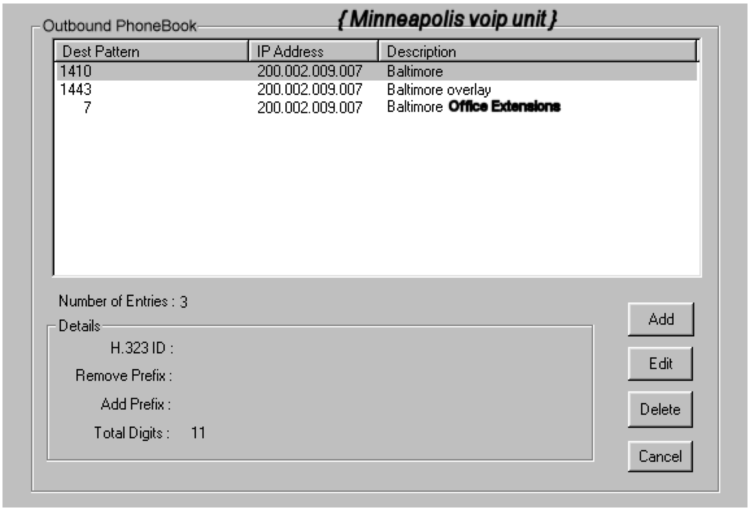
Similarly, the Inbound PhoneBook for the Baltimore VOIP (shown first below) generally matches the Outbound PhoneBook of the Minneapolis VOIP (shown second below).



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.

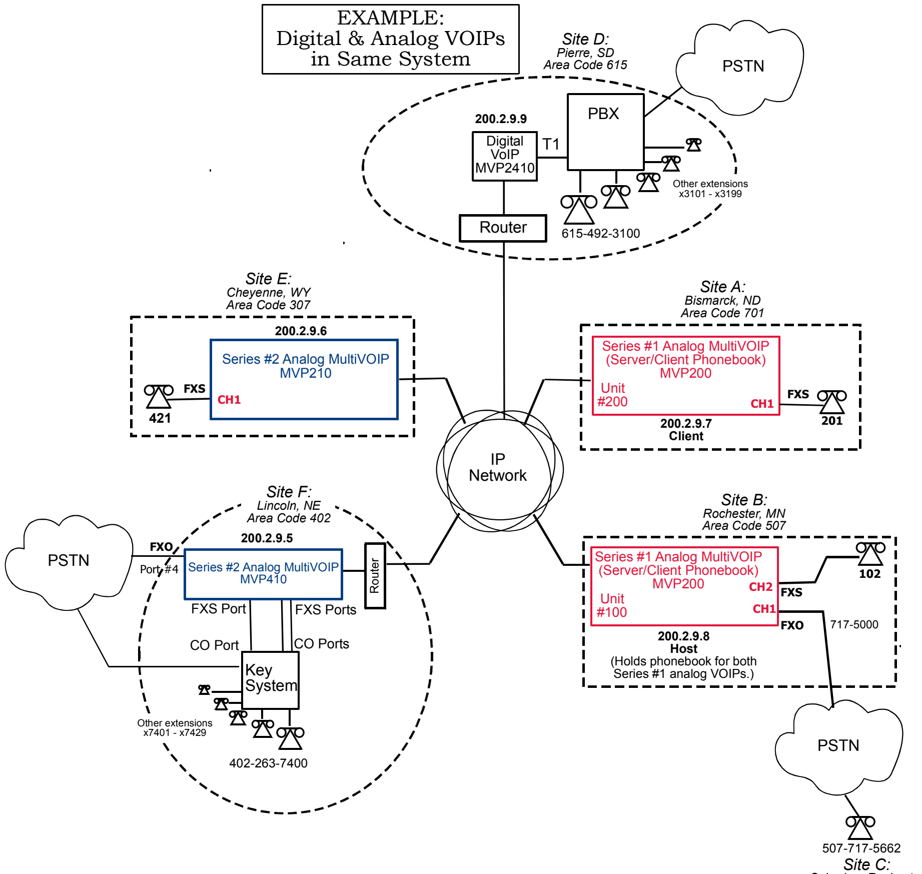
Note also that a comma (as in the entry 9,443) denotes a delay in dialing. A one-second delay is commonly used to allow a second dial tone to be generated for calls going outside of the facility’s PBX system.

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 MVP2410 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 MVP2410) 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 MVP2410 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 MVP2410 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 MVP2410 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 MVP2410: 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 MVP2410 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 MVP2410 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 MVP2410 can be completely transparent to phone users within the company.

Chapter 8: E1 Phonebook Configuration

(European Telephony Standards)

MVP3010 Inbound and Outbound MultiVOIP Phonebooks

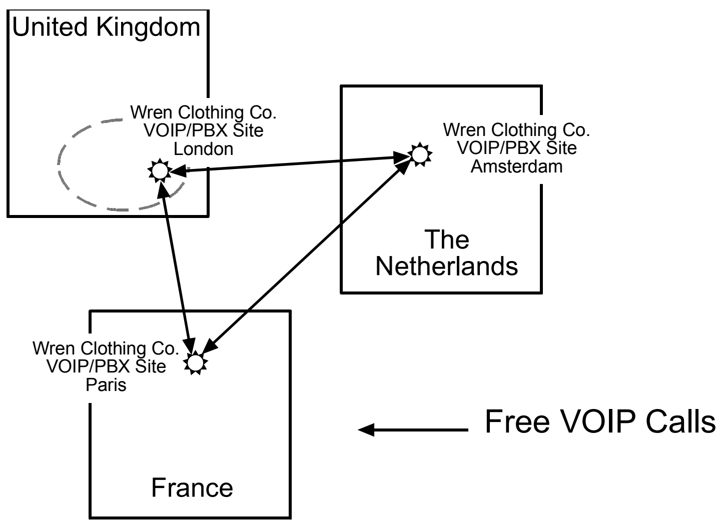
Important Definition:	The MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook describes the dialing sequences that can be used to call that MultiVOIP and how those calls will be directed.

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.

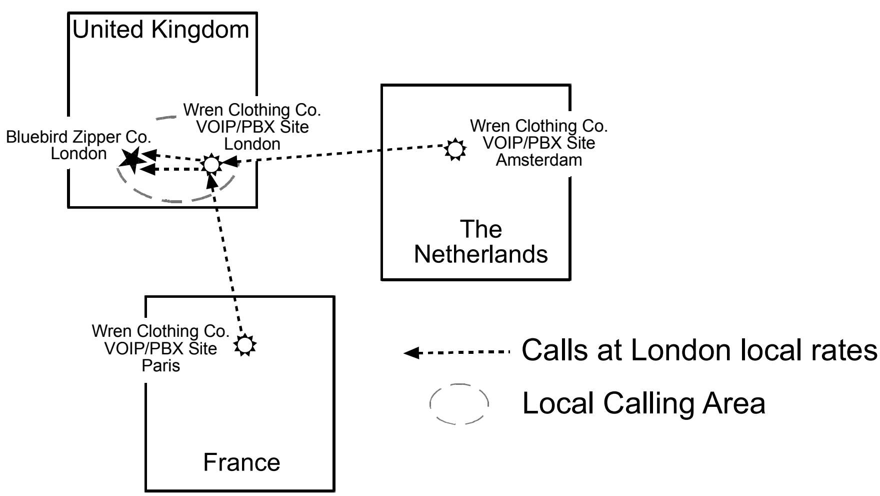
Free Calls: One VOIP Site to Another

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.

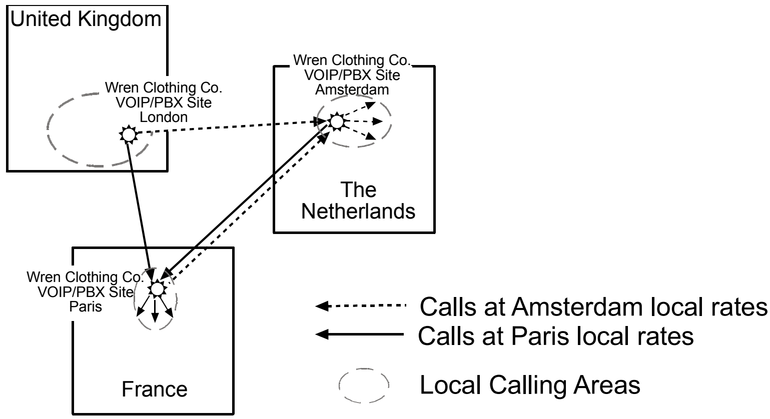


Local Rate Calls: Within Local Calling Area of Remote VOIP

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.)

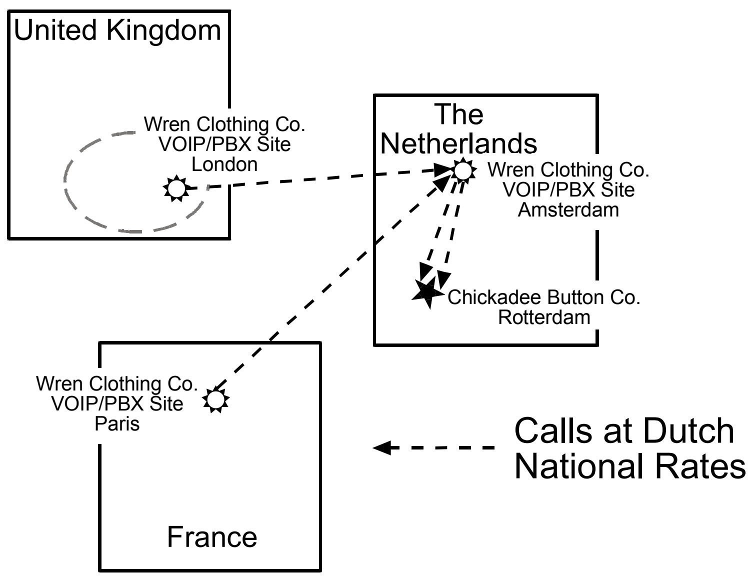


Similarly, the VOIP system allows Wren Clothing employees in London and Amsterdam to call anywhere in Paris at local rates; it allows Wren Clothing employees in Paris and London to call anywhere in Amsterdam at local rates.

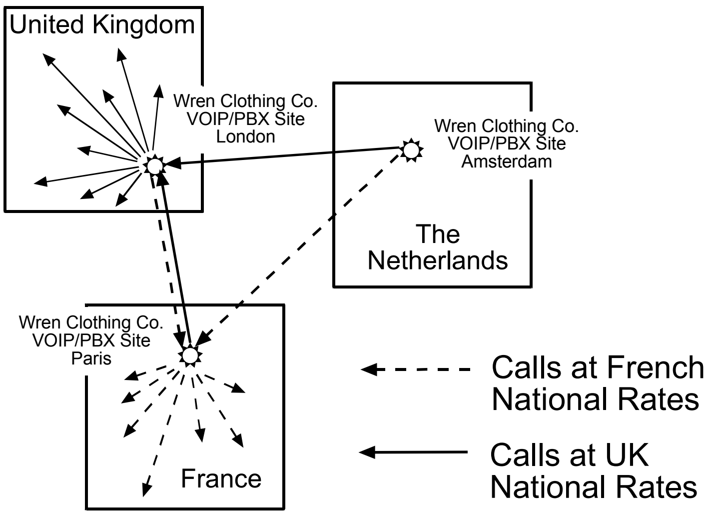


National Rate Calls: Within Nation of Remote VOIP Site

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.

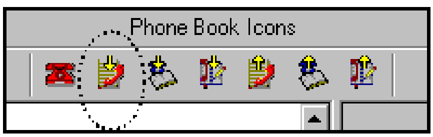
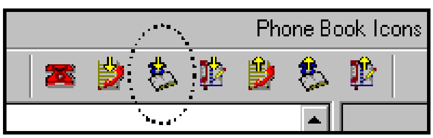
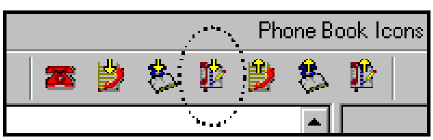
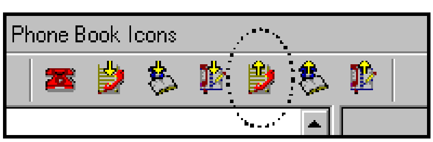
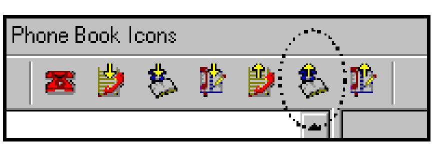
The "Inbound" phonebook for a particular VOIP unit describes the dialing sequences required for a call to originate remotely from any other VOIP sites in the system, and to terminate on that particular VOIP.

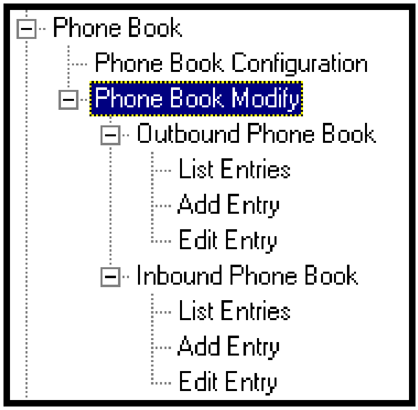
Briefly stated, *the MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook lists the dialing sequences that can be used to call that MultiVOIP*. (Of course, the phone numbers are not literally "listed" individually.) The phone stations that can originate or complete 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.

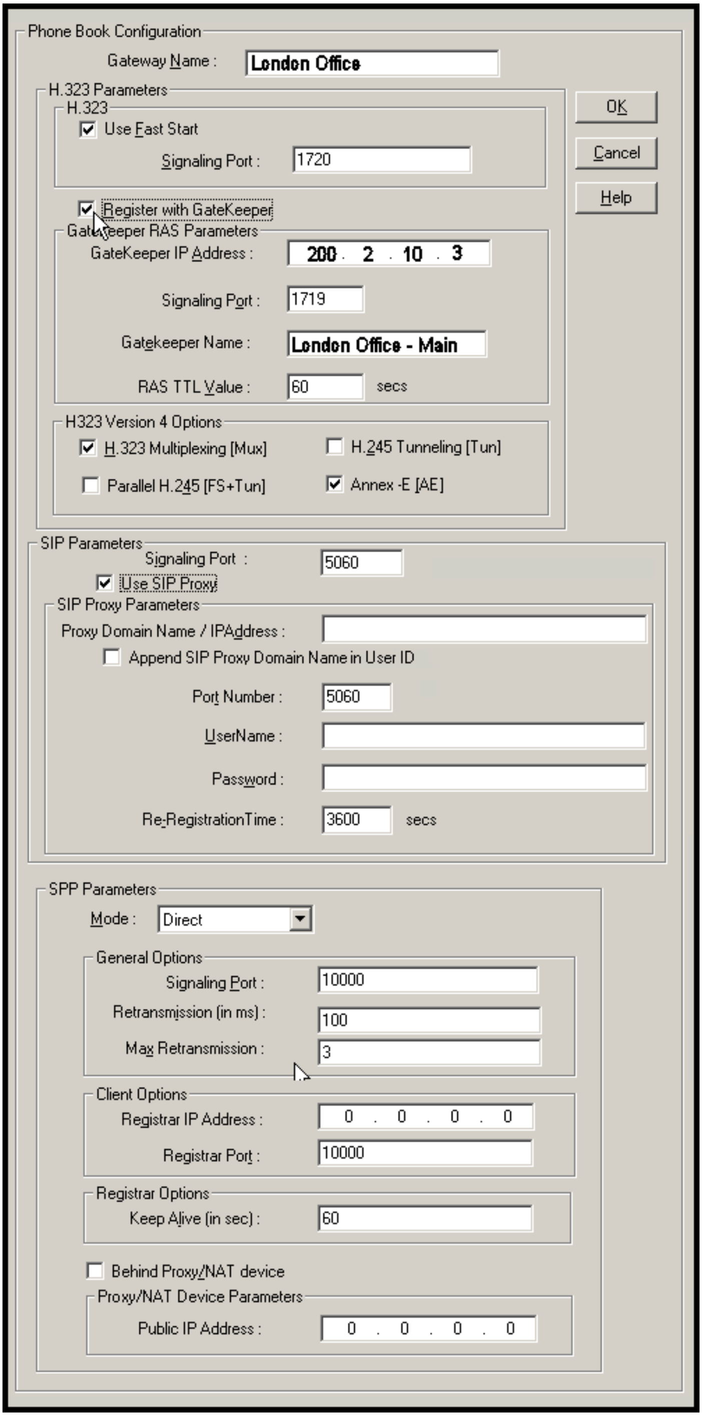
Phonebook configuration screens can be accessed using icons or the sidebar menu.

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).



In consultation with your VOIP administrator, enter the Gateway Name determine which protocol you will use (H.323, SIP, or SPP).

Then fill in the IP address, signaling port, and other parameters, as needed. (The parameters needed for each protocol are different.)

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.
H.323 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.
Signaling Port	port number	Default: 1720 (H.323)
GateKeeper RAS Parameters		
Gatekeeper / IP Address	n.n.n.n, for n = 0 - 255	IP address of the GateKeeper.
Signaling Port	1 - 64000	Well-known port number for GateKeepers. Must match port number of GateKeeper, 1719.
Gatekeeper Name	*alpha-numeric string*	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register.

.

PhoneBook Configuration Parameter Definitions (cont'd)		
GateKeeper RAS Parameters		
Field Name	**Values**	**Description**
RAS TTL Value	*in seconds*	The H.323 Gatekeeper "Time to Live" value. As soon as a MultiVOIP gateway registers with a gatekeeper (allowing the gatekeeper to control its call traffic) a countdown timer begins. The RAS TTL Value is the interval of the countdown timer. Before the TTL countdown expires, the MultiVOIP gateway needs to register with the gatekeeper in order to maintain the connection. If the MultiVOIP does not register before the TTL interval expires, the MultiVOIP gateway's registration with the gatekeeper will expire and the gatekeeper will no longer permit call traffic to or from that gateway. Calls in progress will continue to function even if the gateway becomes de-registered.

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
H.323 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.)	

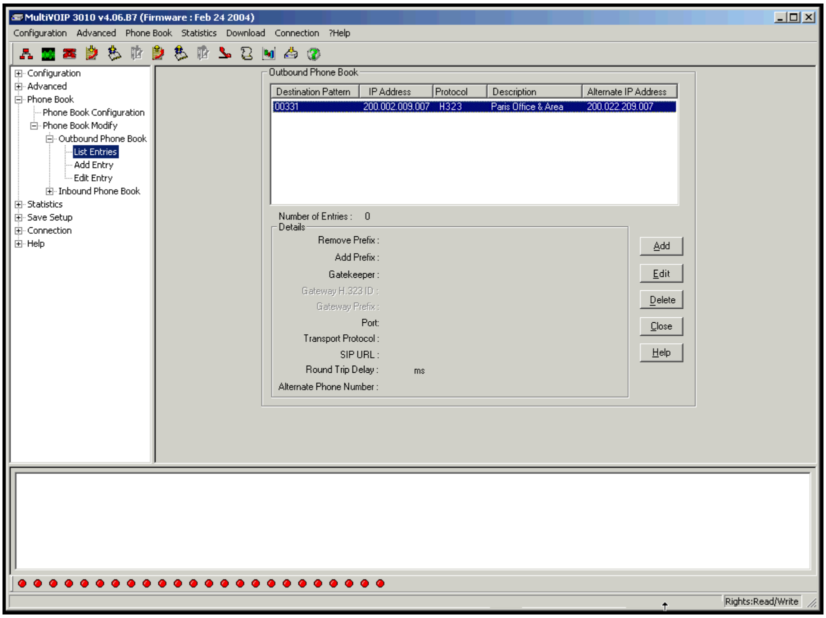
PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
SIP Proxy Parameters		
Signaling Port	1 - 64000	Port number on which the MultiVOIP UserAgent software module will be waiting for any incoming SIP requests.
Use SIP Proxy	Y/N	Allows the MultiVOIP to work in conjunction with a proxy server.
Proxy Domain Name / IP Address	n.n.n.n where n=0-255	Network address of the proxy server that the voip is using.
Append SIP Proxy Domain Name in User ID	Y/N	When checked, the domain name of the SIP Proxy serving the MultiVOIP gateway will be included as part of the User ID for that gateway. If unchecked, the SIP Proxy's IP address will be included as part of the User ID instead of the SIP Proxy's domain name.
Port Number	numeric	Logical port number for proxy communications.
User Name	Values: alphanumeric 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.	

PhoneBook Configuration Parameter Definitions (cont'd)	
Field Name	**Values & Description**
SIP Proxy Parameters	
Password	Values: alphanumeric Description: Password for proxy server function. See "User Name" description above.
Re-Registration Time	Values: numeric (in seconds) Description: This is the timeout interval for registration of the MultiVOIP with a SIP proxy server. The time interval begins the moment the MultiVOIP gateway registers with the SIP proxy server and ends at the time specified by the user in the Re-Registration Time field (this field). When/if registration lapses, call traffic routed to/from the MultiVOIP through the SIP proxy server will cease. However, calls in progress will continue to function until they end.

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)

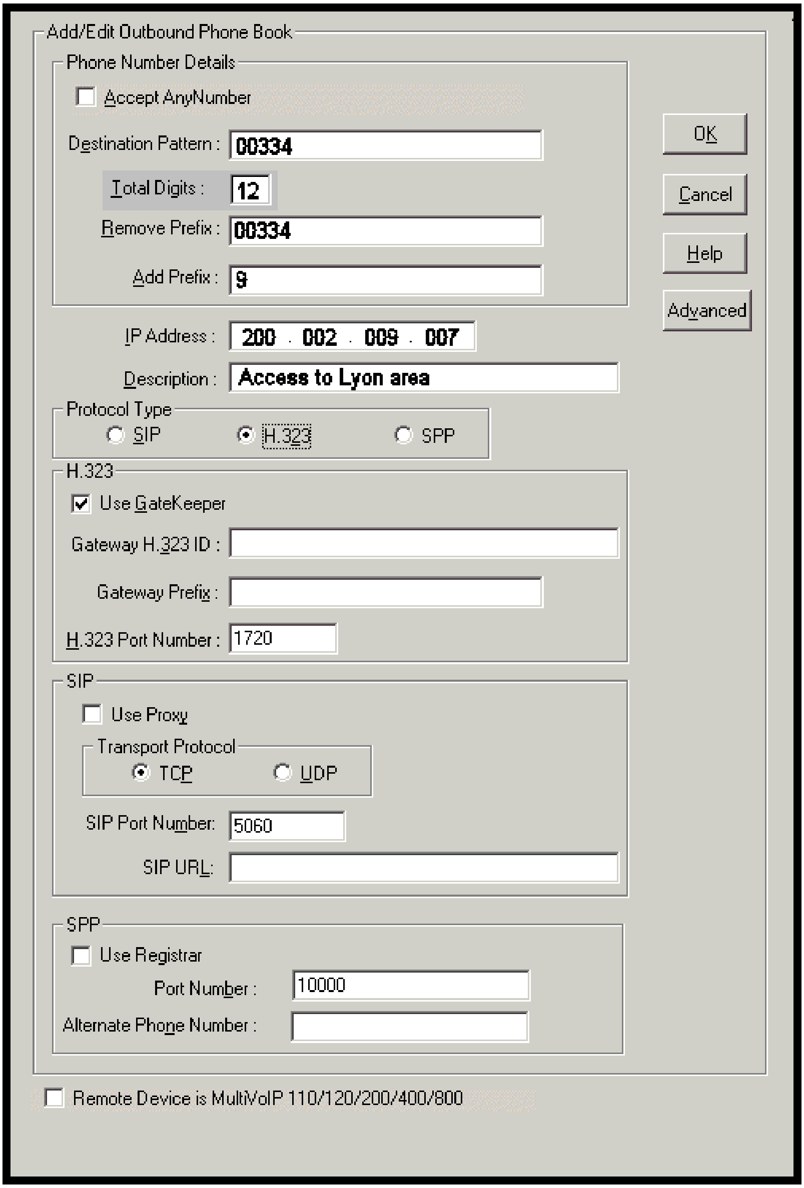
PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP) [continued]		
Client Options		Client Option fields are active only in registrar/client mode and only for client voip units.
Registrar IP Address	n.n.n.n	This is the IP address of the registrar voip to which this client is assigned. (Default value = 0.0.0.0; effectively, there is no useful default value.)
Registrar Port	1-64000	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.)	30 - 300	Time-out duration before a registrar will unregister a client that does not send its "I'm here" signal. Timeout default = 60 seconds.
Proxy/NAT Device Parameters		
Behind Proxy/NAT device	Y/N	Enables MultiVOIP (running in SPP Registrar mode) to operate 'behind' a proxy/NAT device (NAT = Network Address Translation).
Proxy/NAT Device Parameters – Public IP Address	n.n.n.n where n=0-255	The public IP address of the proxy/NAT device which the MultiVOIP is behind.

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**
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Destination Pattern field. The **Any Number** feature works differently depending on whether or not an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the **IP Address** in the Add/Edit Outbound Phone Book screen. "Any Number" can be used in addition to one or more Destination Patterns. **When external routing device is used**. If Any Number is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the external routing device used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). The IP Address of the external routing device must be set in the Phone Book Configuration screen.

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.

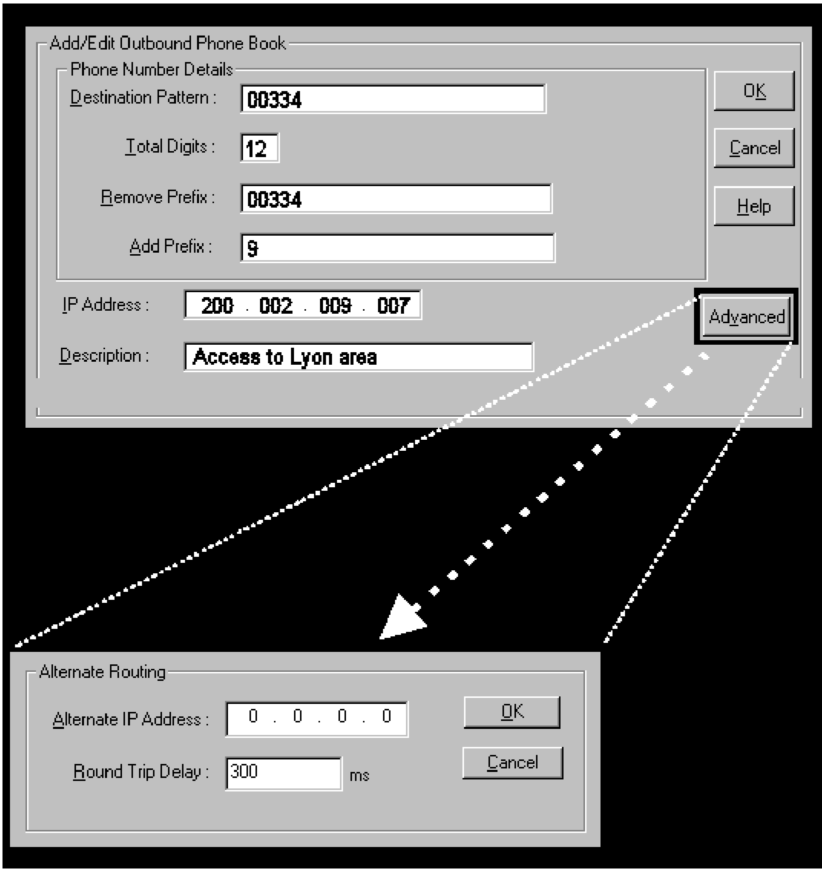
Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 fields		
Use Gatekeepr	Y/N	Indicates whether or not gatekeeper is used.
Gateway H.323 ID	alpha-numeric	The H.323 ID assigned to the destination MultiVOIP. Only valid if "Use Gatekeeper" is enabled for this entry.
Gateway Prefix	numeric	This number becomes registered with the GateKeeper. Call requests sent to the gatekeeper and preceded by this prefix will be routed to the VOIP gateway.
H.323 Port Number	1720	This parameter pertains to Q.931, which is the H.323 call signaling protocol for setup and termination of calls (aka ITU-T Recommendation I.451). H.323 employs only one "well-known" port (1720) for Q.931 signaling. If Q.931 message-oriented signaling protocol is used, 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)

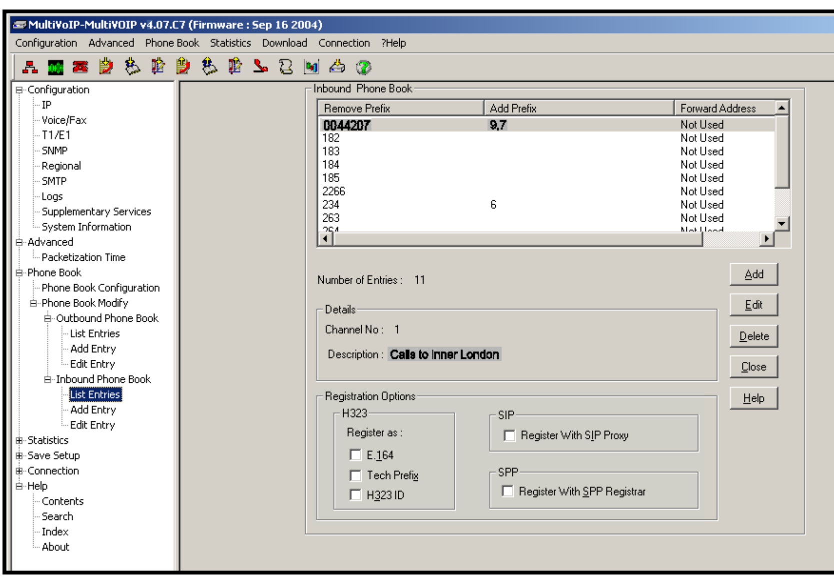
Field Name	Values	Description
SPP Fields		
Use Registrar	**Values:** Y/N **Description:** Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set in Phonebook Configuration screen) has a static IP address and 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.
Remote Device is …	Y/N	Check when system includes 1st-generation MultiVOIPs to allow inter-operation. These include MVP-110/120/200/400/800 MultiVOIP units.
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.

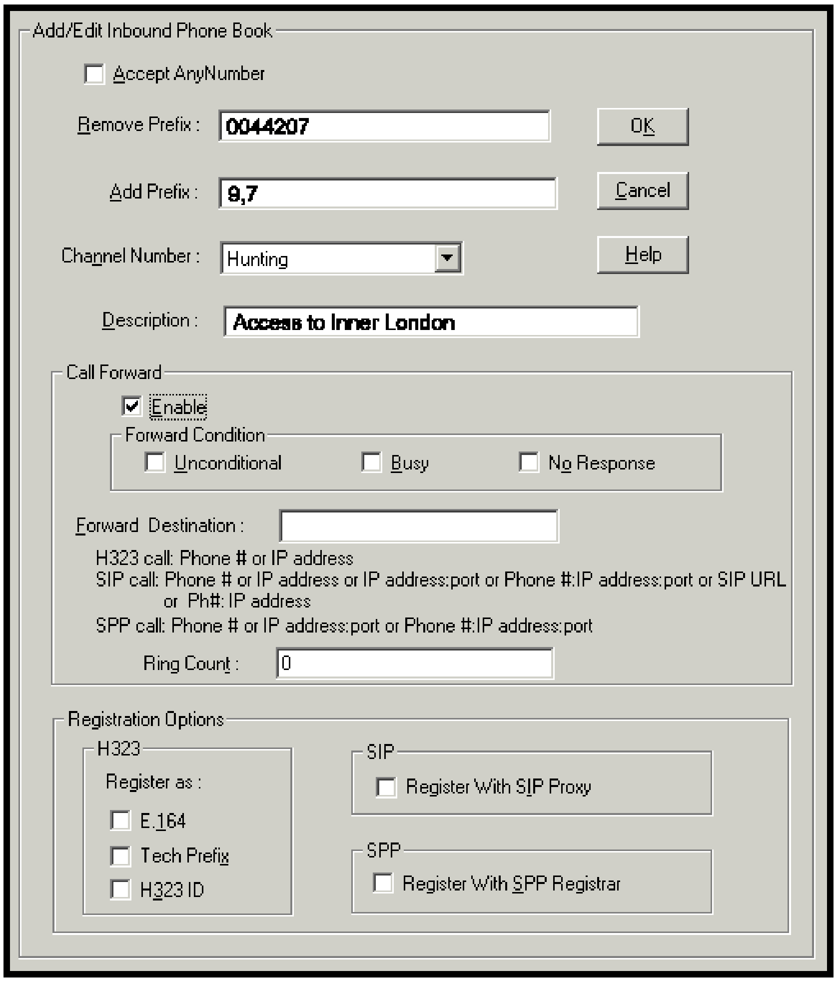


Alternate Routing Field Definitions		
Field Name	**Values**	**Description**
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when 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**
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Remove Prefix field. The **Any Number** feature of the Inbound Phone Book does not work when an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the voip on the channel listed in the **Channel Number** field. "Any Number" can be used in addition to one or more Prefixes.
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
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.
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. Forwarding can be conditioned on both "Busy" and "No Response."

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Forward Destination IP address, phone number, port number, etc.	Phone number or IP address to which calls will be directed. For H.323 calls, the Forward Destination can be either a Phone Number of an IP Address. For SIP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address, (c) IP address: port number, (d) phone number:IP addr: port number, (e) SIP URL, or (f) phone #: IP address. For SPP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address: port, or (c) phone number: IP address: port.	
Ring Count	integer	When No Response is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.
Registration Option Parameters	In an H.323 voip system, gateways can register with the system using one of these identifiers: (a) an E.164 identifier, (b) a Tech Prefix identifier, or (c) an H.323 ID identifier. In a SIP voip system, gateways can register with the SIP Proxy. In an SPP voip system, gateways can register with the SPP Registrar voip unit.	

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 or via the built-in Remote Configuration/Command Modem 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 MultiVOIP web GUI software program or the MultiVOIP program (in conjunction with the built-in modem).

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

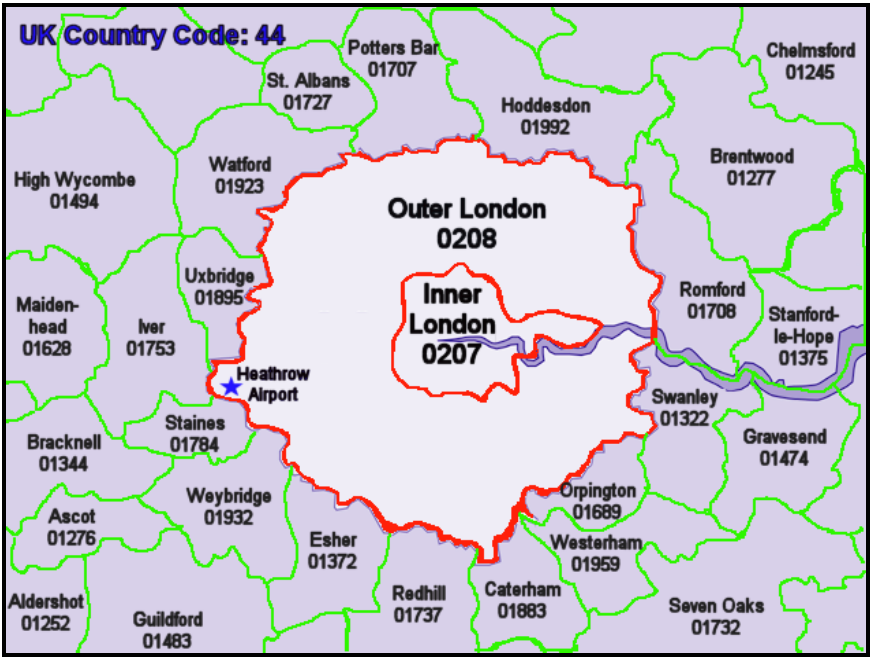
The VOIP system will have the following features:

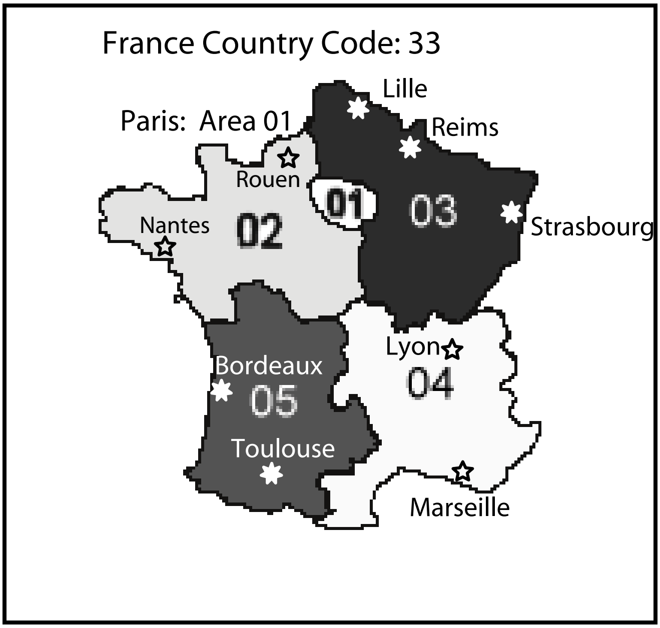
1. Employees in all cities will be able to call each other over the VOIP system using 4-digit extensions.

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.

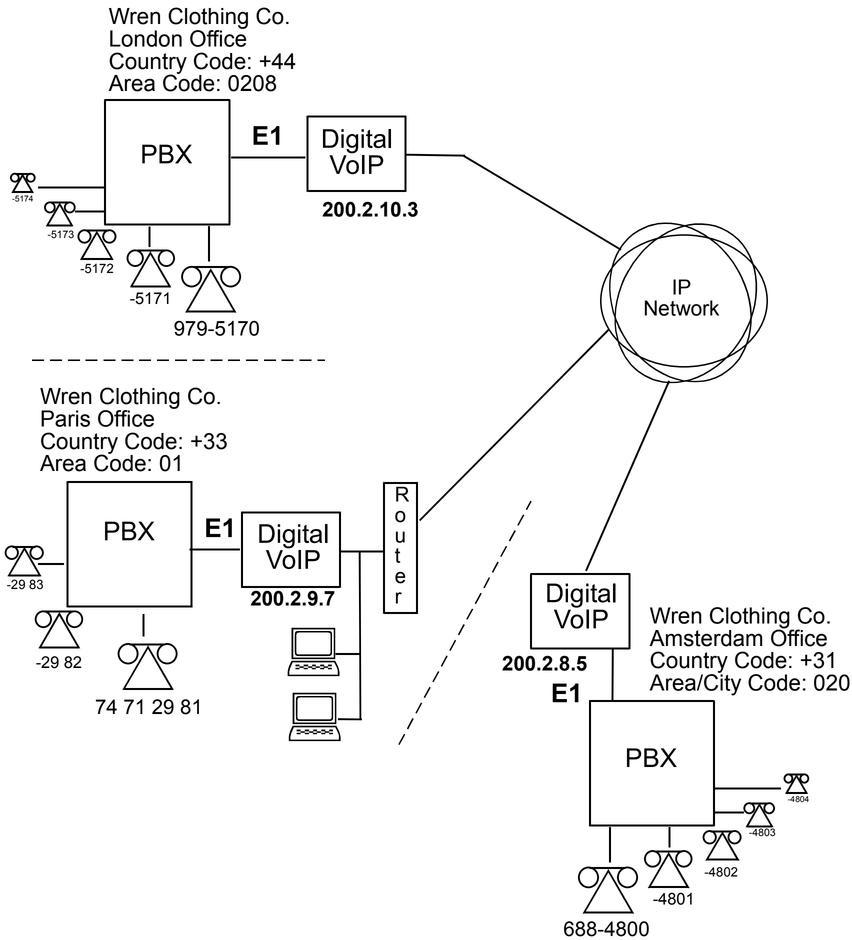
Note that the phonebook entries for Series II analog MultiVOIP used in Euro-type telephony settings will be the same in format as entries for the MVP3010.


UK Country Code: 44
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
Maiden-
head
01628
Iver
01753
Romford
01708
Stanford-
le-Hope
01375
Heathrow
Airport
Swanley
01322
Bracknell
01344
Staines
01784
Gravesend
01474
Weybridge
01932
Orpington
01689
Ascot
01276
Esher
01372
Westerham
01959
Redhill
01737
Caterham
01883
Seven Oaks
01732
Aldershot
01252
Guildford
01483

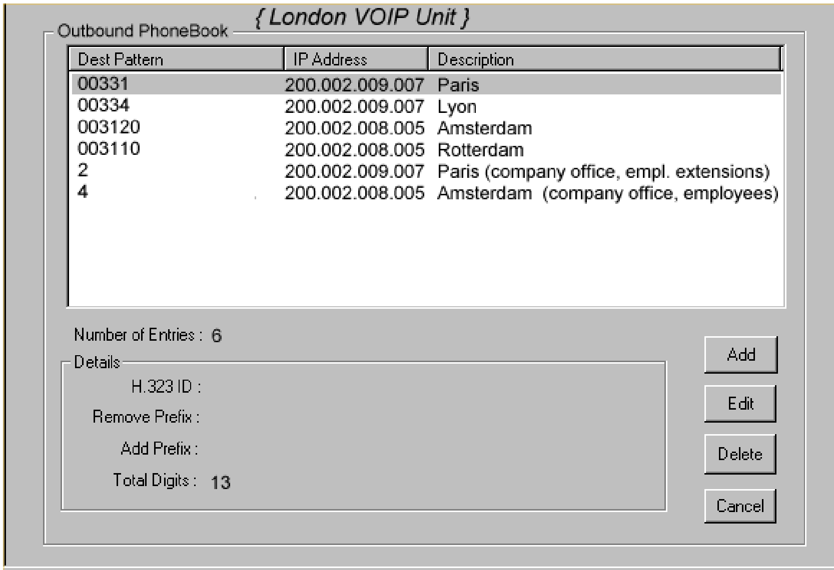

France Country Code: 33
Lille
Paris: Area 01
Reims
Rouen
01
03
Strasbourg
Nantes
02
Lyon
04
Bordeaux
05
Toulouse
Marseille


The Netherlands
Country Code: 31
Texel 0222
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

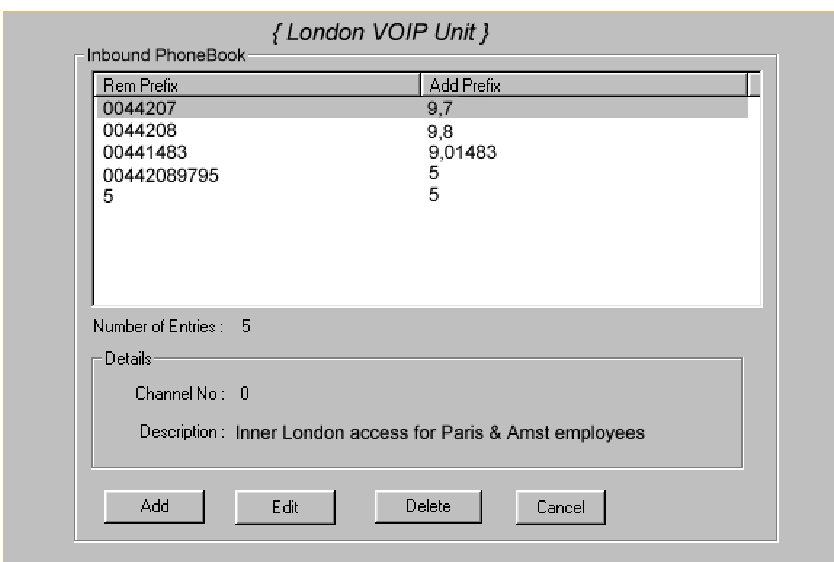
An outline of the equipment setup in these three offices is shown below.



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

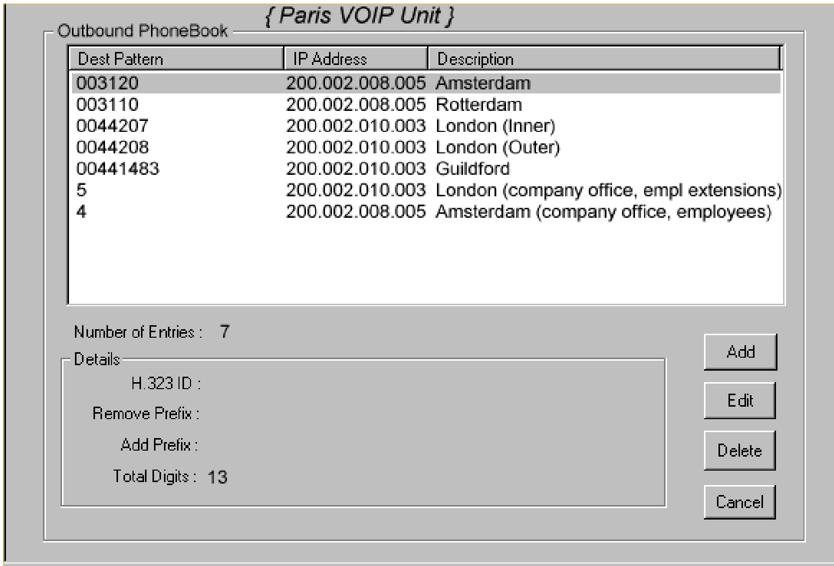


The Inbound PhoneBook for the London VOIP is shown below.

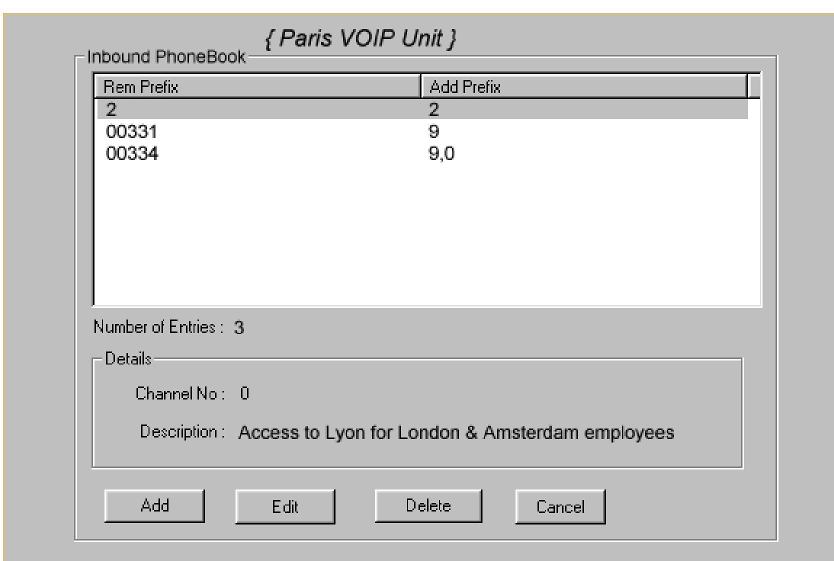


NOTE: Commas are allowed in the Inbound Phonebook, but **not** in the Outbound Phonebook. Commas denote a brief pause for a dial tone, allowing time for the PBX to get an outside line.

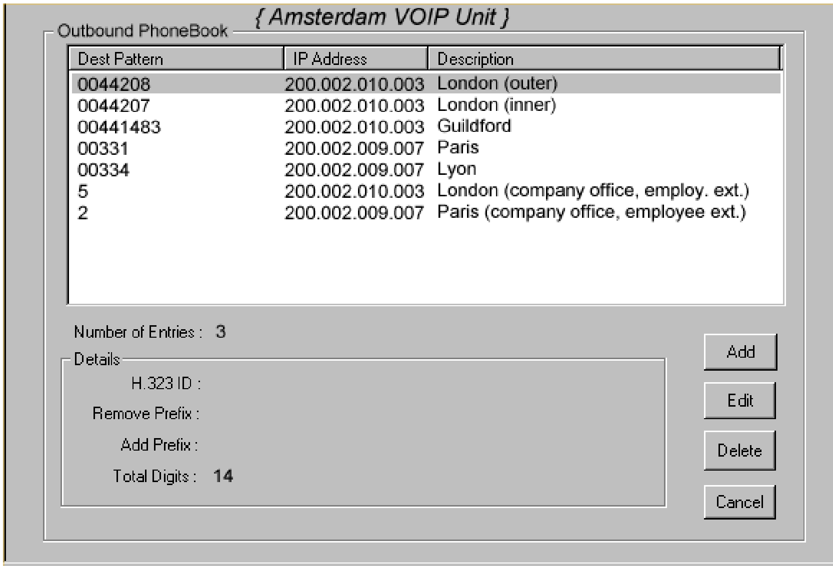
The screen below shows Outbound PhoneBook entries for the VOIP located in the company's Paris facility.



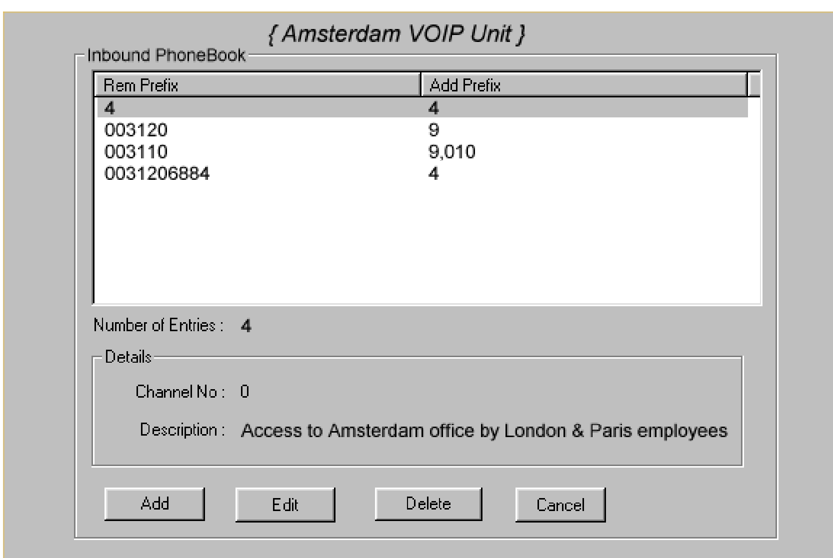
The Inbound PhoneBook for the Paris VOIP is shown below.



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

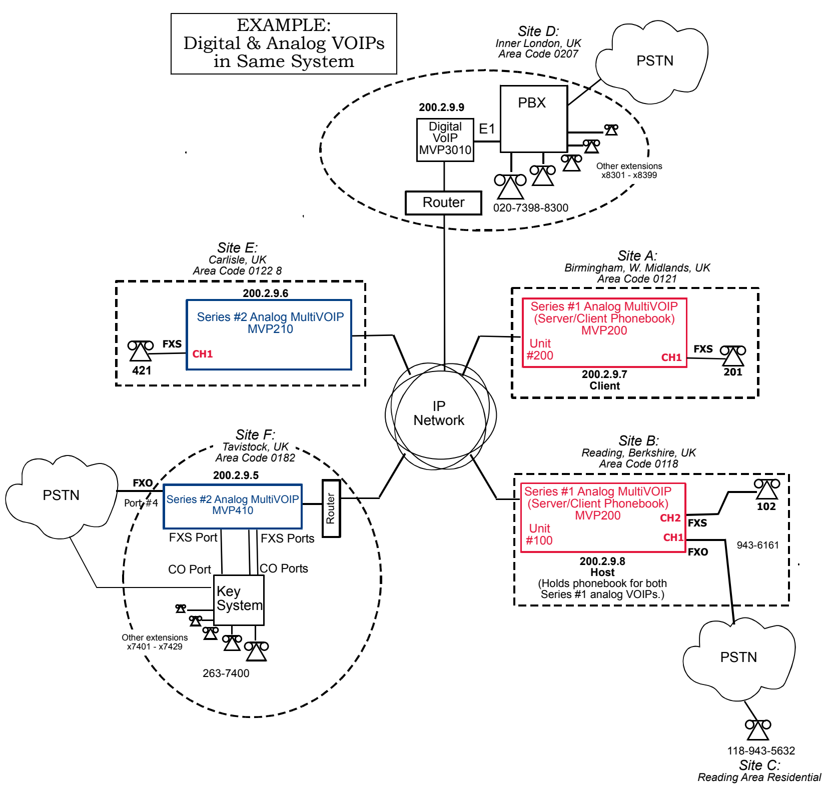


The Inbound PhoneBook for the Amsterdam VOIP is shown below.



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.

<table>
<tr><th colspan="4">Phone Book for Analog VOIP Host Unit (Site B)
(continued)</th></tr>
<tr><th>VOIP Dir #
-OR-
Destination Pattern</th><th>IP Address</th><th>Channel</th><th>Comments</th></tr>
<tr><td>0207
xxx
xxxx</td><td>200.2.9.9</td><td>0
(Note 2.)</td><td>Gives remote voip users access to phone numbers in 0207 area code (Inner London) in which Site D is located.</td></tr>
<tr><td>0208
xxx
xxxx</td><td>200.2.9.9</td><td>0
(Note 2.)</td><td>Gives remote voip users access to phone numbers in 0208 area code (Outer London) for which calls are local from Site D (Inner London).</td></tr>
<tr><td colspan="4">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 402-263-7430 through 402-263-7499 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 0182263740, 0182263741, and 0182263742. In this way, calls to 0182-263-7430 through 0182-263-7499 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.</td></tr>
</table>

The Outbound PhoneBook of the MVP3010 is shown below.

Outbound Phone Book for MVP3010 Digital VOIP (Site D)				
Destin. Pattern	**Remov e 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. Because call is completed at key system, abbreviated dialing (3-digits) is not workable. Human operator or auto-attendant is needed to complete these calls.
0182 263 741	741.	0	
0182 263 742	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 (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

Dial 101.
Hear dial tone from Site B.
Dial 9435632.
Await completion. Talk.

Site A calling Site C, Method 2

Dial 101#9435632
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, MVP130FXS, MVP210, MVP410, and MVP810) 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.

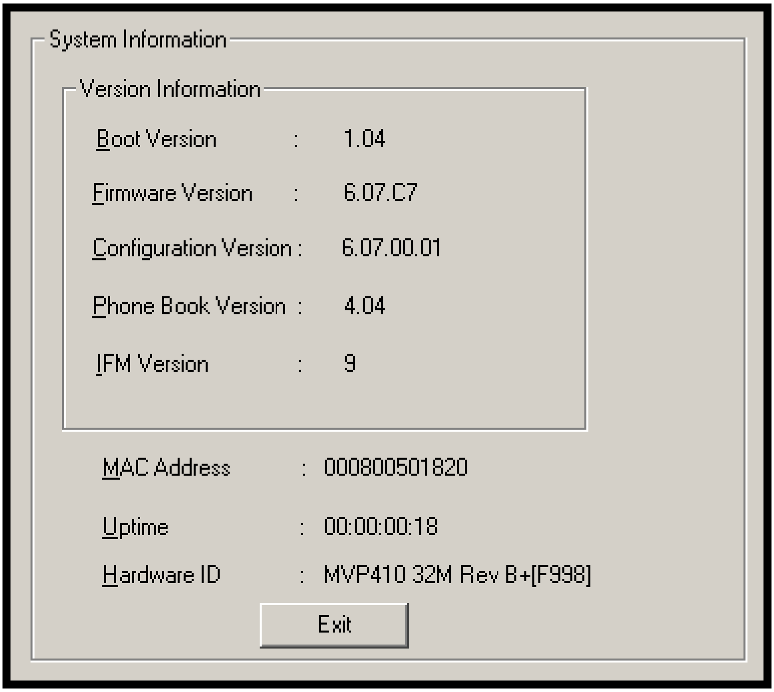
Chapter 10: Operation and Maintenance

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.

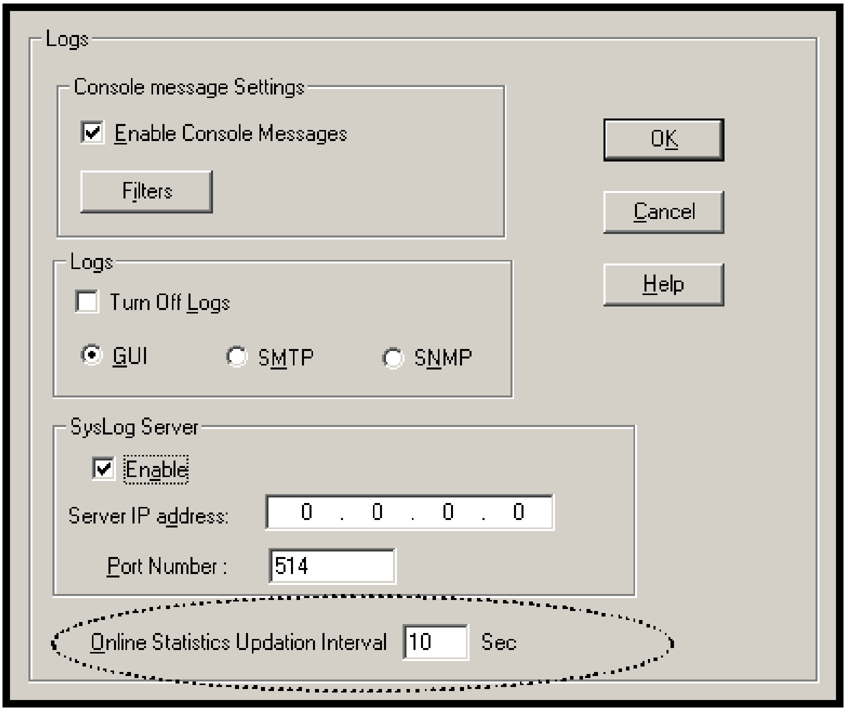
System Information screen

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**.



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

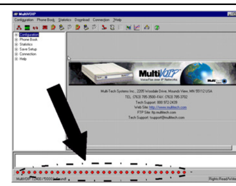
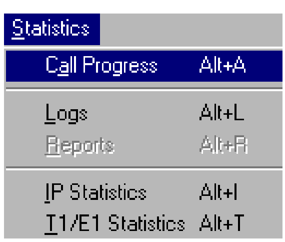
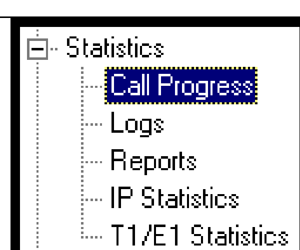
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



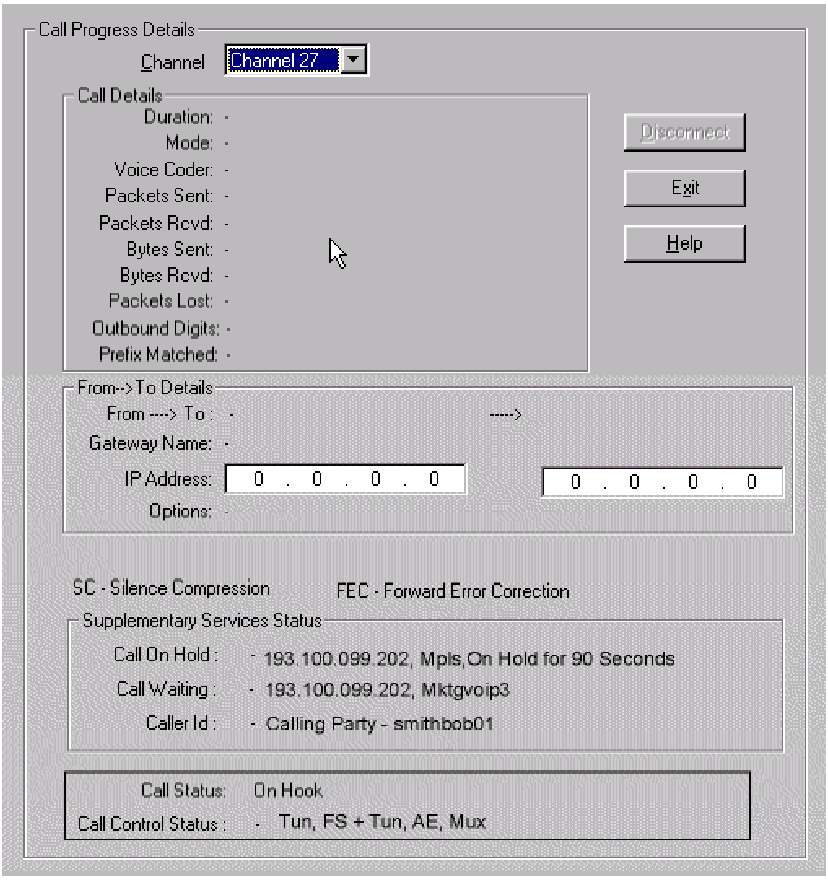
Statistics Screens

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

<table>
<tr><th colspan="2">Accessing Call-Progress Statistics</th></tr>
<tr><td colspan="2">Channel Icons (Main Screen Lower Left)</td></tr>
<tr><td colspan="2">Channel icons are green when data traffic is present, red when idle. </td></tr>
<tr><td colspan="2">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.</td></tr>
<tr><td></td><td></td></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><code>Alt + A</code></td><td>
</td></tr>
</table>

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.

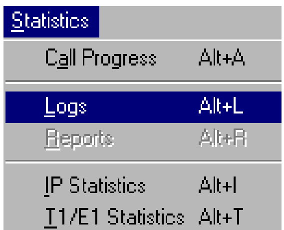
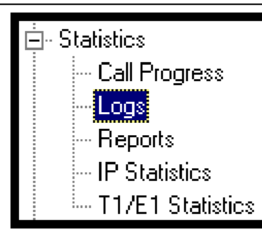
Call Progress Details: Field Definitions (cont'd)		
From – To Details		Description
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that handled this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address from which the call was received.
Options	SC, FEC	Displays VOIP transmission 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.

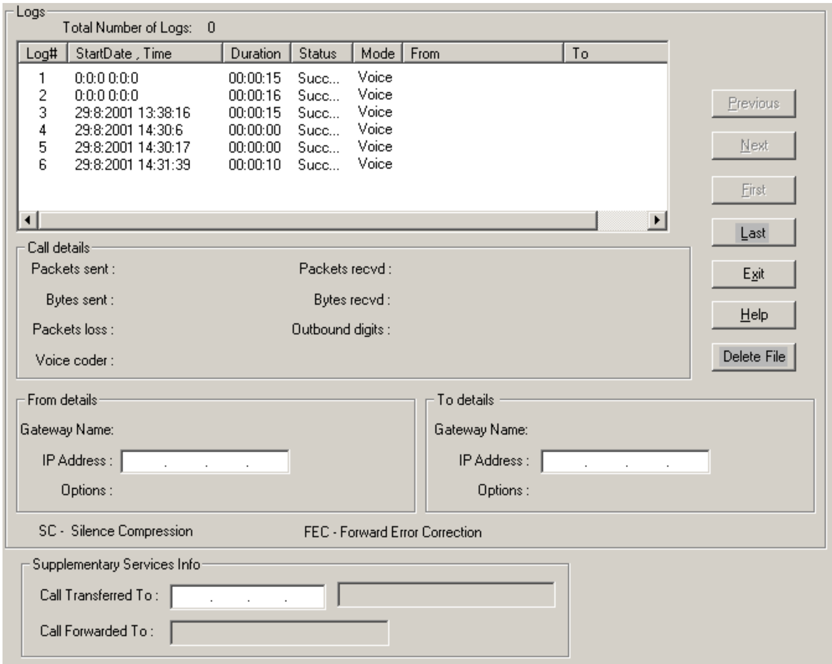
Call Progress Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Supplementary Services Status		
Caller ID	There are four values: "Calling Party + *identifier*"; "Alerting Party + *identifier*"; "Busy Party + *identifier*"; and "Connected Party + *identifier*"	This field shows 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



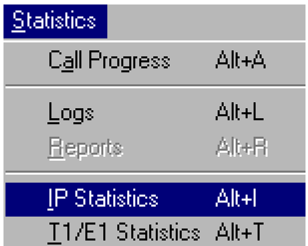
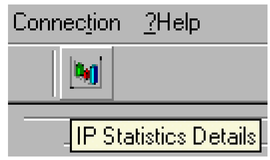
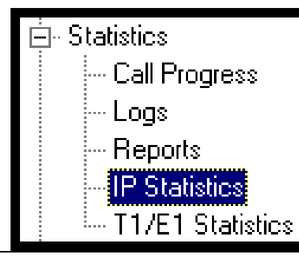
Logs Screen Details: Field Definitions		
Field Name	**Values**	**Description**
Log # column	1 or higher	All calls are assigned an event number in chronological order, with the most recent call having the highest event number.
Start Date,Time column	dd:mm:yyyy hh:mm:ss	The starting time of the call (event). 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		
Previous	--	Displays log entry before currently selected one.
Next	--	Displays log entry after currently selected one.
First	--	Displays first log entry
Last	--	Displays last log entry.
Delete File	--	Deletes selected log file.
Call Details		
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 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.

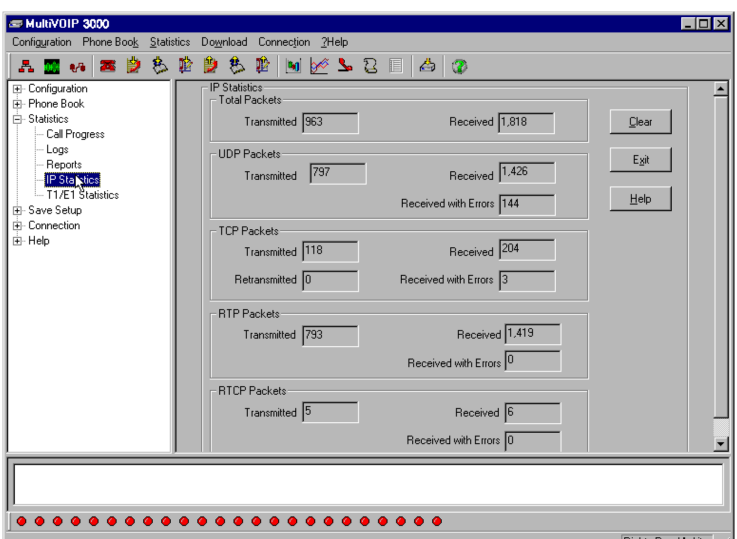
Logs Screen Details: Field Definitions (cont'd)		
Supplementary Services Info		
Call Transferred To	phone number string	Number of party called in transfer.
Call Forwarded To	phone number string	Number of party called in forwarding.

About IP Statistics

Accessing IP Statistics

Pulldown	Icon
	
Shortcut	**Sidebar**
Alt + I	

IP Statistics Screen



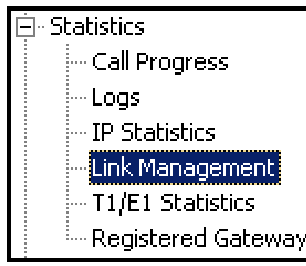
IP Statistics: Field Definitions		
Field Name	**Values**	**Description**
		UDP versus TCP. (User Datagram Protocol versus Transmission Control Protocol). UDP provides unguaranteed, connectionless transmission of data across an IP network. By contrast, TCP provides reliable, connection-oriented transmission 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.

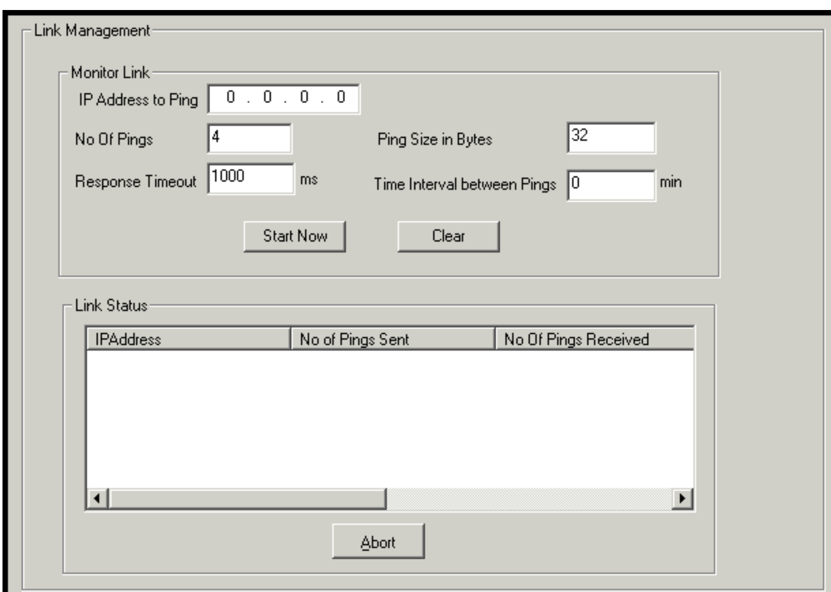
IP Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Total Packets (cont'd)		Sum of data packets of all types.
Received with Errors	integer value	Total number of error-laden packets received by this VOIP gateway since the last "clearing" or resetting of the counter within 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.
Transmit ted	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.
Transmit ted	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 Link Management

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

Accessing Link Management	
Pulldown	
none	
Shortcut/Icon	**Sidebar**
none/none	

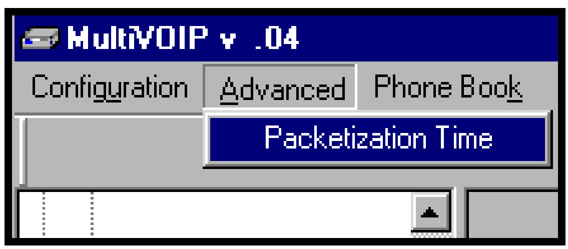
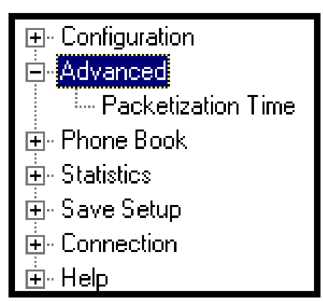


Link Management screen Field Definitions		
Field Name	**Values**	**Description**
Monitor Link fields		
IP Address to Ping	a.b.c.d 0-255	This is the IP address of the target endpoint to be pinged.
No. of Pings	1-999	This field determines how many pings will be generated by the Start Now command.
Response Timeout	500 – 5000 milliseconds	The duration after which a ping will be considered to have failed.
Ping Size in Bytes	32 – 128 bytes	This field determines how long or large the ping will be.
Timer Interval between Pings	0 or 30 – 6000 minutes	This field determines how long of a wait there is between one ping and the next.
Start Now command button	--	Initiates pinging.
Clear command button	--	Erases ping parameters in Monitor Link field group and restores default values.

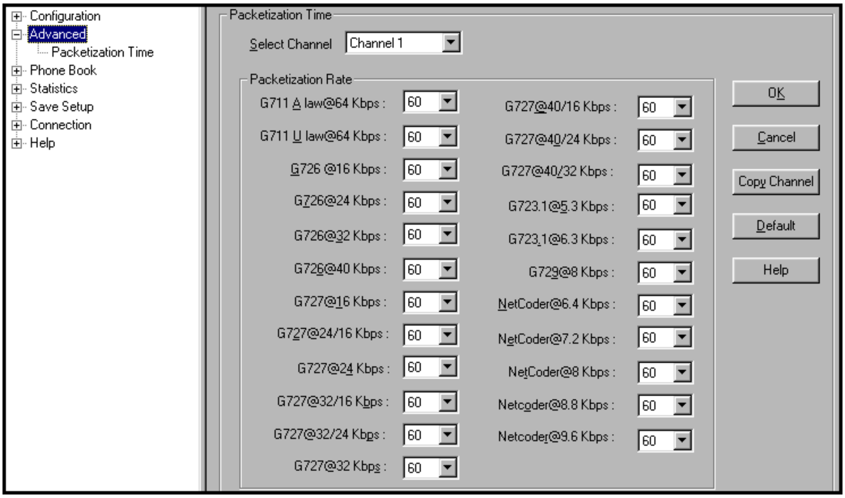
Link Management screen Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Link Status Parameters		These fields summarize the results of pinging.
IP Address column	a.b.c.d 0-255	Target of ping.
No. of Pings Sent	as listed	Number of pings sent to target endpoint.
No. of Pings Received	as listed	Number of pings received by target endpoint.
Round Trip Delay (Min/Max/ Avg)	as listed, in milliseconds	Displays how long it took from time ping was sent to time ping response was received.
Last Error	as listed	Indicates when last data error occurred.

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

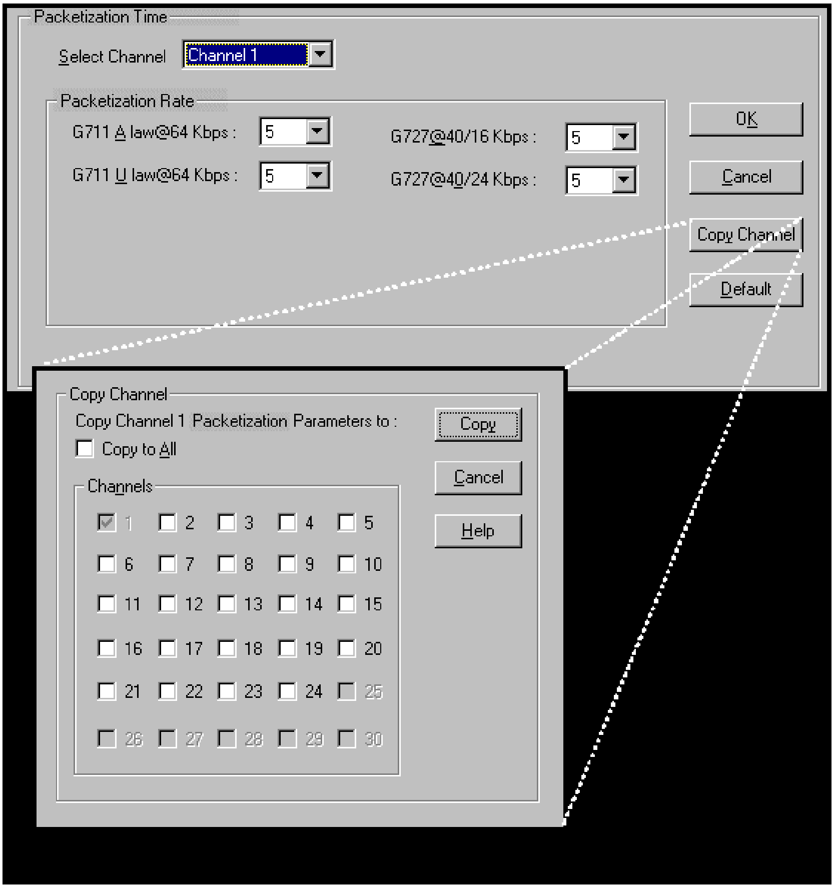


Packetization rates can be set separately for each channel.

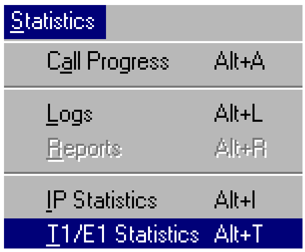
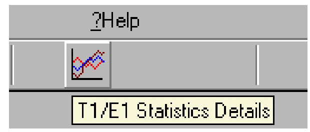
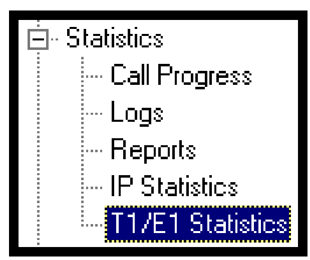
The table below presents the ranges and increments for packetization rates.

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.



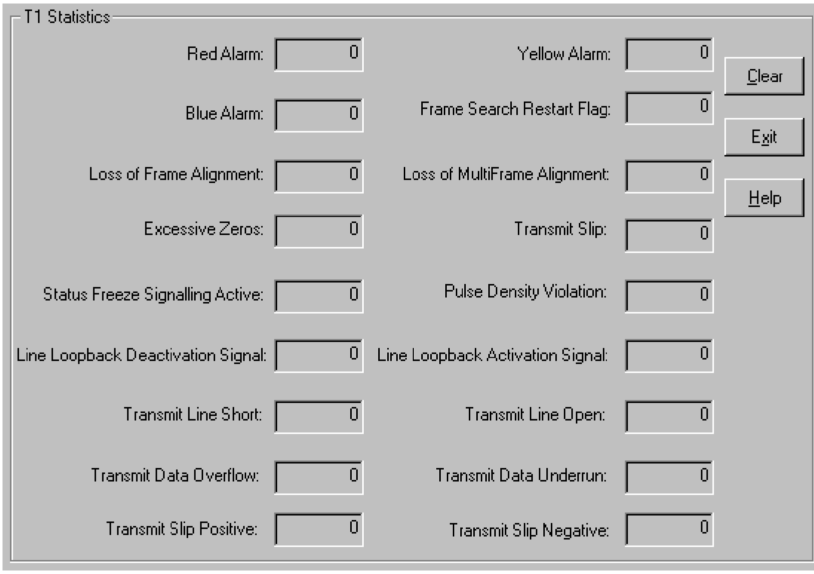
About T1/E1 and BRI Statistics

Accessing T1 Statistics	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Alt + T`	

The T1 and E1 Statistics screens are only accessible and applicable for the MVP2410, and MVP3010.

The BRI statistics screens are only accessible and applicable for the MVP410ST and MVP810ST .

T1 Statistics Screen

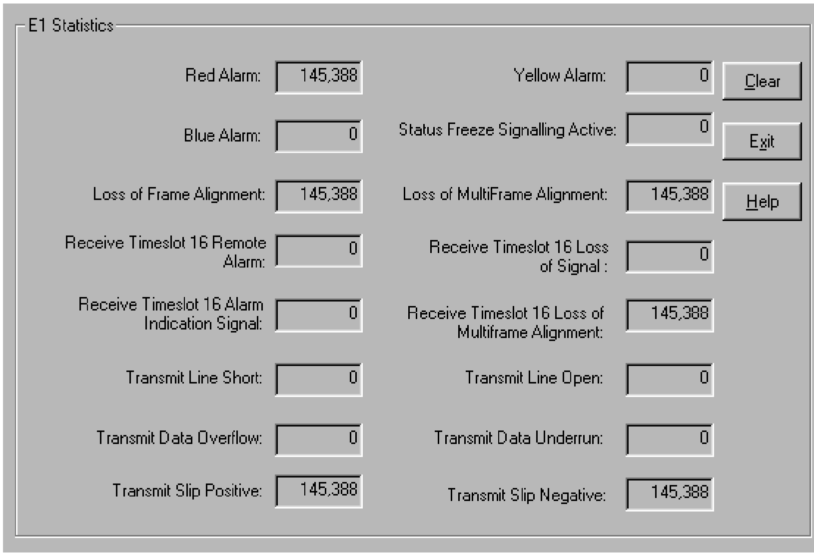


T1 Statistics: Field Definitions		
Field Name	**Values**	**Description**
Red Alarm	Integer tally of alarms counted since last reset.	The alarm condition declared when a device receives no signal or cannot synchronize to the signal being received. A Red Alarm is generated if the incoming data stream has 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

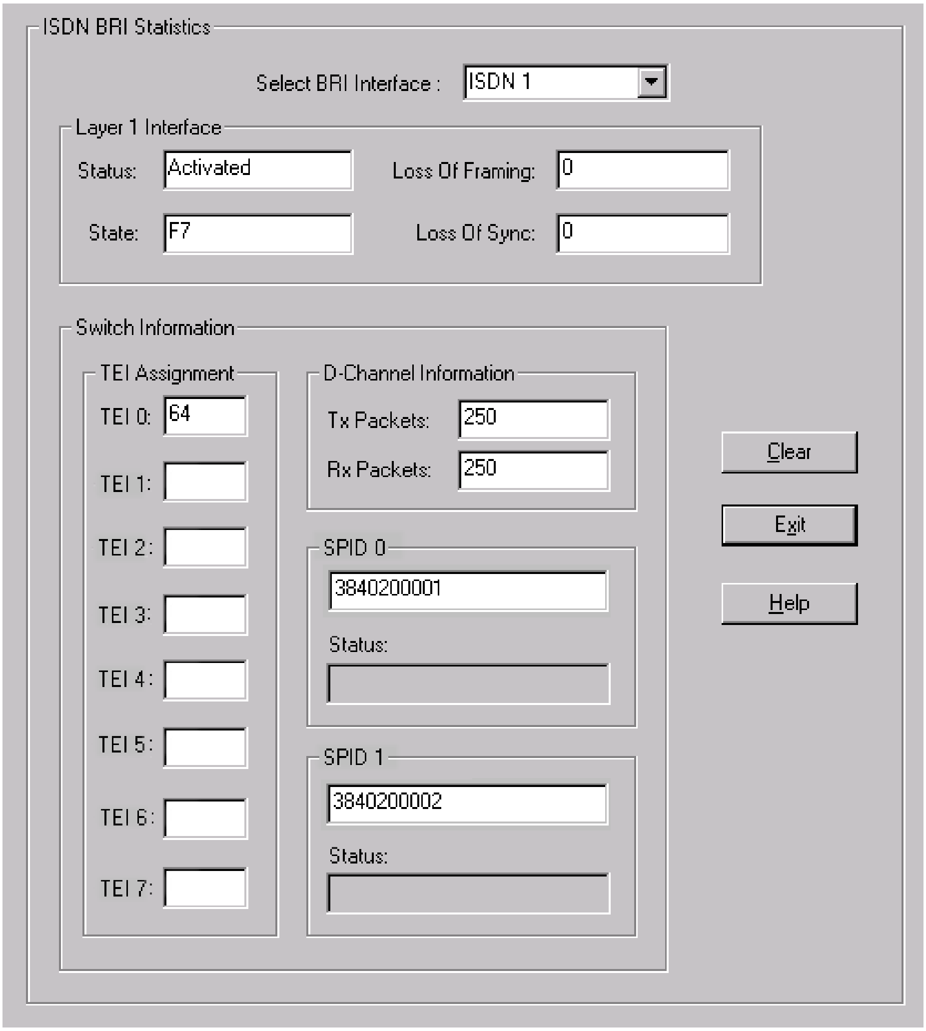


E1 Statistics: Field Definitions		
Field Name	**Values**	**Description**
Red Alarm	Integer tally of alarms counted since last reset.	The alarm condition declared when a device receives no signal or cannot synchronize to the signal being received. A Red Alarm is generated if the incoming data stream has 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.

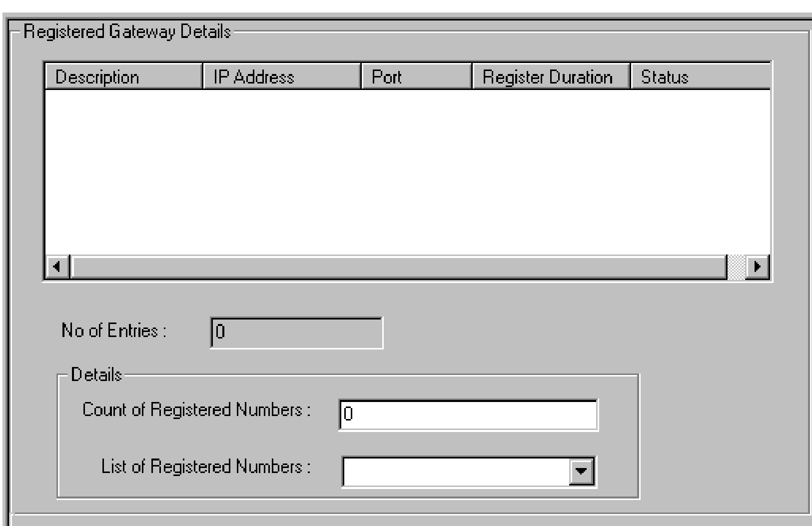
ISDN BRI Statistics: Field Definitions (continued)		
Field Name	**Values**	**Description**
Switch Information: TEI Assignment		
TEI 0 through TEI 7	0-63 (point-to-point assignments) 64-126 (automatic assignments)	Displays the value for each TEI assigned to the BRI port. The TEI (Terminal Endpoint Identifier) 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.

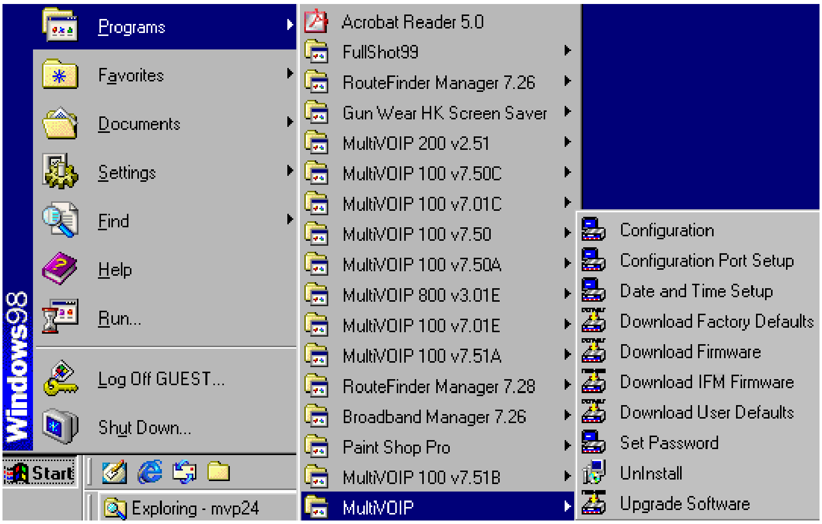
Accessing Registered Gateway Details	
Pulldown	**Icon**
Shortcut	**Sidebar**
	



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.
Configuration Port Setup	Select this to access the **COM Port Setup** screen of the MultiVOIP Configuration program.
Date and Time Setup	Select this for access to set calendar/clock used for data logging.

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 IFM Firmware	Select this to download new versions of IFM firmware as enhancements become available. The Interface Module (IFM) is the telephony interface for analog MultiVOIP units (MVP130, MVP130FXS, MVP210, MVP410, MVP810). There is one IFM for each channel of the MultiVOIP unit. For each channel, the IFM handles the analog signals to and from the attached telephone, PBX or CO line.
Download User Defaults	To be used after a full set of parameter values, values specified by the user, have been saved (using Save Setup). This command loads the saved user defaults into the MultiVOIP.
Set Password	Select this to create a password for access to the MultiVOIP software programs (**Program** group commands, Windows 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.

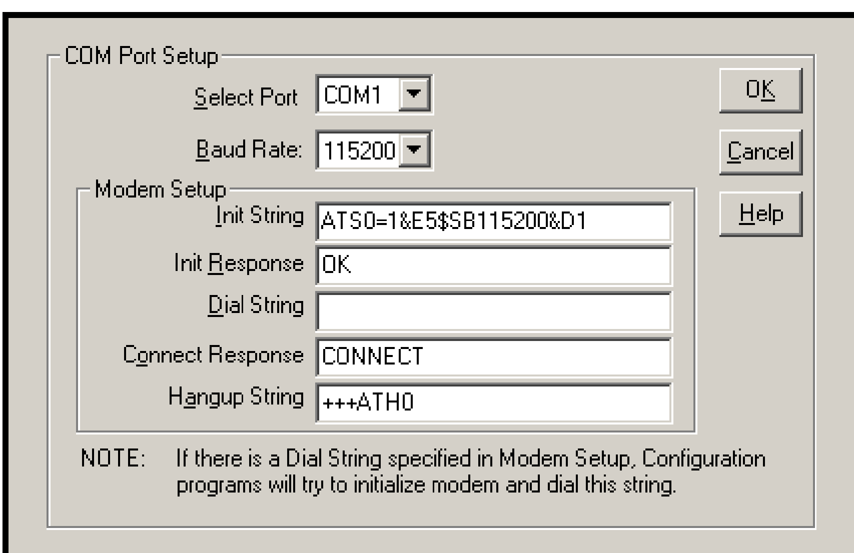
Generally, updated firmware must be downloaded from the MultiTech web/FTP site to the PC before it can be loaded from the PC to the MultiVOIP.

Configuration Option

The "Configuration" option in the MultiVOIP Program menu launches the MultiVOIP Configuration software program.

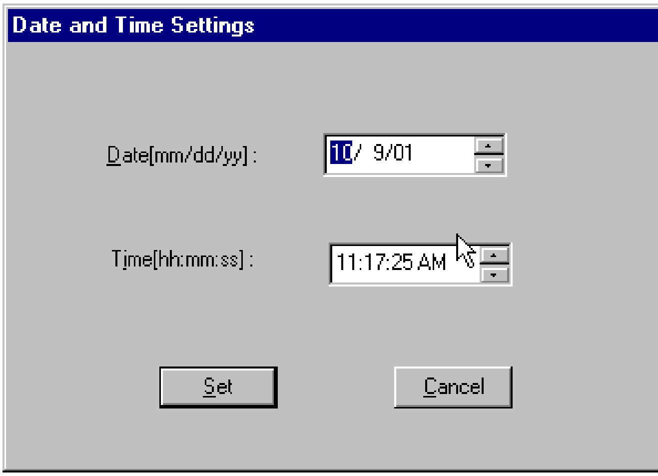
Configuration Port Setup

The Configuration Port Setup option in the MultiVOIP Program menu brings up the **COM Port Setup** screen of the MultiVOIP configuration software.



Date and Time Setup

The dialog box below allows you to set the time and date indicators of the MultiVOIP system.



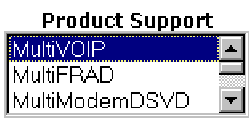
Obtaining Updated Firmware

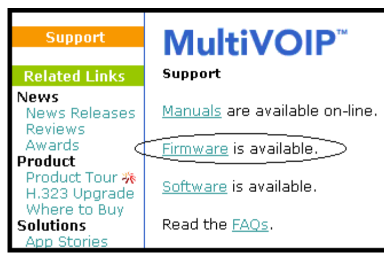
Generally, updated firmware must be downloaded from the MultiTech web/FTP site to the user's PC before it can be downloaded from that PC to the MultiVOIP.

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**.

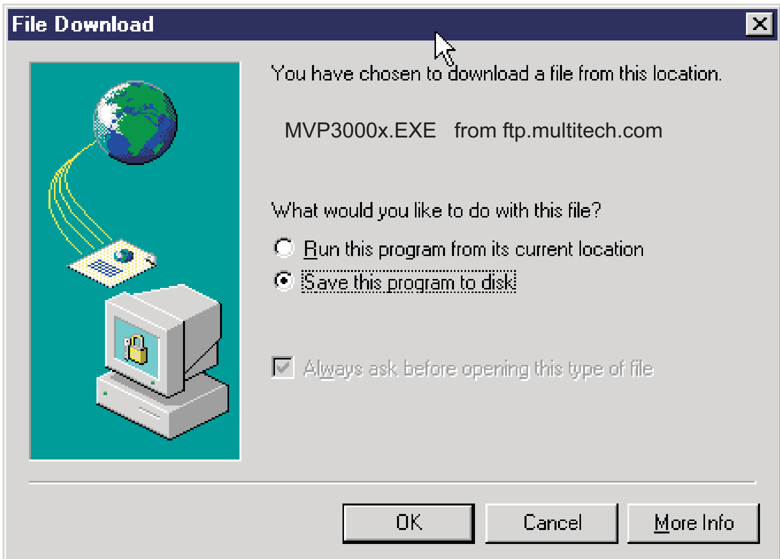
If you conduct a search, for example, on the word "MultiVoip," you will be directed to a list of firmware that can be downloaded.

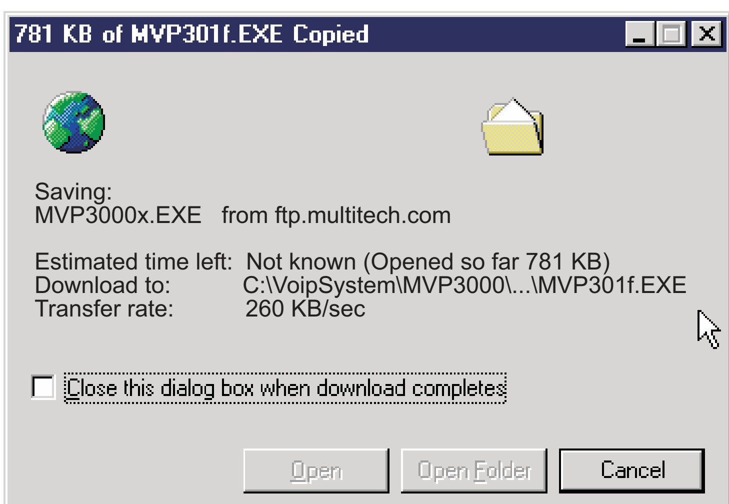
If you choose **Support**, you can select "MultiVoip" in the **Product Support** menu and then click on **Firmware** to find MultiVOIP resources.



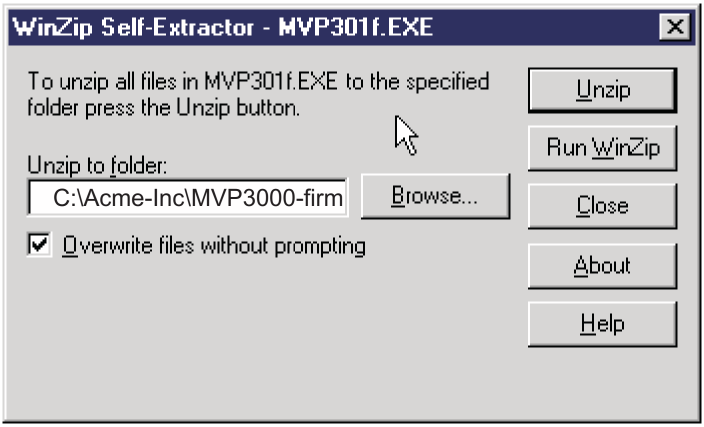


Once the updated firmware has been located, it can be downloaded from the web/ftp site using normal PC/Windows procedures. While the next 3 screens below pertain to the MVP3010, similar screens will appear for any MultiVOIP model described in this manual.





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

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.

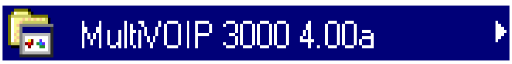
When performing a piecemeal software upgrade (whether from the Windows GUI or web browser GUI), follow these steps in order:

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).



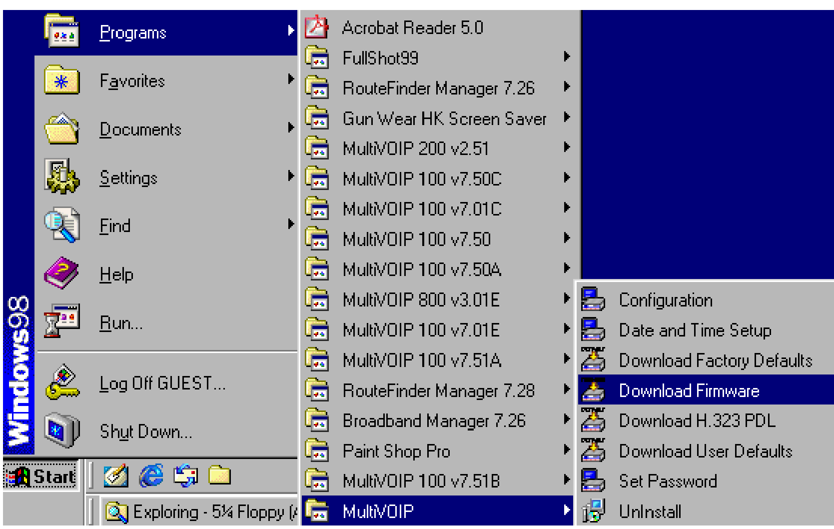
When a new firmware version is installed, the MultiVOIP software can be upgraded in one step using the **Upgrade Software** command, or piecemeal using the **Download Firmware** command and the **Download Factory Defaults** command.

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.

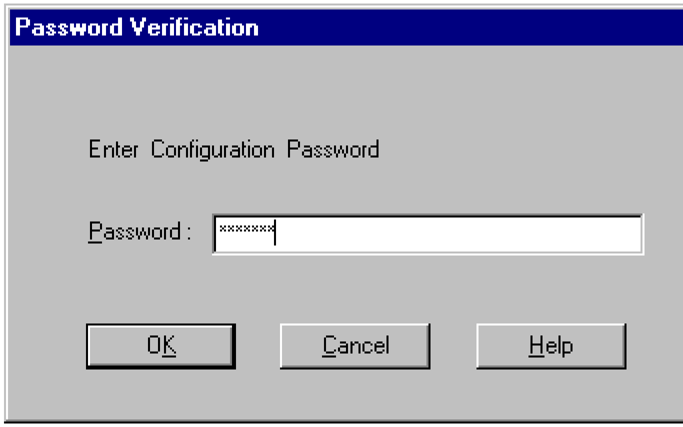
Downloading Firmware

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**.

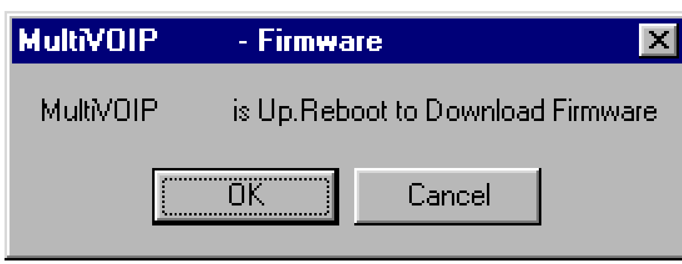


3. If a password has been established, the **Password Verification** screen will appear.



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

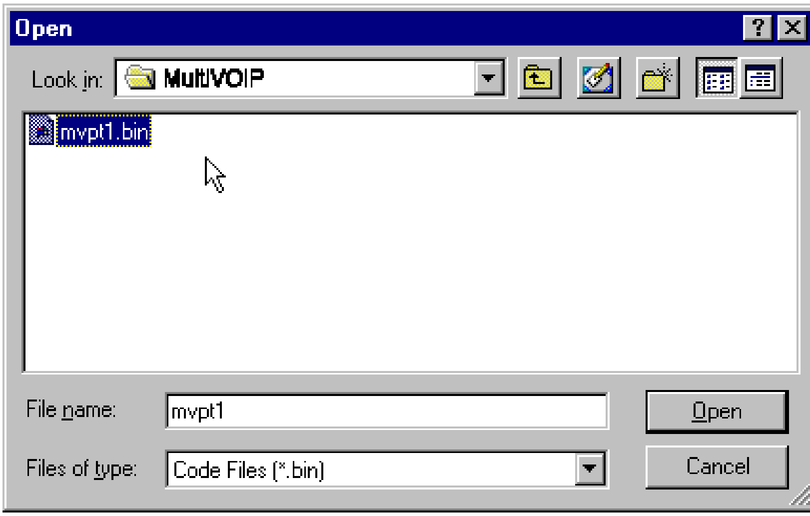
4. The **MultiVOIP ___- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"



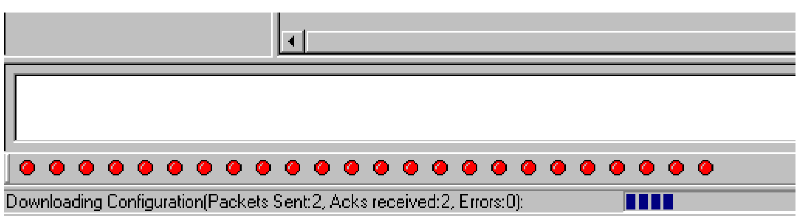
Click **OK** to download the firmware.

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.



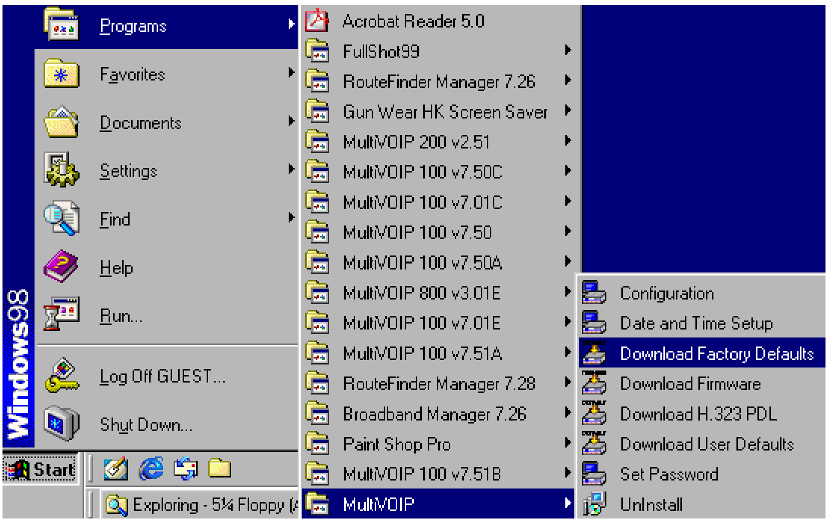
The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

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

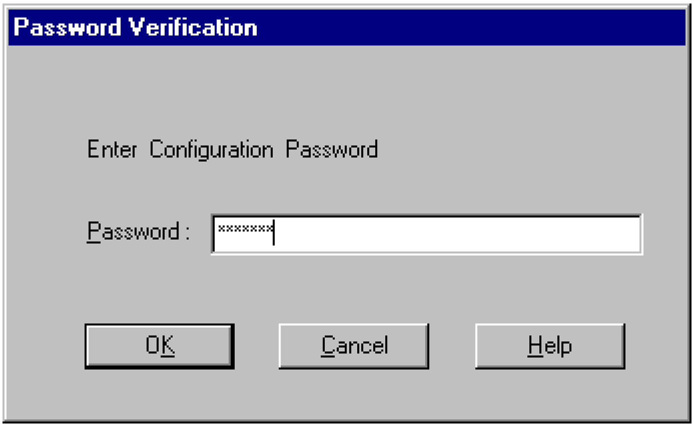
Downloading Factory Defaults

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**.

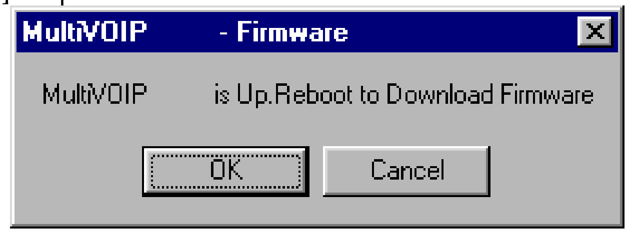


3. If a password has been established, the **Password Verification** screen will appear.



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

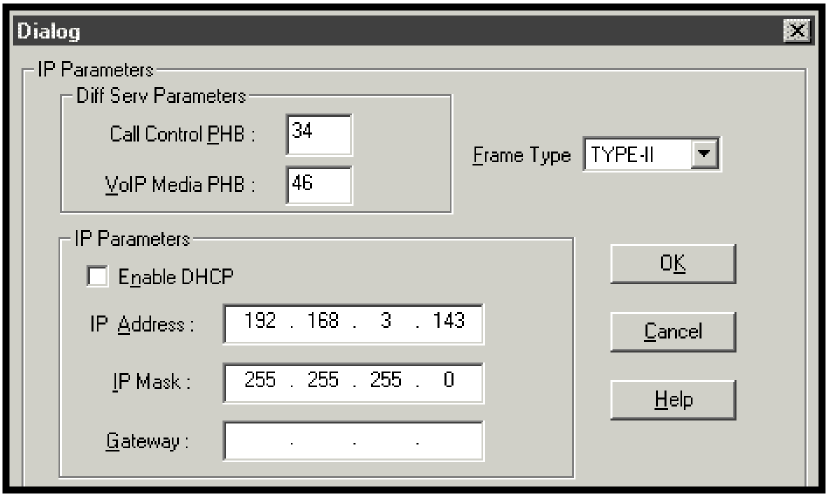
4. The **MVP____- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"



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

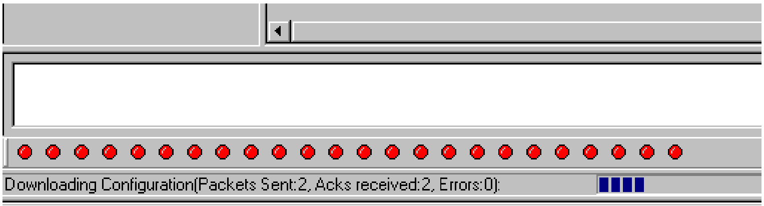
The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

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.

Downloading IFM Firmware (Analog Voips only)

The Interface Module (IFM) is the telephony interface for analog MultiVOIP units (MVP130, MVP130FXS, MVP210, MVP410, MVP810). There is one IFM for each channel of the MultiVOIP unit. For each channel, the IFM handles the analog signals to and from the attached telephone, PBX or CO line. The IFM communicates with the main processor indicating the status of the telephone line. For example, it might indicate that a phone is off hook (FXS) or that an incoming ring is present (FXO). The IFM receives operating instructions from the voip's main processor. For example, the IFM might be instructed to ring the phone (FXS) or seize the line (FXO). The IFM contains a codec (coder/decoder) to convert the incoming audio to a PCM stream (pulse code modulation) which it sends to the DSP (digital signal processor). The IFM's codec also converts outgoing PCM to audio.

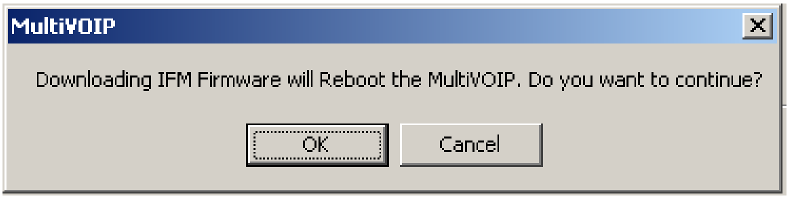
The firmware of the IFMs will change from time to time and you may need to upgrade the firmware on your MultiVOIP unit. To do so, follow these instructions.

1. In the **System Information** screen of the MultiVOIP Configuration software, check the version number of the IFM firmware already installed on the MultiVOIP unit. Write down the version number.

2. Exit the Configuration software program. The MultiVoip Configuration program must be off when invoking the **Download IFM Firmware** command. If it is on, the command will not work.

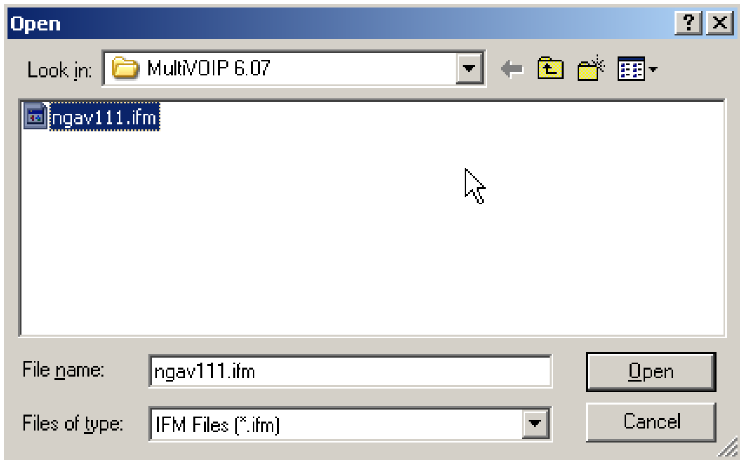
3.To invoke the **Download IFM Firmware** command, go to **Start | Programs | MVP____ x.xx | Download IFM Firmware**.

4. A warning window will appear: "Downloading IFM Firmware will reboot the MultiVOIP. Do you want to continue?" Click **OK**.

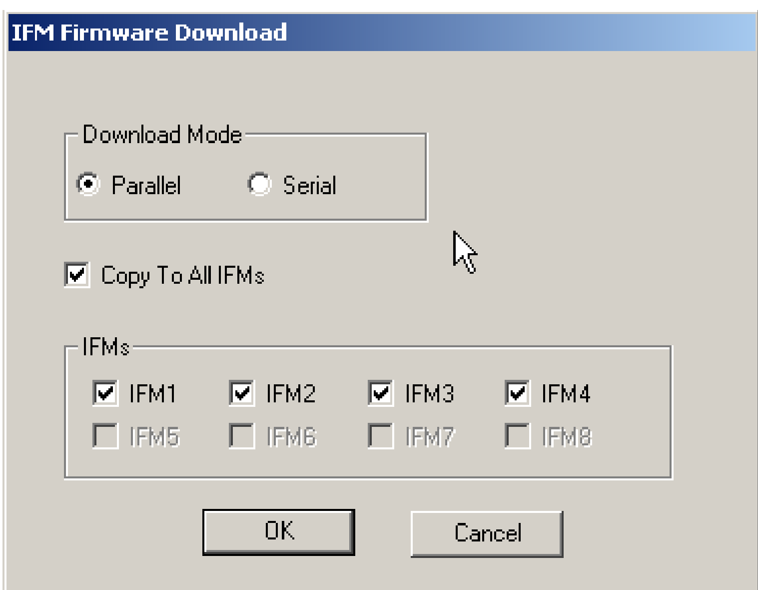


4. The "Boot" LED on the front panel of the MultiVOIP will come on.

5. The software will search for an IFM firmware file to use to upgrade the system. If the file found represents firmware newer than that already installed on the MultiVOIP (or if you want to overwrite the same version of firmware) click **Open**.

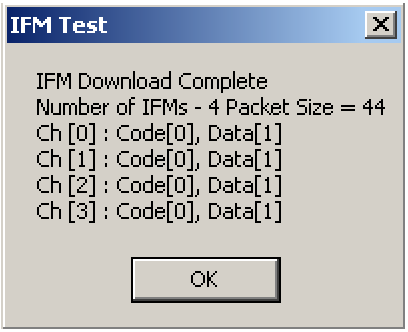


6. The **IFM Firmware Download** screen will appear. Select "Copy to All IFMs" and click **OK**. (Only in very special circumstances would different IFMs in the same voip be loaded with different IFM firmware.)



7. The main MultiVOIP Configuration screen will appear. Progress bars can be seen at the bottom of the screen while files are being copied.

8. Then a completion screen entitled **IFM Test** will appear.



Click **OK**.

9. The MultiVOIP will reboot itself. When the reboot is complete, the MultiVOIP Configuration screen will close.

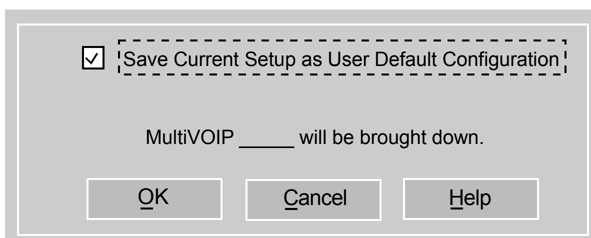
10. The IFM firmware downloading process is complete.

Setting and Downloading User Defaults

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

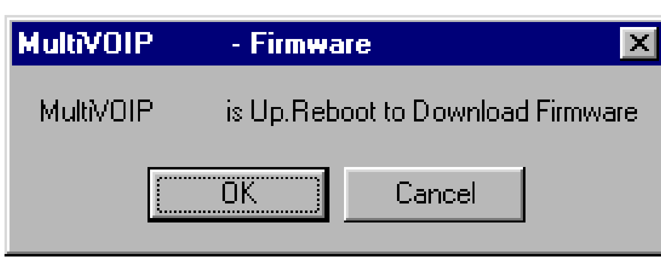
1. Before you can invoke the Download User Defaults command, you must first save a set of configuration parameters by using the **Save Setup** command in the sidebar menu of the MultiVOIP software.

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**.



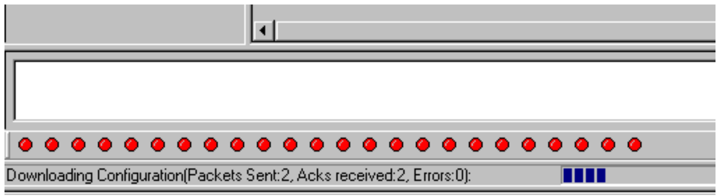
A user default file will be created.

3. The **MVP____- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"

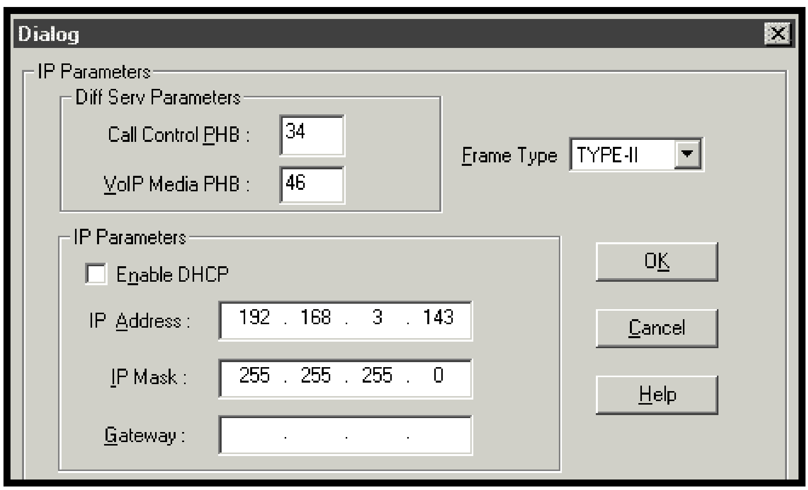


Click **OK** to download the factory defaults. The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

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.

Setting a Password (Windows GUI)

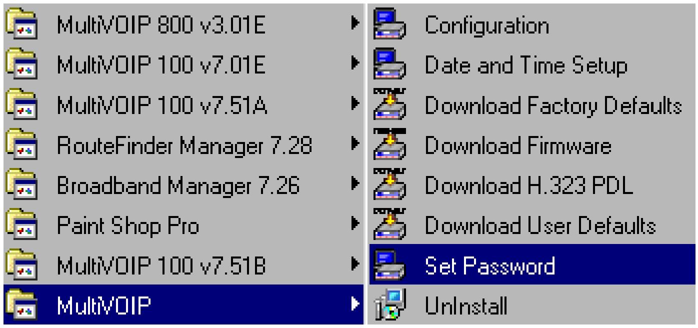
After a user name has been designated and a password has been set, that password is required to gain access to any functionality of the MultiVOIP software. Only one user name and password can be assigned to a voip unit. The user name will be required when communicating with the MultiVOIP via the web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

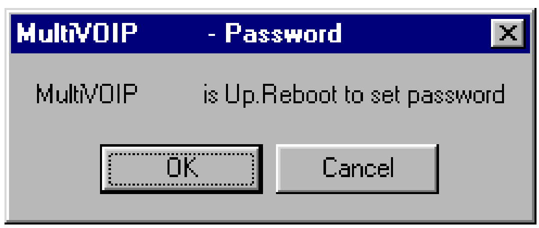
---

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**.



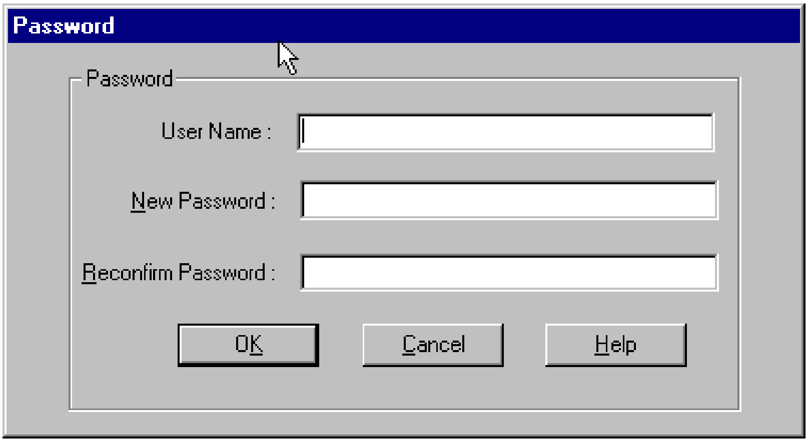
3. You will be prompted to confirm that you want to establish a password, which will entail rebooting the MultiVOIP (which is done automatically).



Click **OK** to proceed with establishing a password.

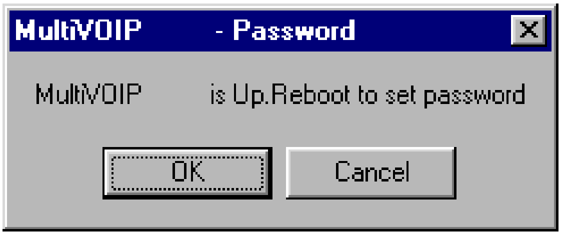
4. The **Password** screen will appear. If you intend to use the FTP Server function that is built into the MultiVOIP, enter a user name. (A User Name is not needed to access the local Windows GUI, the web browser GUI, or the commands in the **Program** group.) Type your password in the **Password** 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.



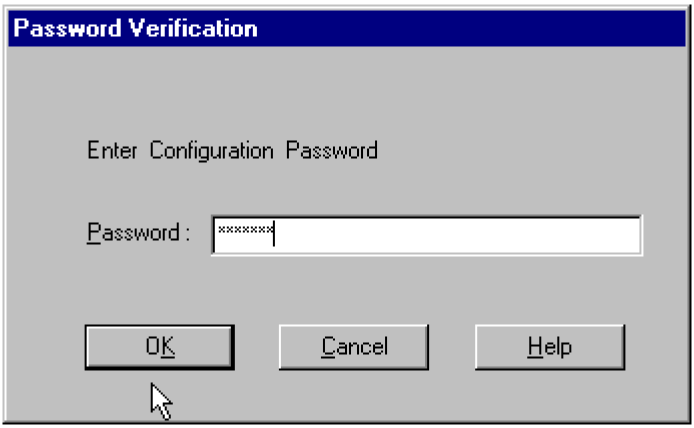
Click **OK**.

5. A message will appear indicating that a password has been set successfully.



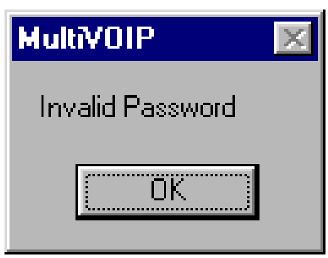
After the password has been set successfully, the MultiVOIP will re-boot itself and, in so doing, its **BOOT** LED will light up.

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.

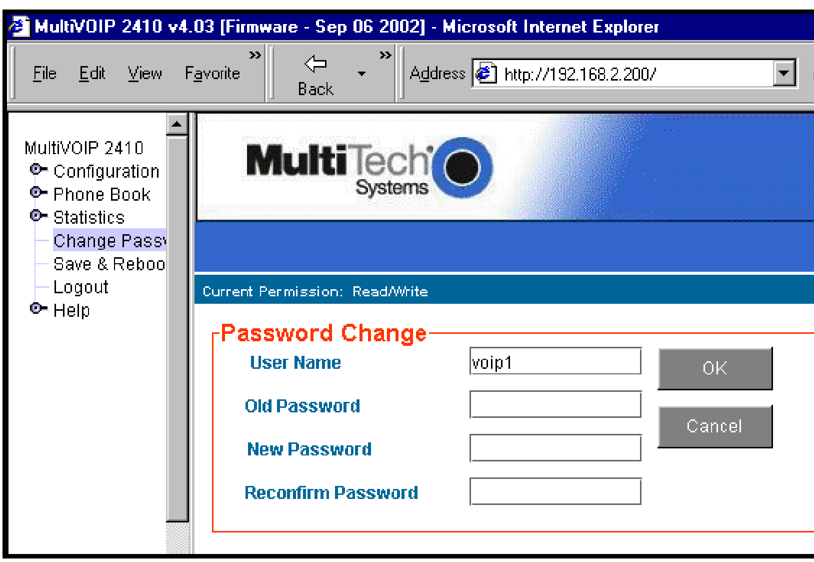
The MultiVOIP program will produce an error message if an invalid password is entered.



Setting a Password (Web Browser GUI)

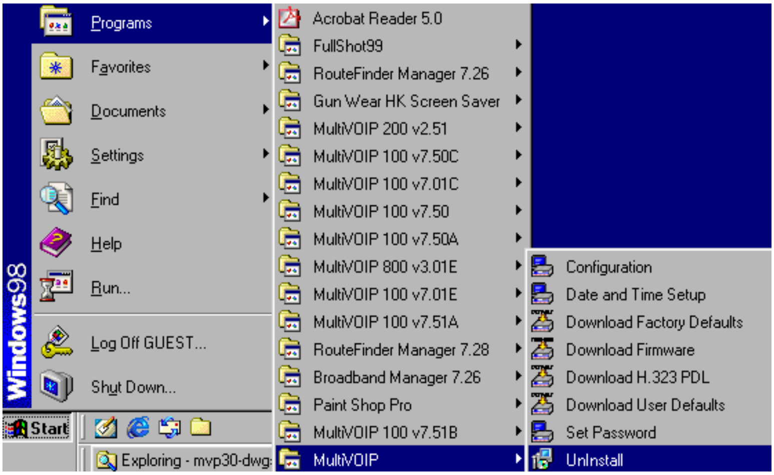
Setting a password is optional when using the MultiVOIP web browser GUI. Only one password can be assigned and it works for all MultiVOIP software functions (Windows GUI, web browser GUI, FTP server, and all Program menu commands, e.g., Upgrade Software – only the FTP Server function requires a User 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.

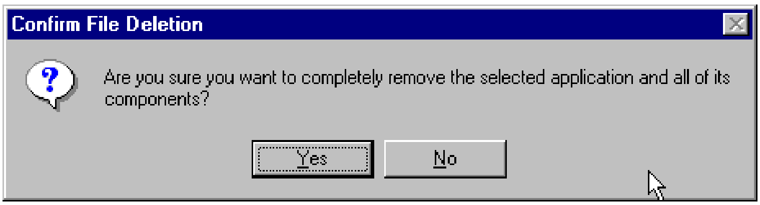


Un-Installing the MultiVOIP Software

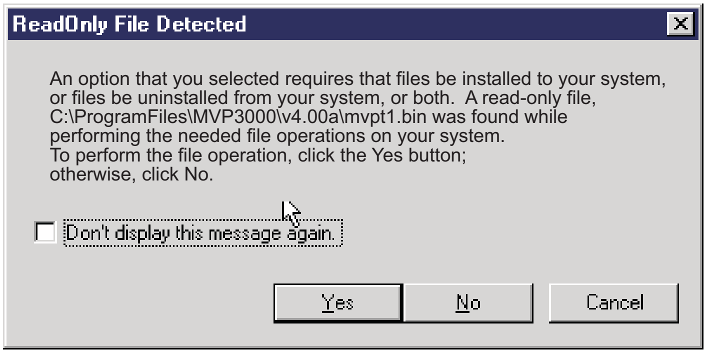
1. To un-install the MultiVOIP configuration software, go to **Start | Programs** and locate the MultiVOIP entry. Select **Uninstall MVP____ vx.xx** (versions may vary).



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**.

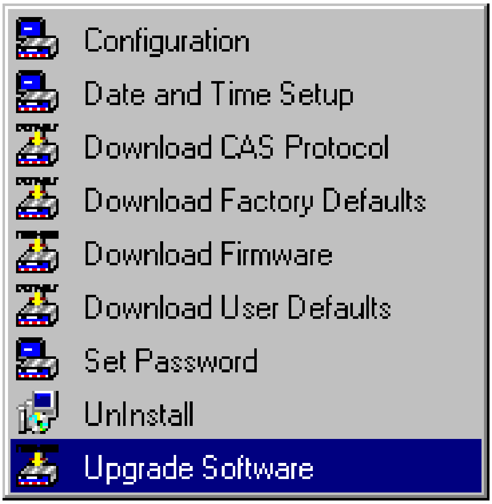


4. A completion screen will appear.

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 H.323 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.

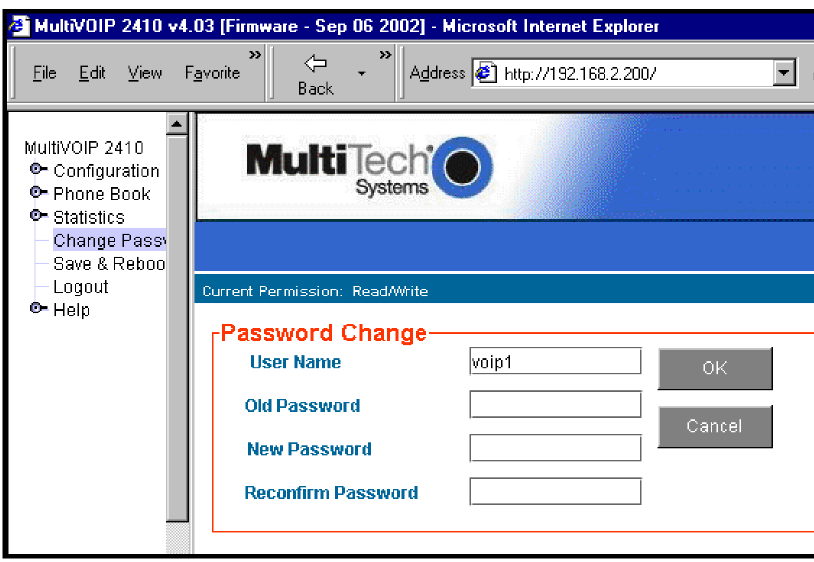
The phonebook downloading feature greatly reduces the data-entry required to establish inbound and outbound phonebooks for the voip units within a system. Although each MultiVOIP unit will require some unique phonebook entries, most will be common to the entire voip system. After the phonebooks for the 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.)

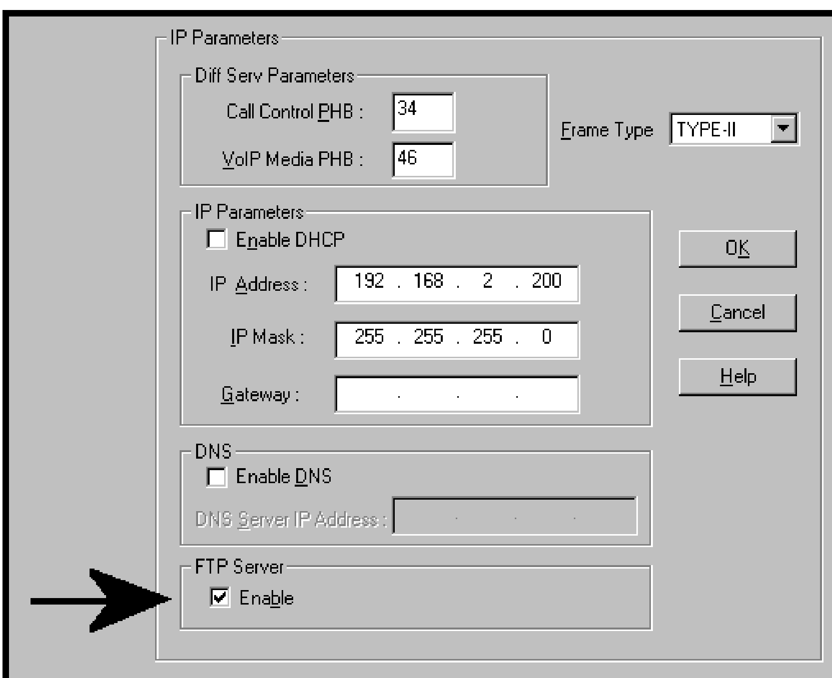


As shown above, the username and password can be set in the web GUI as well as in the Windows GUI.

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.

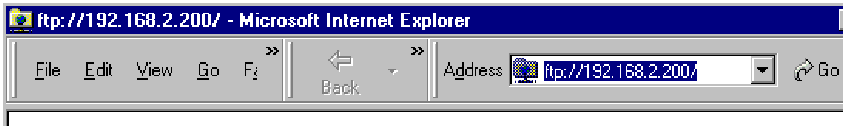
4. **Enable FTP Functionality**. Go to the **IP Parameters** screen and click on the "FTP Server: Enable" box.



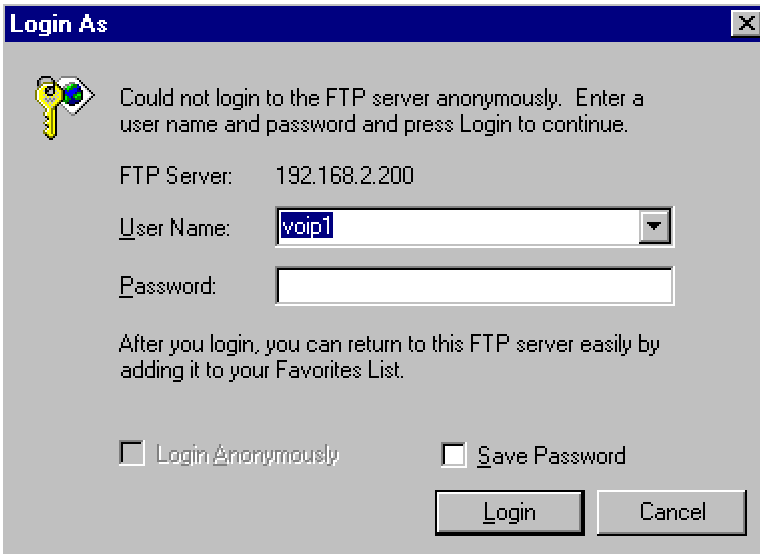
5. **Identify Files to be Updated**. Determine which files you want to update. Six types of files can be updated using the FTP feature. In some cases, the file to be transferred will have "Ftp" as the part of its filename just before the suffix (or extension). So, for example, the file "mvpt1Ftp.bin" 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.

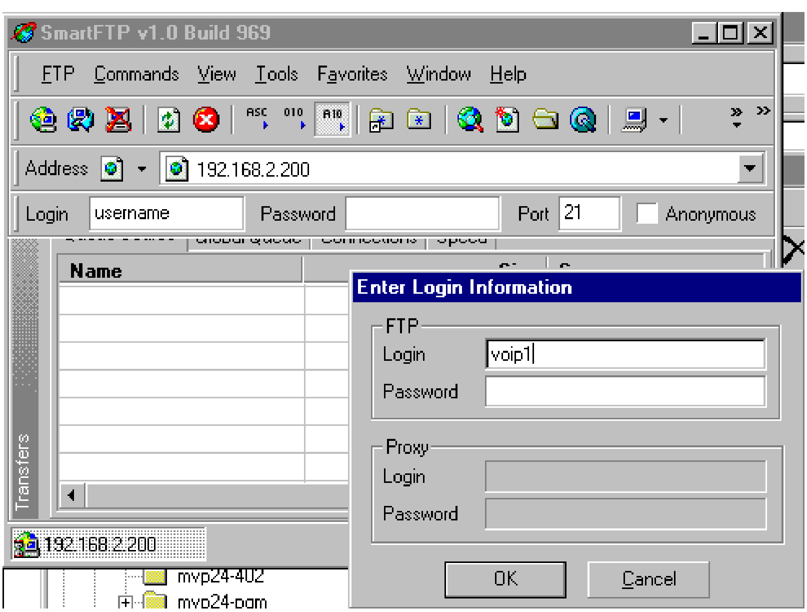
File Type	File Names	Description
firmware "bin" file	mvpt1Ftp.bin	This is the MultiVOIP firmware file. Only one file of this type will be in the directory.
factory defaults	fdefFtp.cnf	This file contains factory default settings for user-changeable configuration parameters. Only one file of this type 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).



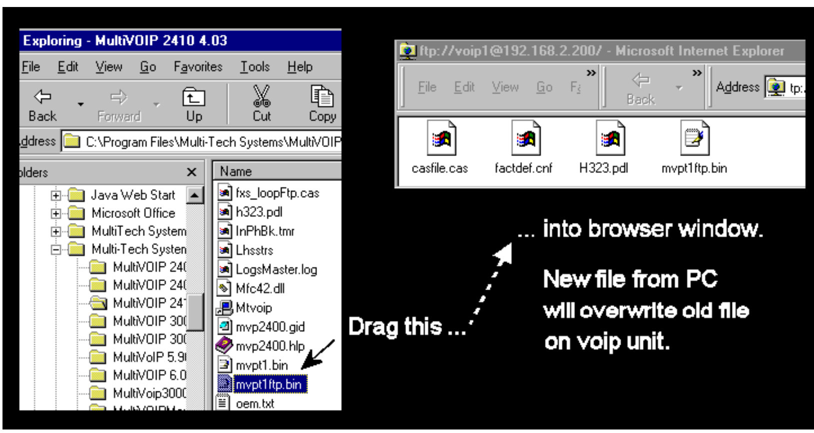


8. **Invoke Download**. Downloading can be done with a web browser or with an FTP client program.

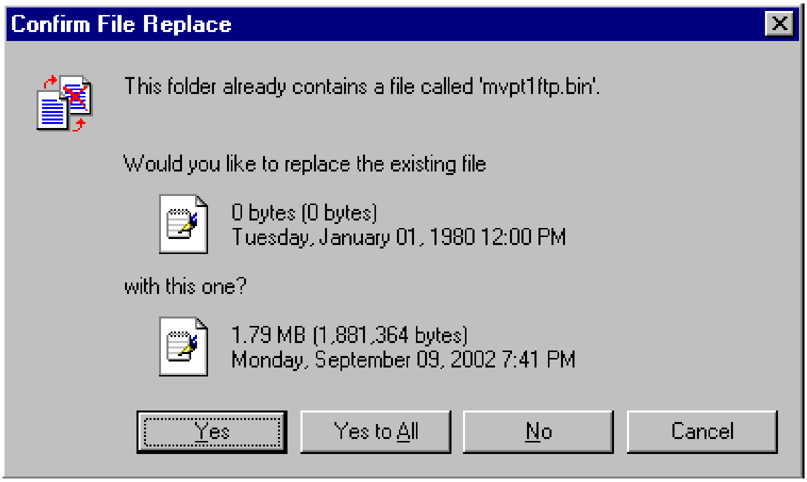
8A. Download with Web Browser.

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).

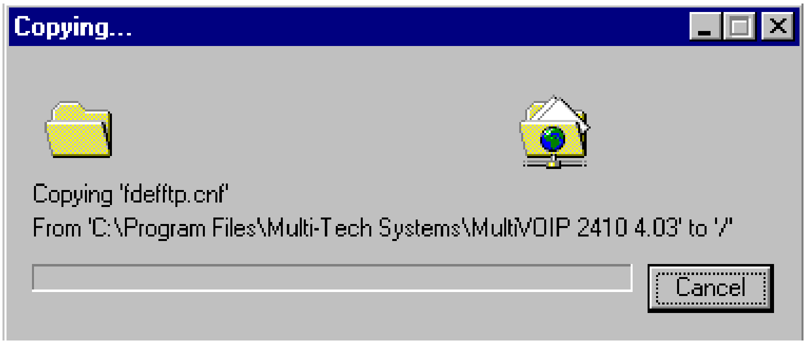
8A2. Drag-and-drop files from the local Windows browser (e.g., Windows Explorer) to the web browser.



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



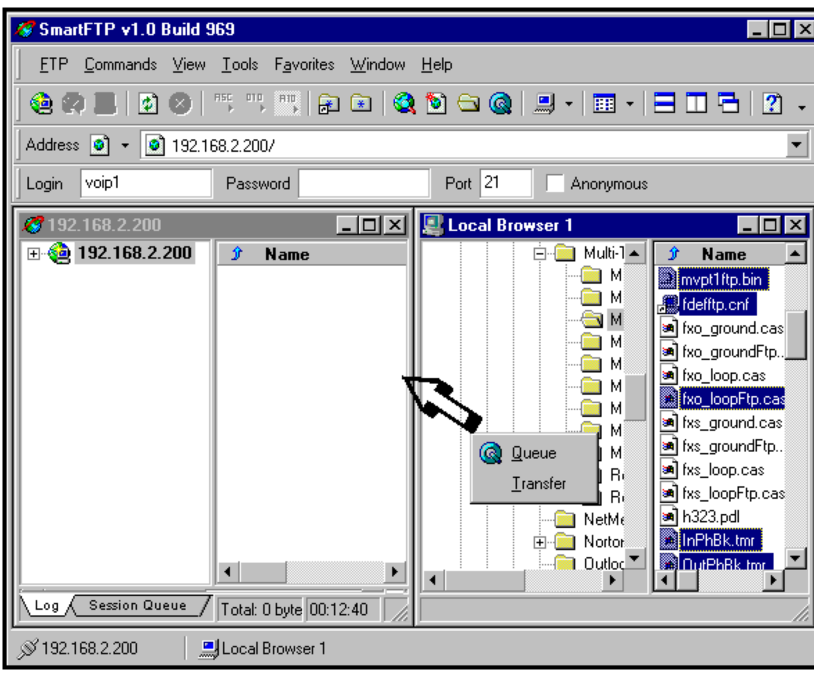
File transfer between PC and voip will look like transfer within voip directories.



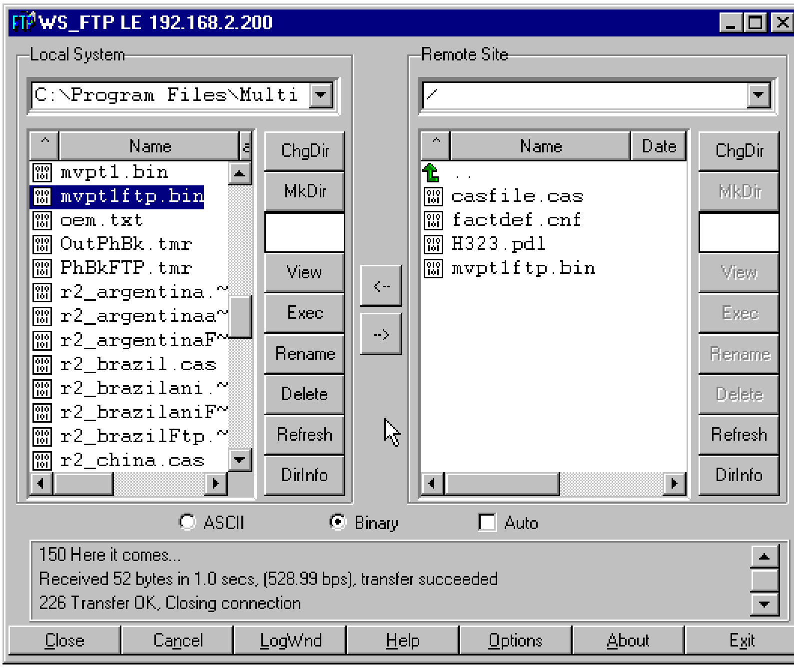
8B. Download with FTP Client Program.

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).

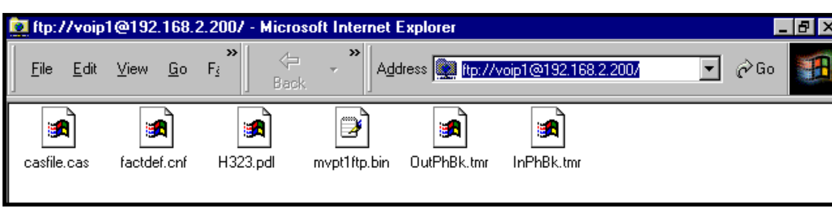
8B2. In the FTP client program window, drag-and-drop files from the local browser pane to the pane for the MultiVOIP FTP server. FTP client GUI operations vary. In some cases, you can choose between immediate and queued transfer. In some cases, there may be automated capabilities to transfer to multiple 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.

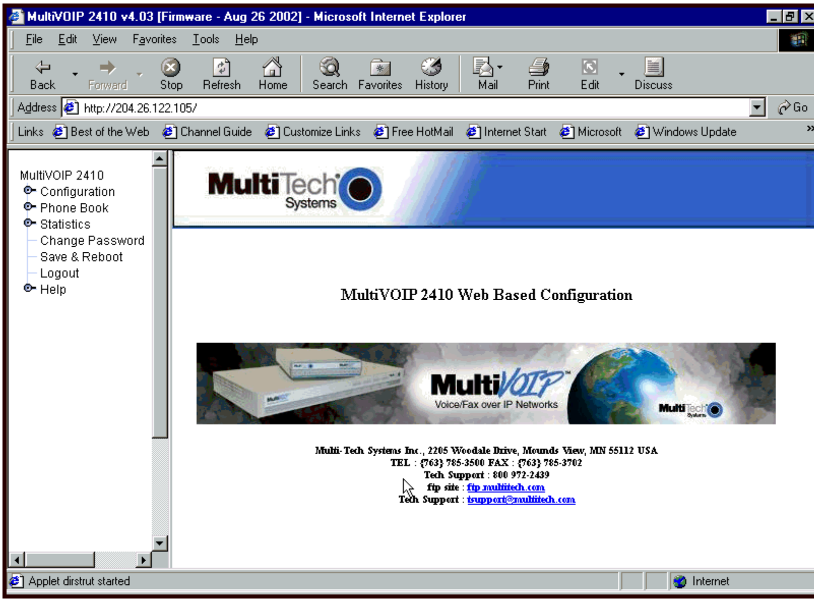


9. **Verify Transfer**. The files transferred will appear in the directory of the MultiVOIP.



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

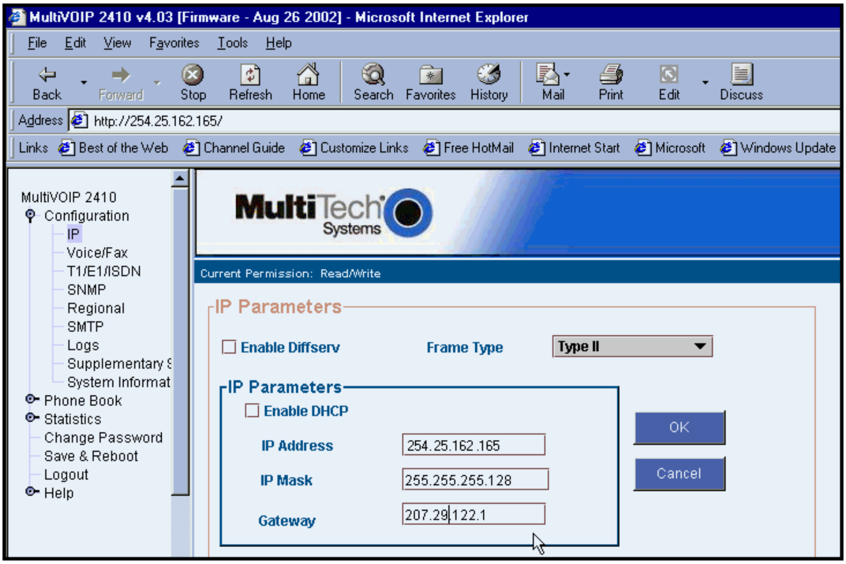


Beginning with the 4.03/6.03 software release, you can control the MultiVOIP unit with a graphic user interface (GUI) based on the common web browser platform. Qualifying browsers are InternetExplorer6 and Netscape6.

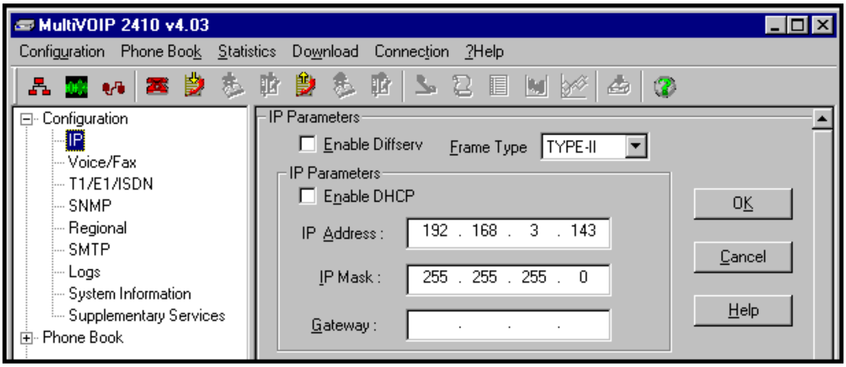
MultiVOIP Web Browser GUI Overview	
Function	Remote configuration and control of MultiVOIP units.
Configuration Prerequisite	Local Windows GUI must be used to assign IP address to MultiVOIP.
Browser Version Requirement	Internet Explorer 6.0 or higher; or Netscape 6.0 or higher
Java Requirement	Java Runtime 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.



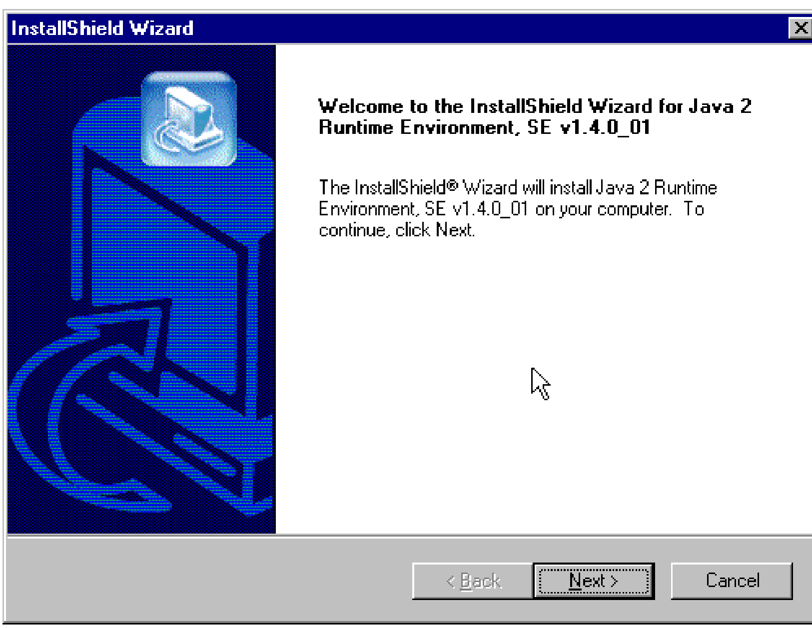
The web GUI, however, cannot perform logging in the same direct mode done in the Windows GUI. However, when the web GUI is used, logging can be done by email (SMTP).

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.

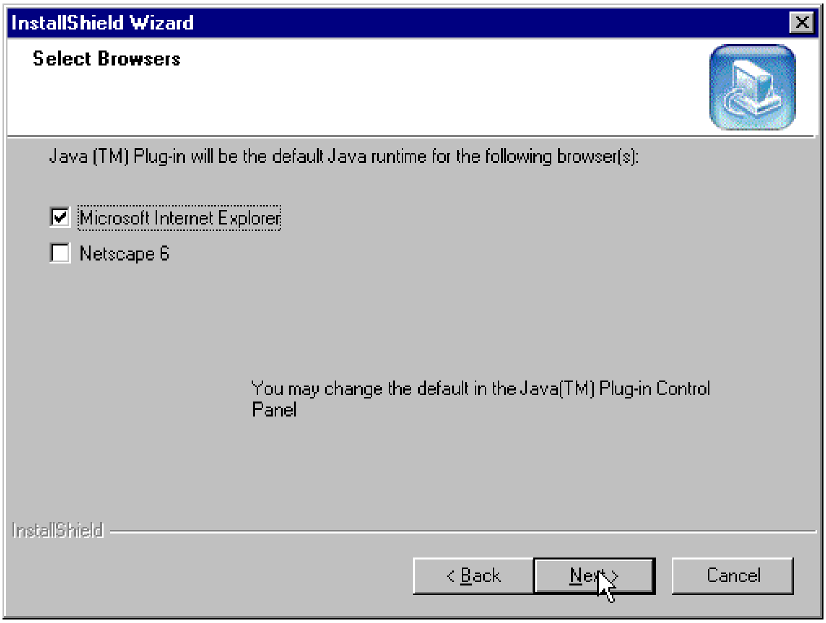
The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

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.

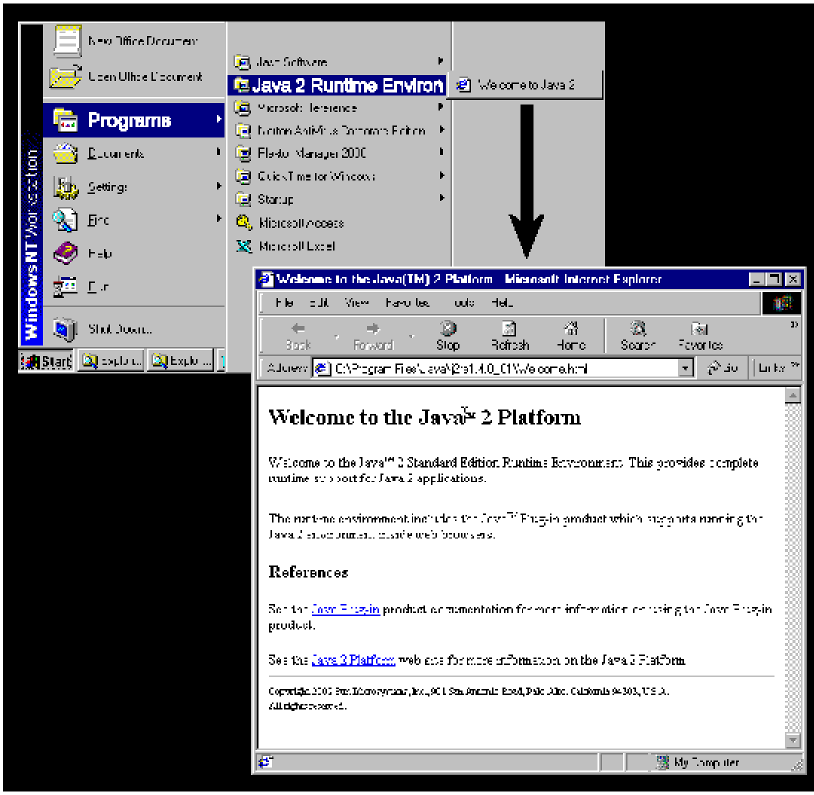
To install the Java program, go to the **Java** directory on the MultiVOIP product CD. Double-click on the EXE file to begin the installation. Follow the instructions on the Install Shield screens.



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

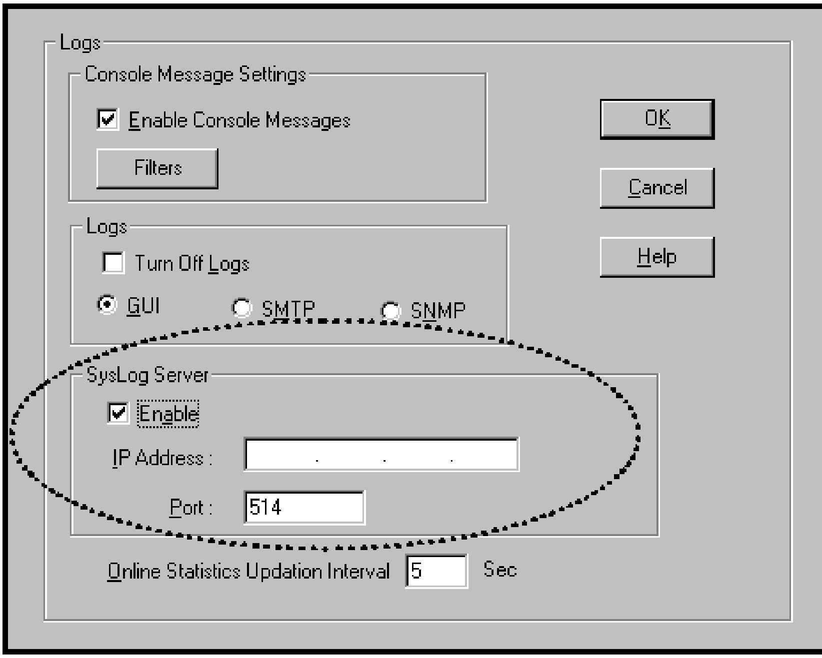
Beginning with the 4.03/6.03 software release, we have built SysLog server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.

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."

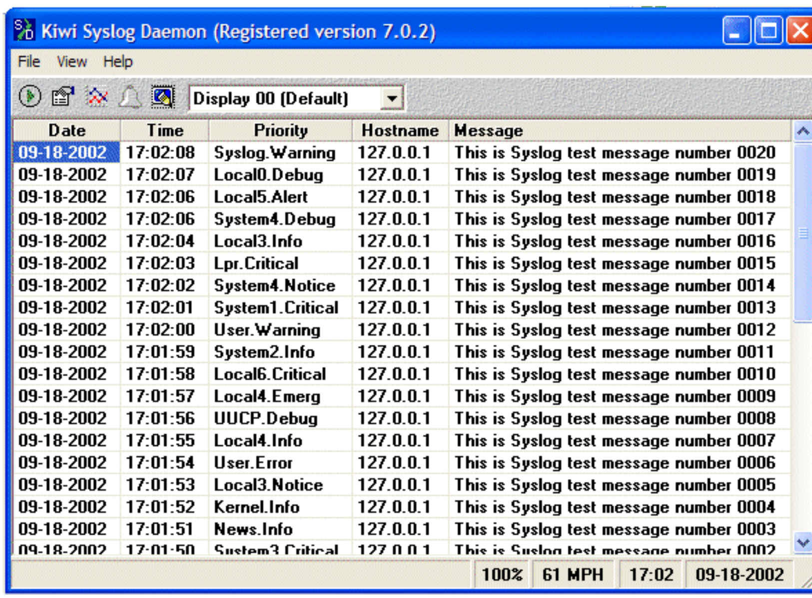
Before a SysLog client program is used, the SysLog functionality must be enabled within the MultiVOIP in the **Logs** menu under **Configuration**.



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.



Date	Time	Priority	Hostname	Message
09-18-2002	17:02:08	Syslog.Warning	127.0.0.1	This is Syslog test message number 0020
09-18-2002	17:02:07	Local0.Debug	127.0.0.1	This is Syslog test message number 0019
09-18-2002	17:02:06	Local5.Alert	127.0.0.1	This is Syslog test message number 0018
09-18-2002	17:02:06	System4.Debug	127.0.0.1	This is Syslog test message number 0017
09-18-2002	17:02:04	Local3.Info	127.0.0.1	This is Syslog test message number 0016
09-18-2002	17:02:03	Lpr.Critical	127.0.0.1	This is Syslog test message number 0015
09-18-2002	17:02:02	System4.Notice	127.0.0.1	This is Syslog test message number 0014
09-18-2002	17:02:01	System1.Critical	127.0.0.1	This is Syslog test message number 0013
09-18-2002	17:02:00	User.Warning	127.0.0.1	This is Syslog test message number 0012
09-18-2002	17:01:59	System2.Info	127.0.0.1	This is Syslog test message number 0011
09-18-2002	17:01:58	Local6.Critical	127.0.0.1	This is Syslog test message number 0010
09-18-2002	17:01:57	Local4.Emerg	127.0.0.1	This is Syslog test message number 0009
09-18-2002	17:01:56	UUCP.Debug	127.0.0.1	This is Syslog test message number 0008
09-18-2002	17:01:55	Local4.Info	127.0.0.1	This is Syslog test message number 0007
09-18-2002	17:01:54	User.Error	127.0.0.1	This is Syslog test message number 0006
09-18-2002	17:01:53	Local3.Notice	127.0.0.1	This is Syslog test message number 0005
09-18-2002	17:01:52	Kernel.Info	127.0.0.1	This is Syslog test message number 0004
09-18-2002	17:01:51	News.Info	127.0.0.1	This is Syslog test message number 0003
09-18-2002	17:01:50	System3.Critical	127.0.0.1	This is Syslog test message number 0002

Chapter 11: 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.

Manufacturer:	Multi-Tech Systems, Inc.
Trade name:	MultiVOIP
Model number:	MVP-2410/3010/810/410/210/130/130FXS
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.

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.

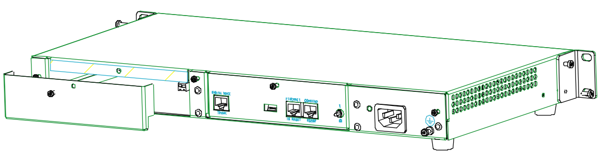


Figure A-1: Remove Plate Covering Expansion Slot

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.

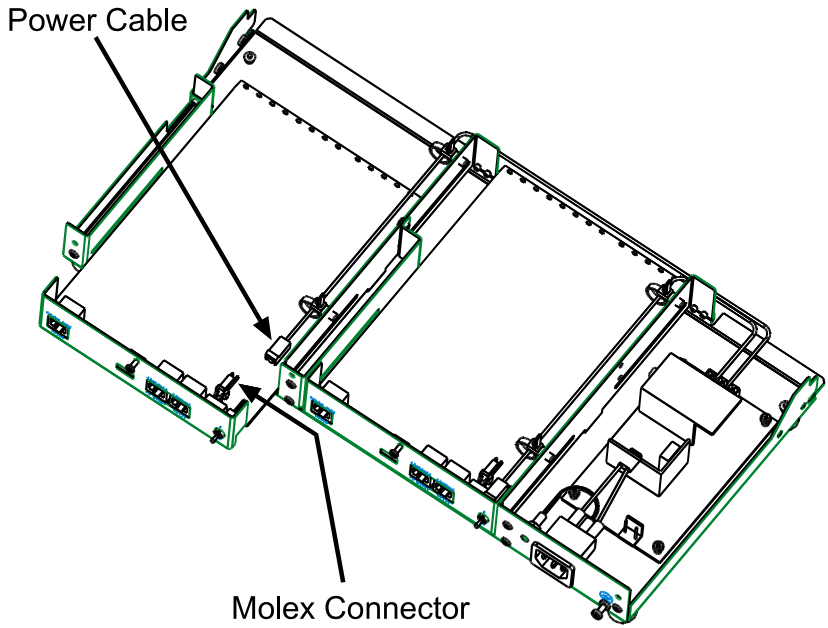


Figure A-2: MVP2410/3010 Chassis (top/rear view)

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.

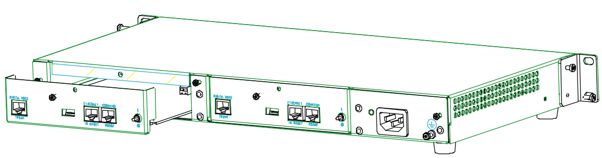


Figure A-3: Sliding Expansion Card into Chassis

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.

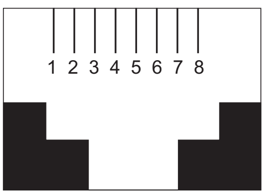
Remember that the expansion card must be configured as though it were simply another complete MultiVOIP unit: it requires its own T1/E1 line; it requires its own connection to a computer running the MultiVOIP configuration software. All of the procedures and operations that apply to the original motherboard of the MVP2410/3010 will also apply to the expansion card. See applicable User Guide chapters for details.

Appendix B: Cable Pinouts

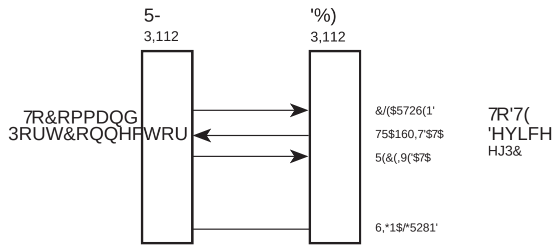
Appendix B: Cable Pinouts

Command Cable

RJ-45 Connector



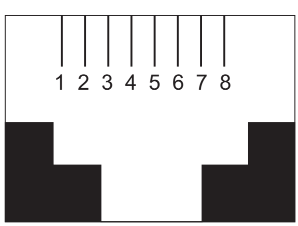
End-to-End Pin Info



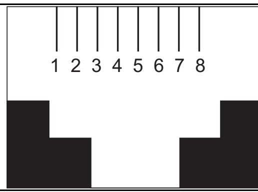
RJ-45 connector plugs into Command Port of MultiVOIP.
DB-9 connector plugs into serial port of command PC (which runs MultiVOIP configuration software).

Ethernet Connector

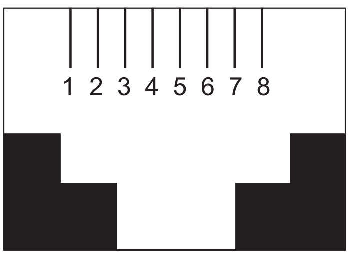
The functions of the individual conductors of the MultiVOIP's Ethernet port are shown on a pin-by-pin basis below.

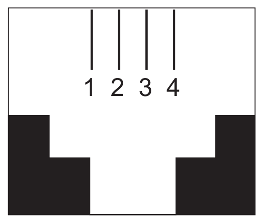
RJ-45 Ethernet Connector	Pin	Circuit Signal Name
	1 2 3 6	TD+ Data Transmit Positive TD- Data Transmit Negative RD+ Data Receive Positive RD- Data Receive Negative

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)

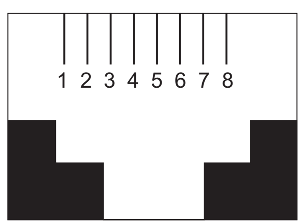
Pin Functions (FXS/FXO Interface)			
FXS Pin	**Description**	**FXO Pin**	**Description**
2	N/C	2	N/C
3	Ring	3	Tip
4	Tip	4	Ring
5	N/C	5	N/C

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.

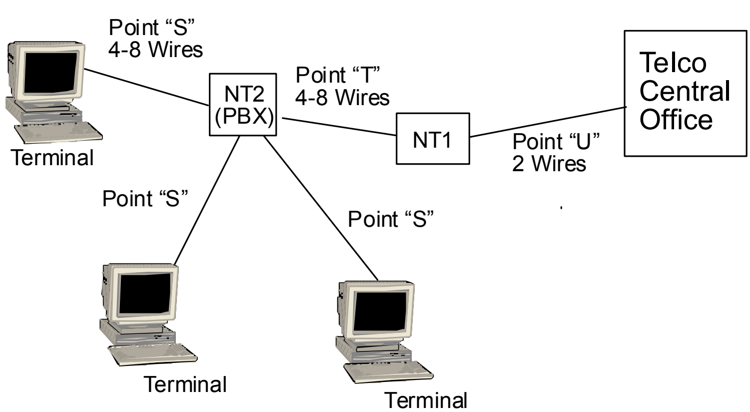
An ISDN Basic Rate (BRI) U-Loop consists of two conductors from the telco central office to the customer premises. The equipment on both sides of the U-loop accommodates the extensive length of the U-loop and the noisy environment in which it may operate. At the customer premises, the U-loop is terminated 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).

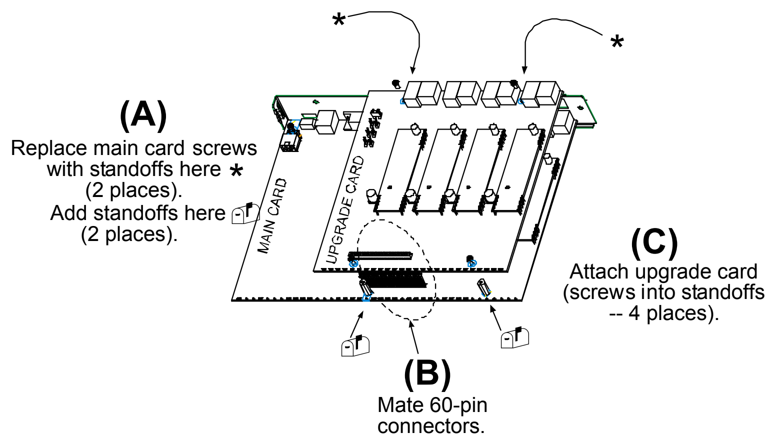


Figure D-1. Installation Summary

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.

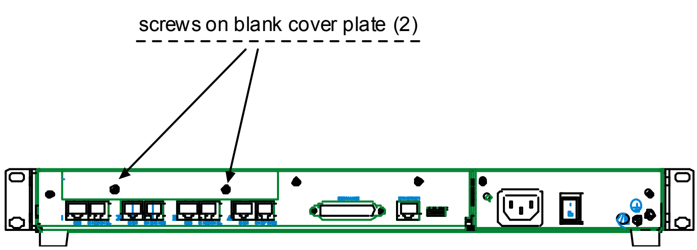


Figure D-2: Removing screws from blank cover plate

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.

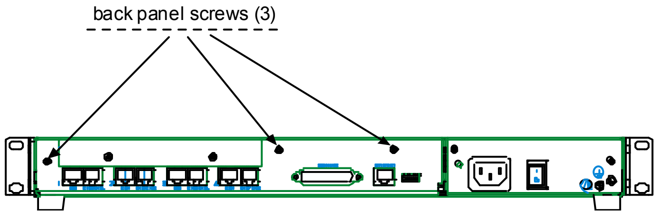


Figure D-3: Removing screws from back panel

4. Slide the main circuit board out of the chassis far enough to unplug the power connector.

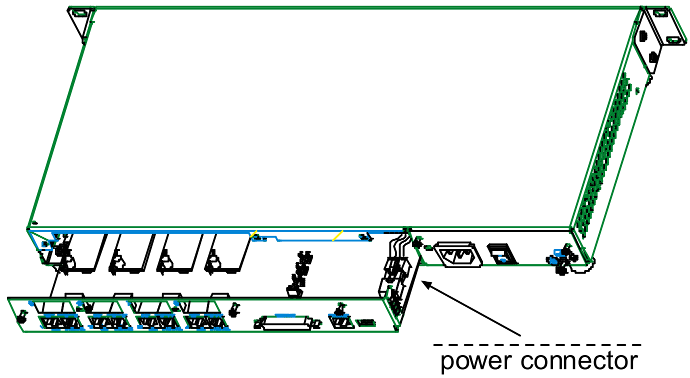


Figure D-4: Accessing power connector

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.

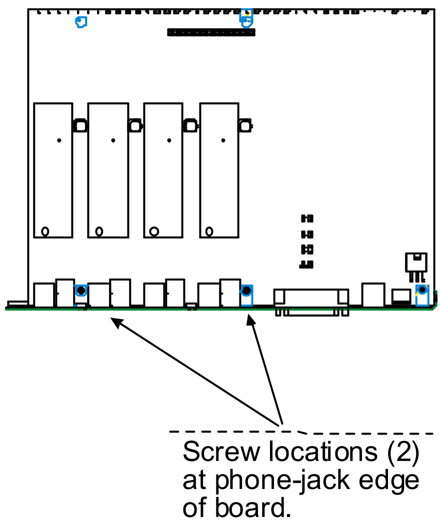


Figure D-5: Screws to be removed and replaced with standoffs (phone-jack edge of board; top view)

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).

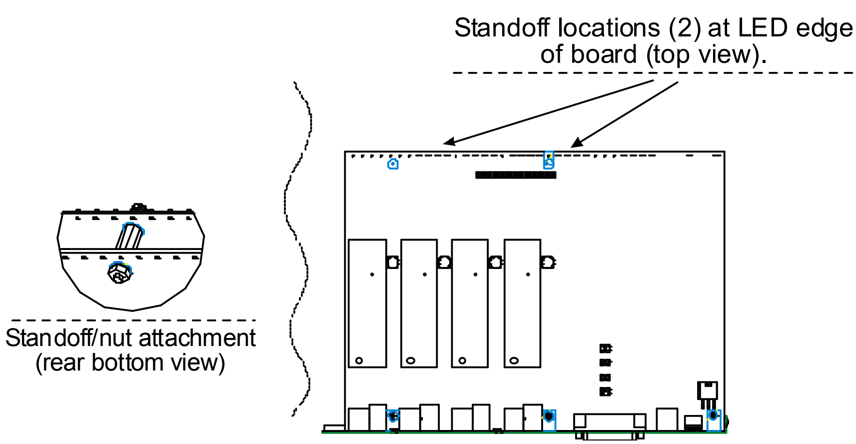


Figure D-6: Standoffs at LED edge of board (top view)

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.

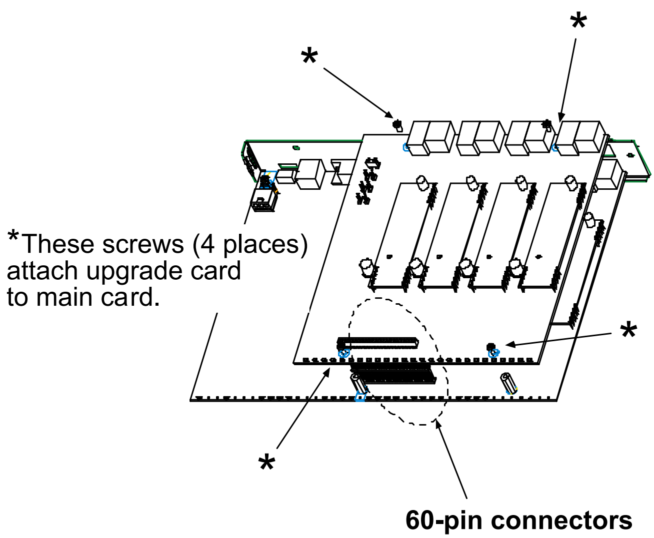


Figure D-7. Attaching upgrade card to main circuit card (secure 4 Phillips screws; mate 60-pin connectors)

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.

INDEX

(

Alternate Phone Number field, SPP
  E1 380
Alternate Phone Number, SPP
  T1 328

A

abbreviated dialing, inter-office
  E1 357
  T1 309
Accept Any Number (inbound)
  E1 384
  T1 333
Accept Any Number (outbound) field
  E1 376
  T1 324
access codes, PBX 66
access codes, types
  PBX 71
  PSTN 71
  special 71
access digits, PBX 66, *See* phonebook digits, types used
access to network
  analog 271
  T1/E1 169
access to remote PSTN
  E1 18
  T1 11
accessing Statistics, Logs screen 420
accessing Call Progress (Statistics) screen 414
accessing configuration parameter groups
  analog *219*
  T1/E1 .... *133*
accessing interface parameters 240
accessing IP Parameters screen
  analog *220*
  T1/E1 .... *134*
accessing IP Statistics screen 425
accessing Logs (Statistics) screen 420
accessing logs screen
  analog 286
  T1/E1 185
accessing Regional Parameters
  analog 272
  T1/E1 170
accessing Registered Gateway Details (Statistics) screen 448
accessing Registered Gateway Details screen 447, 448
accessing RTP Parameters screen 432
accessing SMTP parameters
  analog 280
  T1/E1 178
accessing SNMP parameters
  analog 269
  T1/E1 167
accessing Supplementary Services screen
  analog 290
  T1/E1 189
accessing System Information screen
  analog 302
  T1/E1 201
accessing T1 Statistics screen 435
accessing T1/E1/ISDN Parameters screen 152
accessing Voice/FAX Parameters screen 140, 227
Add Inbound Phonebook Entry icons
  E1 363
  T1 310
Add Outbound Phonebook Entry icon
  E1 363
  T1 310
Add Prefix (inbound) field
  E1 384
  T1 333
Add Prefix (outbound) field
  E1 377
  T1 325
Add/Edit Inbound Phonebook field definitions
  E1 .... 384, 385, 386
  T1 .... 333, 334, 335

Add/Edit Inbound Phonebook screen
E1 384
T1 333
Add/Edit Inbound Phonebook screen fields (E1)
Accept Any Number 384
Add Prefix 384
Channel Number 385
Description (callee location) 385
Enable (Call Forwarding) 385
Forward Condition 385
Forward Destination 386
Registration Option Parameters 386
Remove Prefix 384
Ring Count 386
Add/Edit Inbound Phonebook screen fields (T1)
Accept Any Number 333
Add Prefix 333
Channel Number 333
Description (callee location) 333
Enable (Call Forwarding) 333
Forward Condition 334
Forward Destination 334
Registration Option Parameters 335
Remove Prefix 333
Ring Count 335
Add/Edit Outbound Phonebook field definitions
E1 .... 376, 377, 378, 379, 380
T1 .... 324, 325, 326, 327, 328
Add/Edit Outbound Phonebook fields (E1)
Accept Any Number 376
Add Prefix 377
Advanced button 379
Description 377
destination pattern 377
Gateway H.323 ID 378
Gateway Prefix 378
H.323 Port Number 378
IP Address 377
Protocol Type 377
Remote Device is [legacy] MultiVOIP 380
Remove Prefix 377
SIP Port Number 379
SIP URL 379
Total Digits 377
Transport Protocol (SIP) 379
Use Gatekeeper 378, 380
Use Proxy (SIP) 379
Add/Edit Outbound Phonebook fields (T1)
Accept Any Number 324
Add Prefix 325
Advanced button 327
Description 325
Destination Pattern 325
Gateway H.323 ID 326
Gateway Prefix 326
IP Address 325
Protocol Type 325
Q.931 Port Number 326
Remove Prefix 325
SIP Port Number 327
SIP URL 327
Total Digits 325
Transport Protocol (SIP) 327
Use Gatekeeper 326, 328
Use Proxy (SIP) 327
Add/Edit Outbound Phonebook screen
E1 375
T1 323
Add/Edit Outbound Phonebook SPP Fields
E1 380
T1 328
add-on module (analog, 4-to-8 channel), installation 511
add-on module (T1/E1)
operation 501
add-on module (T1/E1), installation 499
Address (SNMP) field
analog 271
T1/E1 169
Advanced button, Outbound Phonebook
E1 380
T1 328
Advanced Features field group
analog *233*
T1/E1 .... *146*
airflow *91*
Alerting Party

Supplementary Services (analog) 297, 298, 299
Supplementary Services (T1/E1) 196, 197, 198
Allowed Name Type (analog)
Alerting Party........... 297, 298, 299
Calling Party 296
Allowed Name Type (T1/E1)
Alerting Party........... 196, 197, 198
Calling Party 195
Allowed Name Types, Call Name ID (analog)
Alerting Party........................... 297
Busy Party................................ 298
Calling Party 296
Connected Party....................... 299
Allowed Name Types, Call Name ID (T1/E1)
Alerting Party........................... 196
Busy Party................................ 197
Calling Party 195
Connected Party....................... 198
Alternate IP Address field
E1... 382
T1... 330
Alternate IP Routing
E1... 375
T1... 323
Alternate Phone Number, SPP (Add/Edit Outbound Phonebook)
E1... 380
T1... 328
Alternate Routing
PSTN failover feature, and 330
Alternate Routing field definitions
E1... 382
T1... 330
Alternate Routing field definitions (E1)
Alternate IP Address................. 382
Round Trip Delay 382
Alternate Routing field definitions (T1)
Alternate IP Address................. 330
Round Trip Delay 330
analog phonebook......................... 409
using T1 & E1 examples for..... 409
analog phonebook examples......... *207*
analog telephony interface parameters *211*
Annex E field
E1 369
T1 316
Append SIP Proxy Domain Name in User ID (proxy server)
E1 370
T1 317
area codes.......................................70
Auto Disconnect field group
analog.......................................239
T1/E1 151
AutoCall
analog/BRI 234
T1/E1 147
AutoCall (Voice/Fax Params - analog) and Pass Through Enable (FXS Loop Start) 234
AutoCall/Offhook Alert field
analog/BRI *234*, *235*
T1/E1 ....*147*
Automatic Disconnection field
analog.......................................239

B

bandwidth
coder (analog) *232*
coder (T1/E1)............................*145*
battery caution................................85
baud rate, default (MultiVOIP software connection)
T1/E1...................................200
analog..................................301
baud rate, fax
analog.......................................231
T1/E1 144
baud rate, setting
analog.......................................301
T1/E1 200
Behind Proxy/NAT device
E1 373
T1 320
Bipolar Violation (E1 stats) field ..442
Bipolar Violation (T1 stats) field ..439
Blue Alarm (E1 stats) field 440
Blue Alarm (T1 stats) field 437
Boot LED

analog models 32
analog, MVP130 33
BRI models 39
MVP-210 .... *107*
MVP-410/810 .... *99*
MVP-410ST/810ST *103*
on MVP-2410/3010 .... *94*
Boot Version
System Info............................. 412
System Info (ANALOG/BRI)... 303
System Info (T1/E1) 202
booting time
analog.. 32
analog, MVP130 33
BRI 39
E1... 24
T1... 17
box contents
verifying.................................... 86
BRI connector pinout................... 506
BRI interface types
ST and U 507
built-in modem
setup in Regional Parameters
screen 61
built-in modem (analog)
setup in Regional Parameters
screen 214, 273
built-in modem (T1/E1)
setup in Regional Parameters
screen 128, 170, 171
busy & no-response (E1)
forwarding, dual conditions 385
busy & no-response (T1)
forwarding, dual conditions 334
busy tone, custom
analog...................................... 278
T1/E1 .... 175, 176
busy-tones
analog...................................... 277
T1/E1 175
Bytes Received (call progress) field
.... 416
Bytes Received (SMTP logs) field
analog...................................... 283
T1/E1 182
Bytes received (statistics, logs) field
.... 423
Bytes Sent (call progress) field 416
Bytes Sent (SMTP logs) field
analog......................................283
T1/E1 182
Bytes sent (statistics, logs) field....422

C

cable length, maximum span
E1 161
T1 155
cabling diagram, quick
analog models 52, 53, 55, 56, 57
BRI models 54
E1 models..................................52
MVP130....................................56
MVP130FXS..............................57
MVP210....................................55
MVP2410..................................52
MVP3010..................................52
MVP410....................................53
MVP-410ST/810ST 54
MVP810....................................53
T1 models..................................52
cabling problem, fixing
analog models *219*
T1/E1 models...........................*133*
cabling procedure
MVP130..................................*108*
MVP210..................................*104*
MVP2410..................................*93*
MVP3010..................................*93*
MVP410....................................*95*
MVP-410ST.............................*100*
MVP810....................................*95*
MVP-810ST.............................*100*
Cadence 1 (custom) field
analog......................................279
T1/E1 177
Cadence 2 (custom) field
analog......................................279
T1/E1 177
Cadence 3 (custom) field
analog......................................279
T1/E1 177
Cadence 4 (custom) field
analog......................................279
T1/E1 177
Cadence field

analog....................................... 276
T1/E1 174
cadences, custom
T1.E1 .... 177, 279
T1/E1 175
cadences, signaling
analog....................................... 272
T1/E1 170
Call Control PHB field
analog....................................... *222*
T1/E1 .... *136*
Call Control Status
Call Progress Details (statistics)
field..................................... 419
Call Control Status (call progress)
field.. 419
Call Duration field
analog....................................... 239
T1/E1 151
Call Forward Parameters (inbound
phonebook)
E1.. 385
T1.. 334
Call Forwarded To
logs (statistics) field.................. 424
Call Hold
ANALOG 30
BRI 38
E1... 23
T1... 16
Call Hold (analog) 291
Call Hold (T1/E1)......................... 190
Call Hold Enable
analog....................................... 294
T1/E1 193
Call Mode (SMTP logs) field
analog....................................... 283
T1/E1 182
Call Name Identification
ANALOG 30
BRI 38
E1... 23
T1... 16
Call Name Identification (analog)
Alerting Party............ 297, 298, 299
Calling Party 296
Call Name Identification (T1/E1)
Alerting Party............ 196, 197, 198
Calling Party 195
Call Name Identification (analog). 291
Call Name Identification (T1/E1) . 190
Call On Hold
Call Progress Details (statistics)
field 416, 418
Call on Hold (call progress) field.. 418
Call Progress (Statistics) 414
Call Progress Details (statistics)
screen field
Call On Hold 416
Call Waiting 416
Caller ID............................... 416
Call On Hold 418
Call Waiting 418
Caller ID............................... 419
Call Progress Details (statistics)
screen fields
Channel 416
Duration 416
Mode 416
Voice Coder 416
Packets Sent 416
Packets Received................... 416
Bytes Sent 416
Bytes Received...................... 416
Packets Lost 416
Outbound Digits.................... 416
Prefix Matched...................... 416
Gateway Name...................... 417
IP Address 417
Options................................. 417
Silence Compression............ 417
Forward Error Correction...... 417
Status.................................... 419
Call Control Status 419
Call Status (SMTP logs) field
analog....................................... 284
T1/E1 183
Call Transfer
ANALOG.................................. 30
BRI... 38
E1 23
T1 16
Call Transfer (analog) 291
Call Transfer (T1/E1).................... 190
Call Transfer Enable
analog....................................... 293

T1/E1 192
Call Transfer music jingle during hold
analog.. 293
T1/E1 192
Call Transferred To
logs (statistics) field.................. 424
Call Waiting
ANALOG 30
BRI 38
Call Progress Details (statistics)
field.............................. 416, 418
E1 23
T1 16
Call Waiting (analog) 291
Call Waiting (call progress) field.. 418
Call Waiting (T1/E1) 190
Call Waiting Enable
analog.. 294
T1/E1 193
Caller ID
Call Progress Details (statistics)
field.............................. 416, 419
Caller ID (analog)......................... 291
Caller ID (call progress) field 419
Caller ID (Supplementary Services)
field
analog.. *300*
T1/E1 .... *199*
Caller ID (T1/E1).......................... 191
Caller ID enable
FXO 251
FXS Loop Start................. 245, 246
Caller ID examples 252, 253, 254
Caller ID fields
FXO 251
Caller ID Type
FXO 251
FXS Loop Start......................... 245
Caller Name Identification Enable
analog.. 295
T1/E1 194
calling area codes........................... 70
Calling Party
Supplementary Services (analog)
.... 296
Supplementary Services (T1/E1)
.... 195
Canadian Class A requirements 496
Canadian Limitations Notice
(regulatory) 497
CAS Protocol field
E1 162
T1 156
CAS vs. CCS
T1 .... 156, 162
CCS vs. CAS
T1 .... 156, 162
CD
MultiVOIP 44
Channel (call progress) field 416
channel capacity 9
analog.. 26
BRI 34
E1 18
T1 11
Channel Number (inbound) field
E1 385
T1 333
Channel Number (SMTP logs) field
analog.. 283
T1/E1 182
channel tracing on/off (logging)
analog.. 289
T1/E1 188
city codes 70
Clear (button), ISDN BRI Statistics
screen 446
Clear (IP Statistics) button 426
Clear command (Link Management)
button 430
Client Options fields
E1 373
T1 320
Clocking field
E1 165
T1 159
coder (analog)
bandwidth, max......................... *232*
G.711.. *232*
G.723.1..................................... *232*
G.726.. *232*
G.727.. *232*
G.729.. *232*
Net Coder *232*
Coder (SMTP logs) field
analog.. 283

T1/E1 182
coder (T1/E1)
bandwidth, max......................... *145*
G.711 .... *145*
G.723.1 .... *145*
G.726 .... *145*
G.727 .... *145*
G.729 .... *145*
Net Coder................................. *145*
Coder field
analog....................................... *232*
T1/E1 .... *145*
coder options
packetization rates and............. 432
Coder Parameters field group
analog....................................... *232*
T1/E1 .... *145*
coder types (voice/fax, RTP packetization)
T1/E1 433
COM port
on command PC........................ *116*
COM port (analog models)
conflict, resolving *218*
error message........................... *218*
COM port (T1/E1 models)
conflict, resolving *132*
error message........................... *132*
COM port allocation
analog....................................... 301
T1/E1 200
COM port assignments
analog....................................... 301
T1/E1 200
COM port conflict
error message........................... *116*
COM Port Setup screen *116*
COM Port Setup screen (analog models) *218*
COM Port Setup screen (T1/E1 models) *132*
comma
meaning/use in phonebook 72
comma use
and second dial tone.................... 72
command cable pinout.................. 503
command modem
and Regional Parameters screen . 61
and Regional Parameters screen (analog) 214, 273
and Regional Parameters screen (T1/E1).................128, 170, 171
Command Modem
setup for 61
Command Modem (analog)
setup for 214, 273
Command Modem (T1/E1)
setup for 128, 170, 171
command PC
COM port assignment (detailed)*116*
COM port requirement................51
demands upon 51
non-dedicated use........................51
operating system 51
settings 51
specifications...............................51
Command PC
COM port requirement................40
non-dedicated use of 40
operating system 40
community (voip) defined
analog.......................................271
T1/E1 169
Community Name 1 (SNMP) field
analog.......................................271
T1/E1 169
compatibility, Fast Start
E1 366
T1 313
compatibility, H.450 with H.323, not with SIP
analog................................27, 290
BRI...35
E1 19
T1 12
T1/E1 189
compression standard
E1 165
T1 159
compression, silence
analog.......................................*233*
T1/E1 ....*146*
Compression, Silence (SMTP logs)
analog.......................................284
T1/E1 183
computer requirements....................40

configuration of voip (analog)
local versus remote *208*
configuration of voip (T1/E1)
local versus remote *122*, *123*
Configuration option description
(MultiVOIP program menu) 449
Configuration Parameter Groups, accessing
analog...................................... *219*
T1/E1 .... *133*
Configuration Port Setup option description (MultiVOIP program menu)...................................... 449
configuration procedure, local
detailed, analog *215*
detailed, T1/E1.......................... *129*
summary, analog....................... *214*
summary, T1/E1 .... *128*
Configuration Version
System Info (ANALOG/BRI)... 304
System Info (T1/E1) 203
configuration, local
analog/BRI............................... *210*
T1/E1 .... *124*
configuration, phonebook
E1... 362
starter 63
T1... 309
configuration, saving
analog...................................... 305
T1/E1 204
user 466
configuration, starter
phone/IP.................................... 59
configuration, user default
analog...................................... 306
T1/E1 205
Configuring MultiVOIP phonebooks, general
E1... 356
T1... 308
confirming connectivity.................. 81
conflicts
COM port................................. *116*
Connection Problems, Solving
analog...................................... *218*
T1/E1 .... *132*
connectivity
confirmation of........................... 81
confirming with remote voip. 50, 63
pinging and................................. 82
connectivity test 78
Consecutive Packets Lost field
analog...................................... 239
T1/E1 151
Console Message Settings, Filters for
analog...................................... 289
T1/E1 188
console messages 61, 78, 80, 81
console messages, enabling
analog...................................... 287
T1/E1 186
console parameters tracked
analog...................................... 289
T1/E1 188
coordinated phonebook entries
E1... 362
T1... 309
Copy Channel command (Interface Parameters)
analog...................................... *242*
Copy Channel command (Voice/Fax Parameters)
analog...................................... *229*
T1/E1 .... *142*
Copy Channel field
analog...................................... *230*
Copy Channel field (Voice/Fax Parameters)
T1/E1 .... *143*
Copy Channel, Supplementary Services command
analog...................................... *292*
T1/E1 .... *191*
Copy Channel, Supplementary Services field
analog...................................... *300*
T1/E1 .... *199*
Copy Interface command
BRI.. *263*
Count of Registered Numbers field
(Registered Gateway Details) ... 448
country
ISDN type and.......................... 166
switch type and ISDN 166
Country (ISDN) field

E1/ISDN 164
country codes.................................. 70
Country definitions
ISDN-BRI................................. 268
Country field
ISDN-BRI................................. 265
Country field (ISDN)
T1/ISDN 158
Country Selection for Built-In Modem field
analog (except MVP130).......... 276
T1/E1 174
Country/Region (tone schemes) field
analog............................... 274, 275
T1/E1 172
CRC and ESF frame format (T1).. 155
CRC Check field
T1.. 155
Creating a User Default Configuration
analog...................................... 306
T1/E1 205
Current Loss (FXO disconnect criteria) field 249
Current Loss Detect Timer (FXO) field... 249
Current Loss field
FXS Loop Start 244
Custom (tones, Regional)field
analog...................................... 276
T1/E1 174
custom cadences
analog...................................... 279
T1/E1 177
custom DTMF
analog...................................... 278
T1/E1 .... 175, 176
Custom Fields (SMTP) definitions
analog............................... 283, 284
T1/E1 .... 182, 183
Custom Fields, SMTP log email (analog)
Bytes Received 283
Bytes Sent................................ 283
Call Mode 283
Call Status................................ 284
Channel Number...................... 283
Coder 283
Options.................................... 284
Options....................................284
Description (callee)..................284
Description (caller) 284
Duration 283
From Gateway Number............284
From IP Address 284
Outbound Digits.......................284
Packets Lost 283
Packets Received......................283
Packets Sent 283
Prefix Matched.........................284
Select All.................................283
Start Date, Time 283
To Gateway Number................284
To IP Address 284
Custom Fields, SMTP log email (T1/E1)
Bytes Received.........................182
Bytes Sent 182
Call Mode................................182
Call Status 183
Channel Number 182
Coder.......................................182
Options....................................183
Options....................................183
Description (callee)..................183
Description (caller) 183
Duration 182
From Gateway Number............183
From IP Address 183
Outbound Digits.......................183
Packets Lost 182
Packets Received......................182
Packets Sent 182
Prefix Matched.........................183
Select All.................................182
Start Date, Time 182
To Gateway Number................183
To IP Address 183
Custom Tone-Pair Settings (analog) fields
Cadence 1................................279
Cadence 2................................279
Cadence 3................................279
Cadence 4................................279
Custom Tone-Pair Settings (T1/E1) fields
Cadence 1................................177

Cadence 2 177
Cadence 3 177
Cadence 4 177
Custom Tone-Pair Settings definitions
analog............................... 278, 279
T1/E1 .... 176, 177
Custom Tone-Pair Settings fields (analog)
Frequency 1 278
Frequency 2 278
Gain 1 278
Gain 2 278
Tone Pair................................. 278
Custom Tone-Pair Settings fields (T1/E1)
Frequency 1 176
Frequency 2 176
Gain 1 176
Gain 2 176
Tone Pair................................. 176
custom tones, setting
T1/E1 175
customized log email
analog............................... 283, 284
T1/E1 .... 182, 183

D

D Channel Information fields (ISDN BRI Statistics).......................... 445
data capacity 9
analog.. 26
BRI 34
E1... 18
T1... 11
data compression
analog.. 27
BRI 35
E1... 19
T1... 12
Date & Time Setup (program menu option), command..................... 452
Date and Time Setup option description (MultiVOIP program menu)...................................... 449
debugging messages
analog...................................... 288
T1/E1 186
Default (Supplementary Services) field
analog...................................... *300*
T1/E1 .... *199*
Default (Voice/FAX) field
analog...................................... *230*
T1/E1 .... *143*
default baud rate (MultiVOIP software connection)
analog...................................... 301
T1/E1 200
default configuration, user
analog...................................... 306
T1/E1 205
default values, software................ 460
delay, packets
analog...................................... 237
T1/E1 149
delay, versus voice quality
analog...................................... 238
T1/E1 150
Delete File button
Logs (Statistics) screen 422
Description (callee location)
E1 385
T1 333
Description (callee, outbound phonebook)
E1 377
T1 325
Description field (Registered Gateway Details).................................... 448
Description, From Details (SMTP logs) field
analog...................................... 284
T1/E1 183
Description, To Details (SMTP logs) field
analog...................................... 284
T1/E1 183
Destination Pattern (outbound) field
E1 377
T1 325
destination patterns
digits used 70
tips about.................................... 70
destination patterns, discussion
E1 361

T1 308
Detection Flash Hook field
E1 163
T1 157
Detection Range, Flash Hook Options field
E&M 258
FXO 251
FXS Loop Start 245
Detection Time
E1 163
T1 157
Detection Time field
E1 163
T1 157
dial tone, custom
analog 278
T1/E1 .... 175, 176
dial tone, second
and comma use 72
pausing for 72
Dialing Options (E&M) fields 257
Dialing Options (FXO) fields 248, 249
Dialing Options field
ISDN-BRI 265
dialing patterns
digits used 70
inbound/outbound matching 72
tips about 70
dial-tones
analog 277
T1/E1 175
DID interface (MVP210)
uses of *107*
DID interface (MVP-410/810)
uses of *98*
DID Interface Parameter definitions 260
DID Interface Parameter fields
Message Waiting Indication 260
DID Interface Parameters 259
DID jumper
MVP210 104
MVP-410/810 96
DID lines (MVP210)
polarity sensitivity and 107
DID lines (MVP-410/810)
polarity sensitivity and 98
DID-DPO Interface Parameter definitions 259
DID-DPO Interface Parameter fields
Inter Digit Timer (dialing) 260
Start Modes 259
Wink Timer 259
DID-DPO Parameter fields
Inter-Digit Regeneration Timer (dialing) 260
DID-DPO vs. DID-DPT 259
DiffServ and IP datagram
analog 223
T1/E1 137
DiffServ PHB (Per Hop Behavior) value
analog 222
T1/E1 136
digits in phonebook
specialized codes 71
types 70
dimensions
analog models 43
E1 models 42
Disconnect on Call Progress Tone (E&M) field 258
Disconnect on Call Progress Tone (FXO) field 250
Disconnect Tone Sequence (FXO) field 250
disconnection criteria, FXO .. 249, 250
DNS Server IP Address
T1/E1 .... *138*, *224*
Download Factory Defaults (program menu option) , command 460
Download Factory Defaults option description (MultiVOIP program menu) 450
Download Firmware (program menu option), command 456, 457
Download Firmware option description (MultiVOIP program menu) 450
Download IFM Firmware (program menu option) , command... 462, 463
Download IFM Firmware option description (MultiVOIP program menu) 450

Download User Defaults (program menu option) , command 466
Download User Defaults option description (MultiVOIP program menu) 450
downloading firmware, machine perspective 451, 476
downloading IFM firmware 462
downloading user defaults 466
downloads vs. uploads (FTP)........ 476
dropping digits, in phonebook 71
DTMF
  extended................................... 250
  standard.................................... 250
DTMF frequency chart 250
DTMF Gain (High Tones) field
  analog...................................... *230*
  T1/E1 .... *143*
DTMF Gain (Low Tones) field
  analog...................................... *230*
  T1/E1 .... *143*
DTMF Gain field
  analog...................................... *230*
  T1/E1 .... *143*
DTMF In/Out of Band field
  analog...................................... 231
  T1/E1 144
DTMF inband
  analog...................................... *231*
  T1/E1 .... *144*
DTMF out of band
  analog...................................... 231
  T1/E1 144
DTMF Parameters
  T1/E1 .... *143*
DTMF, custom tone pairs
  analog...................................... 278
  T1/E1 .... 175, 176
Duration (call progress) field........ 416
Duration (DTMF) field
  analog...................................... 231
  T1/E1 144
Duration (SMTP logs) field
  analog...................................... 283
  T1/E1 182
Duration (statistics, logs) field...... 422
Dynamic Jitter Buffer field
  analog...................................... *237*
  T1/E1 .... *149*
Dynamic Jitter field group
  analog...................................... *237*
  T1/E1 .... *149*
Dynamic Jitter fields
  analog...................................... 238
  T1/E1 150

E

E&M interface (MVP210)
  matching telco trunk line........... *106*
  uses of *106*
E&M interface (MVP-410/810)
  matching telco trunk line............. *98*
  uses of *98*
E&M Interface Parameter fields
  Detection Range (flash hook).... 258
  Disconnect on Call Progress Tone 258
  Flash Hook 258
  Inter Digit Timer (dialing) 257
  Interface 256
  Message Waiting Indication...... 257
  Pass Through............................ 256
  Regeneration (dialing)............... 257
  Signal 256
  Type 256
  Wink Timer.............................. 256
E&M Parameter definitions . 256, 257, 258
E&M Parameters.......................... 255
E1 Parameter definitions 161, 162, 163, 165
  Clocking.................................. 165
  Line Build-Out 165
  Line Coding 165
  PCM Law 165
  Pulse Shape Level 165
E1 Parameter fields
    CAS Protocol 162
  CRC Check 162
  Detect Flash Hook.................... 162
  Frame Format........................... 162
  Long-Haul Mode....................... 162
E1 Parameters screen 160
E1 Statistics field definitions 440, 441, 442
E1 Statistics fields

Bipolar Variation 442
Blue Alarm 440
Excessive Zeroes 442
Loss of Frame Alignment 440
Loss of MultiFrame Alignment 441
Receive Slip 442
Receive Timeslot 16 Alarm
Indication Signal 441
Receive Timeslot 16 Loss of
MultiFrame Alignment 442
Receive Timeslot 16 Loss of Signal
.... 441
Red Alarm................................ 440
Status Freeze Signalling Active 441
Transmit Data Overflow 441
Transmit Data Underrun 442
Transmit Line Open 442
Transmit Line Short.................. 441
Transmit Slip 442
Transmit Slip Negative 442
Transmit Slip Positive.............. 441
Yellow Alarm 441
E1 telephony parameters.............. *126*
E1/ISDN Parameter definitions 164
E1/ISDN Parameter fields
Country 164
Enable ISDN-PRI 164
Operator 164
Terminal Network.................... 164
Echo Cancellation field
analog...................................... *233*
T1/E1 .... *146*
echo, removing
analog...................................... *233*
T1/E1 .... *146*
Edit selected Inbound Phonebook
Entry icon
E1 363
T1 310
Edit selected Outbound Phonebook
Entry icon
E1 363
T1 310
email account for voip unit
analog...................................... 281
T1/E1 180
email address for voip
analog.............................. *213*, 280
quick.. 50
T1/E1 .... *127*, 178
email log reports
analog...................................... 280
quick.. 61
recipient.................................... 61
reply-to address 61
subject line 61
T1/E1 178
email logs, illustration
analog...................................... 285
T1/E1 184
EMC, Safety, R&TTE Directive
Compliance 495
emergency phone numbers
caution about 72
Enable (Call Fwdg)
E1 385
T1 333
Enable Call Hold
analog...................................... 294
T1/E1 193
Enable Call Transfer
analog...................................... 293
T1/E1 192
Enable Call Waiting
analog...................................... 294
T1/E1 193
Enable Caller Name Identification
analog...................................... 295
T1/E1 194
Enable Console Messages field
analog...................................... 288
T1/E1 186
Enable DNS field
analog...................................... *224*
T1/E1 .... *138*, *224*
Enable ISDN-PRI field
E1/ISDN.................................. 164
T1/ISDN.................................. 158
Enable SMTP field
analog...................................... 281
T1/E1 180
Enable SNMP Agent............ 167, 269
Enable SNMP Agent field
analog...................................... 271
T1/E1 169
enabling SMTP

analog.......................................280
T1/E1 178
enabling web browser GUI
analog.................................59, 226
T1/E1 139
Error Correction (SMTP logs)
analog.......................................284
T1/E1 183
error correction, forward
analog....................................... *233*
T1/E1 .... *146*
error message
COM port conflict..................... *116*
COM port conflict (analog models)
.... *218*
error message (analog models)
MultiVOIP Not Found.............. *219*
Phone Database Not Read......... *219*
error message (T1/E1 models)
MultiVOIP Not Found.............. *133*
Phone Database Not Read......... *133*
ESF and CRC frame format (T1).. 155
ethernet cable pinout..................... 503
Ethernet interface
analog...26
BRI 34
European Community Directives.. 495
Excessive Zeroes (E1 stats) field .. 442
Excessive Zeroes (T1 stats) field .. 437
exchanges, phone
dedicated....................................71
institutional 71
local 71
non-local 71
organizational 71
expansion card (analog, 4-to-8
channel) installation..................511
expansion card (T1/E1) installation
....499
expansion card (T1/E1)operation.. 501

F

factory default software settings ... 460
factory defaults, downloading....... 460
failover (PSTN)
analog models 27
BRI models 35
E1 models 19
T1 models...................................12
failover (PSTN) feature.................330
FAQ for MultiVOIPs 10
fast busy (unobtainable) tones
analog...............................175, 277
Fast Connect... *See* Fast Start, *See* Fast
Start
E1 369
T1 316
Fast Start compatibility
E1 366
T1 313
Fast Start plus H.245 Tunneling field
E1 369
T1 316
fax baud rate, default
analog.......................................231
T1/E1 144
Fax Enable field
analog.......................................231
T1/E1 144
fax machine
connecting to analog voip
(MVP130) *108*
connecting to analog voip
(MVP130FXS)...................... *109*
connecting to analog voip
(MVP210) *106*, *107*
connecting to analog voip (MVP-
410/810)............................ *97*, *98*
FAX Parameters
analog.......................................231
T1/E1 144
fax tones, output level
analog.......................................231
T1/E1 144
Fax Volume field
analog.......................................231
T1/E1 144
FCC Declaration 495
FCC Part 68 Telecom rules...........496
FCC registration number...............497
FCC rules, Part 15.........................495
FDX LED
analog models 32
analog, MVP130 33
BRI models 39
E1 24

T1 17
Filters (Console Message Settings)
analog....................................... 289
T1/E1 188
Filters button (Console Message Settings)
analog....................................... 288
T1/E1 187
firmware upgrade, implementing.. 456
Firmware Version (System Info)
ANALOG/BRI.......................... 303
T1/E1 202
firmware version, identifying 456
firmware, downloading................ 457
firmware, obtaining updated......... 452
Flash Hook Options fields
E&M 258
FXO 251
forgotten password................ 468, 472
Forward Address/Number
T1 334
Forward Condition (Call Fwdg)
E1 385
T1 334
Forward Destination (Inbound PhBk)
E1 386
Forward Error Correction (call progress) field 417
Forward Error Correction (SMTP logs)
analog....................................... 284
T1/E1 183
Forward Error Correction field
analog....................................... *233*
T1/E1 .... *146*
forward on busy
T1 .... 334, 385
Forward upon No Response
E1 385
T1 334
forwarding, dual conditions (E1)
busy & no-response 385
forwarding, dual conditions (T1)
busy & no-response 334
Frame Format field
E1 161
T1 155
frame relay, and fax
analog...................................... 231
T1/E1 144
Frame Search Restart Flag (T1 stats)
field 438
Frame Type field
analog...................................... *222*
T1/E1 .... *136*
free calls
E1 357
T1 308
frequencies, touch tone 250
Frequency 1 (custom tone) field
analog...................................... 278
T1/E1 176
Frequency 1 (tone pair scheme)
analog...................................... 275
T1/E1 173
Frequency 2 (custom tone) field
analog...................................... 278
T1/E1 176
Frequency 2 (tone pair scheme)
analog...................................... 275
T1/E1 173
frequency, power
analog models 43
E1 models.................................. 42
FRF11
analog...................................... 231
T1/E1 144
From (gateway, statistics, logs) field
.... 422
front panel
analog models 32
BRI models 39
E1 24
MVP2400 16
MVP2410 16
MVP3010 24
T1 16
FTP client program 476
FTP client program, obtaining 478
FTP client programs
graphic vs. textual orientation ... 485
FTP file transfers
using FTP client program.......... 478
using web browser 478
FTP Server Enable field
analog...................................... *225*

T1/E1 .... *138*
FTP Server function
as added feature 476
enabling 478
FTP Server, contacting 480
FTP Server, invoking
download/transfer
using FTP client program 484
using web browser 482
FTP Server, logging in.................. 481
FTP Server, logging out................ 485
FTP transfers
file types 476, 479
phonebooks 476
server location........................... 476
function tracing on/off (logging)
analog... 289
T1/E1 188
FXO Disconnect On fields.... 249, 250
FXO disconnection criteria........... 249
FXO disconnection, triggering of 249, 250
FXO interface (MVP130)
uses of....................................... *108*
FXO Interface Parameter definitions 248, 249
FXO Interface Parameter Definitions 250
FXO Interface Parameter fields
Current Loss 249
Current Loss Detect Timer 249
Detection Range (flash hook) ... 251
Disconnect on Call Progress Tone 250
Disconnect Tone Sequence....... 250
Flash Hook................................ 251
Inter Digit Regeneration Timer. 249
Inter Digit Timer (dialing)........ 248
Message Waiting Indication 248
Regeneration (dialing) 248
Ring Count................................ 250
Silence Detection 250
Silence Timer............................ 250
Tone Detection.......................... 249
FXO interface(MVP210)
uses of....................................... *106*
FXO interface(MVP-410/810)
uses of... *97*
FXO Parameter fields
Caller ID enable 251
Caller ID Type 251
FXO Current Detect Timer 249
Tone Detection..........................249
FXO Parameters............................247
FXS interface(MVP130)
uses of *108*
FXS interface(MVP130FXS)
uses of *109*
FXS interface(MVP210)
uses of *106*
FXS interface(MVP-410/810)
uses of *97*
FXS Loop Start Interface parameter definitions 243
FXS Loop Start Interface Parameter fields
Caller ID enable 245, 246
Caller ID Type 245
Current Loss..............................244
Detection Range (flash hook)....245
Inter Digit Regeneration Timer.244
Inter Digit Timer 244
Message Waiting Indication......244
Pass Through Enable.................245
Ring Count 244
FXS Loop Start Parameter fields
Inter Digit Timer 243
Message Waiting Light 243
FXS Loop Start Parameters...........243
FXS/FXO connector
MVP130...................................*108*
MVP210...................................*106*
MVP-410/810 ....*97*

G

G711 coders (RTP packetization, voice/fax)
T1/E1 433
G723 coders (RTP packetization, voice/fax)
T1/E1 433
G726 coders (RTP packetization, voice/fax)
T1/E1 433
G727 coders (RTP packetization, voice/fax)

T1/E1 433
G729 coders (RTP packetization, voice/fax)
T1/E1 433
Gain 1 (custom tone) field
analog....................................... 278
T1/E1 176
Gain 1 (tone pair scheme)
analog....................................... 275
T1/E1 173
Gain 2 (custom tone) field
analog....................................... 278
T1/E1 176
Gain 2 (tone pair scheme)
analog....................................... 275
T1/E1 173
gatekeeper interaction
analog models 27
BRI models 35
E1 models 19, 20
T1 models 12, 13
Gatekeeper IP Address (Gatekeeper RAS) field
E1... 366
T1... 313
Gatekeeper Name (Gatekeeper RAS) field
E1... 366
T1... 313
GateKeeper RAS Parameters
E1.................................... 366, 367
T1.................................... 313, 314
Gateway (IP Parameters) field
analog....................................... *224*
T1/E1 .... *138*
Gateway H.323 ID (Outbound Phonebook) field
T1.................................... 326, 378
Gateway Name (call progress) field
.... 417
Gateway Name (callee, statistics, logs) field................................. 423
Gateway Name (caller, statistics, logs) field... 423
Gateway Name field
E1... 366
T1... 313
Gateway Number, From Details (SMTP logs) field
analog....................................... 284
T1/E1 183
Gateway Number, To Details (SMTP logs) field
analog....................................... 284
T1/E1 183
Gateway Prefix (outbound phonebook) field
E1 378
T1 326
General Options fields
E1 372
T1 319
Generate Local Dial Tone (Voice/FAX – AutoCall/Offhook Alert) field
analog/BRI 235
Generation Flash-Hook Options field
E&M 258
FXO 251
Generation Time
E1 163
T1 157
Generation Time field
E1 163
T1 157
grounding
in rack installations *91*
MVP210.................................. *107*
MVP410ST *103*
MVP810ST *103*
grounding screw, diagrams
(MVP-2410/3010)....................... 52
(MVP-410/810)........................... 53
(MVP-410ST/810ST).................. 54
GUI (log reporting type) button
analog....................................... 288
T1/E1 187

H

H.245 Tunneling field
E1 368
T1 315
H.323
compatibility (analog models)..... 27
compatibility (BRI models)......... 35

compatibility (E1 models) 19
compatibility (T1 models) 12
H.323 Annex E field
E1 369
T1 316
H.323 coder
analog *232*
T1/E1 .... *145*
H.323 fields (Outbound Phonebook)
E1 378
T1 326
H.323 Multiplexing field
E1 368
T1 315
H.323 Parameters
E1 366
T1 313
H.323 Port Number (outbound phonebook) field
E1 378
H.323 version 4 features
analog 27
BRI 35
E1 19
T1 12
H.323 Version 4 Parameters
E1 369
T1 316
H.450 features, incompatible with SIP
analog 27, 290
BRI 35
E1 19
T1 12
T1/E1 189
H.450 functionality
logs for 424
H.450 standard
ANALOG 30
BRI 38
E1 23
T1 16
Hardware ID
System Info (ANALOG/BRI) 304
System Info (T1/E1) 203
Hold Sequence
analog 294
T1/E1 193
Hold Sequence (analog) 291
Hold Sequence (T1/E1) 190
hold, caller on (analog)
musical jingle for 293
hold, caller on (T1/E1)
musical jingle for 192
hookup
MVP130 56
MVP130FXS 57
MVP210 55
MVP2410 52
MVP3010 52
MVP410 53
MVP-410ST/810ST 54
MVP810 53
HyperTerminal program
and connectivity testing 79

I

IANA 509
icon
variable version 10, *113*
icons, phonebook
E1 363
T1 310
identifying current firmware version 456
IFM (interface module, analog voips only) description 462
IFM firmware, downloading 462, 463
IFM Version
System Info (ANALOG/BRI) 304
System Info (T1/E1) 203
implementing firmware upgrade 456
in band, DTMF
analog *231*
T1/E1 .... *144*
inbound phonebook
example 73
Inbound Phonebook Entries List icon
E1 363
T1 310
Inbound Phonebook entries, list
E1 382
T1 331
inbound phonebook example
quick 67
inbound vs. outbound phonebooks
E1 361

T1 308
Industry Canada requirements 496
info sources
analog telephony details...... 48, *211*
BRI telephony details........ 49
E1 details 48
E1 telephony details........ *126*
IP details 47
IP details (analog system)......... *210*
IP details (T1/E1 system).......... *124*
ISDN-BRI telephony details..... *212*
SMTP details 50
T1 details 47
T1 telephony details........ *125*
voip email account........ 50
info sources (analog models)
SMTP details 213
voip email account........ 213
info sources (T1/E1 models)
SMTP details 127
voip email account........ 127
Input Gain field
analog........ *230*
T1/E1 .... *143*
installation
airflow........ *91*
analog prerequisites *210*, *211*
BRI prerequisites 49
E1 prerequisites 48, *126*
expansion card (analog, 4-to-8 channel) 511
expansion card (T1/E1)........ 499
full summary........ 46
in a nutshell........ 44
in rack *90*
IP prerequisites 47
ISDN-BRI prerequisites........ *212*
log reports by email (analog models) *213*
log reports by email (T1/E1 models) *127*
software (detailed) *111*
T1 prerequisites 47, *125*
T1/E1 prerequisites........ *124*
upgrade card (analog, 4-to-8 channel) 511
upgrade card (T1/E1)........ 499
voip email account(analog models) *213*
voip email account(T1/E1 models) *127*
installation preparations (optional)
log reports by email........ 50
voip email account........ 50
installation, mechanical
analog models 26
BRI models 34
E1 models........ 18
T1 models........ 11
installation, quick
log reports by email........ *50*
voip email account........ *50*
installing Java vis-a-vis web GUI . 488
integrated phone/data networks..... 356
Inter Digit Regeneration Time (FXO) field 249
Inter Digit Regeneration Time (FXS Loop Start) field........ 244
Inter Digit Regeneration Timer (E&M) field 257
Inter Digit Timer (dialing) field
DID-DPO 260
E&M 257
FXO 248
FXS Loop Start 244
Intercept Tone (Regional Params) and Offhook Alert (Voice/Fax Params) 274
Intercept Tone and required Interface & Voice/Fax settings
analog........ 274
Interface field (DID-DPO)........ 259
Interface field (E&M) 256
interface parameters, accessing..... 240
interface parameters, setting.......... 240
interface types, BRI
ST and U 507
interfaces
BRI telephony 54
inter-office dialing
E1 357
T1 309
inter-operation (analog)
with T1/E1 voips........ 25
inter-operation (BRI)

with T1/E1/BRI voips................. 34
inter-operation with phone system
analog models............................. 26
BRI models................................ 34
E1 models.................................. 18
T1 models.................................. 11
IP Address (call progress) field..... 417
IP Address (callee, statistics, logs)
field... 423
IP Address (caller, statistics, logs)
field... 423
IP Address (outbound phonebook)
E1.. 377
T1.. 325
IP Address field
analog....................................... *224*
T1/E1....................................... *138*
IP Address field (Registered Gateway
Details).................................... 448
IP Address to Ping (Link
Management) field................... 430
IP Address, From Details (SMTP
logs) field
analog....................................... 284
T1/E1....................................... 183
IP address, SysLog Server
analog....................................... 288
T1/E1....................................... 187
IP Address, To Details (SMTP logs)
field
analog....................................... 284
T1/E1....................................... 183
IP datagram and DiffServ
analog....................................... 223
T1/E1....................................... 137
IP Mask field
analog....................................... *224*
T1/E1....................................... *138*
IP parameter definitions
analog....................................... *222*
T1/E1....................................... *136*
IP Parameter fields (analog)
Diffserv Parameters.................. *222*
DNS Server IP Address............ *224*
Enable DHCP........................... 224
Enable DNS............................. *224*
Frame Type.............................. *222*
FTP Server Enable.................... *224*
Gateway.................................... *224*
IP Address................................ *224*
IP Mask.................................... *224*
IP Parameter fields (T1/E1)
Enable DHCP............................ 138
IP Parameters screen, accessing
analog....................................... *220*
T1/E1....................................... *134*
IP startup configuration................... 59
IP Statistics field definitions .425, 427
IP Statistics fields
Clear... 425
Received (RTCP Packets)......... 428
Received (RTP Packets)............ 428
Received (TCP Packets)............ 427
Received (Total Packets).......... 425
Received (UDP Packets)........... 427
Received with errors (RTCP
Packets)................................ 428
Received with errors (RTP Packets)
.... 428
Received with errors (TCP Packets)
.... 427
Received with errors (Total
Packets)................................ 427
Received with errors (UDP
Packets)................................ 427
Transmitted (RTCP Packets)..... 428
Transmitted (RTP Packets)....... 428
Transmitted (TCP Packets)....... 427
Transmitted (Total Packets)...... 425
Transmitted (UDP Packets)....... 427
IP Statistics function..................... 425
ISDN BRI Interface screen fields
Status, Layer 1 Interface........... 444
Status, SPID0........................... 445
Status, SPID1........................... 446
ISDN BRI Parameters
TEI n Assignment..................... 265
ISDN BRI Parameters fields
Country.................................... 265
Dialing Options......................... 265
Inter Digit Timer....................... 265
Layer 1 Interface....................... 264
Operator................................... 265
Select BRI Interface.................. 264
Switch Information................... 265
ISDN BRI Statistics screen fields

Clear (button) 446
D Channel Information (field group) 445
Layer 1 Interface (field group) 444
Loss of Framing 444
Loss of Sync 444
Rx Packets 445
Select BRI Interface 444
SPID0 445
SPID1 446
State 444
Switch Information (field group) 445
Tx Packets 445
ISDN parameters, setting 166
ISDN-BRI operating modes
MVP-410ST/810ST) *102*
ISDN-BRI Parameter definitions 264
ISDN-BRI telephony interfaces
uses of *102*
ISDN-BRI telephony parameters *212*
ISDN-PRI
types supported 166
ISDN-PRI implementations 166

J

Java
installing 488
web GUI and 488
jitter buffer
analog 237
T1/E1 149
Jitter Value (Fax) field
analog 231
T1/E1 144
Jitter Value field
analog 239
T1/E1 151
jitter, dynamic
analog 237
T1/E1 149
jumper, DID
MVP210 104
MVP-410/810 96

K

Keep Alive field
E1 373
T1 320
key system
connecting to analog voip (MVP130) *108*
connecting to analog voip (MVP210) *106*
connecting to analog voip (MVP-410/810) *97*
Knowledge Base (online, for MultiVOIPs) 10

L

lab voip network
use in setup 72
Last button
Logs (Statistics) screen 422
Last Error (Link Management) field 431
Layer 1 Interface
ISDN-BRI 264
Layer 1 Interface fields (ISDN BRI Statistics) 444
LED definitions
analog models 32
BRI models 39
E1 24
MVP2400 16
MVP2410 17
MVP3010 24
T1 17
LED definitions (analog models)
FDX 32
LNK 32
LED definitions (analog)
Boot 32
Ethernet 32
Power 32
RCV (channel) 32
RSG 32
XMT (channel) 32
XSG 32
LED definitions (analog, E1)
LNK 24
LED definitions (analog, MVP130)
Boot 33
Ethernet 33
FDX 33
LNK 33

Power 33
RCV (channel) 33
RSG 33
XMT (channel) 33
XSG 33
LED definitions (analog, T1)
LNK 17
LED definitions (BRI models)
FDX 39
LNK 39
LED definitions (BRI)
Boot 39
Ethernet 39
Power 39
RCV (channel) 39
XMT (channel) 39
LED definitions (E1)
Boot 24
E1 24
FDX 24
IC 24
LC 24
LS 24
ONL 24
Power 24
PRI 24
LED definitions (T1)
Boot 17
FDX 17
IC 17
LC 17
LS 17
ONL 17
Power 17
PRI 17
LED indicators
E1 23
T1 16
LED indicators (analog)
channel operation 30
general operation 30
LED indicators (BRI)
channel operation 38
general operation 38
LED indicators, active
analog 30
E1 23
T1 16
LED sets (T1/E1), left and right.... 501
LED types
analog models 30
BRI models 38
lifting
precaution about 85
limitations notice (regulatory), Canadian 497
Line Build Out field
E1 165
T1 159
Line Coding field
E1 165
T1 159
Line Loopback Activation Signal (T1 stats) field 438
Line Loopback Deactivation Signal (T1 stats) field 437
Link Management (Statistics) fields
Clear command button 430
IP Address column 431
IP Address to Ping 430
Last Error 431
No. of Pings 430
No. of Pings Received 431
No. of Pings Sent 431
Ping Size in Bytes 430
Response Timeout 430
Round Trip Delay 431
Start Now command button 430
Timer Interval between Pings ... 430
Link Management (Statistics) screen
field definitions 430, 431
IP Address (ping target 431
Link Status fields
Link Management (Statistics) screen 431
List of Registered Numbers field (Registered Gateway Details) ... 448
lithium battery caution 85
LNK LED
analog models 32
analog, MVP130 33
BRI models 39
E1 24
T1 17
loading of weight in rack *91*
local configuration

analog/BRI *210*
T1/E1 .... *124*
local configuration procedure
detailed, analog *215*
detailed, T1/E1 .... *129*
summary, analog *214*
summary, T1/E1 .... *128*
local exchange numbers 71
local voip configuration (analog) .. *208*
local voip configuration (T1/E1) .. *122*
local Windows GUI vs. web GUI
comparison 487
local-rate access (E1)
to remote PSTN 18
local-rate calls to remote voip sites
E1 358
Log # (statistics, logs) field 422
log report email, customizing
analog 283, 284
T1/E1 .... 182, 183
log report email, triggering
analog 282
T1/E1 181
log reporting method, setting
analog 286
T1/E1 185
log reports
analog models *213*
T1/E1 models *127*
log reports & SMTP
analog 280
T1/E1 178
log reports and SMTP
quick 61
log reports by email
analog 280
quick 61
T1/E1 178
log reports, quick 50
logging options
analog 287
T1/E1 186
logging update interval
analog 287
T1/E1 186
logging, web GUI and 487
Login Name (SMTP) field
analog 281
T1/E1 180
Logs (Statistics) fields
Bytes received 423
Bytes Sent 422
Call Forwarded to 424
Call Transferred to 424
CT Ph# 424
Duration 422
From (gateway) 422
Gateway Name (callee) 423
Gateway Name (caller) 423
H.450 functionality 424
IP Address (callee) 423
IP Address (caller) 423
Log # 422
Mode 422
Options (caller) 423
Options callee 423
Outbound digits 423
Packets Lost 423
Packets received 423
Packets Sent 422
Start Date, Time 422
Status 422
Supplementary Services info 424
To (gateway) 422
Voice coder 423
Logs (Statistics) function 420
Logs (Statistics) screen
Delete File button 422
First button 422
Last button 422
Next button 422
Previous button 422
logs and web browser GUI
analog 287
T1/E1 186
logs by email, illustration
analog 285
T1/E1 184
Logs screen definitions
analog 287
T1/E1 186
Logs screen field definitions
analog 288
T1/E1 187
Logs screen parameters (analog)
Enable Console Messages 288

Filters 288
GUI 288
IP Address (SysLog Server) 288
Online Statistics Updation Interval 288
Port (SysLog Server) 288
SMTP 288
SNMP 288
SysLog Server Enable 288
Turn Off Logs 288

Logs screen parameters (T1/E1)
Console Message Settings 187
Enable Console Messages 186
Filters 187
GUI 187
IP Address (SysLog Server) 187
Online Statistics Updation Interval 187
Port (SysLog Server) 187
SMTP 187
SNMP 187
SysLog Server Enable 187
Turn Off Logs 187

logs screen, accessing
analog 286
T1/E1 185

long distance call savings
T1 308

long-distance call savings
E1 356

Long-Haul Mode field
E1 161
T1 155

Loss of Frame Alignment (E1 stats) field 440

Loss of Frame Alignment (T1 stats) field 437

Loss Of Framing field (ISDN BRI Statistics, Layer 1 Interface) 444

Loss of MultiFrame Alignment (E1 stats) field 441

Loss of MultiFrame Alignment (T1 stats) field 438

Loss of Sync field (ISDN BRI Parameters, Layer 1 Interface) 444

lost packets, consecutive
analog 239
T1/E1 151

lost password 468, 472

M

Mac Address
System Info 412
System Info (ANALOG/BRI) 304
System Info (T1/E1) 203

mail criteria (SMTP), records
analog 282
T1/E1 181

Mail Server IP Address (SMTP) field
analog 282
T1/E1 180

Mail Type (SMTP logs) field
analog 282
T1/E1 181

mains frequency
analog models 43
E1 models 42

management (E1 models)
local 20
remote (SNMP) 20
remote (web browser GUI) 20

management of voips, remote
analog 269
T1/E1 167

Max bandwidth (coder)
analog *232*
T1/E1 .... *145*

Max Baud Rate field
analog 231
T1/E1 144

Max Retransmission (SPP, General Options) field
E1 372
T1 319

maximum cable span
E1 161
T1 155

Maximum Jitter Value field
analog 238
T1/E1 .... *150*

Message Waiting Indication (DID-DPO) 260

Message Waiting Indication (E&M) and DID 257

Message Waiting Indication field
DID-DPO 260

E&M 257
FXO 248
FXS Loop Start 244
Minimum Jitter Value field
analog....................................... 237
T1/E1 .... *149*
Mode (call progress) field............ 416
Mode (Fax) field
analog....................................... 231
T1/E1 144
Mode (SPP) field
E1.. 372
T1.. 319
Mode (statistics, logs) field........... 422
model descriptions
E1.. 18
modem relay
analog....................................... 238
T1/E1 150
modem traffic on voip network
analog....................................... 238
T1/E1 150
modem, command
and Regional Parameters Country
Selection 61
modem, command (analog)
and Regional Parameters Country
Selection 214, 273
modem, command (T1/E1)
and Regional Parameters Country
Selection 128, 170, 171
modem, remote
configuration/command
setup for 61
modem, remote
configuration/command (analog)
setup for 214, 273
modem, remote
configuration/command (T1/E1)
setup for 128, 170, 171
Monitor Link fields
Link Management (Statistics)
screen 430
mounting
analog models 26
BRI models 34
E1 models 18
T1 models 11
mounting in rack 90
procedure for *92*
safety 85, *91*
mounting options 9
Multiplexed UDP field
E1 369
T1 316
MultiVOIP configuration software . 58
E1 models.................................. 20
T1 models.................................. 13
MultiVOIP FAQ (on MTS web site)
.... 10
MultiVOIP Program Menu items.. 449
MultiVOIP Program Menu options
Configuration 449
Configuration Port Setup 449
Date & Time Setup 449
Download Factory Defaults 450
Download Firmware 450
Download IFM Firmware 450
Set Password 450
Uninstall 450
Upgrade Software 450
MultiVOIP program menu, option
descriptions 449, 450
MultiVOIP software
installing.................................. *111*
location of files *114*
program icon location *115*
uninstalling....................... *118*, 473
MultiVOIP software (analog)
moving around in *219*
MultiVOIP software (T1/E1)
moving around in *133*
MultiVoipManager.......................... 10
analog...................................... *208*
T1/E1 .... *123*
MultiVoipManager software
E1 models.................................. 20
T1 models.................................. 13
musical jingle during call transfer
analog...................................... 293
T1/E1 192
MVP130
cabling procedure..................... *108*
Introduction................................ 25
unpacking 89
MVP210

grounding *107*
MVP210x
cabling procedure...................... *104*
unpacking.................................... 88
MVP2410
cabling procedure........................ *93*
unpacking.................................... 86
MVP-2410
remote configuration modem...... *94*
MVP3010
cabling procedure........................ *93*
remote configuration modem...... *94*
unpacking.................................... 86
MVP410
cabling procedure........................ *95*
remote configuration modem...... *99*
MVP410ST
grounding *103*
MVP-410ST
cabling procedure...................... *100*
MVP410x
unpacking.................................... 87
MVP810
cabling procedure........................ *95*
remote configuration modem...... *99*
MVP810ST
grounding *103*
MVP-810ST
cabling procedure...................... *100*
MVP810x
unpacking.................................... 87

N

national-rate calls to foreign voip sites
E1 360
Netcoder coders (RTP packetization, voice/fax)
T1/E1 433
network access
analog....................................... 271
T1/E1 169
Network Disconnection field
analog....................................... 239
T1/E1 151
network/terminal settings, voip and PBX
E1/ISDN 164
ISDN-BRI................................. 264
T1/ISDN................................... 158
No. of Entries field (Registered Gateway Details)....................... 448
No. of Pings (Link Management) field 430
No. of Pings Received (Link Management) field 431
No. of Pings Sent (Link Management) field 431
no-response & busy(E1)
forwarding, dual conditions 385
no-response & busy(T1)
forwarding, dual conditions 334
NT1 device
when required for MVP410ST.. 101
when required for MVP810ST.. 101
NT1 device, use of
BRI voip units 49, 101
Number of Days (email log criteria)
analog....................................... 282
T1/E1 181
Number of Records (email log criteria)
analog....................................... 282
T1/E1 181
numbering plan resources 406

O

obtaining updated firmware 452
Offhook alert................................ 234
Offhook Alert (Voice/Fax Params) and Intercept Tone (Regional Params) 234
Offhook Alert Timer (Voice/FAX -- AutoCall/Offhook Alert) field
analog/BRI 236
T1/E1 148
official phone numbers
caution about 72
Online Statistics Updation Interval field (Logs)
analog....................................... 288
T1/E1 187
operating system 40
operating temperature *91*
operating voltage
analog models 43
T1 models................................... 42

operation
expansion card (T1/E1)............. 501
Operator (ISDN) field
E1/ISDN 164
T1/ISDN 158
Operator definitions
ISDN-BRI................................ 268
Operator field
ISDN-BRI................................ 265
Optimization Factor field
analog....................................... *238*
T1/E1 .... *150*
Options (call progress) field 417
Options (callee, statistics, logs) field
.... 423
Options, From Details (SMTP logs) field
analog....................................... 284
T1/E1 183
Options, To Details (SMTP logs) field
analog....................................... 284
T1/E1 183
out of band, DTMF
analog....................................... 231
T1/E1 144
Outbound Digits (call progress) field
.... 416
Outbound Digits (SMTP logs) field
analog....................................... 284
T1/E1 183
Outbound digits (statistics, logs) field
.... 423
outbound phonebook
example..................................... 73
Outbound Phonebook Entries List icon
E1... 363
T1... 310
Outbound Phonebook entries, list
E1... 374
T1... 322
outbound phonebook example
quick 63
outbound vs. inbound phonebooks
E1... 361
T1... 308
Output Gain field
analog....................................... *230*
T1/E1 .... *143*
output level, fax tones
analog....................................... 231
T1/E1 144
outside line, access to................ 71, 73

P

packet priority and DiffServ
analog....................................... 223
T1/E1 137
packetization (RTP), ranges & increments
T1/E1 433
packetization rates
coder options and 432
Packets Lost (call progress) field .. 416
Packets Lost (SMTP logs) field
analog....................................... 283
T1/E1 182
Packets lost (statistics, logs) field . 423
Packets Received (call progress) field
.... 416
Packets Received (SMTP logs) field
analog....................................... 283
T1/E1 182
Packets received (statistics, logs) field
.... 423
Packets Sent (call progress) field .. 416
Packets Sent (SMTP logs) field
analog....................................... 283
T1/E1 182
Packets sent (statistics, logs) field. 422
packets, consecutive lost
analog....................................... 239
T1/E1 151
Parallel H.245 field
E1 369
T1 316
parameters tracked by console
analog....................................... 289
T1/E1 188
Pass Through (E&M) field............ 256
Pass Through Enable (FXS Loop Start interface) and AutoCall (Voice/Fax Params) 245
Password (proxy server) field
E1 371
T1 318

Password (SMTP) field
analog....................................... 282
T1/E1 180
password, lost/forgotten........ 468, 472
password, setting........................... 468
web browser GUI...................... 472
patents 2
patterns, destination
tips about.................................... 70
PBX characteristics, variations in
E1.. 405
T1.. 354
PBX interaction
analog models 26
BRI models 34
E1 models 18
T1 models 11
PC, command
COM port assignment (detailed)*116*
COM port requirement................ 51
demands upon 51
non-dedicated use 51
operating system 51
settings...................................... 51
specifications 51
PCM Law field
E1... 165
T1... 159
Permissions (SNMP) field
analog....................................... 271
T1/E1 169
personnel requirement
for rack installation *91*
to lift during installation *92*
to lift unit during installation 85
Phone Book Version
System Info (ANALOG/BRI)... 304
System Info (T1/E1) 203
phone exchanges
dedicated.................................... 71
institutional 71
local 71
non-local 71
organizational 71
Phone Number (Voice/FAX –
AutoCall/Offhook Alert) field
analog/BRI............................... *236*
T1/E1 .... *148*
Phone Signaling Tones & Cadences
analog.......................................272
T1/E1 170
phone startup configuration.............59
phone switch types
ISDN implementations in..........166
phone/IP details
importance of writing down 46
importance of writing down
(analog) *210*
importance of writing down
(T1/E1)................................*124*
phonebook
FTP remote file transfers...........476
phonebook configuration
starter 63
phonebook configuration (analog)*207*,
409
phonebook configuration (remote) 476
phonebook configuration (T1/E1). *122*
Phonebook Configuration icon
E1 363
T1 310
Phonebook Configuration Parameter
definitions
E1 ...... 366, 367, 368, 369, 370, 371
T1 ...... 313, 314, 315, 316, 317, 318
Phonebook Configuration procedure
T1 309
Phonebook Configuration Procedure
E1 362
Phonebook Configuration screen
E1 364
T1 309
Phonebook Configuration screen (E1)
Mode (SPP Protocol) 372
Phonebook Configuration screen (T1)
Mode (SPP Protocol) 319
Phonebook configuration screen fields
(E1)
Password (proxy server)............371
Re-Registration Time (proxy
server) 371
Phonebook Configuration screen
fields (E1)
Annex E (H.323, UDP
multiplexing).........................369
Client Options 372

Gatekeeper IP Address.............. 366
Gatekeeper Name...................... 366
Gateway Name.......................... 366
General Options........................ 372
H.245 Tunneling....................... 368
H.323 Multiplexing................... 368
Keep Alive............................... 372
Max Retransmission (SPP, General Options) 372
Parallel H.245 (Tunneling with Fast Start) 369
Port (SPP, General Options)..... 372
Proxy Domain Name / IP Address 370
RAS TTL Value........................ 367
Register with GateKeeper......... 366
Registrar IP Address................. 372
Registrar Options...................... 372
Registrar Port............................ 372
Retransmission (SPP, General Options) 372
Signaling Number (proxy server) 370
Signaling Port 366
Signaling Port (Gatekeeper)...... 366
Use Fast Start........................... 366
Use SIP Proxy.......................... 370
User Name (proxy server)......... 370

Phonebook configuration screen fields (T1)
Password (proxy server) 318
Re-Registration Time (proxy server) 318

Phonebook Configuration screen fields (T1)
Annex E (H.323, UDP multiplexing)......................... 316
Client Options.......................... 319
Gatekeeper IP Address.............. 313
Gatekeeper Name...................... 313
Gateway Name.......................... 313
General Options........................ 319
H.245 Tunneling....................... 315
H.323 Multiplexing................... 315
Keep Alive............................... 319
Max Retransmission (SPP, General Options) 319
Parallel H.245 (Tunneling with Fast Start)..................................... 316
Port (SPP, General Options) 319
Proxy Domain Name / IP Address 317
RAS TTL Value........................ 314
Register with GateKeeper......... 313
Registrar IP Address 319
Registrar Options 319
Registrar Port 319
Retransmission (SPP, General Options)................................ 319
Signaling Number (proxy server) 317
Signaling Port........................... 313
Signaling Port (Gatekeeper)...... 313
Use Fast Start 313
Use SIP Proxy 317
User Name (proxy server)......... 317

phonebook destination patterns....... 70
phonebook dialing patterns 70
phonebook digits
dropping 71
leading....................................... 71
non-PSTN type........................... 71
specialized codes........................ 71
types used.................................. 70

phonebook entries, coordinating
E1 362
T1 309

phonebook examples
analog....................................... *207*
mixed digital/analog................... 73

phonebook icons
E1 363
T1 310

phonebook objectives & considerations
E1 361

phonebook sidebar menu
E1 364
T1 311

phonebook tips 70
phonebook worksheet 76, 77
phonebook, analog voips............... 409
phonebook, inbound
example..................................... 73
example, quick 67

phonebook, outbound
example 73
example, quick 63
phonebooks, inbound vs. outbound
E1 361
T1 308
phonebooks, objectives & considerations
T1 308
Phonebooks, objectives & considerations
E1 356
phonebooks, sample 75
Ping Size in Bytes (Link Management) field 430
pinging and connectivity 82
pinout
BRI connector 506
command cable 503
ethernet cable 503
T1/E1 connector 504
Voice/FAX connector 504
polarity sensitivity
DID lines and (MVP210) 107
DID lines and (MVP-410/810) ... 98
Port (SPP, General Options) field
E1 372
T1 319
Port field (Registered Gateway Details) 448
Port field, SysLog Server
analog 288
T1/E1 187
Port Number (Gatekeeper RAS) field
E1 366
T1 313
Port Number (proxy server) field
E1 370
T1 317
Port Number (SMTP) field
analog 282
T1/E1 180
port number (SNMP) field
analog 271
T1/E1 169
Port Number field, SPP (Outbound Phonebook)
E1 380
T1 328
power consumption
analog models 43
E1 models 42
power frequency
analog models 43
E1 models 42
Power LED
analog models 32, 33
BRI models 39
powering of ISDN-BRI phones
MVP-410ST/810ST 102
Prefix Matched (call progress) field 416
Prefix Matched (SMTP logs) field
analog 284
T1/E1 183
prerequisites
for technical configuration (analog) *210*
for technical configuration (T1/E1) *124*
prerequisites for installation
BRI info 49
E1 info 48
IP info 47
T1 info 47
PRI
ISDN implementations 166
product CD 44
use in software installation .. 58, *111*
Product CD
E1 models 20
T1 models 13
product family 9, 10
product groups 8
Program Menu items 449
Protocol Type (outbound phonebook)
E1 377
T1 325
Proxy Domain Name / IP Address field
E1 370
T1 317
Proxy/NAT Device Parameters – Public IP Address
E1 373
T1 320

PSTN failover feature
Alternate Routing, and............. 330
analog models 27
BRI models 35
E1 models 19
T1 models 12
Pulse Density Violation (T1 stats) field... 438
Pulse Shape Level field
E1.. 165
T1.. 159

Q

Q.931 Port Number (outbound phonebook) field
T1.. 326
quality-of-service
analog... 27
BRI 35
E1... 19
T1... 12

R

rack mounting
grounding.................................. *91*
safety.................................. 85, *91*
rack mounting instructions............. 90
rack mounting procedure *92*
rack, equipment
weight capacity of...................... *91*
rack-mountable voip models........... 85
RAS TTL Value (Gatekeeper RAS) field
E1.. 367
T1.. 314
RCV (channel) LED
analog models 32
analog, MVP130 33
BRI models............................... 39
Receive Slip (E1 Stats) field......... 442
Receive Slip (T1 Stats) field......... 439
Receive Timeslot 16 Alarm Indication Signal (E1 stats) field............... 441
Receive Timeslot 16 Loss of MultiFrame Alignment (E1 stats) field... 442
Receive Timeslot 16 Loss of Signal (E1 stats) field.......................... 441
Received (RTCP Packets, IP Stats) field 428
Received (RTP Packets, IP Stats) field 428
Received (TCP Packets, IP Stats) field 427
Received (Total Packets, IP Stats) field 426
Received (UDP Packets, IP Stats) field 427
Received with Errors (RTCP Packets, IP Stats) field............................428
Received with Errors (RTP Packets, IP Stats) field............................428
Received with Errors (TCP Packets, IP Stats) field............................427
Received with Errors (Total Packets, IP Stats) field............................427
Received with Errors (UDP Packets, IP Stats) field............................427
Recipient Address (email logs) field
T1/E1 181
Recipient Address (email logs)field
analog......................................282
recovering voice packets
analog......................................*233*
T1/E1 ....*146*
Red Alarm (E1 stats) field 440
Red Alarm (T1 stats) field 437
Regeneration (dialing, FXO) field 248
Regional Parameter definitions
analog.......................274, 275, 276
T1/E1 ....172, 173, 174
Regional Parameter fields (analog)
Cadence...................................276
Country/Region (tone schemes) 274
Custom (tones) 276
Frequency 1..............................275
Frequency 2..............................275
Gain 1......................................275
Gain 2......................................275
Pulse Generation Ratio.............276
type (of tone)............................275
Regional Parameter fields (T1/E1)
Cadence...................................174
Country/Region (tone schemes) 172
Custom (tones) 174
Frequency 1..............................173

Frequency 2 173
Gain 1 173
Gain 2 173
type (of tone)............................ 172
Regional Parameters fields (analog, *except MVP130*)
Country Selection for Built-In Modem................................ 276
Regional Parameters fields (T1/E1)
Country Selection for Built-In Modem................................ 174
regional parameters, setting
analog...................................... 272
T1/E1 170
Register Duration field (Registered Gateway Details) 448
Registered Gateway Details (Statistics) screen, accessing..... 448
Registered Gateway Details 'Statistics' function......... 447, 448
Registered Gateway Details screen448
Registered Gateway Details screen fields
Description............................... 448
IP Address............................... 448
No. of Entries........................... 448
Port 448
Register Duration...................... 448
Status 448
Registered Gateway Details screen fields: 448
Registrar IP Address field
E1... 373
T1... 320
Registrar Options fields
E1... 373
T1... 320
Registrar Port field
E1... 373
T1... 320
Registration Option Parameters (Inbound Phone Book)
E1... 386
T1... 335
remote configuration modem
MVP-2410 .... *94*
MVP3010.................................. *94*
MVP410..................................... *99*
MVP810..................................... *99*
Remote Configuration/Command Modem
setup for 61
Remote Configuration/Command Modem (analog)
setup for 214, 273
Remote Configuration/Command Modem (T1/E1)
setup for 128, 170, 171
remote control/configuration
web GUI and 488
Remote Device is [legacy voip] (Outbound Phonebook)
T1 328
Remote Device is [legacy] MultiVOIP checkbox 380
remote phonebook configuration .. 476
remote voip
using to confirm configuration... 50, 63
remote voip configuration (analog) *208*
remote voip configuration (T1/E1) *122*
Remote Voip Management
analog...................................... 269
T1/E1 167
Remove Prefix (inbound) field
E1 384
T1 333
Remove Prefix (outbound) field
E1 377
T1 325
re-order tone, custom
T1/E1 175
Reply-To Address (email logs) field
T1/E1 181
Reply-To Address (email logs)field
analog...................................... 282
Requires Authentication (SMTP) field
analog...................................... 281
T1/E1 180
Re-Registration Time (proxy server)
E1 371
T1 318
Resolutions (MultiVOIP troubleshooting) 10

Response Timeout (Link Management) field.................... 430
Retransmission (SPP, General Options) field
E1 372
T1 319
Retrieve Sequence
analog...................................... 294
T1/E1 193
Retrieve Sequence (analog) 291
Retrieve Sequence (T1/E1)........... 190
RFC768....................................... 509
RFC793....................................... 509
ring cadences, custom
analog...................................... 279
T1/E1 .... 175, 177
Ring Count (FXO) field................ 250
Ring Count field
FXS Loop Start......................... 244
Ring Count forwarding condition
E1 386
T1 335
ring tone, custom
analog...................................... 278
T1/E1 .... 175, 176
ring-tones
analog...................................... 277
T1/E1 175
Round Trip Delay (Link Management) field.................... 431
Round Trip Delay field
E1 382
T1 330
RSG LED
analog models 32
analog, MVP130 33
RTP packetization, ranges & increments................................ 433
RTP Parameters screen 433
Rx Packets field (ISDN BRI Statistics, D-Channel Information) 445

S

Safety Recommendations for Rack Installations................................ *91*
safety warnings 85
Safety Warnings Telecom.......... 85
sample phonebooks 75
Save Setup command
analog...................................... 305
T1/E1 204
saving configuration
analog...................................... 305
T1/E1 204
user... 466
Saving the MultiVOIP Configuration
analog...................................... 305
T1/E1 204
savings on toll calls
E1 356
T1 308
scale-ability
E1 18
T1 11
second dial tone
and comma use........................... 72
Select All (SMTP logs) field
analog...................................... 283
T1/E1 182
Select BRI Interface field............. 444
Select BRI Interface ISDN-BRI field
BRI... 264
Select Channel field
analog...................................... *230*
T1/E1 .... *143*
Select Channel, Supplementary Services field
analog...................................... *293*
T1/E1 .... *192*
Selected Coder field
analog...................................... *232*
T1/E1 .... *145*
Set Baud Rate
analog...................................... 301
T1/E1 200
Set Custom Tones & Cadences
T1/E1 175
Set ISDN Parameters 166
Set Log Reporting Method
analog...................................... 286
T1/E1 185
Set Password (program menu option) , command................................. 468
Set Password (web browser GUI) , command................................. 472

Set Password option description
(MultiVOIP program menu) 450
Set Regional Parameters
analog....................................... 272
T1/E1 170
Set SMTP Parameters
analog....................................... 280
T1/E1 178
Set SNMP Parameters
analog....................................... 269
T1/E1 167
Set Supplementary Services
Parameters
analog....................................... 290
T1/E1 189
Set T1/E1/ISDN Parameters 152
Set Telephony Interface Parameters
.... 240
Set Voice/FAX Parameters
analog....................................... 227
T1/E1 140
setting IP parameters
analog....................................... *220*
T1/E1 .... *134*
setting password........................... 468
web browser GUI..................... 472
setting RTP Parameters................ 433
setting user defaults 466
setup, saving
analog....................................... 305
T1/E1 204
user 466
setup, saving user values............... 466
Signal (type, E&M) field 256
signaling cadences
analog....................................... 272
T1/E1 170
signaling parameters (analog
telephony) 240
Signaling Port (proxy server) field
E1... 370
T1... 317
Signaling Port field
E1... 366
T1... 313
signaling tones
analog....................................... 272
T1/E1 170
signaling types
analog telephony (MVP130)..... *108*
analog telephony (MVP130FXS)
.... *109*
analog telephony (MVP210).... *106*,
107
analog telephony (MVP-410/810)
.... *97*, *98*
Silence Compression (call progress)
field 417
Silence Compression (SMTP logs)
analog...................................... 284
T1/E1 183
Silence Compression field
analog...................................... *233*
T1/E1 .... *146*
Silence Detection (FXO) field 250
Silence Timer (FXO) field 250
simulated voip network
use in startup 72
Single-Port Protocol, general
description
analog.. 27
BRI... 35
E1 19
T1 12
SIP
compatibility
analog models 27
BRI models 35
E1 models............................... 19
T1 models............................... 12
SIP Fields (Outbound Phonebook)
E1 379
T1 327
SIP incompatibility with H.450
Supplementary Services
analog................................ 27, 290
BRI... 35
E1 19
T1 12
T1/E1 189
SIP Port Number field
E1 379
T1 327
SIP port number, standard
E1 379
T1 327

SIP Proxy Parameters
E1 ... 370
T1 ... 317
SIP URL field
E1 ... 379
T1 ... 327
SMTP
quick setup ... 61
SMTP (log reporting type) button
analog ... 288
T1/E1 ... 187
SMTP logs by email, illustration
analog ... 285
T1/E1 ... 184
SMTP Parameters definitions
analog ... 282
T1/E1 ... 179
SMTP Parameters fields (analog)
Enable SMTP ... 281
Login Name ... 281
Mail Server IP Address ... 282
Mail Type ... 282
Number of Days ... 282
Number of Records ... 282
Password ... 282
Port Number ... 282
Recipient Address ... 282
Reply-To Address ... 282
Requires Authentication ... 281
Subject ... 282
SMTP Parameters fields (T1/E1)
Enable SMTP ... 179
Login Name ... 179
Mail Server IP Address ... 179
Mail Type ... 181
Number of Days ... 181
Number of Records ... 181
Password ... 179
Port Number ... 179
Recipient Address ... 181
Reply-To Address ... 181
Requires Authentication ... 179
Subject ... 181
SMTP parameters, accessing
analog ... 280
T1/E1 ... 178
SMTP parameters,setting
analog ... 280
T1/E1 ... 178
SMTP port, standard
analog ... 282
T1/E1 ... 180
SMTP prerequisites
analog models ... 213
quick ... 50
T1/E1 models ... 127
SMTP, enabling
analog ... 280
T1/E1 ... 178
SNMP (log reporting type) button
analog ... 288
T1/E1 ... 187
SNMP agent program
analog ... *208*
T1/E1 ... *123*
SNMP agent, enabling
analog ... 269
T1/E1 ... 167
SNMP Parameter Definitions
T1/E1 ... 169
SNMP Parameter fields (analog)
Address ... 271
Community Name (2) ... 271
Community Name 1 ... 271
Enable SNMP Agent ... 271
Permissions (1) ... 271
Permissions (2) ... 271
Port Number ... 271
SNMP Parameter fields (T1/E1)
Address ... 169
Community Name (2) ... 169
Community Name 1 ... 169
Enable SNMP Agent ... 169
Permissions (1) ... 169
Permissions (2) ... 169
Port Number ... 169
SNMP Parameters, setting
analog ... 269
T1/E1 ... 167
software
control ... 58
uninstalling (detailed) ... *118*
updates (analog) ... *208*
updates (T1/E1) ... *123*
software (MultiVOIP)
uninstalling ... 473

software configuration
summary *111*
software installation
detailed..................................... *111*
quick 58
software loading........................... *111*
software loading, quick.................. 58
software version numbers *113*
software, MultiVOIP (analog)
screen-surfing in *219*
software, MultiVOIP (T1/E1)
moving around in...................... *133*
screen-surfing in *133*
software, MultiVOIP(analog)
moving around in...................... *219*
software, on command PC 58
Solving Common Connection
Problems
analog...................................... *218*
T1/E1 .... *132*
sound quality, improving
analog...................................... *233*
T1/E1 .... *146*
specialized codes, in dialing 71
specifications
E1 models 42
T1 models 41
SPID0 field (ISDN BRI Statistics,
Switch Information).................. 445
SPID1 field (ISDN BRI Statistics,
Switch Information).................. 446
SPP Fields (Outbound Phonebook)
E1.. 380
T1.. 328
SPP Fields (Phonebook Configuration
screen)
E1.. 372
T1.. 319
SPP, general description
analog... 27
BRI 35
E1.. 19
T1.. 12
SPP, strengths & compatibilities of
analog... 27
BRI 35
E1.. 19
T1.. 12
ST interface (ISDN-BRI)
description.................................507
Start Date, Time (SMTP logs) field
analog.......................................283
T1/E1 182
Start Date,Time (statistics, logs) field
....422
Start Modes (DID-DPO) field259, 260
Start Now command (Link
Management) button 430
starter configuration
inbound phonebook.....................67
outbound phonebook...................63
phone/IP 59
startup tasks....................................46
State field (ISDN BRI Statistics,
Layer 1 Interface)......................444
Options (caller...............................423
Status (call progress) field.............419
Status (statistics, logs) field 422
Status field (ISDN BRI Statistics,
Layer 1 Interface)......................444
Status field (ISDN BRI Statistics,
SPID0) 445
Status field (ISDN BRI Statistics,
SPID1) 446
Status field (Registered Gateway
Details).....................................448
Status Freeze Signalling Active (E1
stats) field.................................441
Status Freeze Signalling Active (T1
stats) field.................................437
Subject (email logs) field
analog.......................................282
T1/E1 181
supervisory signaling (analog)......241
supervisory signaling parameters
(analog telephony).....................240
supervisory signaling types
MVP130...................................*108*
MVP130FXS............................*109*
MVP210...........................*106*, *107*
MVP-410/810 ....*97*, *98*
Supplementary (Telephony) Services
ANALOG....................................30
BRI...38
E1 23
T1 16

Supplementary Services (analog)
Alerting Party............ 297, 298, 299
Call Hold.................................. 291
Call Hold Enable....................... 294
Call Name Identification........... 291
Call Transfer 291
Call Transfer Enable 293
Call Waiting............................. 291
Call Waiting Enable.................. 294
Caller Name Identification Enable
.... 295
Calling Party 296
Enable Call Hold....................... 294
Enable Call Transfer 293
Enable Call Waiting.................. 294
Enable Caller Name Identification
.... 295
Hold Sequence 294
Retrieve Sequence 294
Select Channel 293
Transfer Sequence 293
Supplementary Services (T1/E1)
Alerting Party............ 196, 197, 198
Call Hold.................................. 190
Call Hold Enable....................... 193
Call Name Identification........... 191
Call Transfer 190
Call Transfer Enable 192
Call Waiting............................. 190
Call Waiting Enable.................. 193
Caller Name Identification Enable
.... 194
Calling Party 195
Enable Call Hold....................... 193
Enable Call Transfer 192
Enable Call Waiting.................. 193
Enable Caller Name Identification
.... 194
Hold Sequence 193
Retrieve Sequence 193
Select Channel 192
Transfer Sequence 192
Supplementary Services Info
logs for..................................... 424
Supplementary Services Parameter
buttons (analog)
Copy Channel *300*
Default *300*
Supplementary Services Parameter
buttons (T1/E1)
Copy Channel............................ *199*
Default *199*
Supplementary Services Parameter
Definitions
analog*293*, *294*, *295*, *296*, *297*, *298*, *299*, *300*
T1/E1 *192*, *193*, *194*, *195*, *196*, *197*, *198*, *199*
Supplementary Services Parameter
fields (analog)
Call Transfer Enable *293*
Call Waiting Enable *294*
Hold Sequence *294*
Retrieve Sequence..................... *294*
Transfer Sequence..................... *293*
Supplementary Services Parameter
fields (analog)
Select Channel *293*
Supplementary Services Parameter
fields (analog)
Call Hold Enable....................... *294*
Supplementary Services Parameter
fields (analog)
Call Name Identification Enable*295*
Supplementary Services Parameter
fields (analog)
Calling Party *296*
Supplementary Services Parameter
fields (analog)
Allowed Name Types................ *296*
Supplementary Services Parameter
fields (analog)
Alerting Party............................ *297*
Supplementary Services Parameter
fields (analog)
Allowed Name Types................ *297*
Supplementary Services Parameter
fields (analog)
Busy Party................................. *298*
Supplementary Services Parameter
fields (analog)
Allowed Name Types................ *298*
Supplementary Services Parameter
fields (analog)
Connected Party *299*

Supplementary Services Parameter fields (analog)
Allowed Name Types *299*
Supplementary Services Parameter fields (analog)
Caller ID *300*
Supplementary Services Parameter fields (T1/E1)
Call Transfer Enable *192*
Call Waiting Enable.................. *193*
Hold Sequence.......................... *193*
Retrieve Sequence *193*
Transfer Sequence *192*
Supplementary Services Parameter fields (T1/E1)
Select Channel *192*
Supplementary Services Parameter fields (T1/E1)
Call Hold Enable....................... *193*
Supplementary Services Parameter fields (T1/E1)
Call Name Identification Enable*194*
Supplementary Services Parameter fields (T1/E1)
Calling Party *195*
Supplementary Services Parameter fields (T1/E1)
Allowed Name Types *195*
Supplementary Services Parameter fields (T1/E1)
Alerting Party........................... *196*
Supplementary Services Parameter fields (T1/E1)
Allowed Name Types *196*
Supplementary Services Parameter fields (T1/E1)
Busy Party................................ *197*
Supplementary Services Parameter fields (T1/E1)
Allowed Name Types *197*
Supplementary Services Parameter fields (T1/E1)
Connected Party....................... *198*
Supplementary Services Parameter fields (T1/E1)
Allowed Name Types *198*
Supplementary Services Parameter fields (T1/E1)
Caller ID.................................. *199*
Supplementary Services Parameters screen, accessing
analog...................................... *290*
T1/E1 .... *189*
Supplementary Services parameters, setting
analog...................................... *290*
T1/E1 .... *189*
Supplementary Services, incompatible with SIP
analog................................. 27, 290
BRI.. 35
E1 19
T1 12
T1/E1 189
Switch Information fields (ISDN BRI Statistics).................................. 445
switch types (phone) and ISDN-PRI
.... 166
SysLog client
ANALOG.................................. 29
BRI.. 37
E1 22
T1 15
SysLog client programs
availability............................... 491
features & presentation types.... 493
SysLog functionality
ANALOG.................................. 29
BRI.. 37
E1 22
T1 15
SysLog server
ANALOG.................................. 29
BRI.. 37
E1 22
T1 15
SysLog Server Enable field
analog...................................... 288
T1/E1 187
SysLog Server function
as added feature........................ 491
capabilities of........................... 493
enabling.................................... 492
location of 491
SysLog Server IP Address field
analog...................................... 288

T1/E1 187
SysLog Server, enabling
analog.. 287
T1/E1 186
System Information screen
for op & maint *411*
System Information screen, accessing
analog.. *302*
T1/E1 .... *201*
System Information update interval, setting
analog.. *302*
for op & maint *413*
T1/E1 .... *201*

T

T1 model descriptions..................... 11
T1 Parameter definitions...... 155, 156, 157, 159
Clocking.................................... 159
Line Build-Out.......................... 159
Line Coding 159
PCM Law.................................. 159
Pulse Shape Level..................... 159
Yellow Alarm Format............... 159
T1 Parameter fields
CAS Protocol 156
CRC Check 155
Detect Flash Hook 157
Frame Format............................ 155
Long-Haul Mode 155
T1/E1/ISDN.............................. 155
T1 Parameters screen..................... 154
T1 Statistics field definitions 438, 439
T1 Statistics fields
Bipolar Violation 439
Frame Search Restart Flag........ 438
Line Loopback Activation Signal 438
Loss of MultiFrame Alignment 438
Pulse Density Violation 438
Receive Slip 439
Transmit Data Underrun 438
Transmit Line Open.................. 438
Transmit Slip 438
Transmit Slip Negative 438
Yellow Alarm 438
T1 telephony parameters............... *125*
T1/E1 connector pinout.................504
T1/E1 Statistics function...........435
T1/E1/ISDN field
E1 161
T1 155
T1/E1/ISDN Parameters screen, accessing 152
T1/E1/ISDN parameters, setting ...152
T1/ISDN Parameter definitions.....158
T1/ISDN Parameter fields
Country 158
Enable ISDN-PRI......................158
Operator 158
Terminal Network 158
table-top voip models......................85
TCP/UDP compared
E1 379
IP Statistics context...................426
T1 327
technical configuration
startup...59
technical configuration (analog)
prerequisites to *210*
summary...................................*207*
technical configuration (T1/E1)
prerequisites to *124*
summary...................................*122*
technical configuration procedure
detailed, analog *215*
detailed, T1/E1 ....*129*
summary, analog *214*
summary, T1/E1 ....*128*
TEI Assignment fields (ISDN BRI Statistics, Switch Information)..445
TEI n Assignment
ISDN-BRI 265
TEIn fields (ISDN BRI Statistics, Switch Information) 445
telco authorities and ISDN 166
telecom safety warnings.............85
telephony interface parameters, setting.......................................240
telephony interfaces
uses of ... *97*, *98*, *106*, *107*, *108*, *109*
telephony interfaces, BRI................54
telephony signaling cadences
analog.......................................272
T1/E1 170

telephony signaling tones
analog ... 272
T1/E1 ... 170
telephony startup configuration ... 59
telephony toning schemes
analog ... 277
T1/E1 ... 175
temperature
operating ... *91*
terminal mode (ISDN-BRI) & D-channel support
MVP-410ST/810ST ... 101
Terminal Network field
E1/ISDN ... 164
T1/ISDN ... 158
terminal/network settings, voip and PBX
E1/ISDN ... 164
ISDN-BRI ... 264
T1/ISDN ... 158
timeout interval
voips under H.323 gatekeeper .. 314, 367
voips under SIP proxy server ... 318, 371
Timer Interval between Pings (Link Management) field ... 430
tips, phonebook ... 70
To (gateway, statistics, logs) field 422
toll-call savings
E1 ... 356
T1 ... 308
toll-free access (T1)
to remote PSTN ... 11
within voip network ... 11
toll-free access (within voip network)
E1 ... 18
T1 ... 11
Tone Detection (FXO disconnect criteria) field ... 249
Tone Pair (custom) field
analog ... 278
T1/E1 ... 176
tone pairs, custom
T1/E1 ... 175
tones, signaling
analog ... 272
T1/E1 ... 170
Total Digits (outbound) field
E1 ... 377
T1 ... 325
touch tone frequencies ... 250
trace on/off (logging)
analog ... 289
T1/E1 ... 188
Transfer Sequence
analog ... 293
T1/E1 ... 192
Transfer Sequence (analog) ... 291
Transfer Sequence (T1/E1) ... 190
Transmit Data Overflow (E1 stats) field ... 441
Transmit Data Overflow (T1 stats) field ... 437
Transmit Data Underrun (E1 stats) field ... 442
Transmit Data Underrun (T1 stats) field ... 438
Transmit Line Open (E1 stats) field ... 442
Transmit Line Open (T1 stats) field ... 438
Transmit Line Short (E1 stats) field ... 441
Transmit Line Short (T1 stats) field ... 437
Transmit Slip (E1 stats) field ... 442
Transmit Slip (T1 stats) field ... 438
Transmit Slip Negative (E1 stats) field ... 442
Transmit Slip Negative (T1 stats) field ... 438
Transmit Slip Positive (E1 stats) field ... 441
Transmit Slip Positive (T1 stats) field ... 437
Transmitted (RTCP Packets, IP Stats) field ... 428
Transmitted (RTP Packets, IP Stats) field ... 428
Transmitted (TCP Packets, IP Stats) field ... 427
Transmitted (Total Packets, IP Stats) field ... 426
Transmitted (UDP Packets, IP Stats) field ... 427

Transport Protocol (SIP) field
E1 ... 379
T1 ... 327
trap manager parameters (SNMP)
T1/E1 ... 169
triggering log report email
analog ... 282
T1/E1 ... 181
troubleshooting ... 82
Troubleshooting Resolutions for MultiVOIPs ... 10
Turn Off Logs field
analog ... 288
T1/E1 ... 187
Tx Packets field (ISDN BRI Statistics, D-Channel Information) ... 445
Type (E&M type) field ... 256
Type (of tone) field
analog ... 275
T1/E1 ... 172
Type-of-Service IP header field & DiffServ
analog ... 223
T1/E1 ... 137

U

U interface (ISDN-BRI)
description ... 507
UDP multiplexed (H.323 Annex E) field
E1 ... 369
T1 ... 316
UDP/TCP compared
E1 ... 379
IP Statistics context ... 426
T1 ... 327
unconditional forwarding
E1 ... 385
T1 ... 334
Uninstall (program menu option) , command ... 473
Uninstall option description (MultiVOIP program menu) ... 450
uninstalling MultiVOIP software. *118*, 473
unobtainable tone, custom
analog ... 278
T1/E1 ... 175, 176
unobtainable tones
analog ... 175, 277
unpacking
MVP130 ... 89
MVP210x ... 88
MVP2410 ... 86
MVP3010 ... 86
MVP410x ... 87
MVP810x ... 87
Up Time
System Info ... 412
System Info (ANALOG/BRI) ... 304
System Info (T1/E1) ... 203
update interval (logging)
analog ... 287
T1/E1 ... 186
updated firmware, obtaining ... 452
upgrade
E1 ... 18
T1 ... 11
upgrade card (analog, 4-to-8 channel) installation ... 511
upgrade card (T1/E1) installation .. 499
Upgrade Software option description
MultiVOIP program menu ... 450
upgrade, firmware ... 456
uploads vs. downloads (FTP) ... 476
Use Fast Start (Q.931) field
E1 ... 366
T1 ... 313
Use Gatekeeper (Outbound Phonebook) field
E1 ... 378
T1 ... 326
Use Proxy (SIP) field
E1 ... 379
T1 ... 327
Use Registrar field (Outbound Phonebook)
E1 ... 380
T1 ... 328
Use SIP Proxy field
E1 ... 370
T1 ... 317
user default configuration, creating
analog ... 306
T1/E1 ... 205
user defaults, downloading ... 466

user defaults, setting 466
user name
Windows GUI........................... 468
User Name (proxy server) field
E1... 370
T1... 317
user values (software), saving....... 466

V

variations in PBX characteristics
E1... 405
T1... 354
version numbers.............................. 10
version numbers (software) *113*
version, firmware.......................... 456
Voice Coder (call progress) field.. 416
Voice coder (statistics, logs) field. 423
VOICE connector
MVP130FXS *108*
voice delay
analog................................ 237, 238
T1/E1 .... 149, 150
Voice Gain field
analog....................................... *230*
T1/E1 .... *143*
voice packets (analog)
recovering lost/corrupted *233*
voice packets (T1/E1)
recovering lost/corrupted *146*
voice packets, consecutive lost
analog....................................... 239
T1/E1 151
voice packets, delayed
analog................................ 237, 238
T1/E1 .... 149, 150
voice packets, re-assembling
analog....................................... 231
voice packets, re-assembly
T1/E1 144
voice quality, improving
analog....................................... *233*
T1/E1 .... *146*
voice quality, versus delay
analog....................................... 238
T1/E1 150
Voice/FAX connector pinout........ 504
Voice/FAX Parameter definitions
analog................................ 238, 239
T1/E1 .... 150, 151
Voice/FAX Parameter Definitions
analog........ 230, 231, *232*, *233*, *237*
T1/E1 .... *143*, 144, *145*, *146*, *149*
Voice/FAX Parameter fields (analog)
Automatic Disconnection..........239
Call Duration............................239
Consecutive Packets Lost..........239
Copy Channel...........................*230*
Default *230*
DTMF Gain..............................*230*
DTMF Gain (High Tones) *230*
DTMF Gain (Low Tones).........*230*
DTMF In/Out of Band *230*
Duration (DTMF)......................*230*
Dynamic Jitter Buffer................237
Echo Cancellation *233*
Fax Enable 231
Fax Volume..............................231
Forward Error Correction..........*233*
Input Gain *230*
Jitter Value...............................239
Jitter Value (Fax) 231
Max Baud Rate (Fax)................231
Maximum Jitter Value 238
Minimum Jitter Value 237
Mode (Fax)...............................231
Network Disconnection.............239
Optimization Factor 238
Output Gain..............................*230*
Select Channel *230*
Silence Compression................*233*
Voice Gain *230*
Voice/FAX Parameter fields (analog/BRI)
AutoCall/Offhook Alert *234*, *235*
AutoCall/Offhook Alert fields .234, 235
Generate Local Dial Tone 235
Offhook Alert Timer 236
Phone Number (Auto Call/Offhook Alert)...................................*236*
Voice/FAX Parameter fields (T1/E1)
AutoCall/Offhook Alert *147*
AutoCall/Offhook Alert fields .. 147
Automatic Disconnection.......... 151
Call Duration............................ 151
Consecutive Packets Lost.......... 151

Copy Channel *143*
Default *143*
DTMF Gain *143*
DTMF Gain (High Tones)........ *143*
DTMF Gain (Low Tones)......... *143*
DTMF In/Out of Band............. *143*
Duration (DTMF) 143
Dynamic Jitter Buffer 149
Echo Cancellation.................... *146*
Fax Enable 144
Fax Volume 144
Forward Error Correction *146*
Input Gain *143*
Jitter Value............................... 151
Jitter Value (Fax) 144
Max Baud Rate 144
Maximum Jitter Value 150
Minimum Jitter Value............... 149
Mode (Fax) 144
Network Disconnection 151
Offhook Alert Timer................ 148
Optimization Factor................. 150
Output Gain *143*
Phone Number 148
Phone Number (AutoCall/Offhook Alert).................................. *148*
Select Channel *143*
Silence Compression *146*
Voice Gain............................... *143*
Voice/FAX Parameters screen, accessing
analog...................................... *227*
T1/E1 .... *140*
Voice/FAX parameters, setting
analog...................................... *227*
T1/E1 .... *140*
Voip Caller ID Case #1 –telco standard CID enters voip system 252
Voip Caller ID Case #2 – H.323 voip system, no telco CID................ 252
Voip Caller ID Case #3 –SPP 253
Voip Caller ID Case #4 – Remote FXS call on H.323 voip system 253
Voip Caller ID Case #5 –DID channel in H.323 voip system 254
voip dialing digits
non-PSTN type 71
types used.................................. 70
voip email account
analog...................................... 281
T1/E1 180
voip management, remote
analog...................................... 269
T1/E1 167
Voip Media PHB field
analog...................................... *222*
T1/E1 .... *136*
voip network, lab/simulated
use in startup 72
voip software
host PC 40, 51
voip software (analog)
host PC *208*
voip software (T1/E1)
host PC *123*
voip system example, conceptual (E1)
calls to remote PSTN 358
foreign calls, national rates 360
voip site to voip site 357
voip system example, digital & analog, with phonebook details
E1 394
T1 342
voip system example, digital only, with phonebook details
E1 387
T1 336
voip(E1)
basic functions of 19
voip(T1)
basic functions of 12
voltage, operating
analog models 43
E1 models.................................. 42

W

warnings, safety 85
web browser GUI and logs
analog...................................... 287
T1/E1 186
web browser GUI, enabling
analog................................. 59, 226
T1/E1 139
web browser interface

browser version requirement ... 486, 489
general 486
Java requirement 486
prerequisite local assigning of IP address 487
video useability 486
web GUI
Java and 488
remote control/configuration and 488
web GUI vs. local Windows GUI
comparison 487
web GUI vs. Windows GUI
BRI 36
web GUI, logging and 487
weight
analog models 43
E1 models 42
T1 models 41
weight loading
in rack *91*
weight of unit
lifting precaution 85
personnel requirement 85
Well Known Ports 509
well-known port number, SMTP
analog 282
T1/E1 180
well-known port, gatekeeper registration
E1 366
T1 313
well-known port, H.323 params
E1 378
T1 326
well-known port, Q.931 params, H.323
E1 366
T1 313
well-known port, SIP
E1 379
T1 327
well-known port, SNMP
analog 271
T1/E1 169
Windows GUI vs. web GUI
BRI 37
wink signaling (DID-DPO) 260
wink signaling (E&M) 256
Wink Timer (DID-DPO) field 260
Wink Timer (E&M) field 256
worksheet
phonebook 76, 77

X

XMT (channel) LED
analog models 32
analog, MVP130 33
BRI models 39
XSG LED
analog models 32
analog, MVP130 33

Y

Yellow Alarm (E1 stats) field 441
Yellow Alarm (T1 stats) field 438
Yellow Alarm Format field (T1) ... 159

S000249K