
Multi*VOIP*TM

Voice / Fax over IP Networks

User Guide for Voice/IP Gateways

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

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



User Guide

S000249H

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

This publication may not be reproduced, in whole or in part, without prior expressed written permission from Multi-Tech Systems, Inc. All rights reserved.
Copyright © 2003, by Multi-Tech Systems, Inc.

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) Add embedded gatekeeper models, ISDN-BRI models, MultiVantage Apx., SPP protocol, & Call State Apx.
E	Remove MultiVantage. (04/18/03)
F	Update ISDN-BRI info in SW version 5.02c. (06/04/03)
G	Add MVP130 information. (06/30/03)
H	Revisions to ISDN-BRI & MVP130 content. (08/15/03)

Patents

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

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
<http://www.multitech.com>

CONTENTS

CHAPTER 1: OVERVIEW	8
ABOUT THIS MANUAL.....	9
INTRODUCTION TO TI MULTIVOIPS (MVP2400, MVP2410, & MVP24-48).....	12
<i>TI Front Panel LEDs</i>	17
INTRODUCTION TO EI MULTIVOIPS (MVP3010 & MVP30-60).....	19
<i>EI Front Panel LEDs</i>	24
<i>EI LED Descriptions</i>	25
INTRODUCTION TO ANALOG MULTIVOIPS (MVP130, MVP-210/410/810 & MVP428)	26
<i>Analog MultiVOIP Front Panel LEDs</i>	31
INTRODUCTION TO ISDN-BRI MULTIVOIPS (MVP410ST & MVP810ST).....	35
<i>ISDN BRI MultiVOIP Front Panel LEDs</i>	39
<i>ISDN-BRI MultiVOIP LED Descriptions</i>	40
COMPUTER REQUIREMENTS.....	41
SPECIFICATIONS.....	42
<i>Specs for Digital TI MultiVOIP Units</i>	42
<i>Specs for Digital EI MultiVOIP Units</i>	43
<i>Specs for Analog/BRI MultiVOIP Units</i>	44
INSTALLATION AT A GLANCE.....	45
RELATED DOCUMENTATION.....	45
CHAPTER 2: QUICK START INSTRUCTIONS	46
INTRODUCTION.....	47
MULTIVOIP STARTUP TASKS.....	47
<i>Phone/IP Details *Absolutely Needed* Before Starting the Installation</i>	48
Gather IP Information.....	48
Gather Telephone Information (T1).....	48
Gather Telephone Information (E1).....	49
Gather Telephone Information (Analog).....	49
Gather Telephone Information (ISDN BRI).....	50
Obtain Email Address for VOIP (for email call log reporting).....	51
Identify Remote VOIP Site to Call.....	51
Identify VOIP Protocol to be Used.....	51
<i>Placement</i>	52
<i>The Command/Control Computer (Specs & Settings)</i>	52
<i>Quick Hookups</i>	53
<i>Load MultiVOIP Control Software onto PC</i>	58
<i>Phone/IP Starter Configuration</i>	59
<i>Phonebook Starter Configuration (with remote voip)</i>	66
Outbound Phonebook.....	66
Inbound Phonebook.....	70
<i>Phonebook Tips</i>	73
<i>Phonebook Example</i>	76
<i>Connectivity Test</i>	81
<i>Troubleshooting</i>	85

CHAPTER 3: MECHANICAL INSTALLATION AND CABLING.....	87
INTRODUCTION.....	88
SAFETY WARNINGS.....	88
<i>Lithium Battery Caution</i>	88
<i>Safety Warnings Telecom</i>	88
UNPACKING YOUR MULTIVOIP.....	89
<i>Unpacking the MVP2410/3010</i>	89
<i>Unpacking the MVP2400</i>	90
<i>Unpacking the MVP-410x/810x</i>	91
<i>Unpacking the MVP210x</i>	92
<i>Unpacking the MVP130</i>	93
RACK MOUNTING INSTRUCTIONS FOR MVP-2410/3010 & MVP-410x/810x.....	94
<i>Safety Recommendations for Rack Installations</i>	95
<i>19-Inch Rack Enclosure Mounting Procedure</i>	96
CABLING.....	97
<i>Cabling Procedure for MVP2410/3010</i>	97
<i>Cabling Procedure for MVP2400</i>	98
<i>Cabling Procedure for MVP-410/410G/810/810G</i>	99
<i>Cabling Procedure for MVP-410ST/810ST</i>	101
<i>Cabling Procedure for MVP210x</i>	105
<i>Cabling Procedure for MVP130</i>	107
CHAPTER 4: SOFTWARE INSTALLATION.....	108
INTRODUCTION.....	109
LOADING MULTIVOIP SOFTWARE ONTO THE PC.....	109
UN-INSTALLING THE MULTIVOIP CONFIGURATION SOFTWARE.....	116
CHAPTER 5: TECHNICAL CONFIGURATION FOR DIGITAL T1/E1	
MULTIVOIPS (MVP2400, MVP2410, MVP3010).....	119
CONFIGURING THE DIGITAL T1/E1 MULTIVOIP.....	120
LOCAL CONFIGURATION.....	122
<i>Pre-Requisites</i>	122
IP Parameters.....	122
T1 Telephony Parameters (for MVP2400 & MVP2410).....	123
E1 Telephony Parameters (for MVP3010).....	124
SMTP Parameters (for email call log reporting).....	125
<i>Local Configuration Procedure (Summary)</i>	126
<i>Local Configuration Procedure (Detailed)</i>	127
<i>Modem Relay</i>	144
CHAPTER 6: TECHNICAL CONFIGURATION FOR ANALOG/BRI	
MULTIVOIPS (MVP130, MVP-210/210G, MVP-410/410G, MVP-810/810G &	
MVP-410ST/810ST).....	195
CONFIGURING THE ANALOG/BRI MULTIVOIP.....	196
LOCAL CONFIGURATION.....	199
<i>Pre-Requisites</i>	199
IP Parameters.....	199

Analog Telephony Interface Parameters (for MVP130/210/410/810).....	200
ISDN-BRI Telephony Parameters (for MVP-410ST/810ST).....	201
SMTP Parameters (for email call log reporting).....	202
<i>Local Configuration Procedure (Summary)</i>	203
<i>Local Configuration Procedure (Detailed)</i>	204
<i>Modem Relay</i>	221
CHAPTER 7: T1 PHONEBOOK CONFIGURATION	277
CONFIGURING THE MVP2400/2410 MULTIVOIP PHONEBOOKS.....	278
T1 PHONEBOOK EXAMPLES.....	301
<i>3 Sites, All-T1 Example</i>	301
<i>Configuring Mixed Digital/Analog VOIP Systems</i>	307
<i>Call Completion Summaries</i>	316
<i>Variations in PBX Characteristics</i>	319
CHAPTER 8: E1 PHONEBOOK CONFIGURATION	320
MVP3010 INBOUND AND OUTBOUND MULTIVOIP PHONEBOOKS	321
<i>Free Calls: One VOIP Site to Another</i>	322
<i>Local Rate Calls: Within Local Calling Area of Remote VOIP</i>	323
<i>National Rate Calls: Within Nation of Remote VOIP Site</i>	325
<i>Inbound versus Outbound Phonebooks</i>	326
PHONEBOOK CONFIGURATION PROCEDURE.....	330
E1 PHONEBOOK EXAMPLES.....	349
<i>3 Sites, All-E1 Example</i>	349
<i>Configuring Digital & Analog VOIPs in Same System</i>	356
<i>Call Completion Summaries</i>	365
<i>Variations in PBX Characteristics</i>	368
<i>International Telephony Numbering Plan Resources</i>	369
CHAPTER 9: ANALOG/BRI PHONEBOOK CONFIGURATION	371
CHAPTER 10: OPERATION AND MAINTENANCE	373
OPERATION AND MAINTENANCE	374
<i>System Information screen</i>	374
<i>Statistics Screens</i>	376
<i>About Call Progress</i>	376
<i>About Logs</i>	382
<i>About Reports</i>	385
<i>About IP Statistics</i>	386
<i>About Packetization Time</i>	390
<i>About T1/E1 and BRI Statistics</i>	393
<i>About Registered Gateway Details</i>	405
MULTIVOIP PROGRAM MENU ITEMS.....	407
<i>Date and Time Setup</i>	409
<i>Obtaining Updated Firmware</i>	409
<i>Implementing a Software Upgrade</i>	413
Identifying Current Firmware Version	413
Downloading Firmware.....	414
Downloading CAS Protocols.....	417

Downloading Factory Defaults.....	419
<i>Setting and Downloading User Defaults</i>	421
<i>Downloading IFM Firmware</i>	423
<i>Setting a Password (Windows GUI)</i>	424
<i>Setting a Password (Web Browser GUI)</i>	427
<i>Un-Installing the MultiVOIP Software</i>	428
<i>Upgrading Software</i>	430
FTP SERVER FILE TRANSFERS (“DOWNLOADS”)	431
WEB BROWSER INTERFACE	441
SYSLOG SERVER FUNCTIONS	446
CHAPTER 11: EMBEDDED GATEKEEPER (FOR MVP-210G/410G/810G)	449
.....	
INTRODUCTION TO EMBEDDED GATEKEEPER	450
GETTING STARTED WITH THE GATEKEEPER-EQUIPPED MULTIVOIP	451
EMBEDDED GATEKEEPER SYSTEM EXAMPLE	454
GATEKEEPER BASICS.....	481
<i>Introduction</i>	481
<i>Mandatory Gatekeeper Functions</i>	481
Address Translation.....	481
Admission Control.....	481
Bandwidth Control	481
Zone Management.....	482
<i>Optional Gatekeeper Functions</i>	482
Call Control Signaling.....	482
Call Authorization	482
Bandwidth Management.....	482
Call Management	483
FEATURES.....	483
THE GATEKEEPER PROTOCOLS	484
MULTIVOIP GATEKEEPER SOFTWARE SCREENS.....	487
GK DEFINED SERVICE TYPES.....	516
<i>Example of a Gatekeeper Service</i>	516
<i>Built-in Gatekeeper-Defined Services</i>	517
Service Types: Zone Prefixes (1 and 2).....	517
Service Types: Forward.....	519
GATEKEEPER LOG DATA DATA FILES	520
GATEKEEPER SOFTWARE USER LICENSE AGREEMENT	521
CHAPTER 12 WARRANTY, SERVICE, AND TECH SUPPORT	523
LIMITED WARRANTY.....	524
REPAIR PROCEDURES FOR U.S. AND CANADIAN CUSTOMERS	524
TECHNICAL SUPPORT	526
<i>Contacting Technical Support</i>	526
CHAPTER 13: REGULATORY INFORMATION	527
<i>EMC, Safety, and R&TTE Directive Compliance</i>	528
FCC DECLARATION	528
<i>Industry Canada</i>	529

<i>FCC Part 68 Telecom</i>	529
<i>Canadian Limitations Notice</i>	530
APPENDIX A: EXPANSION CARD INSTALLATION (MVP24-48 & MVP30-60)	531
INSTALLATION.....	532
OPERATION.....	534
APPENDIX B: CABLE PINOUTS	535
APPENDIX B: CABLE PINOUTS.....	536
<i>Command Cable</i>	536
<i>Ethernet Connector</i>	536
<i>T1/E1 Connector</i>	537
<i>Voice/Fax Channel Connectors</i>	537
<i>ISDN BRI RJ-45 Pinout Information</i>	539
<i>ISDN Interfaces: "ST" and "U"</i>	540
APPENDIX C: TCP/UDP PORT ASSIGNMENTS	541
WELL KNOWN PORT NUMBERS	542
PORT NUMBER ASSIGNMENT LIST	542
APPENDIX D: INSTALLATION INSTRUCTIONS FOR MVP428 UPGRADE CARD	543
INSTALLATION INSTRUCTIONS FOR MVP428 UPGRADE CARD.....	544
APPENDIX E: CALL STATES & REASONS FOR EMBEDDED GATEKEEPERS	548
CALL STATES AND CALL REASONS	549
<i>Possible Call States of which the Embedded Gatekeeper Software can be notified</i>	549
<i>Call Reasons sent to Embedded Gatekeeper Software with respect to a Call State</i>	552
INDEX	556

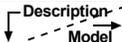
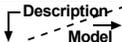
Chapter 1: Overview

About This Manual

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

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

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

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

How to Use This Manual. *In short, use the index and the examples.*

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

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

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



Figure 1-1: Variable Model/Version Icon

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

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

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

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

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

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

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

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

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

SPP (Single-Port Protocol) is a non-standard protocol developed by Multi-Tech. SPP is not compatible with the “Proprietary” protocol used in Multi-Tech’s earlier generation of voip gateways. SPP offers advantages in certain situations, especially when firewalls are used and when dynamic IP address assignment is needed. However, when SPP is used, certain features of SIP and H.323 will not be available and SPP will not inter-operate with voip systems using H.323 or SIP.

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

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

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

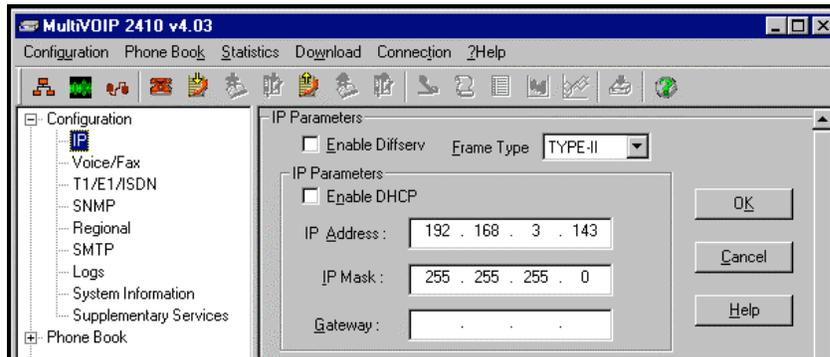
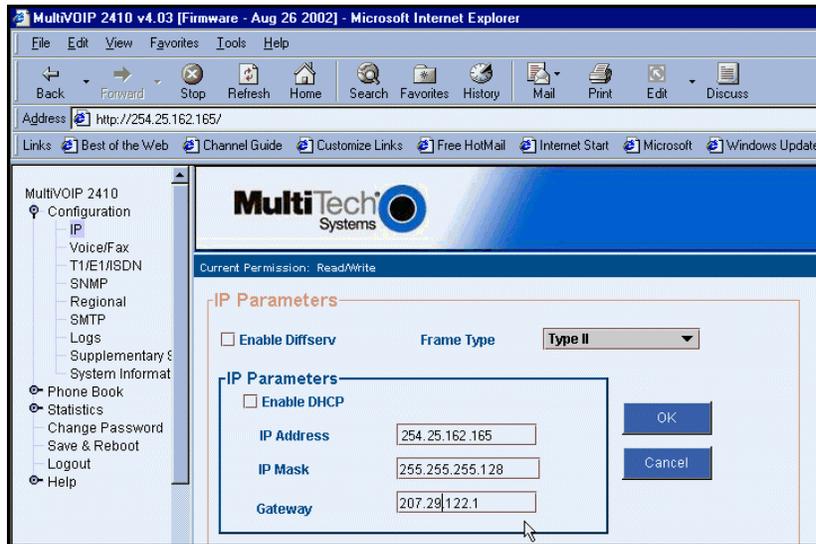
PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

Gatekeeper. T1 voip systems can have gatekeeper functionality either by adding, as an endpoint, either a Multi-Tech standalone gatekeeper (special software residing in separate hardware), or an analog gateway with embedded gatekeeper functionality (MVP210G, MVP410G, or MVP810G). Gatekeepers are optional but useful within voip systems. The gatekeeper acts as the

'clearinghouse' for all calls within its zone. MultiTech's embedded and stand-alone gatekeeper software packages both perform all of the standard gatekeepers functions (address translation, admission control, bandwidth control, and zone management) and also support many valuable optional functions (call control signaling, call authorization, bandwidth management, and call management). The stand-alone gatekeeper is, however, slightly more feature-rich than the embedded gatekeeper. For more details, see the "Embedded Gatekeeper" chapter of this manual and the manual on MultiTech's stand-alone gatekeeper.

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

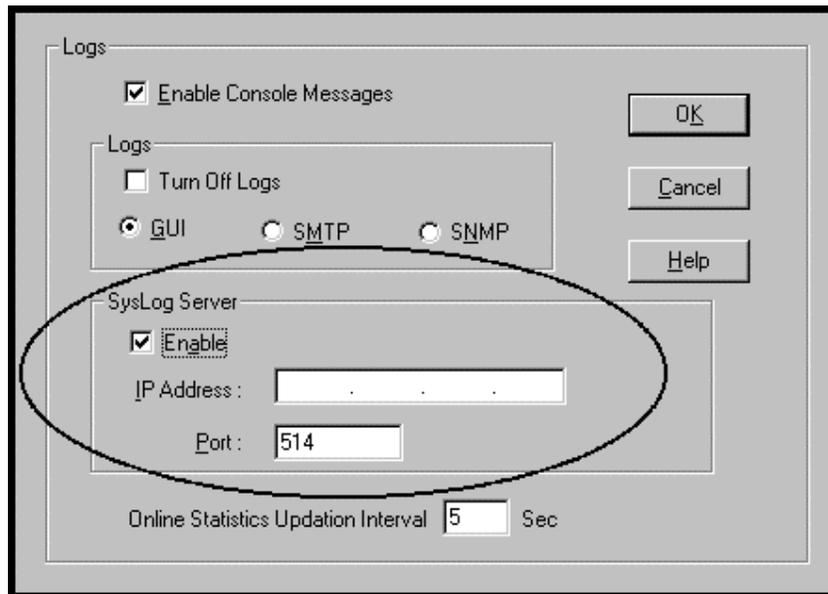
While the web GUI's appearance differs slightly, its content and organization are essentially the same as that of the Windows GUI (except for logging).



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

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

Logging of System Events. MultiTech has built SysLog Server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.



The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a “daemon”). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

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

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

T1 Front Panel LEDs

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

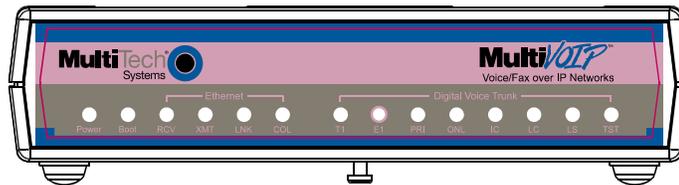


Figure 1-2. MultiVOIP MVP2400 Front Panel

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

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



Figure 1-3. MultiVOIP MVP2410x Chassis

T1 LED Descriptions

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

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

Introduction to E1 MultiVOIPs (MVP3010 & MVP30-60)

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

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

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

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

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

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

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.

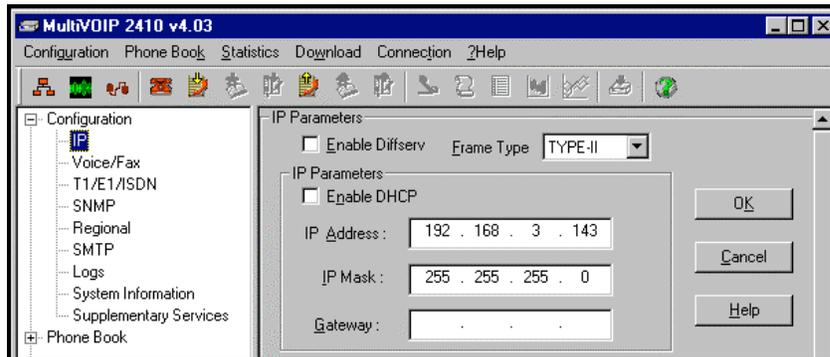
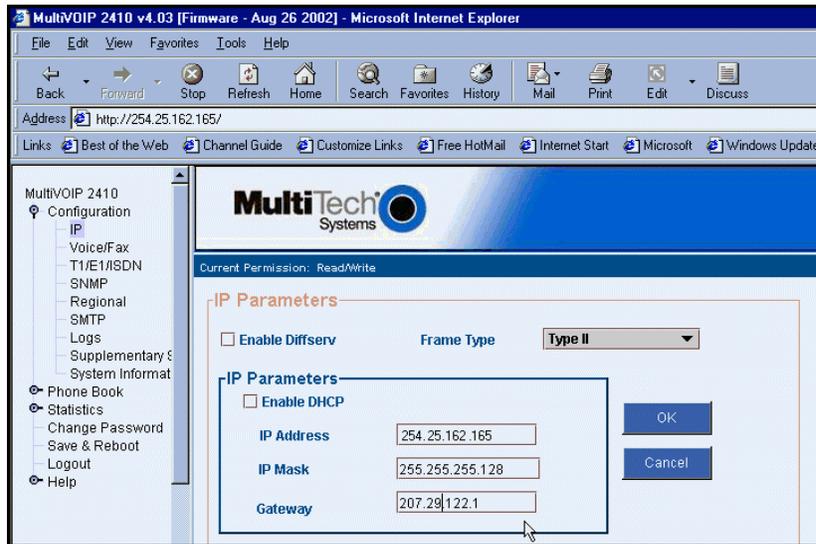
Ports. The MVP3010 also has a 10/100 Mbps Ethernet LAN interface, and a Command port for configuration. An MVP3010 upgraded with the MVP30-60 kit will have two Ethernet LAN interfaces and two Command ports.

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

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

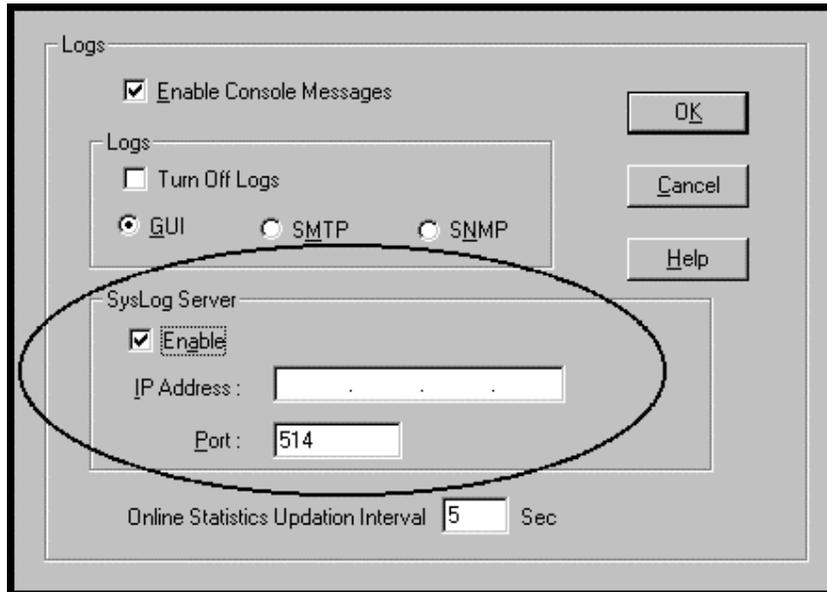
While the web GUI's appearance differs slightly, its content and organization are essentially the same as that of the Windows GUI (except for logging).



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

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

Logging of System Events. MultiTech has built SysLog Server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.



The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a “daemon”). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

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

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

E1 Front Panel LEDs

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



Figure 1-4. MultiVOIP MVP3010 Chassis

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

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

E1 LED Descriptions

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

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

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

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

Mounting. Mechanically, the MVP410 and MVP810 MultiVOIPs are designed for a one-high industry-standard EIA 19-inch rack enclosure. By contrast, MVP130 and the MVP210 are tabletop units. The product must be installed by qualified service personnel in a restricted-access area, in accordance with Articles 110-16, 10-17, and 110-18 of the National Electrical Code, ANSI/NFPA 70.

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

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

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

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

SPP (Single-Port Protocol) is a non-standard protocol developed by Multi-Tech. SPP is not compatible with the “Proprietary” protocol used in Multi-Tech’s earlier generation of voip gateways. SPP offers advantages in certain situations, especially when firewalls are used and when dynamic IP address assignment is needed. However, when SPP is used, certain features of SIP and H.323 will not be available and SPP will not inter-operate with voip systems using H.323 or SIP.

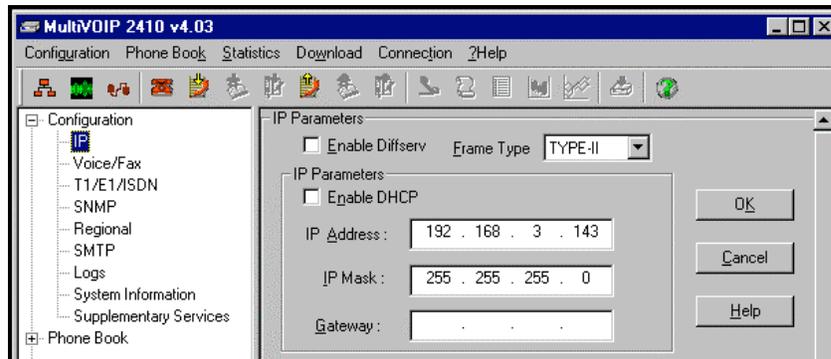
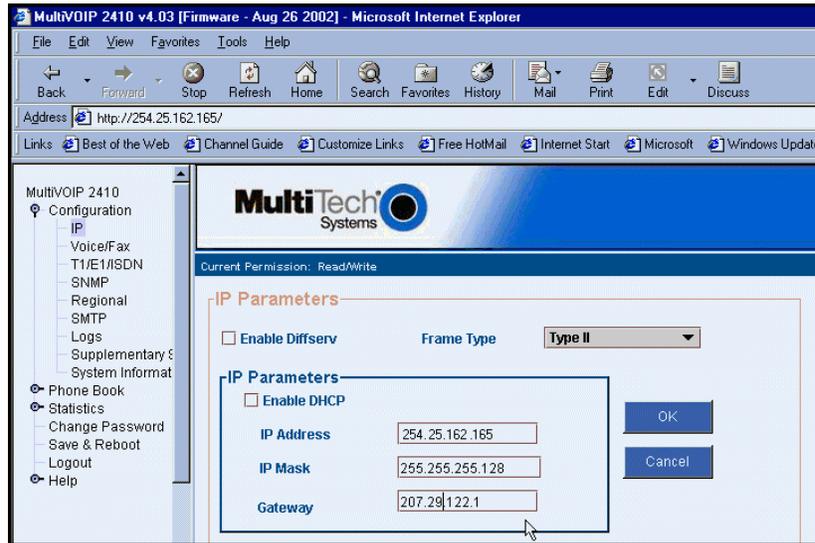
Data Compression & Quality of Service. The analog MultiVOIP unit comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

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

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

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

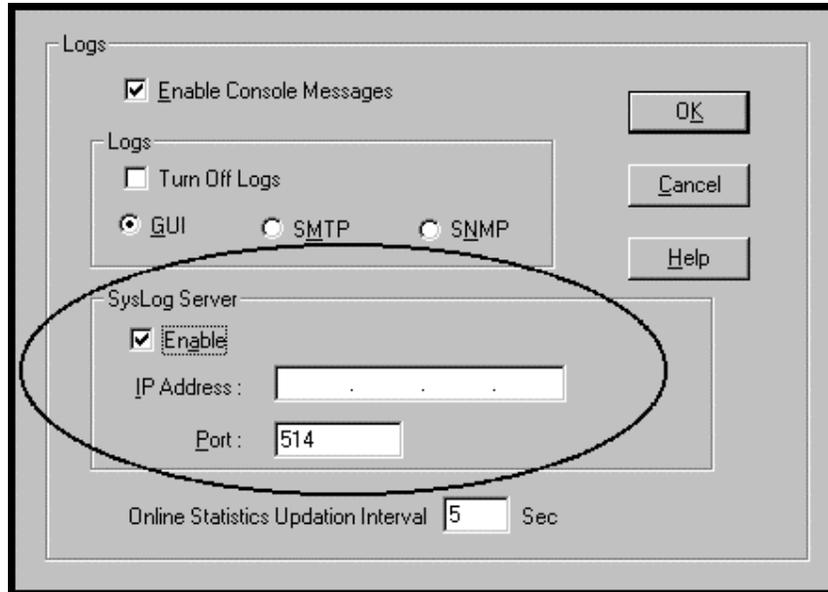
While the web GUI's appearance differs slightly, its content and organization are essentially the same as that of the Windows GUI (except for logging).



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

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

Logging of System Events. MultiTech has built SysLog Server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.



The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a "daemon"). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

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

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



Figure 1-5: MVP-410/810 Chassis



Figure 1-6: MVP-210 Chassis



Figure 1-7. MultiVOIP MVP130Chassis

Analog MultiVOIP Front Panel LEDs

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

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

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



Figure 1-8. MVP410/810 Front Panel

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



Figure 1-9. MVP210 Front Panel

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

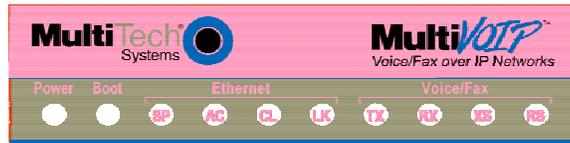


Figure 1-10. MVP130 Front Panel

Analog MultiVOIP LED Descriptions

MVP210/410/810 Front Panel LED Definitions	
LED NAME	DESCRIPTION
General Operation LEDs (one set on each MultiVOIP model)	
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on briefly while the MultiVOIP is booting. It lights whenever the MultiVOIP is booting or downloading a setup configuration data set.
Ethernet	<p>RCV. Receive. Lights (blinks) when receiving data on Ethernet port.</p> <p>XMT. Transmit. Lights (blinks) when transmitting data on Ethernet port. ..</p> <p>LNK. Link. When lit, VOIP “sees” the hub or network via the Ethernet connection. ..</p> <p>COL. Collision. Lit when data collisions occur. ..</p>
Channel-Operation LEDs (one set for each channel)	
XMT	Transmit. This indicator blinks when voice packets are being transmitted to the local area network.
RCV	Receive. This indicator blinks when voice packets are being received from the local area network.
XSG	Transmit Signal. This indicator lights when the FXS-configured channel is off-hook, the FXO-configured channel is receiving a ring from the Telco, or the M lead is active on the E&M configured channel. That is, it lights when the MultiVOIP is receiving a ring from the PBX.
RSG	Receive Signal. This indicator lights when the FXS-configured channel is ringing, the FXO-configured channel has taken the line off-hook, or the E lead is active on the E&M-configured channel.

MVP130 Front Panel LED Definitions	
LED NAME	DESCRIPTION
General Operation LEDs	
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on briefly while the MultiVOIP is booting. It lights whenever the MultiVOIP is booting or downloading a setup configuration data set.
Ethernet	<p>SP. During normal operation, the SP LED lights to indicate 100Mbps is selected.</p> <p>AC. During normal operation, the AC LED lights when transmitting or receiving. It will flash at a rate of 50ms high and 50ms low when active.</p> <p>CL. During normal operation, the CL LED lights to indicate a collision. It will flash at a rate of 50ms high and 50ms low when active.</p> <p>LK. During normal operation, the LK LED lights to indicate a good link is detected.</p>
Channel-Operation LEDs	
TX	Transmit. This indicator blinks when voice packets are being transmitted to the local area network.
RX	Receive. This indicator blinks when voice packets are being received from the local area network.
XS	Transmit Signal. This indicator lights when the FXS-configured channel is off-hook or the FXO-configured channel is receiving a ring from the Telco or PBX.
RS	Receive Signal. This indicator lights when the FXS-configured channel is ringing or the FXO-configured channel has taken the line off-hook.

Introduction to ISDN-BRI MultiVOIPs (MVP410ST & MVP810ST)

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

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

Mounting. Mechanically, the MVP410ST and MVP810ST MultiVOIPs are designed for a one-high industry-standard EIA 19-inch rack enclosure. The product must be installed by qualified service personnel in a restricted-access area, in accordance with Articles 110-16, 10-17, and 110-18 of the National Electrical Code, ANSI/NFPA 70.

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

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

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

SPP (Single-Port Protocol) is a non-standard protocol developed by Multi-Tech. SPP is not compatible with the “Proprietary” protocol used in Multi-Tech’s earlier generation of voip gateways. SPP offers advantages in certain situations, especially when firewalls are used and when dynamic IP address assignment is needed. However, when SPP is used, certain features of SIP and H.323 will not be available and SPP will not inter-operate with voip systems using H.323 or SIP.

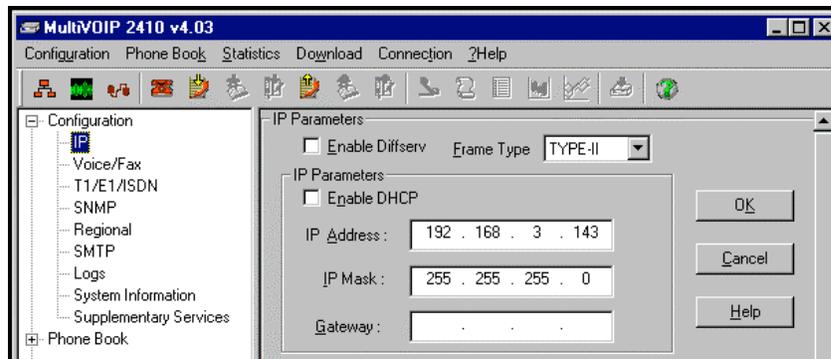
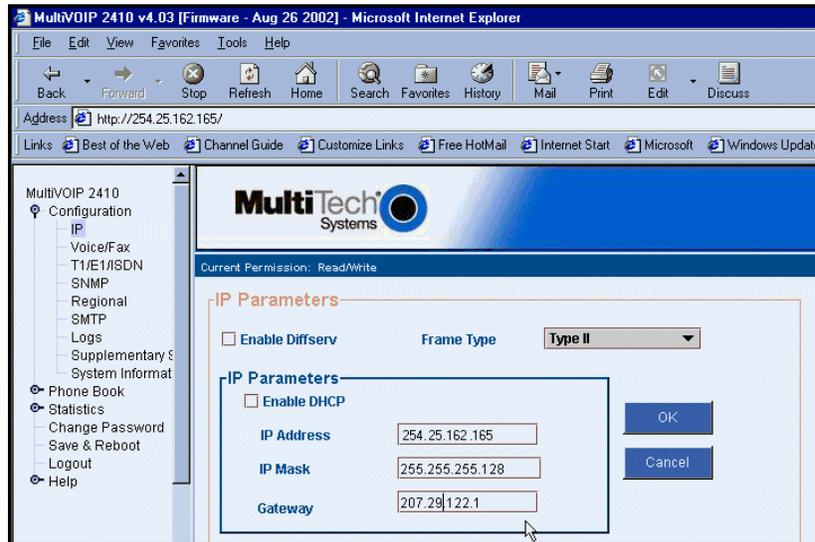
Data Compression & Quality of Service. The BRI MultiVOIP unit comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

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

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

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

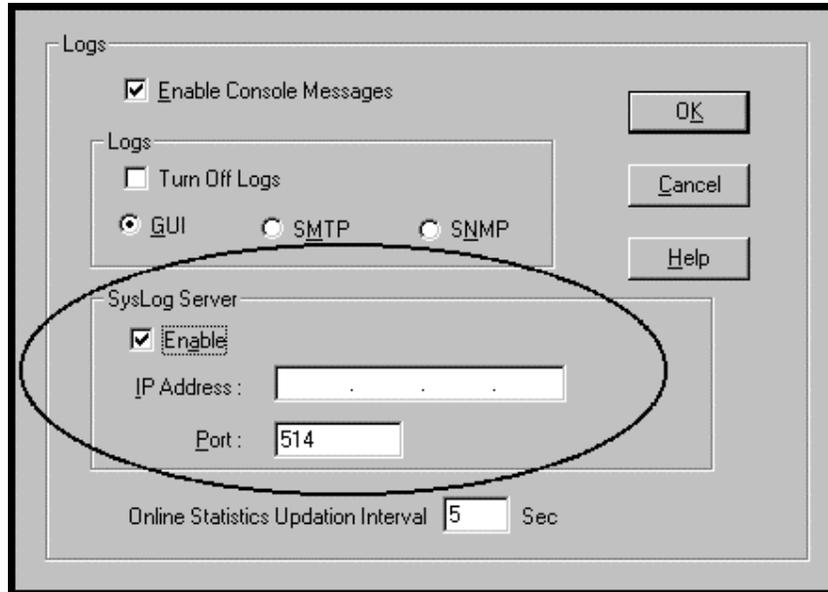
While the web GUI's appearance differs slightly, its content and organization are essentially the same as that of the Windows GUI (except for logging).



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

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

Logging of System Events. MultiTech has built SysLog Server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.



The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a "daemon"). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

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

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

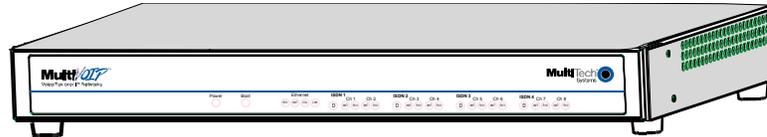


Figure 1-11: MVP-410ST/810ST Chassis

ISDN BRI MultiVOIP Front Panel LEDs

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

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

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



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

ISDN-BRI MultiVOIP LED Descriptions

MVP-410ST/810ST Front Panel LED Definitions	
LED NAME	DESCRIPTION
General Operation LEDs (one set on each MultiVOIP model)	
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on briefly while the MultiVOIP is booting. It lights whenever the MultiVOIP is booting or downloading a setup configuration data set.
Ethernet	<p>RCV. Receive. Lights (blinks) when receiving data on Ethernet port.</p> <p>XMT. Transmit. Lights (blinks) when transmitting data on Ethernet port. ..</p> <p>LNK. Link. When lit, VOIP “sees” the hub or network via the Ethernet connection. ..</p> <p>COL. Collision. Lit when data collisions occur. ..</p>
D-Channel Operation LEDs (one for each ISDN line)	
D	<p>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.</p> <p>*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.</p>
B-Channel Operation LEDs (one for each B-channel)	
XMT	Transmit. This indicator blinks when voice packets are being transmitted onto the B-channel.
RCV	Receive. This indicator blinks when voice packets are being received on the B-channel.

Computer Requirements

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

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

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

Specifications

Specs for Digital T1 MultiVOIP Units

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

Specs for Digital E1 MultiVOIP Units

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

Specs for Analog/BRI MultiVOIP Units

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

Installation at a Glance

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

Related Documentation

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

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

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

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

<http://www.multitech.com>

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

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

Chapter 2: Quick Start Instructions

Introduction

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

MultiVOIP Startup Tasks

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

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

Gather IP Information

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

Gather Telephone Information (T1)

➤	T1 Phone Parameters <i>Ask phone company or PBX maintainer.</i>	<i>Info needed to operate:</i> MVP2400 MVP2410
	T1 Telephony Parameters: Record for this VOIP Site	
	• Which frame format is used? ESF ___ or D4 ___	
	• Which CAS or PRI protocol is used? _____	
	• Clocking: Does the PBX or telco switch use internal or external clocking? _____ Note that the setting used in the voip unit will be the opposite of the setting used by the telco/PBX.	
	• Which line coding is used? AMI ___ or B8ZS ___	
	• Pulse shape level?: (most commonly 0 to 40 meters)	

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

Gather Telephone Information (E1)

➤	E1 Phone Parameters <i>Ask phone company or PBX maintainer.</i>	<i>Info needed to operate:</i> MVP3010
	 E1 Telephony Parameters: Record for this VOIP Site	
	<ul style="list-style-type: none"> Which frame format is used? Double Frame _____ MultiFrame w/ CRC4 _____ MultiFrame w/ CRC4 modified _____ 	
	<ul style="list-style-type: none"> Which CAS or PRI protocol is used? _____ 	
	<ul style="list-style-type: none"> 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. 	
	<ul style="list-style-type: none"> Which line coding is used? AMI ___ or HDB3 ___ 	
	<ul style="list-style-type: none"> Pulse shape level?: (most commonly 0 to 40 meters) 	

Gather Telephone Information (Analog)

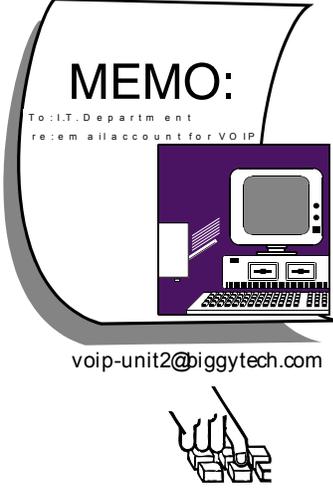
➤	Analog Phone Parameters <i>Ask phone company or telecom manager.</i>	<i>Needed for:</i> MVP810 MVP410 MVP210 MVP130
	 Analog Telephony Interface Parameters: Record for this VOIP Site	
	<ul style="list-style-type: none"> Which interface type (or “signaling”) is used? E&M _____ FXS/FXO _____ 	
	<ul style="list-style-type: none"> If FXS, determine whether the line will be used for a phone, fax, or KTS (key telephone system) 	
	<ul style="list-style-type: none"> If FXO, determine if line will be an analog PBX extension or an analog line from a telco central office 	
	<ul style="list-style-type: none"> If E&M, determine these aspects of the E&M trunk line from the PBX: <ul style="list-style-type: none"> 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 <i>Ask phone company or telecom manager.</i>	<i>Needed for:</i> MVP810ST MVP410ST
	 ISDN-BRI Telephony Interface Parameters: Record them for this VOIP Site	
	<ul style="list-style-type: none"> • In which country is this voip installed? 	
	<ul style="list-style-type: none"> • Which operator (switch type) is used? 	
	<ul style="list-style-type: none"> • What type of line coding use required, A-law or u-law? 	
	<ul style="list-style-type: none"> • Determine which BRI ports will be network side and which BRI ports will be terminal side. 	
	<ul style="list-style-type: none"> • 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)

<i>required if log reports of VOIP call traffic are to be sent by email</i>	Optional
<p>SMTP Parameters Preparation Task:</p> <p>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.</p> <p>Get the IP address of the mail server computer, as well.</p>	

Identify Remote VOIP Site to Call

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

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

Identify VOIP Protocol to be Used

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

Placement

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

The Command/Control Computer (Specs & Settings)

The computer used for command and control of the MultiVOIP

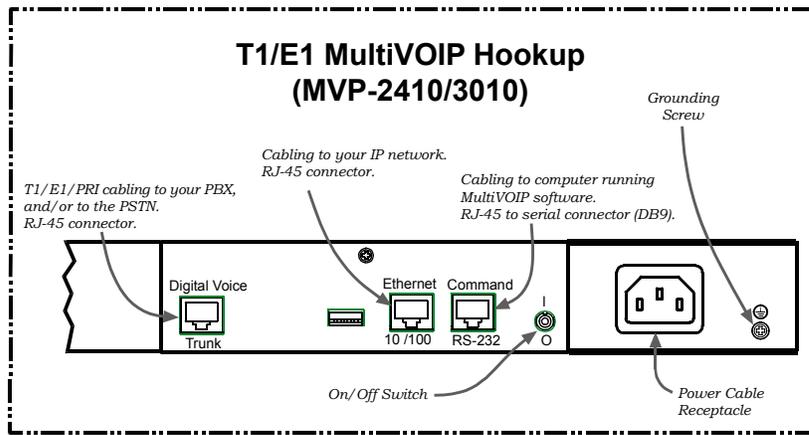
- (a) must be an IBM-compatible PC,
- (b) must use a Microsoft operating system,
- (c) must be connected to your local network (Ethernet) system, and
- (d) must have an available serial COM port.

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

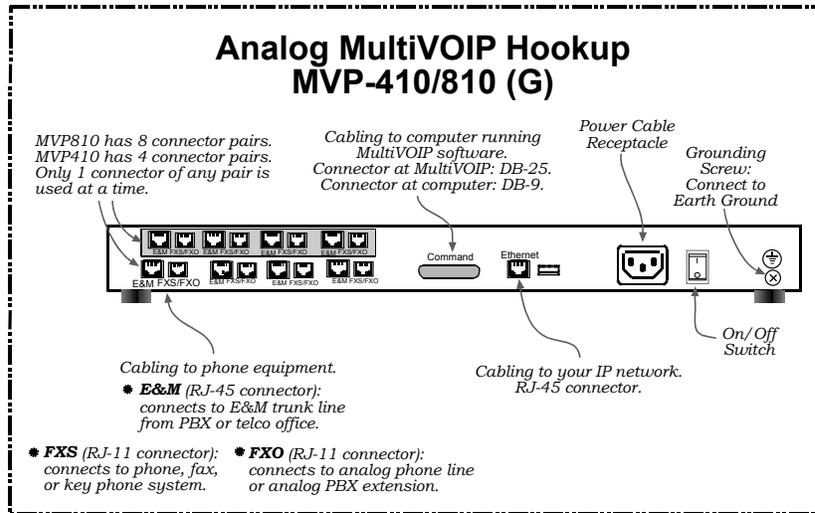
COM port on controller PC. You'll need an available COM port on the controller PC. You'll need to know which COM port is available for use with the MultiVOIP (COM1, COM2, etc.).

Quick Hookups

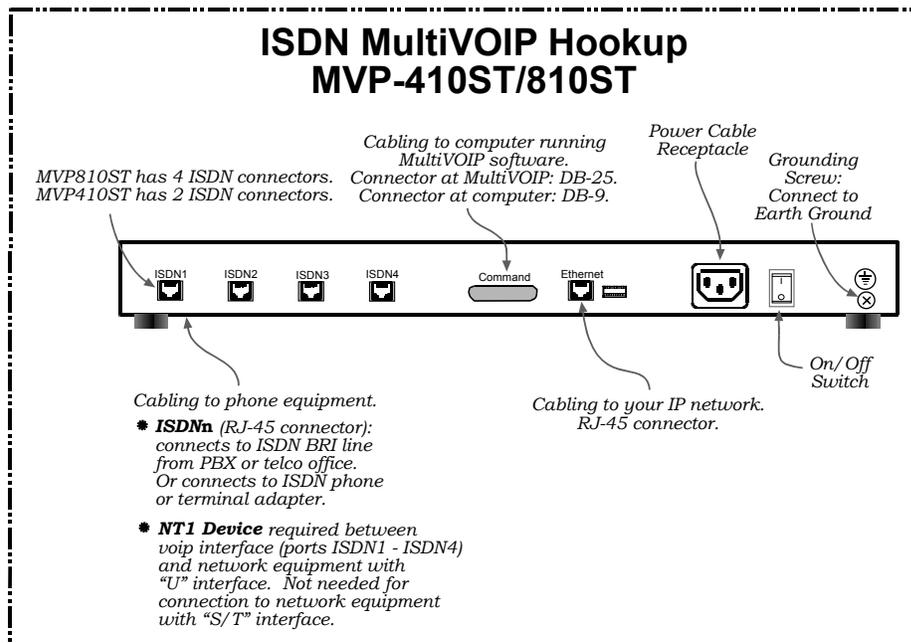
Hookup for MVP2410 & MVP3010



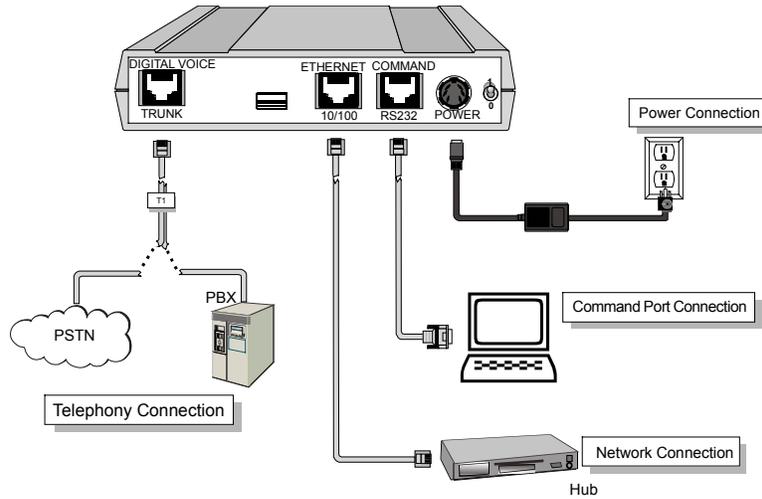
Hookup for MVP-410/410G & MVP-810/810G



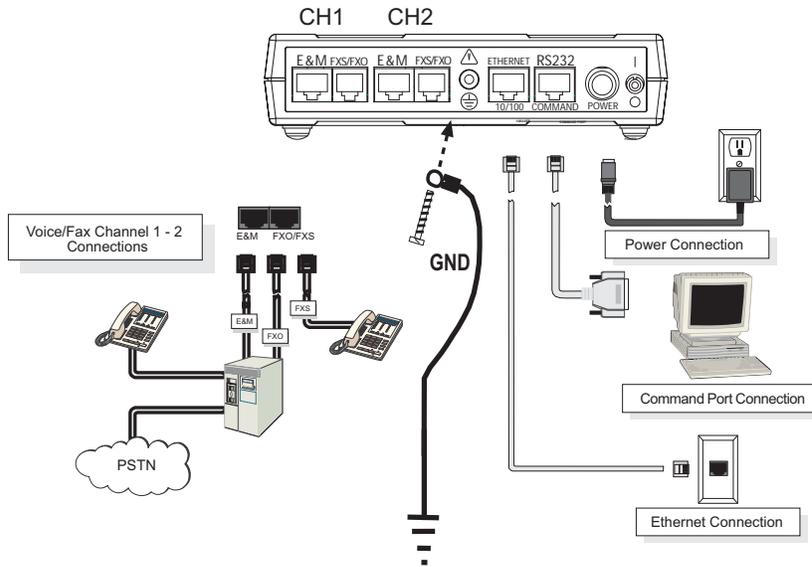
Hookup for MVP410ST & MVP810ST



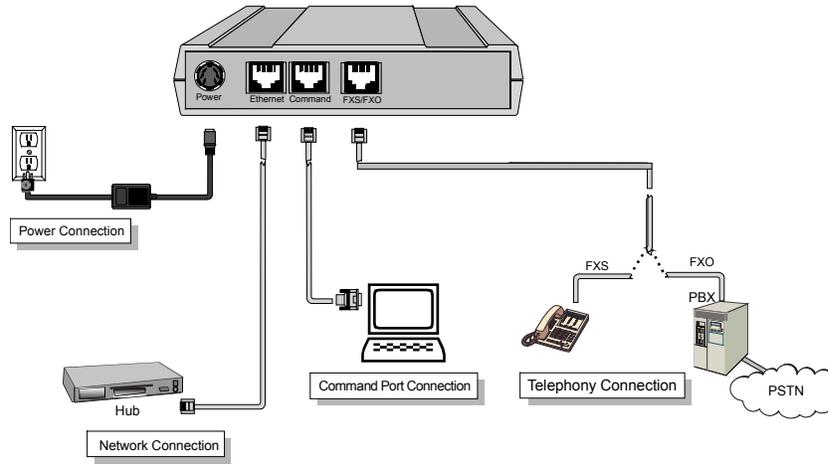
Hookup for MVP2400



Hookup for MVP210x



Hookup for MVP130



Load MultiVOIP Control Software onto PC

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

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

Phone/IP Starter Configuration

Full details here:

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

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

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

<p style="text-align: center;">Analog MultiVOIPs MVP130, MVP-210/410/810 MVP-210G/410G/810G</p> <p>Go to Configuration Interface. Enter parameters obtained from phone company or PBX administrator.</p>	<p style="text-align: center;">Digital MultiVOIPs MVP-2400/2410x/3010</p> <p>Go to Configuration T1/E1/ISDN. Enter parameters obtained from phone company or PBX administrator.</p>
<p>ISDN-BRI MultiVOIPs MVP-410ST/810ST</p> <p>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.</p>	

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

9. Go to **Configuration | SMTP**.

SMTP lets you send phone-call log records to the Voip Administrator by email. Select **Enable SMTP**.

You should have already obtained an email address for the MultiVOIP itself (this serves as the origination email account for email logs that the MultiVOIP can email out automatically).

Enter this email address in the “Login Name” field.

Type the password for this email account.

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

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

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

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

10. Go to **Configuration | Logs**.

Select “Enable Console Messages.” (*Not applicable if using Web GUI.*)

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

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

Phone/IP Starter Configuration (continued)

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

Go to **Supplementary Services**. Select any features to be used.

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

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

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

12. **(For analog gatekeeper-equipped models only. These have model numbers with a "G" suffix.**

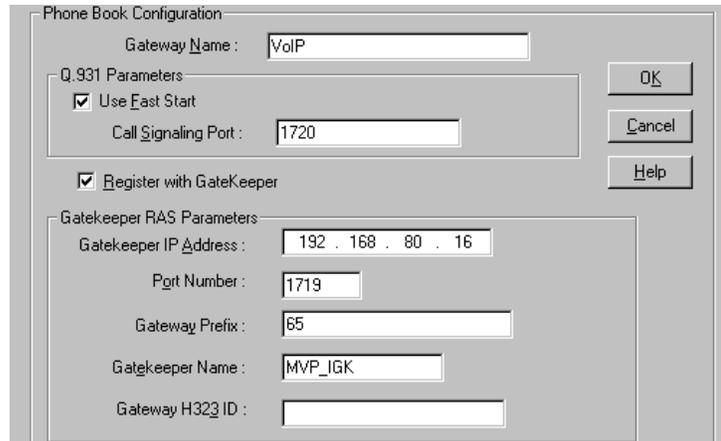
For MVP2410G, skip to step 13 and see User Guide for embedded gatekeeper info.

For units without embedded gatekeeper, skip to step 13.)

For quick-start purposes, we will arrange for the gatekeeper-equipped voip unit to register itself as a client of its own gatekeeper capability. Then we will set up a gatekeeper-controlled call from one channel to another of that self-same gatekeeper-equipped voip unit to demonstrate that the gatekeeper functionality is active. Thereafter, you can register additional voip units (and other endpoints) with the gatekeeper-equipped voip per instructions in the **User Guide**.

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

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



The image shows a "Phone Book Configuration" dialog box with the following fields and options:

- Gateway Name: VoIP
- Q.931 Parameters:
 - Use Fast Start
 - Call Signaling Port: 1720
- Register with GateKeeper
- Gatekeeper RAS Parameters:
 - Gatekeeper IP Address: 192 . 168 . 80 . 16
 - Port Number: 1719
 - Gateway Prefix: 65
 - Gatekeeper Name: MVP_IGK
 - Gateway H323 ID: (empty)

Buttons: OK, Cancel, Help

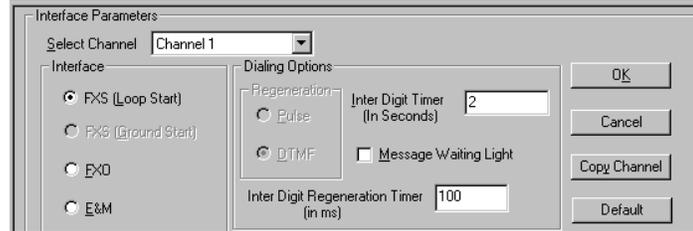
12B. In the "Destination Pattern" field of the **Add/Edit Outbound Phonebook** screen, enter 65. Click on "Use Gatekeeper." In the "Gateway Prefix" field, enter 65. Click **OK**.

The screenshot shows the "Add/Edit Outbound Phone Book" dialog box. The "Phone Number Details" section has "Destination Pattern" set to 65, "Total Digits" set to 0, and empty "Remove Prefix" and "Add Prefix" fields. The "IP Address" and "Description" fields are empty. The "Protocol Type" section has radio buttons for SIP, H.323 (selected), and SPP. The "H.323" section has "Use GateKeeper" checked, "Gateway H.323 ID" empty, "Gateway Prefix" set to 65, and "Q.931 Port Number" set to 1720. On the right side, there are buttons for OK, Cancel, Help, and Advanced.

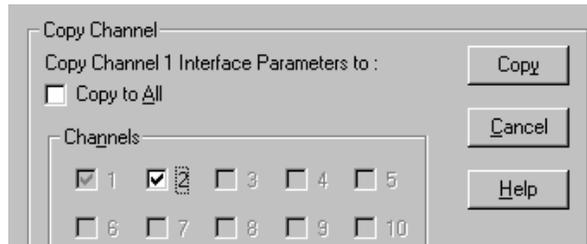
12C. In the "Remove Prefix" field of the **Add/Edit Inbound Phonebook** screen, enter 65. Click **OK**.

The screenshot shows the "Add/Edit Inbound Phone Book" dialog box. The "Remove Prefix" field contains 65, and the "Add Prefix" field is empty. The "Channel Number" dropdown menu is set to "Hunting". On the right side, there are buttons for OK, Cancel, and Help.

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



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



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

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

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

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

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

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

END OF PROCEDURE.

Phonebook Starter Configuration (*with remote voip*)

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

To do this part of the quick setup, you need to know of another voip that you can call to conduct a test. It should be at a remote location, typically somewhere outside of your building. You must know the phone number and IP address for that site. We are assuming here that the MultiVOIP will operate in conjunction with a PBX.

You must configure both the Outbound Phonebook and the Inbound Phonebook. A starter configuration only means that two voip locations will be set up to begin the system and establish voip communication.

Outbound Phonebook

1. Open the MultiVOIP program
(Start | MultiVOIP xxx | Configuration)
2. Go to **Phone Book | PhoneBook Modify | Outbound Phonebook | Add Entry**.
3. On a sheet of paper, write down the calling code of the remote voip (area code, country code, city code, etc.) that you'll be calling.

Follow the example that best fits your situation.

North America, Long-Distance Example

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

Answer: Write down **312**.

Euro, National Call Example

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

Answer: write down **0121**.

Euro, International Call Example

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

Answer: write down **3305**.

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

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

**North America,
Long-Distance Example**

Seattle-Chicago system.

Seattle voip works with PBX that uses “8” for all voip calls. “1” must immediately precede area code of dialed number.

Answer: write down **81**.

**Euro, National Call
Example**

London/Birming. system.

London voip works with PBX that uses “9” for all out-of-building calls whether by voip or by PSTN. “0” must immediately precede area code of dialed number.

Answer: write down **90**.

Euro, International Call Example

Rotterdam/Bordeaux system.

Rotterdam voip works with PBX where “9” is used for all out-of-building calls. “0” must precede all international calls.

Answer: write down **90**.

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

**North America,
Long-Distance Example**

Seattle-Chicago system.

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

**Euro, National Call
Example**

London/Birming. system.

Leading zero of Birmingham
area code is dropped when
combined with national-
dialing access code. (Such
practices vary by country.)

Answer: enter **90121** as
Destination Pat-tern
in Outbound
Phonebook of
London voip.
Not 900121.

Euro, International Call Example

Rotterdam/Bordeaux system.

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

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

<p>North America, Long-Distance Example Seattle-Chicago system.</p> <p>To complete Seattle-to-Chicago call, 81312 must be followed by the 7-digit local phone number in Chicago.</p> <p>Answer: enter 12 as number of Total Digits in Outbound Phone book of Seattle voip.</p>	<p>Euro, National Call Example London/Birming. system.</p> <p>To complete London-to-Birmingham call, 90121 must be followed by the 7-digit local phone number in Birmingham.</p> <p>Answer: enter 12 as number of Total Digits in Outbound Phone book of London voip.</p>
<p>Euro, International Call Example Rotterdam/Bordeaux system.</p> <p>To complete Rotterdam-to-Bordeaux call, 903305 must be followed by 8-digit local phone number in Bordeaux.</p> <p>Answer: enter 14 as number of Total Digits in Outbound Phonebook of Rotterdam voip.</p>	

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

<p>North America, Long-Distance Example Seattle-Chicago system.</p> <p>Answer: enter 8 in “Remove Prefix” field of Seattle Outbound Phonebook.</p>	<p>Euro, National Call Example London/Birming. system.</p> <p>Answer: enter 9 in “Remove Prefix” field of London Outbound Phonebook.</p>
<p>Euro, International Call Example Rotterdam/Bordeaux system.</p> <p>Answer: enter 9 in “Remove Prefix” field of Outbound Phonebook for Rotterdam voip.</p>	

Some PBXs will not ‘hand off’ the “8” or “9” to the voip. But for those PBX units that do, it’s important to enter the “8” or “9” in the “Remove Prefix” field in the Outbound Phonebook. This precludes the problem of having to make two inbound

phonebook entries at remote voips, one to account for situations where “8” is used as the PBX access digit, and another for when “9” is used.

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

Inbound Phonebook

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

North America, Long-Distance Example

Seattle-Chicago system.

Seattle is area 206. Chicago employees must dial 81 before dialing any Seattle number on the voip system.

Answer: **1206** is prefix to be removed by local (Seattle) voip.

Euro, National Call Example

London/Birming. system.

Inner London is 0207 area. Birmingham employees must dial 9 before dialing any London number on the voip system.

Answer: **0207** is prefix to be removed by local (London) voip.

Euro, International Call Example

Rotterdam/Bordeaux system.

Rotterdam is country code 31, city code 010. Bordeaux employees must dial 903110 before dialing any Rotterdam number on the voip system.

Answer: **03110** is prefix to be removed by local (Rotterdam) voip.

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

<p>North America, Long-Distance Example</p> <p>Seattle-Chicago system.</p> <p>On Seattle PBX, “8” is used to get an outside line.</p> <p>Answer: 8 is the prefix to be added by local (Seattle) voip.</p>	<p>Euro, National Call Example</p> <p>London/Birming. system.</p> <p>On London PBX, “9” is used to get an outside line.</p> <p>Answer: 9 is the prefix to be added by local (London) voip.</p>
<p>Euro, International Call Example</p> <p>Rotterdam/Bordeaux system.</p> <p>On Rotterdam PBX, “9” is used to get an outside line.</p> <p>Answer: 9 is prefix to be added by local (Rotterdam) voip.</p>	

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.)

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

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.

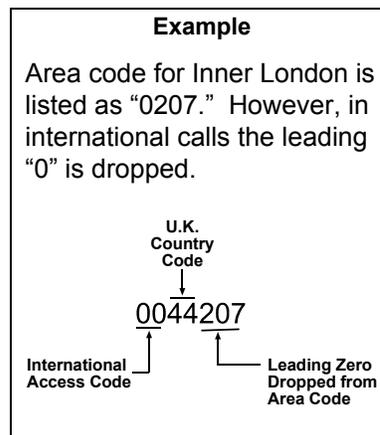
(c) **local exchange numbers.** Within any calling area there will be many local exchange numbers. A single exchange may be used for an entire small town. In cities, an exchange may be used for a particular neighborhood (although exchanges in cities do not always cover easily discernible areas). Organizations 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.

Phonebook Entry ➡



4. Using a Comma.

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

Detail

; = 1-second pause

In many PBX systems
(not needed in all)

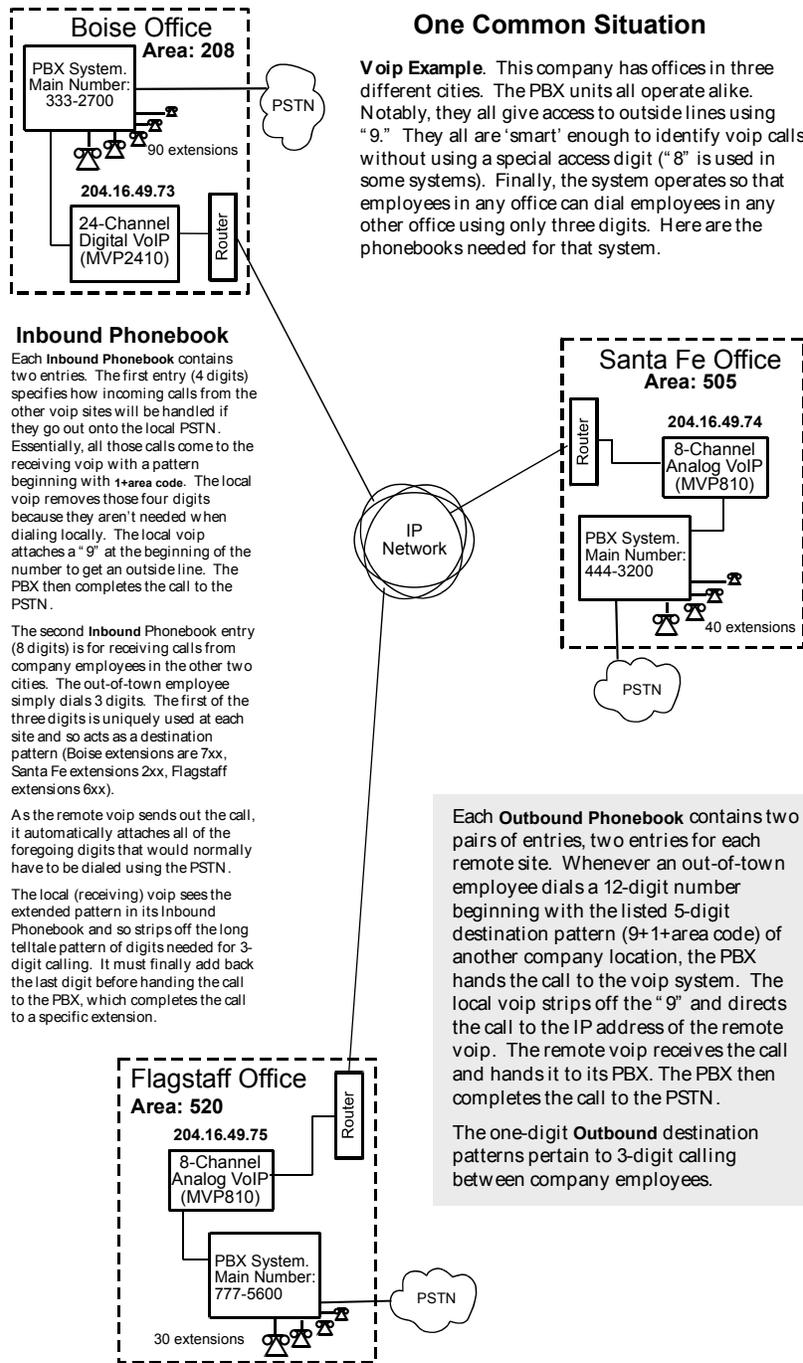
5. **Ease of Use.** The phonebook setup determines how easy the voip system is to use. Generally, you’ll want to make it so dialing a voip call is very similar to dialing any other number (on the PSTN or through the PBX).

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

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

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

Phonebook Example



One Common Situation

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

Inbound Phonebook

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

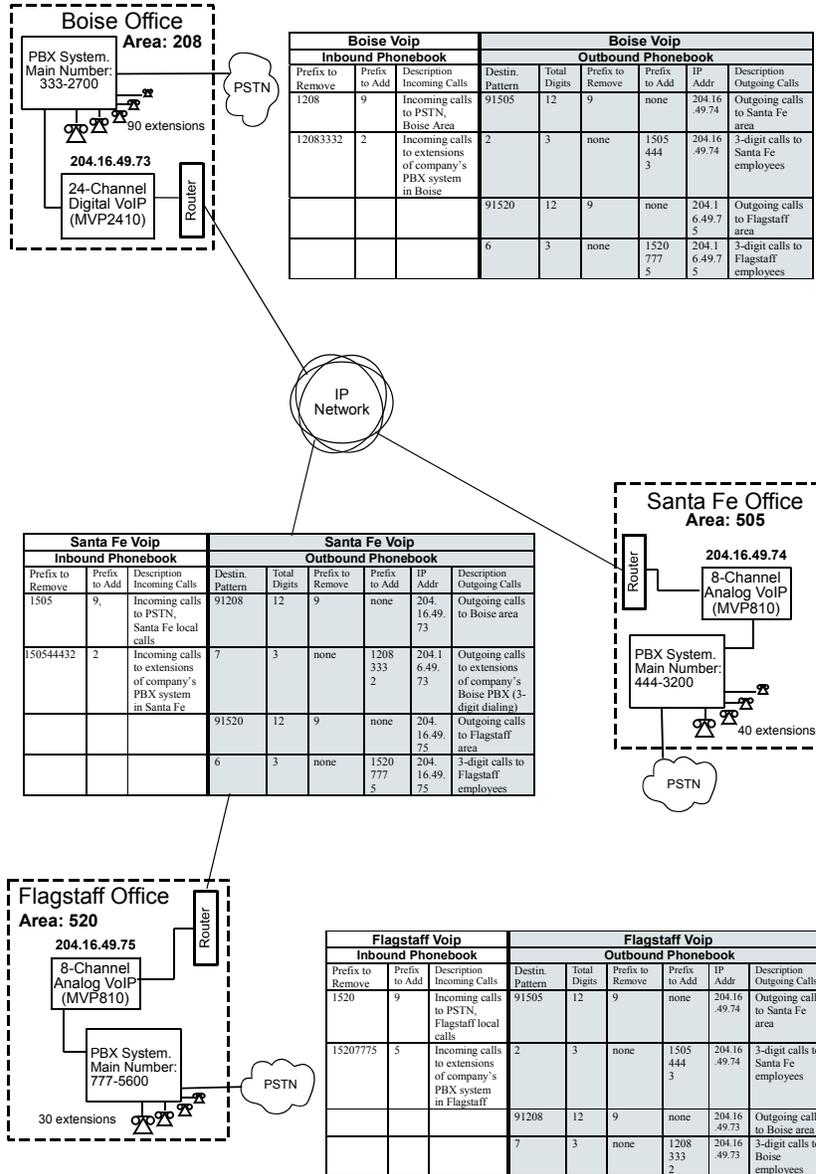
The second **Inbound Phonebook** entry (8 digits) is for receiving calls from company employees in the other two cities. The out-of-town employee simply dials 3 digits. The first of the three digits is uniquely used at each site and so acts as a destination pattern (Boise extensions are 7xx, Santa Fe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

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

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

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

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



Sample Phonebooks Enlarged

Boise Voip Inbound Phonebook			Boise Voip Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
1208	9,	Incoming calls to PSTN, Boise Area	91505	12	9	none	204.16.49.74	Outgoing calls to Santa Fe area
120833327	7	Incoming calls to extensions of company's PBX system in Boise	2	3	none	15054443	204.16.49.74	3-digit calls to Santa Fe employees (extensions 200 to 240)
			91520	12	9	none	204.16.49.75	Outgoing calls to Flagstaff area
			6	3	none	15207775	204.16.49.75	3-digit calls to Flagstaff employees (extensions 600-630)

Santa Fe Voip Inbound Phonebook			Santa Fe Voip Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
1505	9,	Incoming calls to PSTN, Santa Fe local calls	91208	12	9	none	204.16.49.73	Outgoing calls to Boise area
150544432	2	Incoming calls to extensions of company's PBX system in Santa Fe	7	3	none	12083332	204.16.49.73	3-digit calls to Boise employees (extensions 700-790)
			91520	12	9	none	204.16.49.75	Outgoing calls to Flagstaff area
			6	3	none	15207775	204.16.49.75	3-digit calls to Flagstaff employees (extensions 600-630)

Flagstaff Voip Inbound Phonebook			Flagstaff Voip Outbound Phonebook					
Prefix to Remove	Prefix to Add	Description Incoming Calls	Destin. Pattern	Total Digits	Prefix to Remove	Prefix to Add	IP Addr	Description Outgoing Calls
1520	9,	Incoming calls to PSTN, Flagstaff local calls	91505	12	9	none	204.16.49.74	Outgoing calls to Santa Fe area
152077756	6	Incoming calls to extensions of company's PBX system in Flagstaff	2	3	none	15054443	204.16.49.74	3-digit calls to Santa Fe employees (extensions 200-240)
			91208	12	9	none	204.16.49.73	Outgoing calls to Boise area
			7	3	none	12083332	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:

Connectivity Test

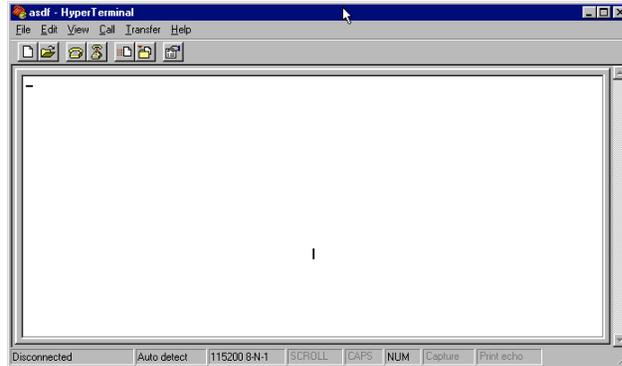
The procedures “Phone/IP Starter Configuration” and “Phonebook Starter Configuration” must be completed before you can do this procedure.

1. These connections must be made:

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

2. Inbound Phonebook and Outbound Phonebook must both be set up with at least one entry in each. These entries must allow for connection between two voip units.
3. Console messages must be enabled. (If this has not been done already, go, in the MultiVOIP GUI, to Configuration | Logs and select the “Console Messages” checkbox.
4. You now need to free up the COM port connection (currently being used by the MultiVOIP program) so that the HyperTerminal program can use it. To do this, you can either (a) click on **Connection** in the sidebar and select “Disconnect” from the drop-down box, or (b) close down the MultiVOIP program altogether.

5. Open the **HyperTerminal** program.



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

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

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

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

On the “Connection” tab, set connection preferences to:

Data bits: 8

Parity: none

Stop bits: 1

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

7. Make VOIP call.

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

8. Read console messages recorded on HyperTerminal.

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

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

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

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

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

“PSTN: pstn call connected on X”

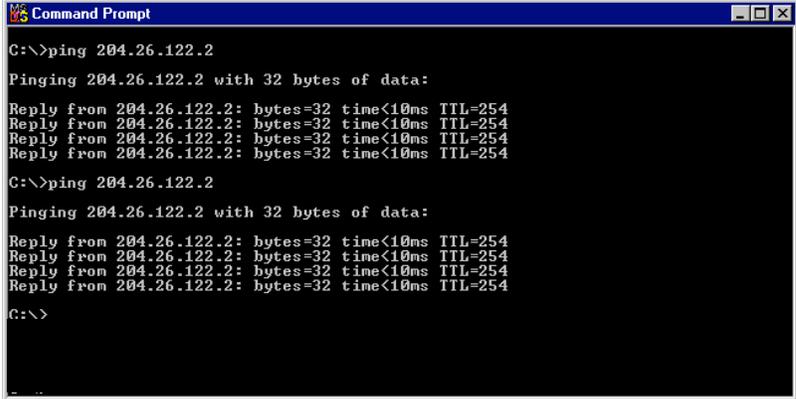
where x is the number of the voip channel carrying the call

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

Troubleshooting

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

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



```
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 MVP2400, MVP2410, or MVP3010.

Check cabling. Are connections well seated? To correct receptacle?

Is the **ONL** LED on?

(If on, ONL indicates that the MultiVOIP is online on the network.)

Are T1/E1/PRI Parameter settings correct?

B. For MVP130, MVP210, MVP410, or MVP810.

Check cabling. Are connections well seated? To correct receptacle?

Are telephone Interface Parameter settings correct?

C. For MVP410ST or MVP810ST.

Check cabling. Are connections well seated? To correct receptacle?

If terminal equipment is connected to the voip, then "Network" should be selected for that BRI interface in the **ISDN BRI Parameters** screen.

Note: Each BRI interface is separately configurable.

If network equipment such as an ISDN BRI PBX or an ISDN BRI line from a phone company is connected to the voip, then "Terminal" should be selected for that BRI interface in the **ISDN BRI Parameters** screen.

Was the proper country and operator chosen?

Was the proper type of line coding (A-law or u-law) chosen?

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

Chapter 3: Mechanical Installation and Cabling

Introduction

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

Please read the safety notices before beginning installation.

Safety Warnings

Lithium Battery Caution

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

When the battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for battery replacement.

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

Safety Warnings Telecom

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

Unpacking Your MultiVOIP

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

Unpacking the MVP2410/3010

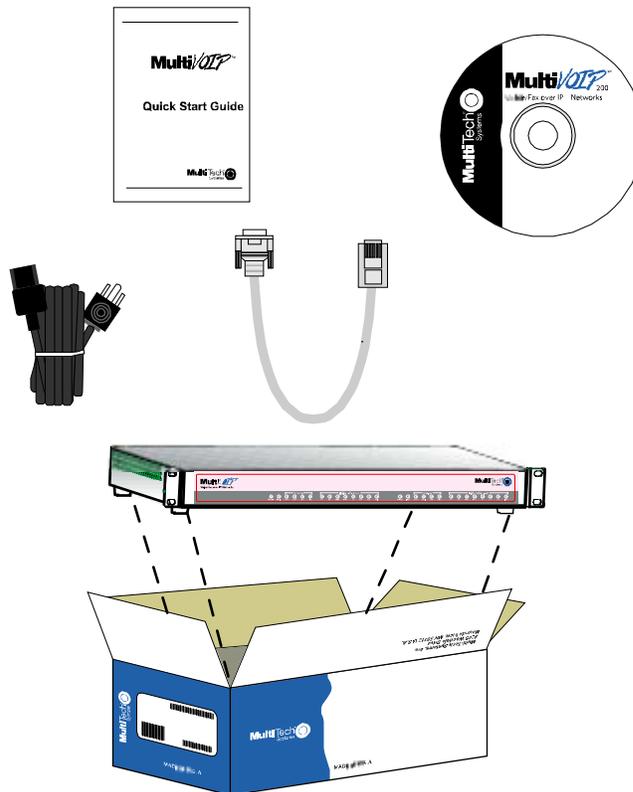


Figure 3-1: Unpacking the MVP2410/3010

Unpacking the MVP2400



Figure 3-2: Unpacking the MVP2400

Unpacking the MVP-410x/810x

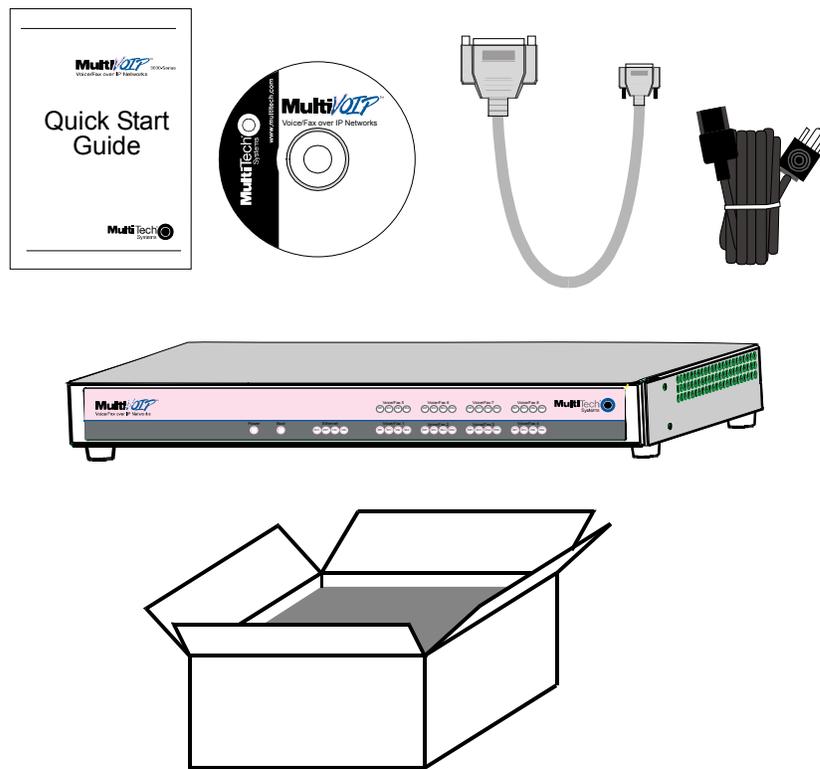


Figure 3-3: Unpacking the MVP-410x/810x

Unpacking the MVP210x



Figure 3-4: Unpacking the MVP210x

Unpacking the MVP130



Figure 3-5: Unpacking the MVP130

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

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

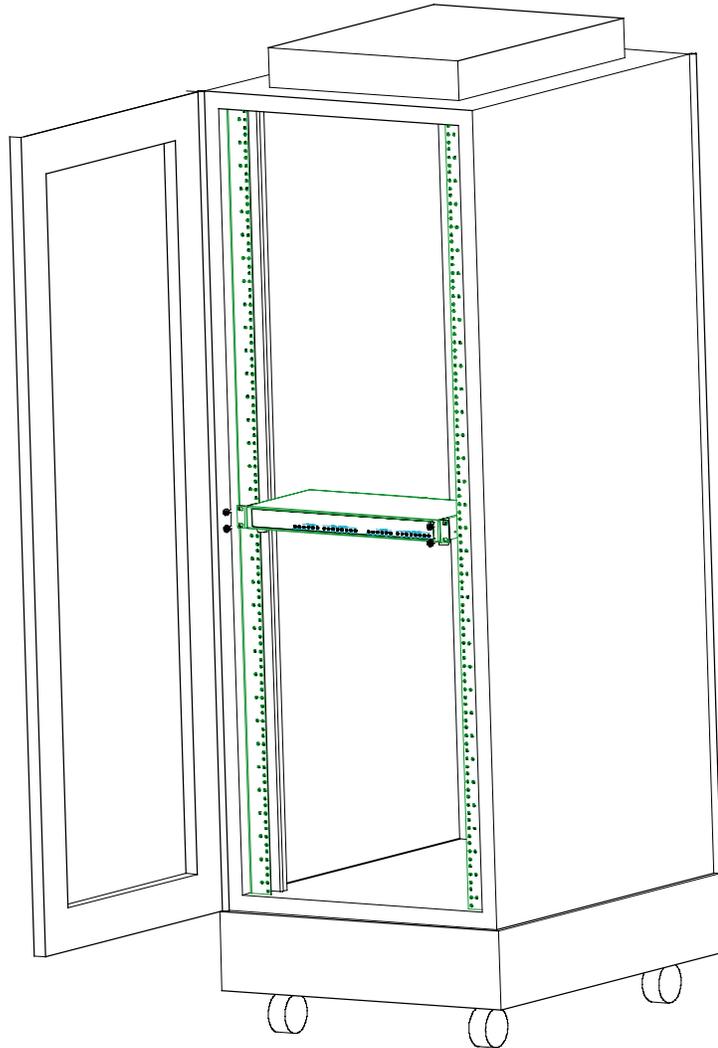


Figure 3-6: Rack-Mounting (MVP2410/3010 or MVP410x/810x)

Safety Recommendations for Rack Installations

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

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

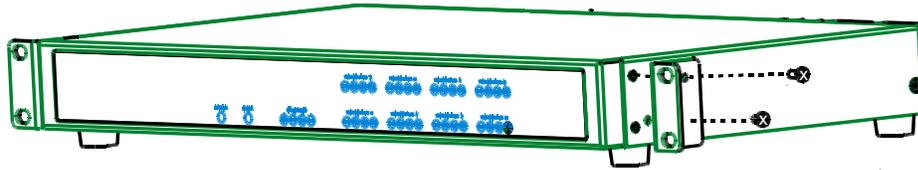
Ensure that the mains supply circuit is capable of handling the load of the equipment. See the power label on the equipment for load requirements (full specifications for MultiVOIP models are presented in chapter 1 of this manual).

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

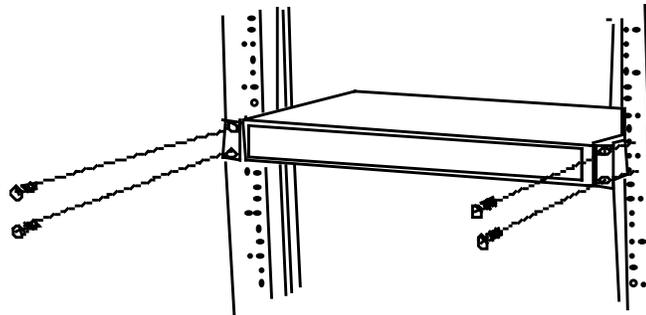
19-Inch Rack Enclosure Mounting Procedure

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

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



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



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

Cabling

Cabling Procedure for MVP2410/3010

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

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

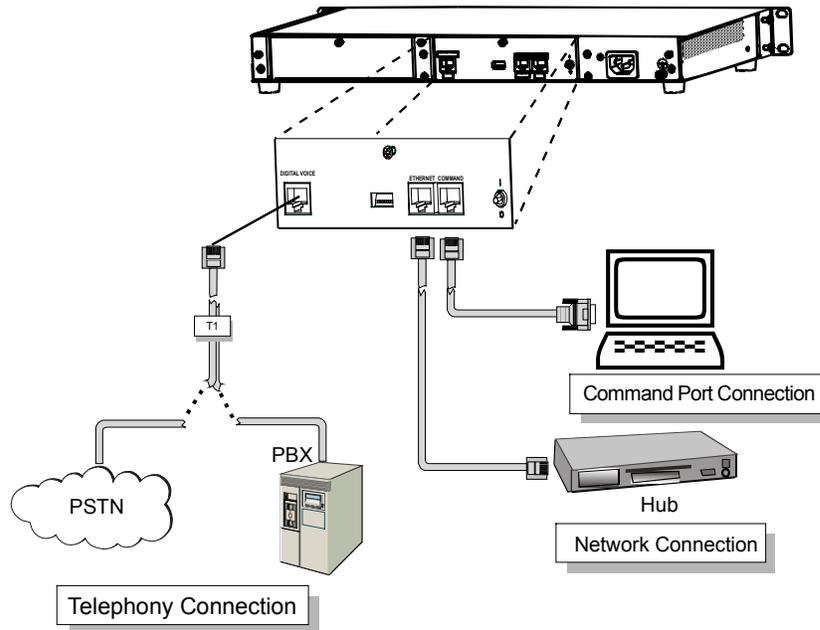


Figure 3-9. Cabling for MVP2410/3010

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

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

Proceed to Chapter 4 “Software Installation.”

Cabling Procedure for MVP2400

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

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

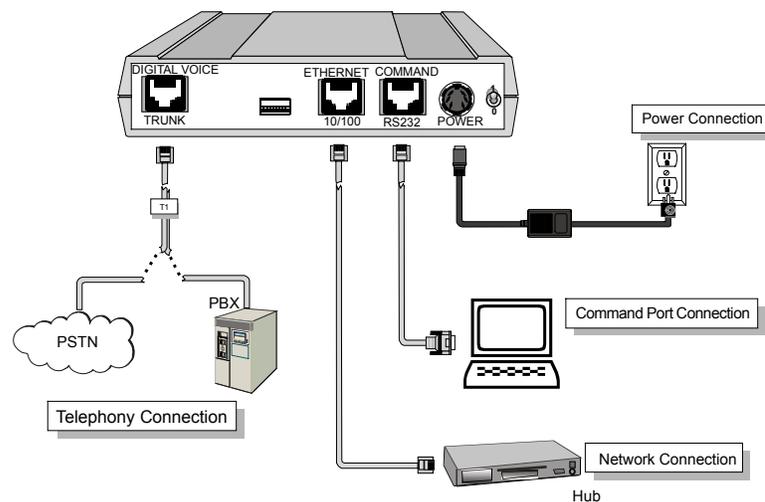


Figure 3-10: Cabling for MVP2400

- Connect the MultiVOIP to the PC (the computer that will hold the MultiVOIP software) using the RJ-45 to DB9 (female) cable provided with your unit. Plug the RJ-45 end of the cable into the **Command** port of the MultiVOIP and connect the other end (the DB9 connector) to the PC serial port you are using (typically COM1 or COM2). See Figure 3-10.
- Connect a network cable to the **Ethernet** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.
- Turn on power to the MultiVOIP by setting the power switch on the right side panel to the **ON** position. Wait for the **Boot** LED on the MultiVOIP to go off before proceeding. This may take a couple of minutes.

Proceed to Chapter 4 “Software Installation.”

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

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

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

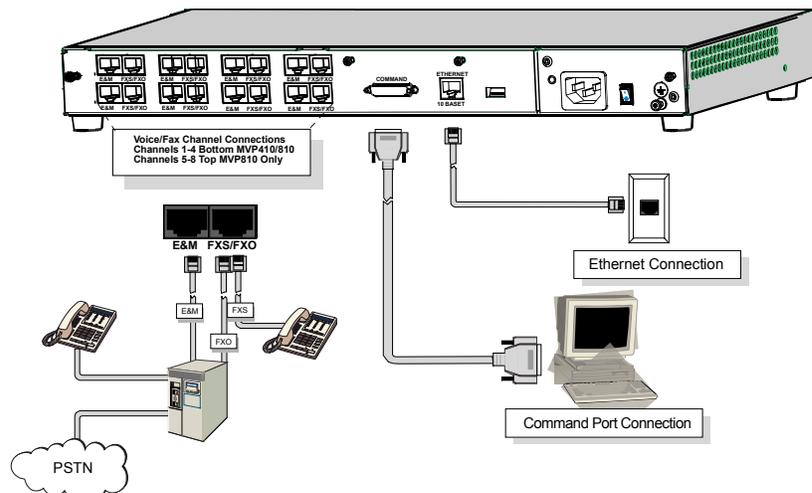


Figure 3-11: Cabling for MVP-410/410G/810/810G

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

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

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

This can be accomplished by connecting a grounding wire between the chassis and a metallic object that will provide an electrical ground.

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

Proceed to Chapter 4 to load the MultiVOIP software.

Cabling Procedure for MVP-410ST/810ST

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

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

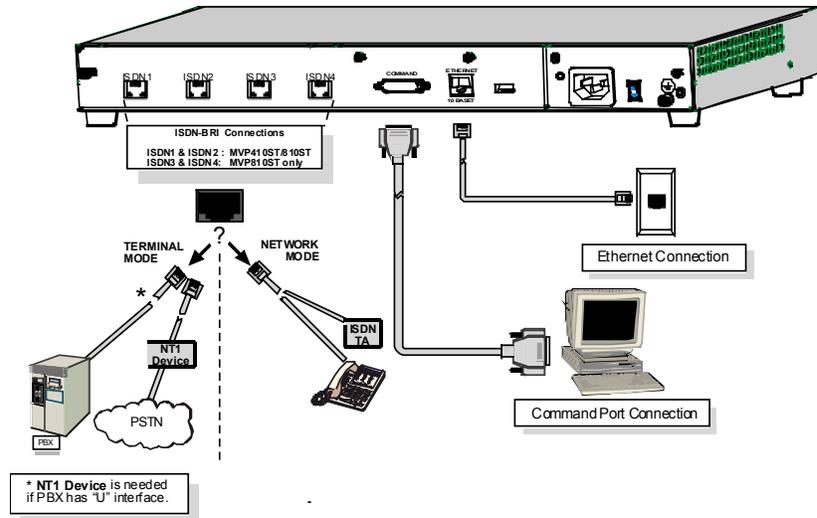
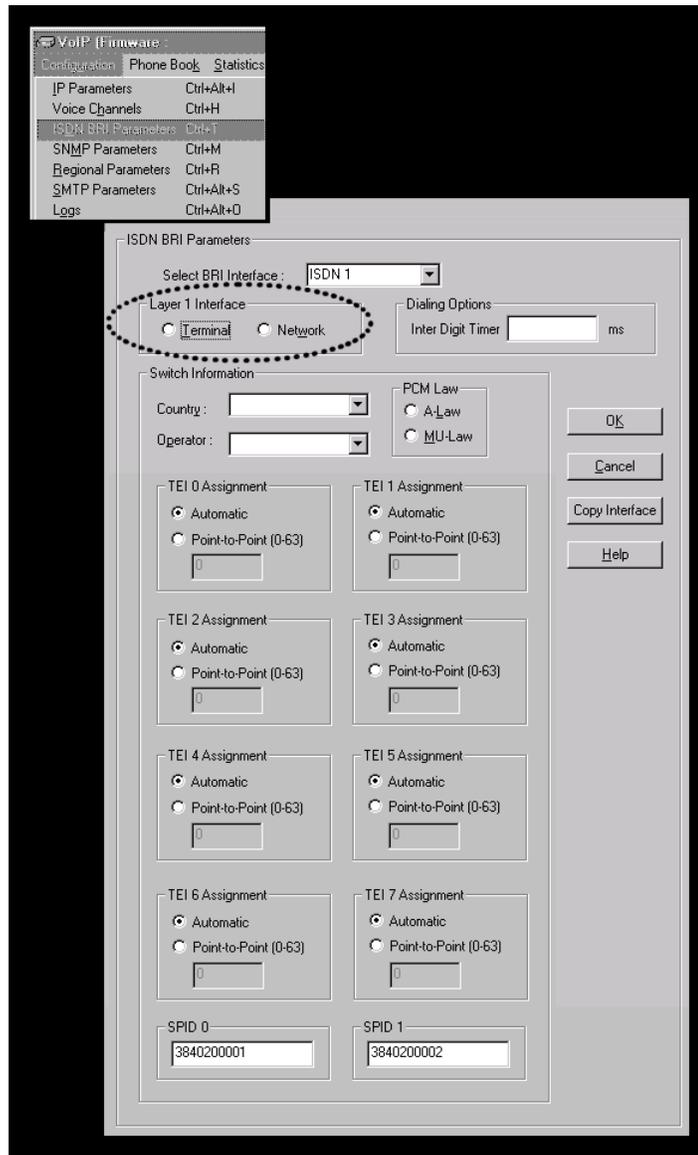


Figure 3-12: Cabling for MVP-410ST/810ST

2. Connect the MultiVOIP to a PC by using a DB-25 (male) to DB-9 (female) cable. Plug the DB-25 end of the cable into the Command port of the MultiVOIP and the other end into the PC serial port. See Figure 3-12.
3. Connect a network cable to the **ETHERNET 10BASE-T** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.

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

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



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

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

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

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

Proceed to Chapter 4 to load the MultiVOIP software.

Cabling Procedure for MVP210x

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

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

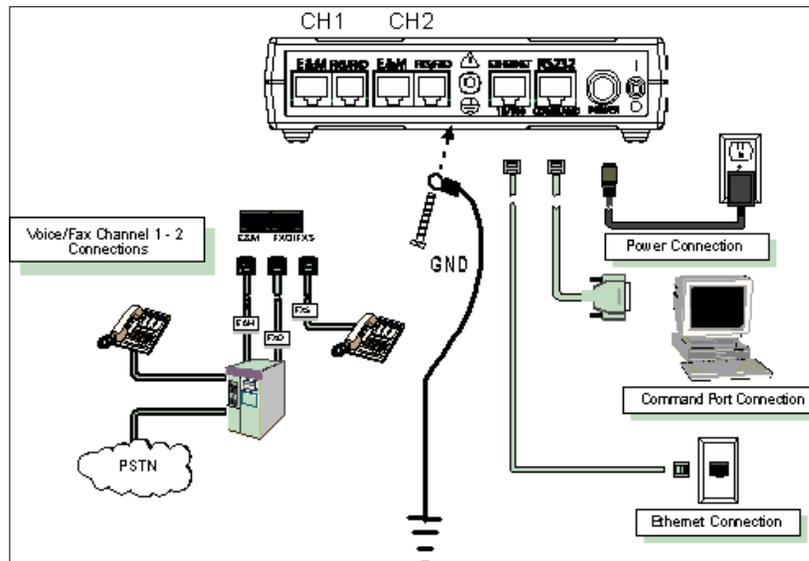


Figure 3-13: Cabling for MVP210x

2. Connect the MultiVOIP to a PC by using a RJ-45 (male) to DB-9 (female) cable. Plug the RJ-45 end of the cable into the Command port of the MultiVOIP and the other end into the PC serial port. See Figure 3-13.
3. Connect a network cable to the **ETHERNET 10/100** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.
4. If you are connecting a station device such as an analog telephone, a fax machine, or a Key Telephone System (KTS) (FXS interface), or a PBX extension (FXO interface) to your MultiVOIP, connect one end of an RJ-11 phone cord to the Channel 1 **FXS/FXO** connector on the back MultiVOIP and the other end to the device or phone jack. You will define the interface in the Interface dialog box in the software when you configure the unit.

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

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

This can be accomplished by connecting a grounding wire between the chassis and a metallic object that will provide an electrical ground.

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

Proceed to Chapter 4 to load the MultiVOIP software.

Cabling Procedure for MVP130

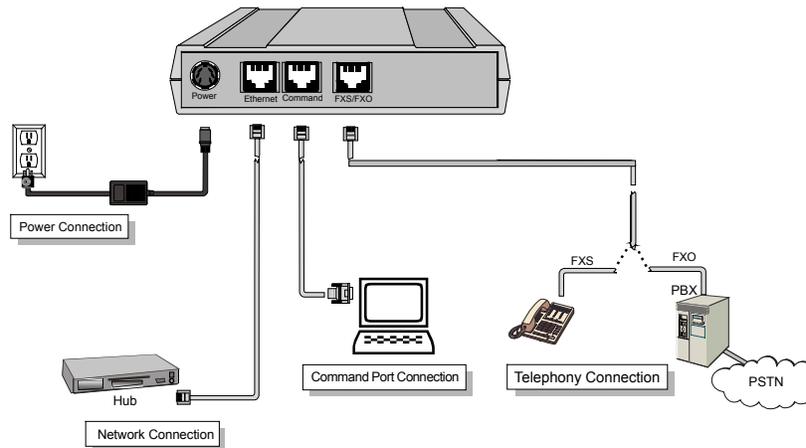


Figure 3-14: Cabling for MVP130

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

1. Connect the power cord supplied with your MultiVOIP to the power connector on the back of the MultiVOIP and to a live AC outlet as shown in Figure 3-14.
2. Connect the MultiVOIP to a PC by using a RJ-45 (male) to DB-9 (female) cable. Plug the RJ-45 end of the cable into the Command port of the MultiVOIP and the other end into the PC serial port. See Figure 3-14.
3. Connect a network cable to the **ETHERNET 10/100** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.
4. If you are connecting a station device such as an analog telephone, a fax machine, or a Key Telephone System (KTS) (FXS interface), or a PBX extension (FXO interface) to your MultiVOIP, connect one end of an RJ-11 phone cord to the **Channel 1 FXS/FXO** connector on the back MultiVOIP and the other end to the device or phone jack. You will define the interface in the Interface dialog box in the software when you configure the unit.

Proceed to Chapter 4 to load the MultiVOIP software.

Chapter 4: Software Installation

Introduction

Configuring software for your MultiVOIP entails three tasks:

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

Loading MultiVOIP Software onto the PC

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

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

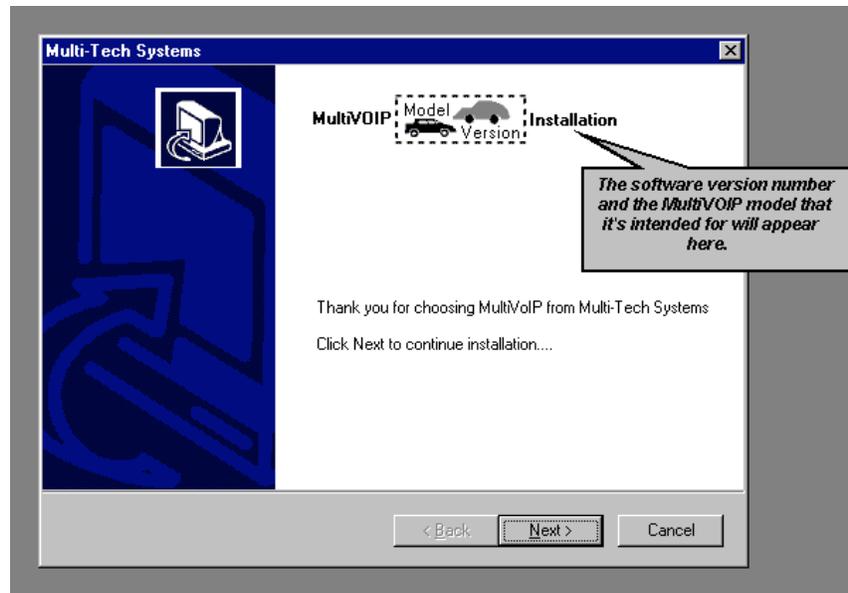
1. Be sure that your MultiVOIP has been properly cabled and that the power is turned on.

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



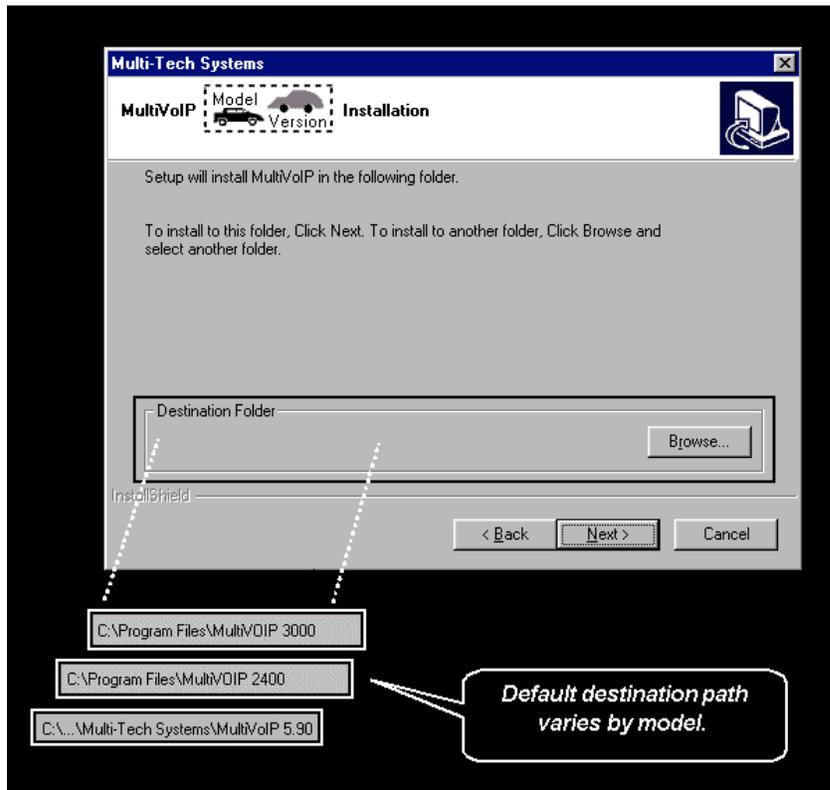
- If the Multi-Tech Installation CD window does not display automatically, click **My Computer**, then right click the **CD ROM drive** icon, click **Open**, and then click the **Autorun** icon.
3. When the Multi-Tech Installation CD dialog box appears, click the **Install Software** icon.

4. A 'welcome' screen appears.



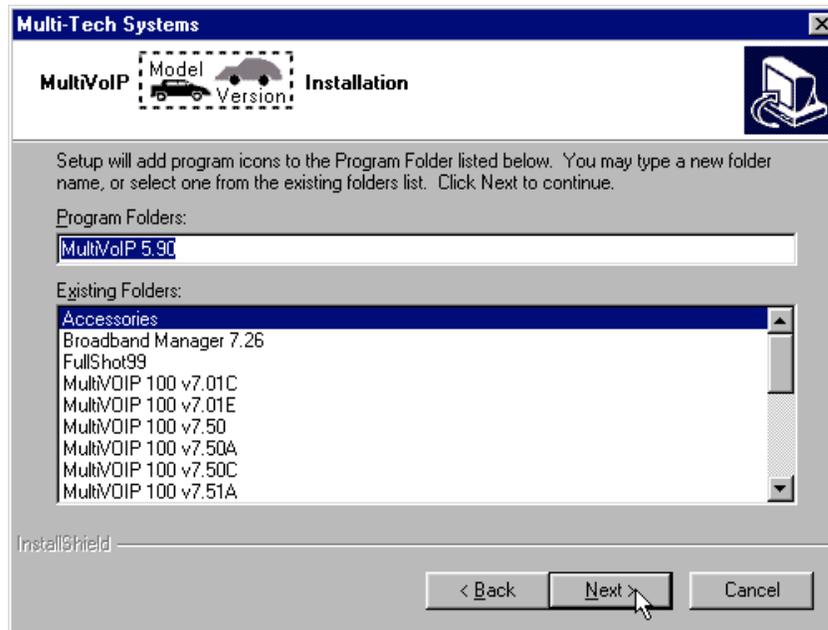
Press **Enter** or click **Next** to continue.

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



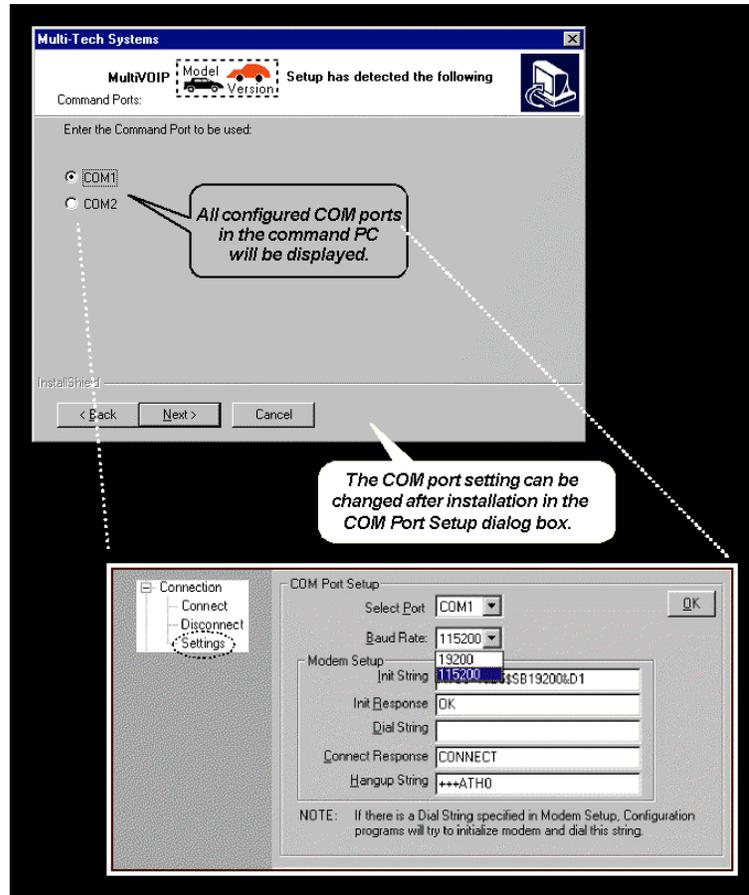
Choose a location and click Next.

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



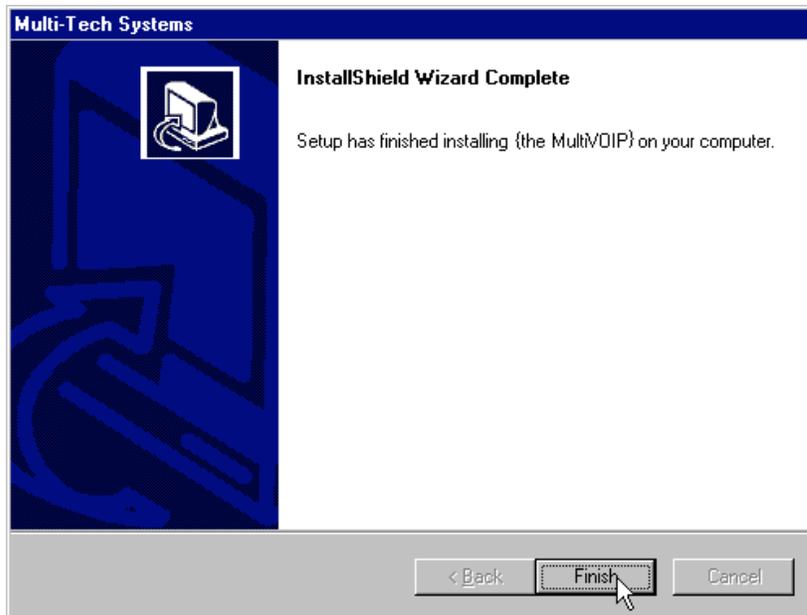
Click **Next**. Transient progress screens will appear while files are being copied.

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



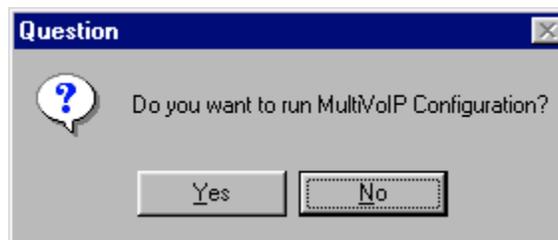
<p>NOTE: If the COM port setting made here conflicts with the actual COM port resources available in the command PC, this error message will appear when the MultiVOIP program is launched. If this occurs, you must reset the COM port.</p>	
---	--

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.

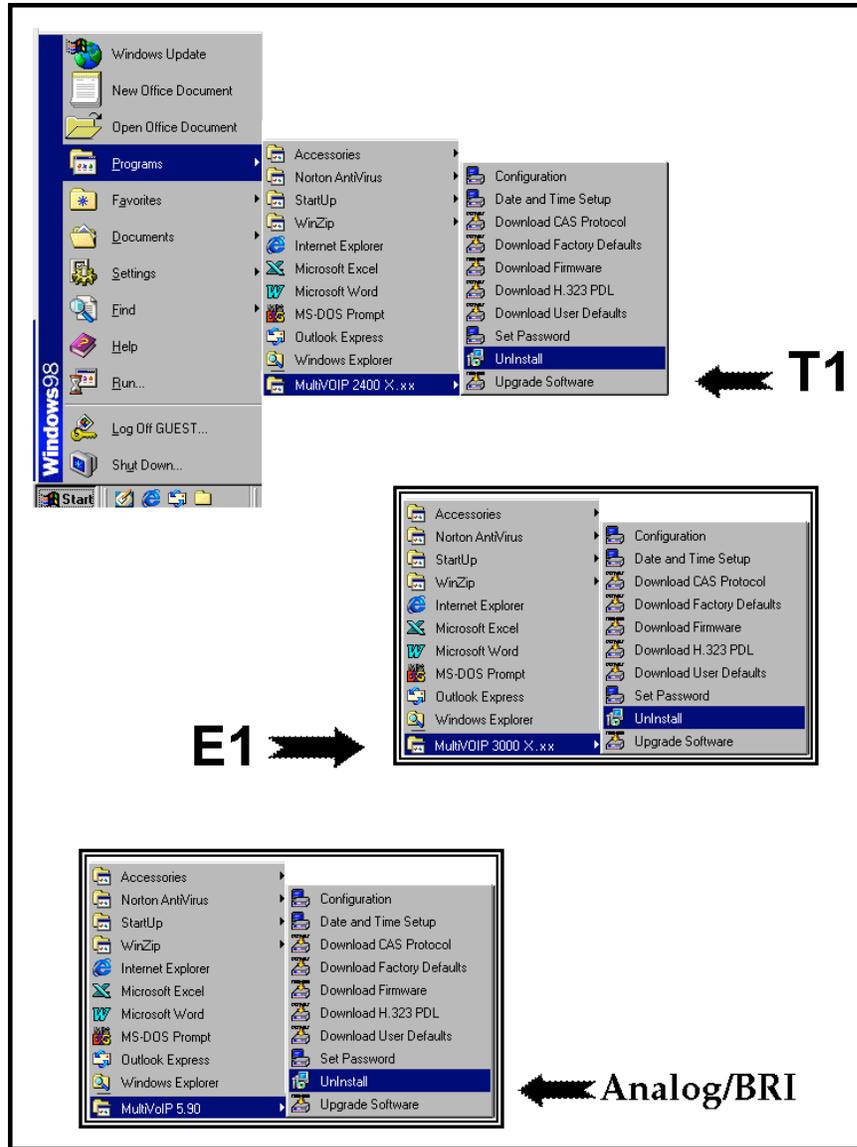


Software installation is complete at this point. You may proceed with Technical Configuration now or not, at your convenience.

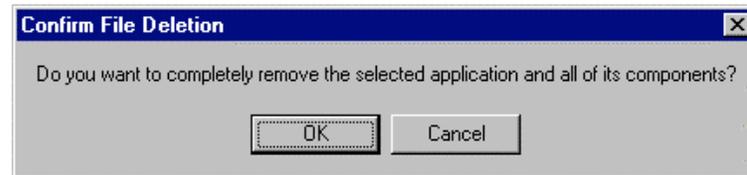
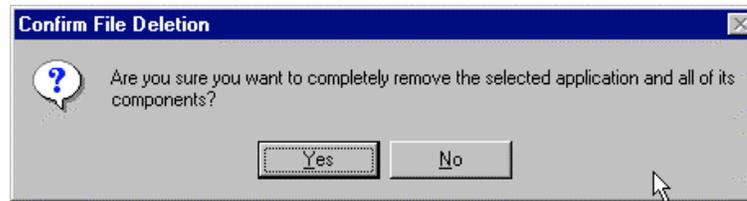
Technical Configuration instructions are in the next two chapters of this manual: Chapter 5 for T1/E1 MultiVOIP units and Chapter 6 for Analog MultiVOIP units.

Un-Installing the MultiVOIP Configuration Software

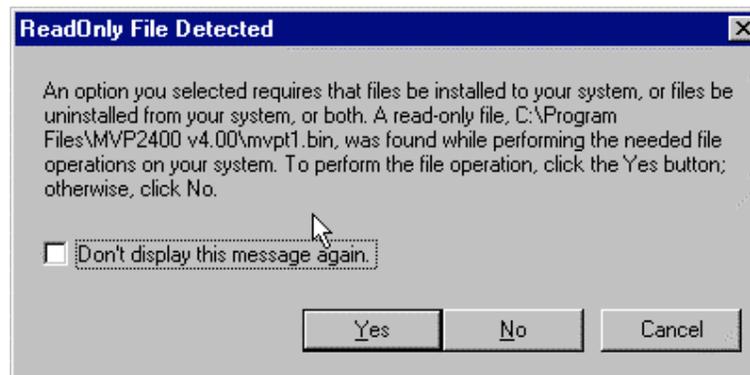
1. To un-install the MultiVOIP configuration software, go to **Start | Programs** and locate the entry for the MultiVOIP program. Select **Uninstall**.



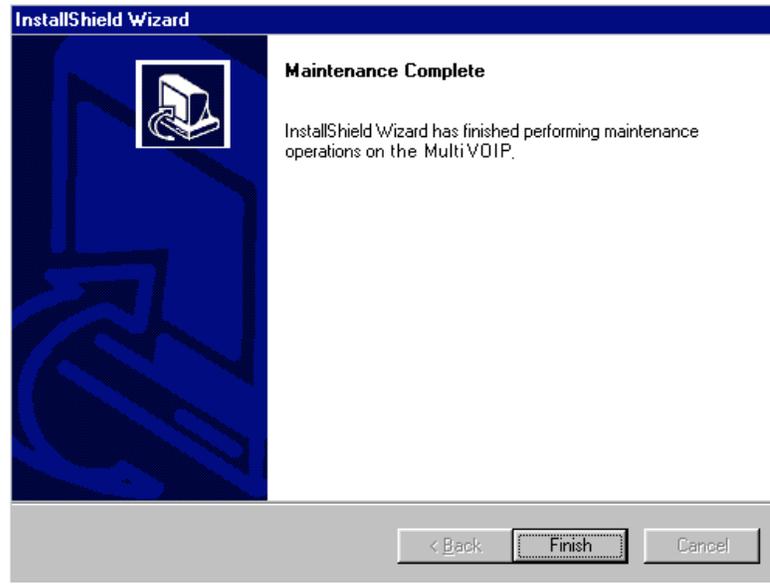
- Two confirmation screens will appear. Click **Yes** and **OK** when you are certain you want to continue with the uninstallation process.



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



4. A completion screen will appear.



Click **Finish**.

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

Configuring the Digital T1/E1 MultiVOIP

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

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

These technical parameters pertain to

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

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

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

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

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

Remote configuration is done through a connection between the MultiVOIP’s Ethernet (network) port and a computer connected to the same network. The computer could be miles or continents away from the MultiVOIP itself. There are two ways of doing remote configuration and operation of the MultiVOIP

unit: (1) using the MultiVoipManager SNMP program, or (2) using the MultiVOIP web browser interface program.

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

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

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

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

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

Local Configuration

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

Pre-Requisites



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

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

IP Parameters

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

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

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

T1 Telephony Parameters (for MVP2400 & MVP2410)

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

➤	T1 Phone Parameters <i>Ask phone company or PBX maintainer.</i>	<i>Info needed to operate:</i> MVP2400 MVP2410
	 T1 Telephony Parameters: Record for this VOIP Site	
	<ul style="list-style-type: none"> • Which frame format is used? ESF ___ or D4 ___ 	
	<ul style="list-style-type: none"> • Which CAS or PRI protocol is used? _____ 	
	<ul style="list-style-type: none"> • 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. 	
	<ul style="list-style-type: none"> • 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.

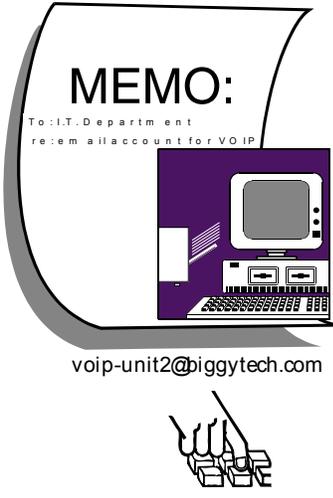
E1 Telephony Parameters (for MVP3010)

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

➤	E1 Phone Parameters <i>Ask phone company or PBX maintainer.</i>	<i>Info needed to operate:</i> MVP3010
	 E1 Telephony Parameters: Record for this VOIP Site	
	<ul style="list-style-type: none"> • Which frame format is used? Double Frame _____ MultiFrame w/ CRC4 _____ MultiFrame w/ CRC4 modified _____ 	
	<ul style="list-style-type: none"> • Which CAS or PRI protocol is used? _____ 	
	<ul style="list-style-type: none"> • 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. 	
	<ul style="list-style-type: none"> • Which line coding is used? AMI ___ or HDB3 ___ 	
	<ul style="list-style-type: none"> • 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)

<i>required if log reports of VOIP call traffic are to be sent by email</i>	Optional
<p>SMTP Parameters Preparation Task:</p> <p>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.</p> <p>Get the IP address of the mail server computer, as well.</p>	 <p>MEMO: To: I.T. Department re: email account for VOIP</p> <p>voip-unit2@biggytech.com</p>

Local Configuration Procedure (Summary)

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

1. Check Power and Cabling.
2. Start MultiVOIP Configuration Program.
3. Confirm Connection.
4. Solve Common Connection Problems.
 - A. Fixing a COM Port Problem.
 - B. Fixing a Cabling Problem.
5. Familiarize yourself with configuration parameter screens and how to access them.
6. Set IP Parameters.
7. Enable web browser GUI (optional).
8. Set Voice/Fax Parameters.
9. Set T1/E1 Parameters.
10. Set ISDN Parameters (if applicable).
11. Set SNMP Parameters (applicable if MultiVoipManager remote management software is used).
12. Set Regional Parameters (Phone Signaling Tones and Cadences).
13. Set Custom Tones and Cadences (optional).
14. Set SMTP Parameters (applicable if Log Reports are via Email).
15. Set Log Reporting Method (GUI, locally in MultiVOIP Configuration program; SNMP, remotely in MultiVoipManager program; or SMTP, via email).
16. Set Supplementary Services Parameters. The Supplementary Services screen allows voip deployment of features that are normally found in PBX or PSTN systems (e.g., call transfer and call waiting).
17. Set Baud Rate (of COM port connection to 'Command' PC).
18. View System Information and set updating interval (optional).
19. Save the MultiVOIP configuration.
20. Create a User Default Configuration (optional).

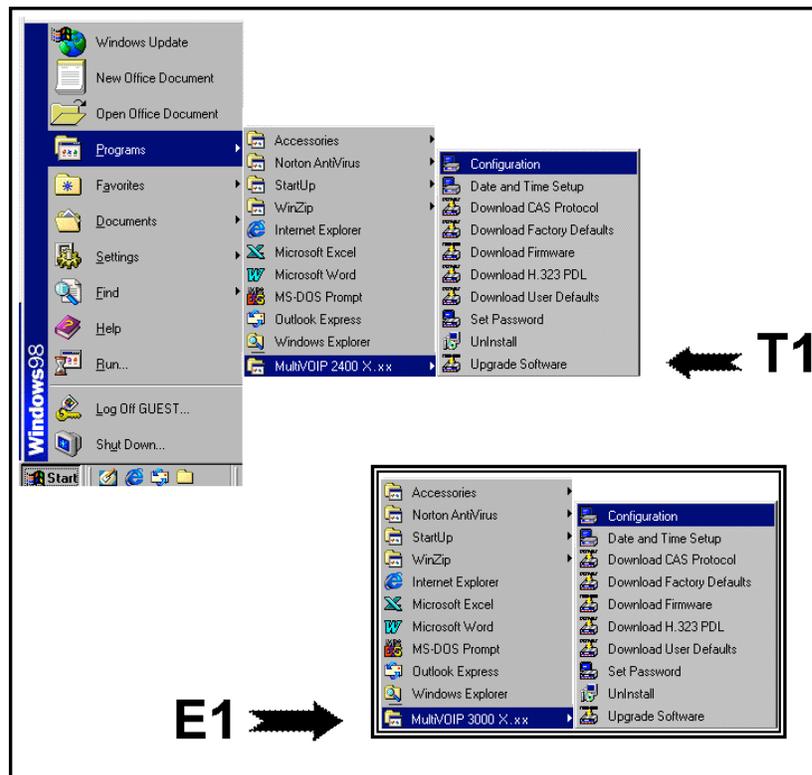
Local Configuration Procedure (Detailed)

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

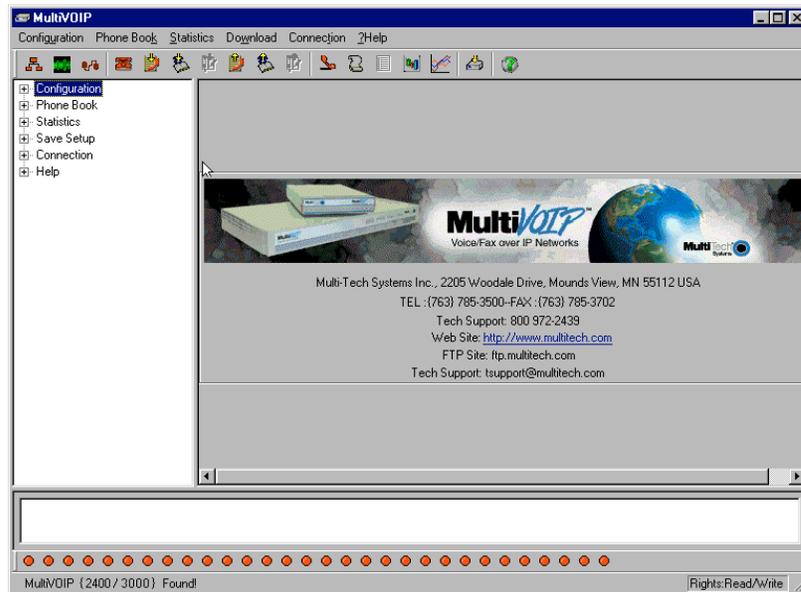
1. **Check Power and Cabling.** Be sure the MultiVOIP is turned on and connected to the computer via the MultiVOIP's Command Port (DB9 connector at computer's COM port; RJ45 connector at MultiVOIP).

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

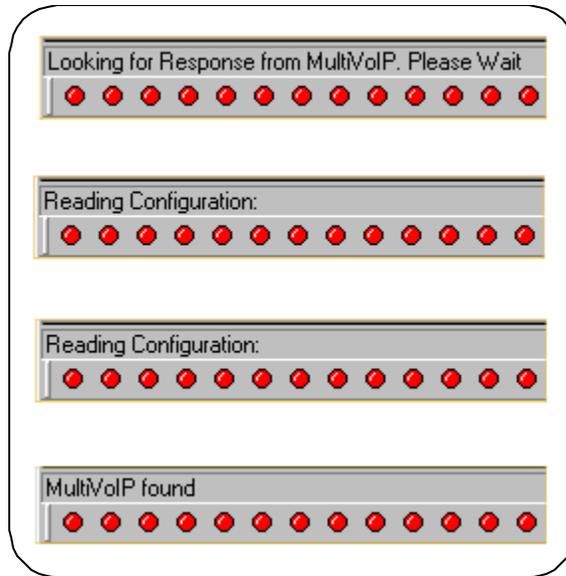
2. **Start MultiVOIP Configuration Program.** Launch the MultiVOIP program from the Windows **Start** menu (from the folder location determined during installation).



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



In the lower left corner of the screen, the connection status of the MultiVOIP will be displayed. The messages in the lower left corner will change as detection occurs. The message “MultiVOIP Found” confirms that the MultiVOIP is in contact with the MultiVOIP configuration program. Skip to step 5.

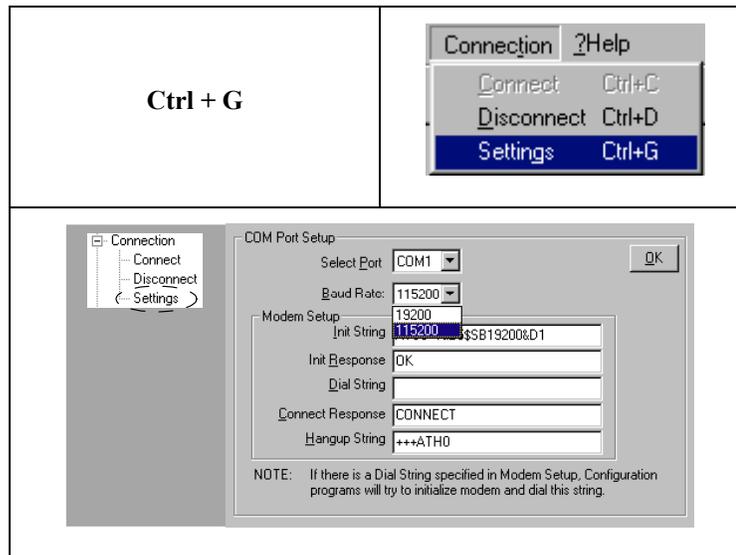


4. Solving Common Connection Problems.

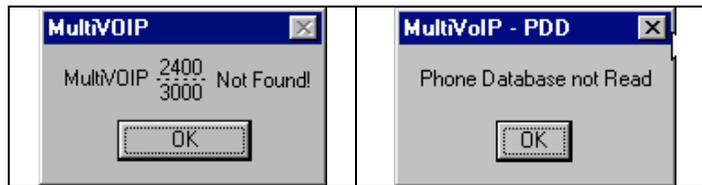
A. **Fixing a COM Port Problem.** If the MultiVOIP main screen appears but is grayed out and seems inaccessible, the COM port that was specified for its communication with the PC is unavailable and must be changed. An error message will appear.



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



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”).

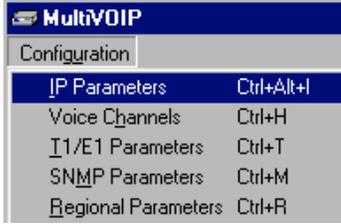
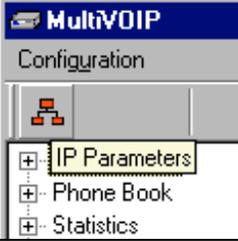


In this case, the MultiVOIP is simply disconnected from the network. For instructions on MultiVOIP cable connections, see the “Cabling” section of Chapter 3.

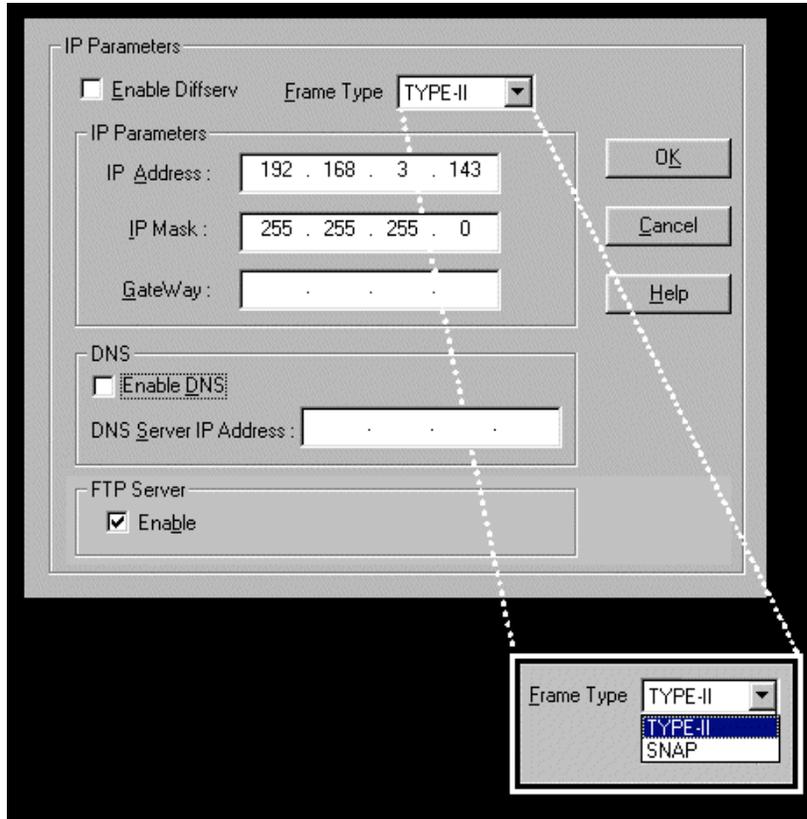
5. Configuration Parameter Groups: Getting Familiar, Learning About Access. The first part of configuration concerns IP parameters, Voice/FAX parameters, T1/E1 parameters, SNMP parameters, Regional parameters, SMTP parameters, Supplementary Services parameters, Logs, and System Information. In the MultiVOIP 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.

Accessing "IP Parameters"	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + Alt + I</p>	

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



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

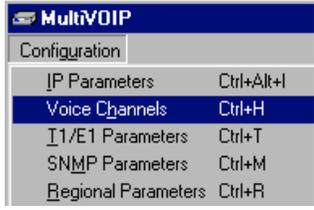
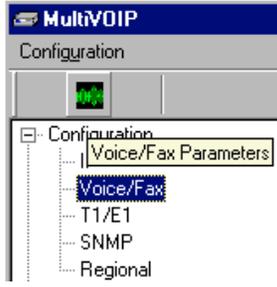
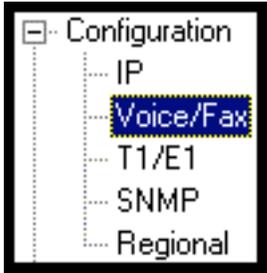
IP Parameter Definitions		
Field Name	Values	Description
Enable Diffserv	Y/N	Diffserv is used for QoS (quality of service). When enabled, the TOS (Type of Service) bits in the IP header are configured so that routers supporting Diffserv can give priority to the VOIP's IP packets. Disabled by default.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

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

Voice/Fax Parameters

Select Channel

Voice Gain
Input dB Output dB

Dtmf
Gain
High dB Low dB

DTMF Out of Band

Coder
 Manual Automatic
Selected Coder
Max bandwidth kbps

Fax
 Fax Enable
Max Baud Rate
Fax Volume dB
Jitter Value ms
Mode

Advanced Features
 Silence Compression
 Echo Cancellation
 Forward Error Correction

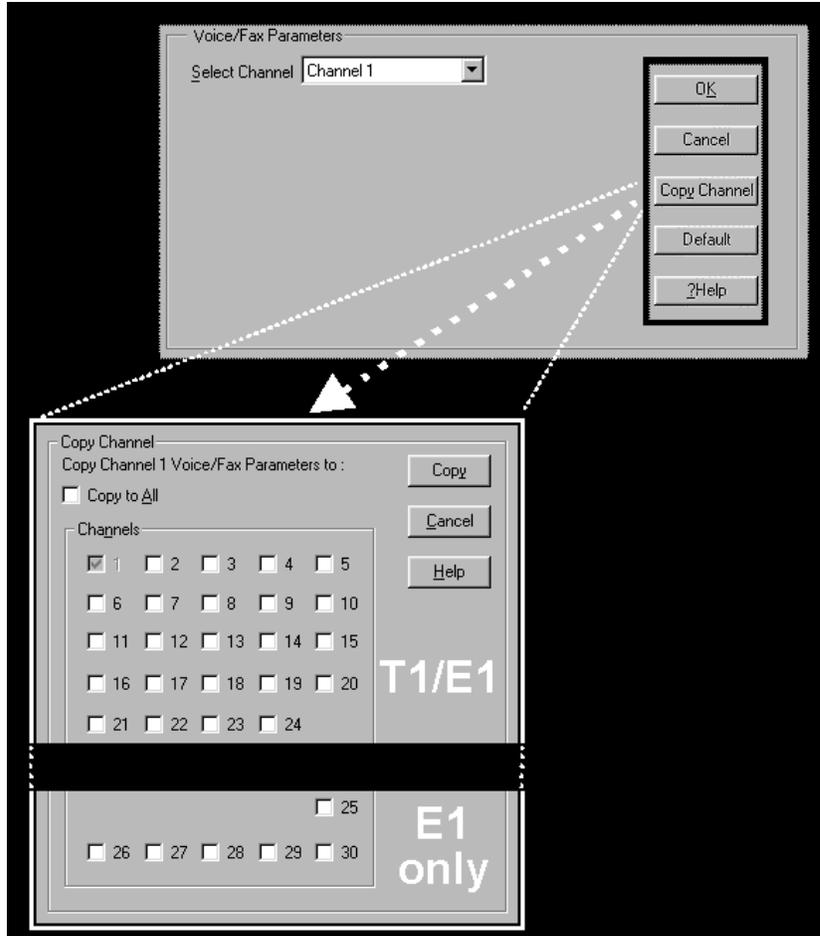
Auto Call
 Auto Call Enable
Phone Number

Dynamic Jitter Buffer
Minimum Jitter Value ms
Maximum Jitter Value ms
Optimization Factor

Automatic Disconnection
 Jitter Value ms Consecutive Packets Lost
 Call Duration seconds Network Disconnection seconds

OK
Cancel
Copy Channel
Default
Help

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	<p>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.</p> <p>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.</p> <p>To make selections from the Selected Coder drop-down list, the Manual option must be enabled.</p>
Max bandwidth (coder)	11 – 128 kbps	<p>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.</p> <p>If coder selected automatically, then enter a value for maximum bandwidth, as directed by VOIP administrator.</p>

Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
Advanced Features		
Silence Compression	Y/N	Determines whether silence compression is enabled (checked) for this voice channel. With Silence Compression enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. Default = off.
Echo Cancellation	Y/N	Determines whether echo cancellation is enabled (checked) for this voice channel. Echo Cancellation removes echo and improves sound quality. Default = on.
Forward Error Correction	Y/N	Determines whether forward error correction is enabled (checked) for this voice channel. Forward Error Correction enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off
Auto Call Enable	Y/N	The Auto Call option enables the local MultiVOIP to call a remote MultiVOIP without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote MultiVOIP identified in the Phone Number box of this option.
Phone No. (Auto Call)	--	Phone number used for Auto Call function. A corresponding phone number must be listed in the Outbound Phonebook.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
Dynamic Jitter		
Dynamic Jitter Buffer		<p>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.</p> <p>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.</p>
Minimum Jitter Value	60 to 400 ms	The default minimum dynamic jitter buffer of 60 milliseconds is the minimum delay that would be acceptable over a low jitter network. Default = 60 msec

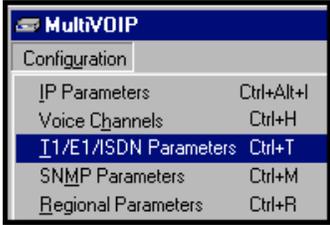
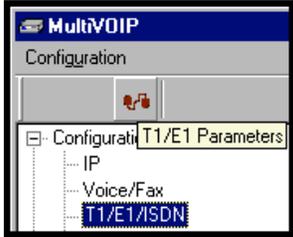
Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
Dynamic Jitter		
Maximum Jitter Value	60 to 400 ms	The default maximum dynamic jitter buffer of 300 milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 msec
Optimization Factor	0 to 12	The Optimization Factor determines how quickly the length of the Dynamic Jitter Buffer is changed based on actual jitter encountered on the network. Selecting the minimum value of 0 means low voice delay is desired, but increases the possibility of jitter-induced voice quality problems. Selecting the maximum value of 12 means highest voice quality under jitter conditions is desired at the cost of increased voice delay. Default = 7.

Modem Relay

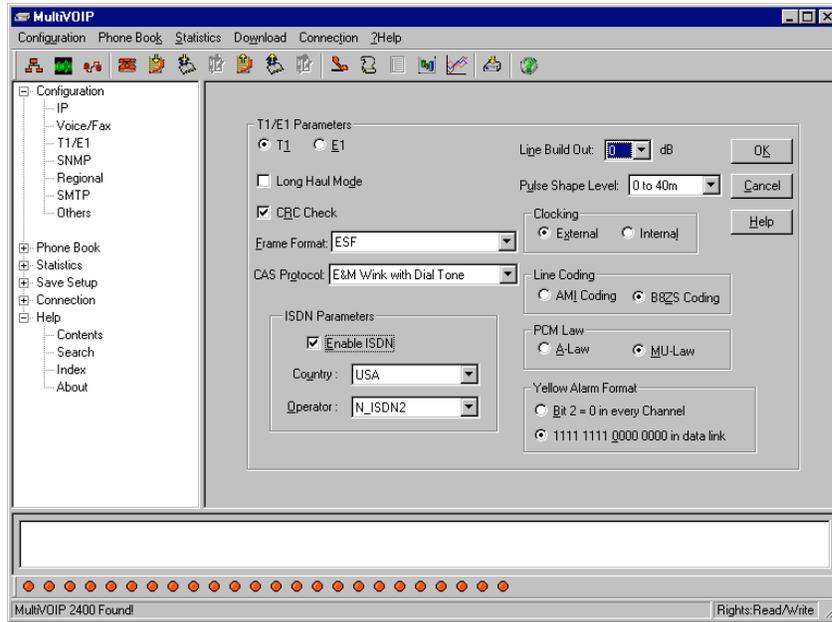
To place modem traffic onto the voip network (an application called “modem relay”), use Coder G.711 mu-law at 64kbps.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
Auto Disconnect		
Automatic Disconnection	--	The Automatic Disconnection group provides four options which can be used singly or in any combination.
Jitter Value	1-65535 milliseconds	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 Disconnection	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
<p>Ctrl + T</p>	

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/E1/ISDN Parameters

T1 E1

Long Haul Mode

CRC Check

Line Build Out: 0 dB

Frame Format: ESF

Pulse Shape Level: 0 to 40m

CAS Protocol: E&M Wink

Clocking: External Internal

Line Coding: AMI Coding B8ZS Coding

ISDN Parameters:

Enable ISDN-PRI

Terminal Network

Country: []

Operator: []

PCM Law: A-Law MU-Law

Yellow Alarm Format:

Bit 2 = 0 in every Channel

1111 1111 0000 0000 in data link

OK Cancel Help

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	<p>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).</p> <p>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.</p> <p>Match this parameter to the setting of PBX or central-office switch.</p>

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: 0 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.

T1/E1/ISDN Parameters

T1 E1

Long Haul Mode

CRC Check

Line Build Out: 0 dB

Frame Format: MultiFrame with CRC4(modified)

CAS Prglocat: E&M Wink with Dial Tone

ISDN Parameters

Enable ISDN-PR!

Terminal Network

Country: []

Operator: []

Pulse Shape Level: 0 to 40m

Clocking: External Internal

Line Coding: AMT Coding HDB3 Coding

PCM Law: A-Law MU-Law

Buttons: OK, Cancel, Help

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	<p>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).</p> <p>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.</p> <p>Match this parameter to the setting of PBX or central-office switch.</p>

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: 0 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.

ISDN Parameters

Enable ISDN-PRI

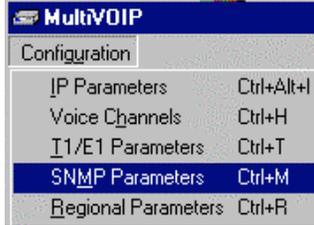
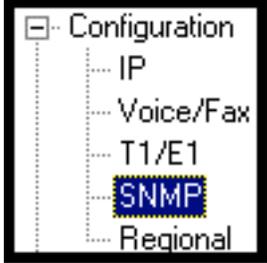
Terminal Network

Country : USA

Operator : N_ISDN2

Country :	Operator :
Australia	AUSTEL_1
Belgium	BG_V1
Europe	ETSI ECMA_QSIG FT_VN6 RITA
France	FT_VN2 FT_VN3 FT_VN6
Germany	DT_1TR6
HongKong	HK_TEL
Italy	ETSI
Japan	NTT KDD
Korea	KOREAN_OP
NewZealand	TEL_NZ
Sweden	SWD_TVKT
USA	N_ISDN1 N_ISDN2 ATT_4ESS ATT_5E5 ATT_5E9 ATT_5E10 BELLCORE_PRI NT_DMS100
UK	BT ISDN2

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.

Accessing “SNMP Parameters”	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + M</p>	

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

SNMP Parameters

Enable SNMP Agent

Trap Manager

Address : 0 . 0 . 0 . 0

Community Name :

Port Number : 162

Community Name - 1 : public

Permissions : Read Only

Community Name - 1 : public

Permissions : Read/Write

Read Only

Read/Write

OK

Cancel

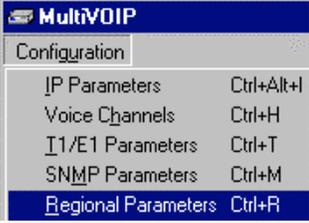
Help

The SNMP Parameter fields are described in the table below.

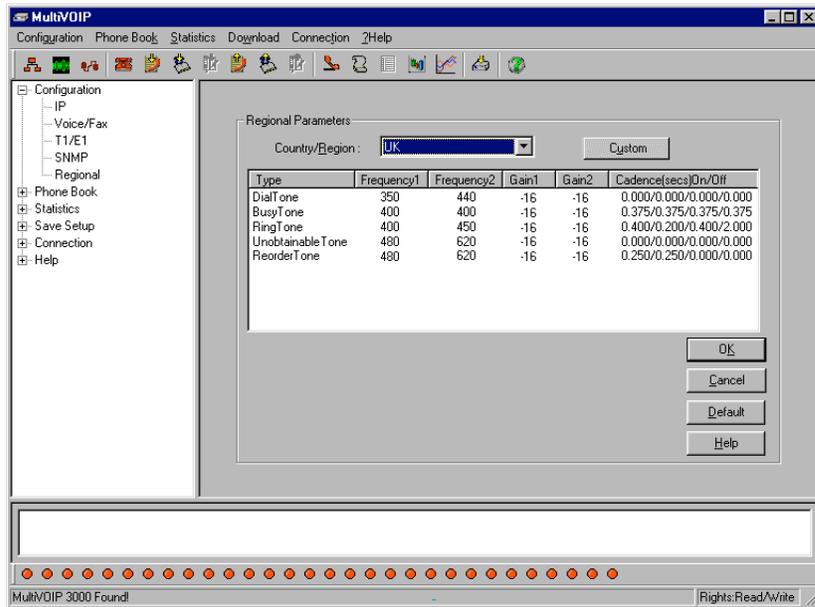
SNMP Parameter Definitions		
Field Name	Values	Description
Enable SNMP Agent	Y/N	Enables the SNMP code in the firmware of the MultiVOIP. This must be enabled for the MultiVOIP to communicate with and be controllable by the MultiVoipManager software. Default: disabled
Trap Manager Parameters		
Address	4 places; n.n.n.n n = 0-255	IP address of MultiVoipManager PC.
Community Name	--	A "community" is a group of VOIP endpoints that can communicate with each other. Often "public" is used to designate a grouping where all end users have access to entire VOIP network. However, calling permissions can be configured to restrict access as needed.
Port Number	162	The default port number of the SNMP manager receiving the traps is the standard port 162.
Community Name 1	Length = 19 characters (max.) Case sensitive.	First community grouping.
Permissions	Read-Only, Read/Write	If this community needs to change MultiVOIP settings, select Read/Write. Otherwise, select Read-Only to view settings.
Community Name 2	Length = 19 characters (max.) Case sensitive.	Second community grouping
Permissions	Read-Only, Read/Write	If this community needs to change MultiVOIP settings, select Read/Write. Otherwise, select Read-Only to view settings.

12. **Set Regional Parameters** (Phone Signaling Tones & Cadences).

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

Accessing "Regional Parameters"	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + R</p>	

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



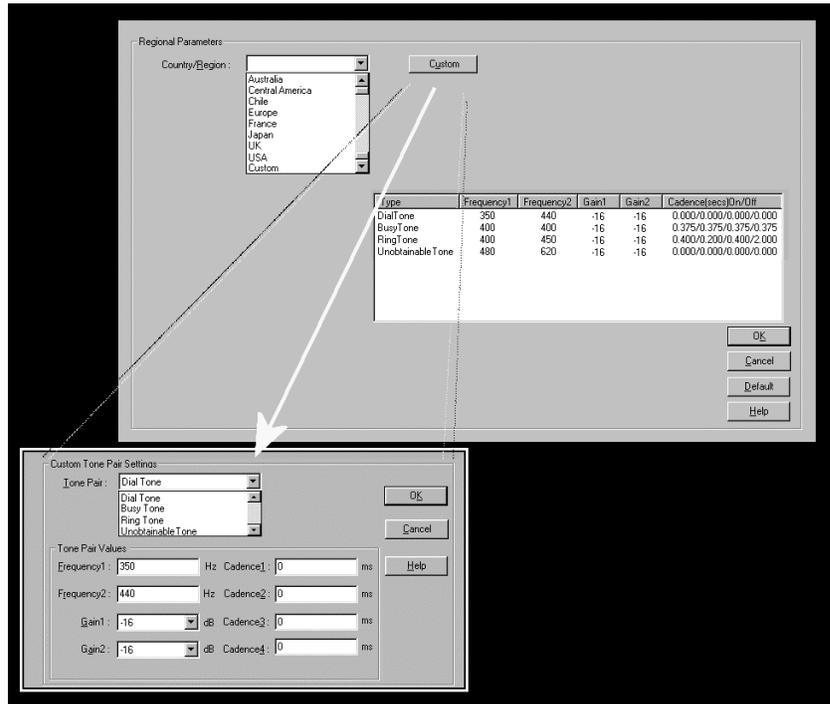
In each field, enter the values that fit your particular system.

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

“Regional Parameter” Definitions		
Field Name	Values	Description
Country/ Region	USA, Japan, UK, Custom	Name of a country or region that uses a certain set of tone pairs for dial tone, ring tone, busy tone, ‘unobtainable’ tone (fast busy tone) and re-order tone (a tone pattern indicating the need for the user to hang up the phone). In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The “Custom” option (button) assures that any tone-pairing scheme worldwide can be accommodated.
Type column	dial tone, ring tone, busy tone, unobtainable tone (fast busy), & re-order tone	Type of telephony tone-pair for which frequency, gain, and cadence are being presented.
Frequency 1	frequency in Hertz	Lower frequency of pair.
Frequency 2	frequency in Hertz	Higher frequency of pair.
Gain 1	gain in dB +3dB to -31dB and “mute” setting	Amplification factor of lower frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the T1 port. Default: -16dB
Gain 2	gain in dB +3dB to -31dB and “mute” setting	Amplification factor of higher frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the T1 port. Default: -16dB

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

13. **Set Custom Tones and Cadences** (optional) . The **Regional Parameters** dialog box has a secondary dialog box that allows you to customize DTMF tone pairs to create unique ring-tones, dial tones, busy-tones “unobtainable” tones (fast busy signal) or “re-order” tones (telling the user that they must hang up an off-hook phone) for your system. This screen allows the user to specify tone-pair attributes that are not found in any of the standard national/regional telephony toning schemes. To access this customization feature, click on the **Custom** button on the **Regional Parameters** screen.

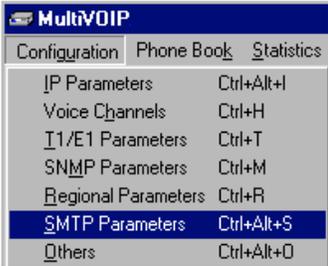
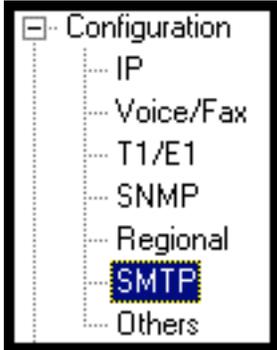


The **Custom Tone-Pair Settings** fields are described in the table below.

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

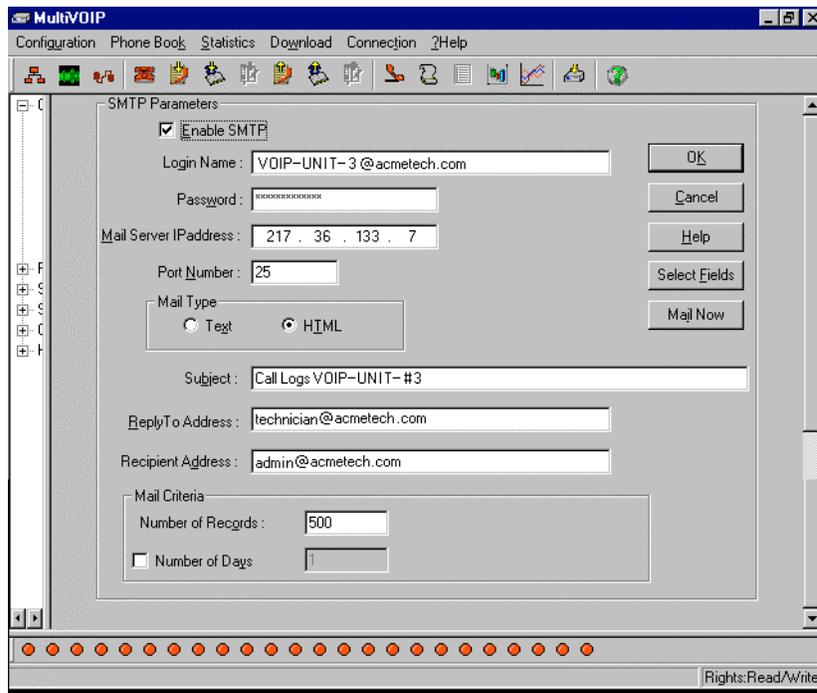
Custom Tone-Pair Settings Definitions		
Field Name	Values	Description
Cadence 1	integer time value in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable tone (fast busy), dial tone (which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

14. **Set SMTP Parameters** (Log Reports by Email). The **SMTP Parameters** screen is applicable when the VOIP administrator has chosen to receive log reports by email (this is done by selecting the “SMTP” checkbox in the **Others** screen and selecting “Enable SMTP” in the **SMTP Parameters** screen.). The **SMTP Parameters** screen can be reached by pulldown menu, keyboard shortcut, or sidebar.

Accessing “SMTP Parameters”	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + Alt + S</p>	

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

The SMTP Parameters screen is shown below.

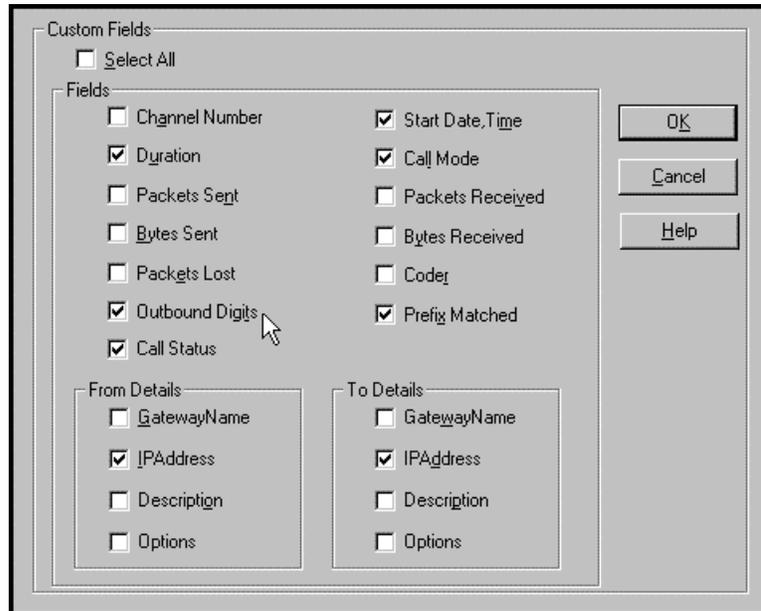


"SMTP Parameters" Definitions		
Field Name	Values	Description
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select "SMTP" in the Logs screen.
Login Name	alpha-numeric, per email domain	This is the User Name for the MultiVOIP unit's email account.
Password	alpha-numeric	Login password for MultiVOIP unit's email account.
Mail Server IP Address	n.n.n.n for n= 0 to 255	This mail server must be accessible on the IP network to which the MultiVOIP is connected.
Port Number	25	25 is a standard port number for SMTP.

.....

“SMTP Parameters” Definitions (cont’d)		
Field Name	Values	Description
Mail Type	text or html	Mail type in which log reports will be sent.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP unit.
Reply-To Address	email address	User specified. This email address functions as a source email identifier for the MultiVOIP, which, of course, cannot usefully receive email messages. The Reply-To address provides a destination for returned messages indicating the status of messages sent by the MultiVOIP (esp. to indicate when log report email was undeliverable or when an error has occurred).
Recipient Address	email address	User specified. Email address at which VOIP administrator will receive log reports.
Mail Criteria		Criteria for sending log summary by email. The log summary email will be sent out either when the user-specified number of log messages has accumulated, or once every day or multiple days, <i>which ever comes first</i> .
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.

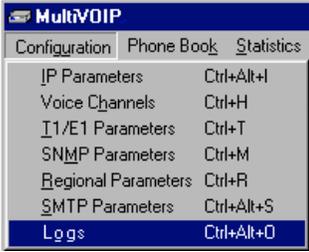
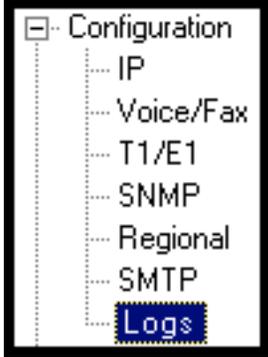


"Custom Fields" Definitions			
Field	Description	Field	Description
Select All	Log report to include all fields shown.		
Channel Number	Data channel carrying call.	Start Date, Time	Date and time the phone call began.
Duration	Length of call.	Call Mode	Voice or fax.
Packets Sent	Total packets sent in call.	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.

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
<p>Ctrl + Alt + O</p>	

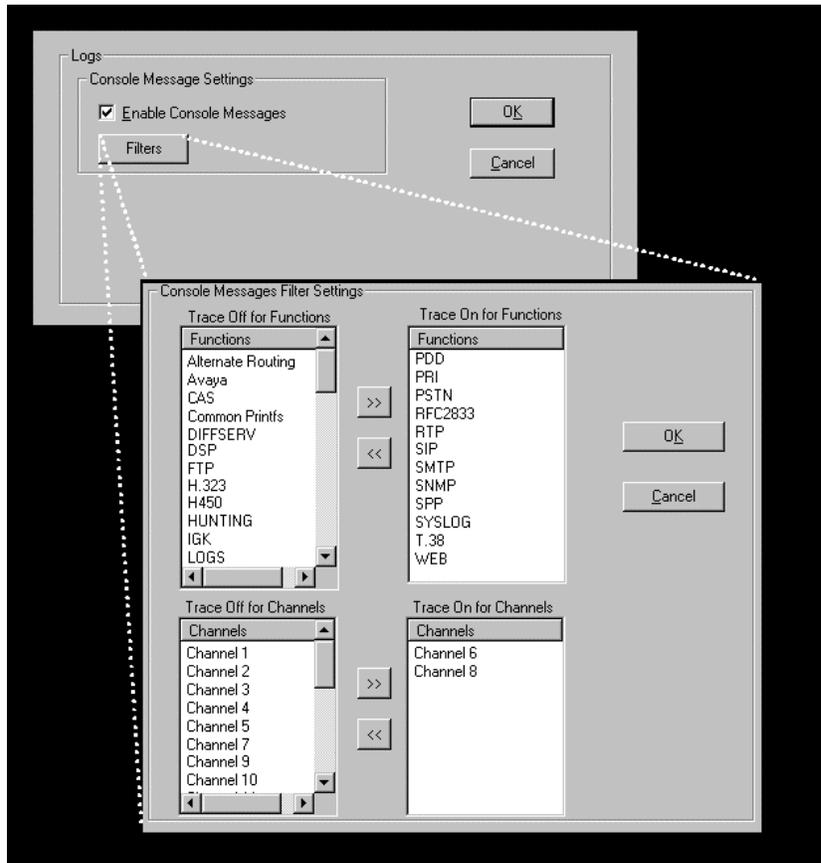
If you enable console messages, you can customize the types of messages to be included/excluded in log reports by clicking on the "Filters" button and using the **Console Messages Filter Settings** screen (see subsequent page). If you use the logging function, select the logging option that applies to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

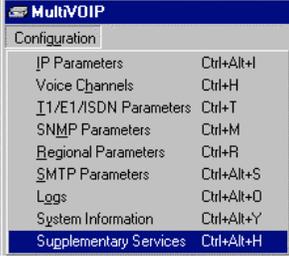
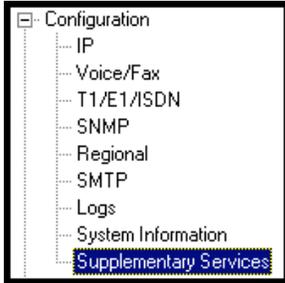
"Logs" Screen Definitions		
Field Name	Values	Description
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic tele-communications program like HyperTerminal™ or similar application. Normally, this should be disabled because it consumers MultiVOIP pro-cessing resources. 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 <i>Operation & Maintenance</i> chapter.
IP Address	n.n.n.n for n= 0-255	IP address of computer, connected to voip network, on which SysLog Server program is running.
Port	514	Logical port for SysLog Server. 514 is commonly used.
Online Statistics Updation Interval	integer	Set the interval (in seconds) at which logging information will be updated.

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



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

Accessing “Supplementary Services Parameters”	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + Alt +H</p>	

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.

The screenshot shows a configuration window titled "Supplementary Services Parameters". At the top, there is a "Select Channel" dropdown menu set to "Channel 1". Below this are four main sections:

- Call Transfer:** Includes a checked "Enable" checkbox and a "Transfer Sequence" text field containing "#*1".
- Call Hold:** Includes a checked "Enable" checkbox and a "Hold Sequence" text field containing "#*2".
- Call Waiting:** Includes a checked "Enable" checkbox and a "Retrieve Sequence" text field containing "#*3".
- Call Name Identification:** Includes a checked "Enable" checkbox, an "Allowed Name Type" section with four unchecked checkboxes ("Calling Party", "Busy Party", "Alerting Party", "Connected Party"), and a "Caller Id" text field.

At the bottom right of the window are four buttons: "OK", "Default", "Cancel", and "Copy Channel".

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

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

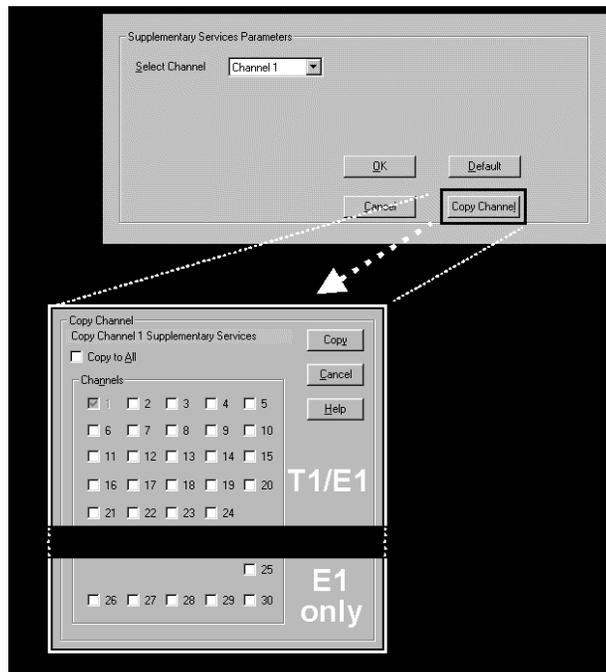
Call Hold. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function. Invoked by keypad sequence.

Call Waiting. Call Waiting notifies an engaged caller of an incoming call and allows them to receive a call from a third party while the party with whom they have been speaking is put on hold. Invoked by keypad sequence.

Call Name Identification. When enabled for a given voip unit (the 'home' voip), this feature gives notice to remote voips involved in calls. Notification goes to the remote voip administrator, not to individual phone stations. When the home voip is the caller, a plain English descriptor will be sent to the remote (callee) voip identifying

the channel over which the call is being originated (for example, “Calling Party - Omaha Sales Office Line 2”). If that voip channel is dedicated to a certain individual, the descriptor could say that, as well (for example “Calling Party - Harold Smith in Omaha”). When the home voip receives a call from any remote voip, the home voip sends a status message back to that caller. This message confirms that the home voip’s phone channel is either busy or ringing or that a connection has been made (for example, “Busy Party - Omaha Sales Office Line ”). These messages appear in the **Statistics – Call Progress** screen of the remote voip.

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



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

Supplementary Services Parameter Definitions		
Field Name	Values	Description
Select Channel	1-2 (210); 1-4 (410); 1-8 (810)	The channel to be configured is selected here.
Call Transfer Enable	Y/N	Select to enable the Call Transfer function in the voip unit. This is a “blind” transfer and the sequence of events is as follows: Callers A and B are having a conversation. Caller A wants to put B into contact with C. Caller A dials call transfer sequence. Caller A hears dial tone and dials number for caller C. Caller A gets disconnected while Caller B gets connected to caller C.
Transfer Sequence	any phone keypad character	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call transfer. The call-transfer sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). The sequences for call transfer, call hold, and call waiting can be from 1 to 4 digits in length consisting of any combination of digits 1234567890*#.

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Call Hold Enable	Y/N	Select to enable Call Hold function in voip unit. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function.
Hold Sequence	phone keypad characters	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call hold. The call-hold sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #).
Call Waiting Enable	Y/N	Select to enable Call Waiting function in voip unit.
Retrieve Sequence	phone keypad characters, two characters in length	The numbers and/or symbols that the caller must press on the phone keypad to initiate retrieval of a waiting call. The call-waiting retrieval sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). This is the phone keypad sequence that a user must press to retrieve a waiting call. Customize-able. Sequence should be distinct from sequence that might be used to retrieve a waiting call via the PBX or PSTN.

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Call Name Identification Enable		<p>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).</p> <p>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.</p> <p>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.)</p>

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Calling Party, Allowed Name Type (CNI)		<p>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.</p> <p>This field is applicable only when the 'home' voip unit is originating the call.</p> <p>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.</p> <p>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.</p>

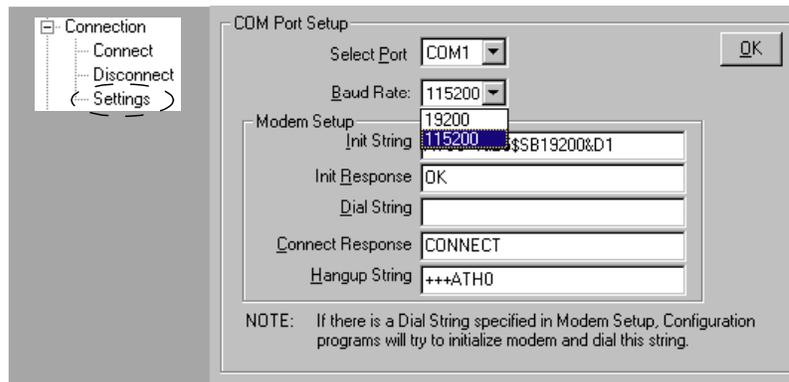
Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Alerting Party, Allowed Name Type (CNI)		<p>If the 'home' voip unit is receiving the call and Alerting Party is selected, then the identifier (from the Caller Id field) will tell the originating remote voip unit that the call is ringing.</p> <p>This field is applicable only when the 'home' voip unit is receiving the call.</p> <p>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.</p> <p>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.</p>

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Busy Party, Allowed Name Type (CNI)		<p>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.</p> <p>This field is applicable only when the 'home' voip unit is receiving the call.</p> <p>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.</p> <p>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.</p>

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Connected Party, Allowed Name Type (CNI)		<p>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.</p> <p>This field is applicable only when the 'home' voip unit is receiving the call.</p> <p>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.</p> <p>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.</p>

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.

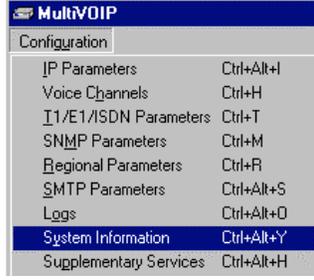
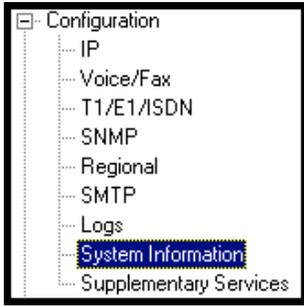


First, it is important to note that the default COM port established by the MultiVOIP program is COM1. ***Do not accept the default value until you have checked the COM port allocation on your PC.*** To do this, check for COM port assignments in the system resource dialog box(es) of your Windows operating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

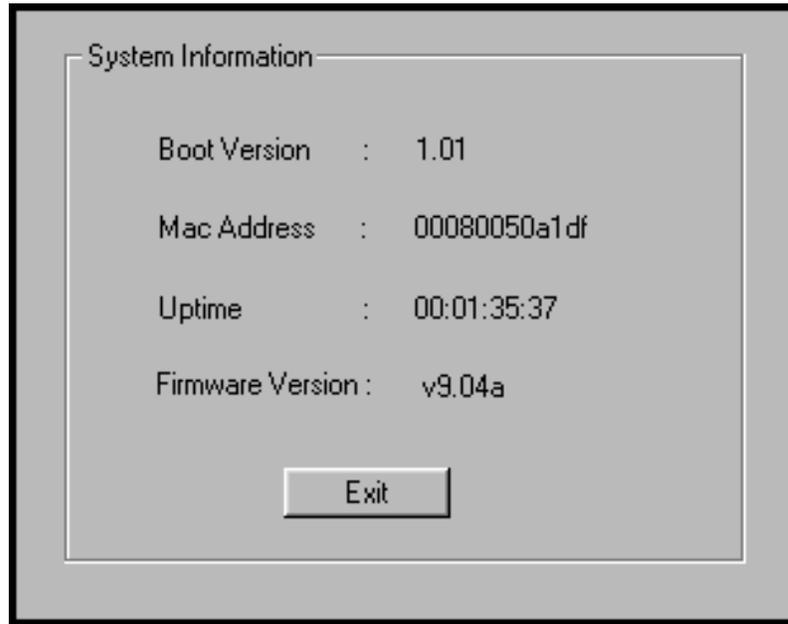
The default baud rate is 115,200 bps.

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

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

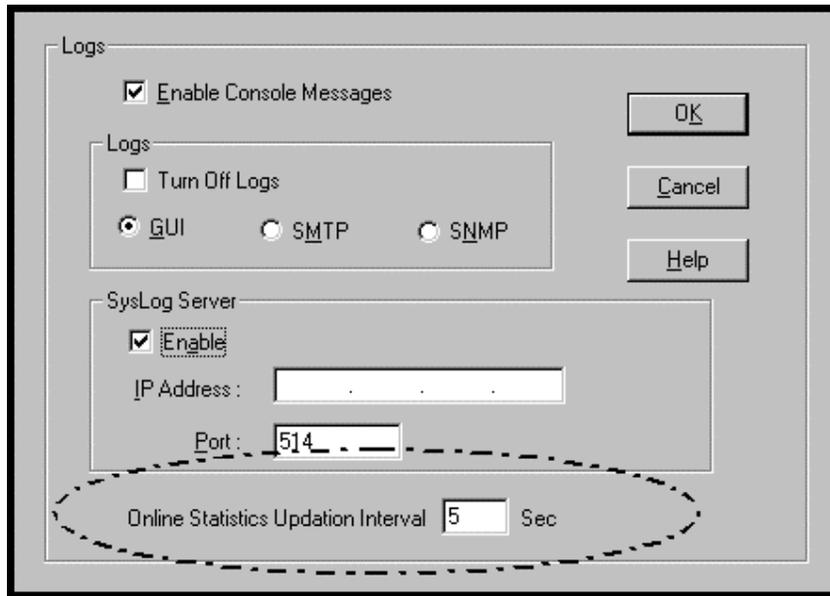
Accessing the “System Information” Screen	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + Alt + Y</p>	

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

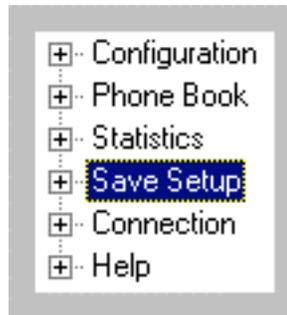


System Information Parameter Definitions		
Field Name	Values	Description
Boot Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Mac Address	alpha-numeric	Denotes the number assigned as the voip unit's unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.
Firmware Version	alpha-numeric	Indicates version of MultiVOIP firmware.

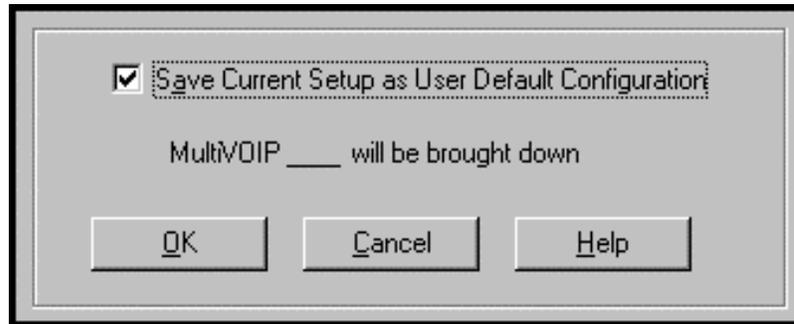
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



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



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



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

Configuring the Analog/BRI MultiVOIP

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

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

These technical parameters pertain to

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

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

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

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

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

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

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

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

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

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

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

Local Configuration

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

Pre-Requisites



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

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

IP Parameters

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

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

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

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

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

➤	<p>Analog Phone Parameters</p> <p><i>Ask phone company or telecom manager.</i></p>	<p><i>Needed for:</i></p> <p>MVP810</p> <p>MVP410</p> <p>MVP210</p> <p>MVP130</p>
 <p>Analog Telephony Interface Parameters: Record for this VOIP Site</p>		
<ul style="list-style-type: none"> • Which interface type (or “signaling”) is used? E&M _____ FXS/FXO _____ 		
<ul style="list-style-type: none"> • If FXS, determine whether the line will be used for a phone, fax, or KTS (key telephone system) 		
<ul style="list-style-type: none"> • If FXO, determine if line will be an analog PBX extension or an analog line from a telco central office 		
<ul style="list-style-type: none"> • If E&M, determine these aspects of the E&M trunk line from the PBX: <ul style="list-style-type: none"> • What is its Type (1, 2, 3, 4, or 5)? • Is it 2-wire or 4-wire? • Is it Dial Tone or Wink? 		

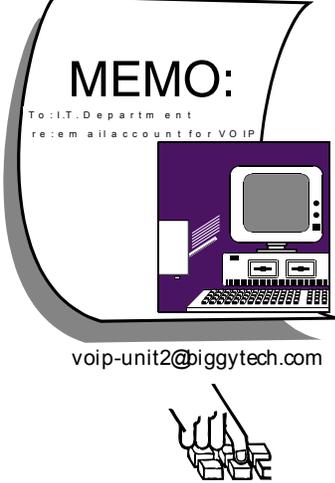
ISDN-BRI Telephony Parameters (for MVP-410ST/810ST)

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

➤	ISDN-BRI Phone Parameters <i>Ask phone company or telecom manager.</i>	<i>Needed for:</i> MVP810ST MVP410ST
	 ISDN-BRI Telephony Interface Parameters: Record them for this VOIP Site	
	<ul style="list-style-type: none"> • In which country is this voip installed? 	
	<ul style="list-style-type: none"> • Which operator (switch type) is used? 	
	<ul style="list-style-type: none"> • What type of line coding use required, A-law or u-law? 	
	<ul style="list-style-type: none"> • 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)

<p><i>required if log reports of VOIP call traffic are to be sent by email</i></p>	<p>Optional</p>
<p>SMTP Parameters Preparation Task:</p> <p>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.</p> <p>Get the IP address of the mail server computer, as well.</p>	

Local Configuration Procedure (Summary)

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

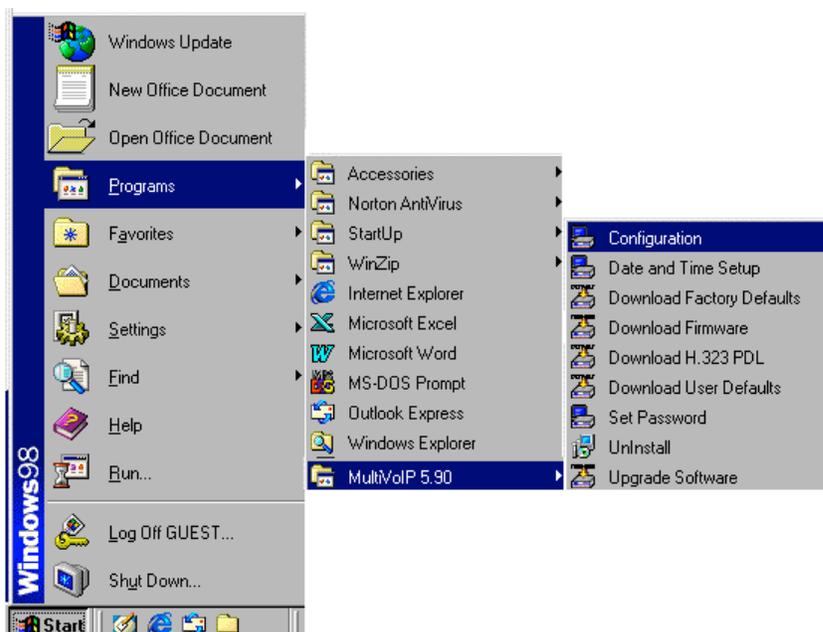
1. Check Power and Cabling.
2. Start MultiVOIP Configuration Program.
3. Confirm Connection.
4. Solve Common Connection Problems.
 - A. Fixing a COM Port Problem.
 - B. Fixing a Cabling Problem.
5. Familiarize yourself with configuration parameter screens and how to access them.
6. Set IP Parameters.
7. Enable web browser GUI (optional).
8. Set Voice/Fax Parameters.
9. Set Telephony Interface Parameters (analog) or ISDN Parameters (ISDN/BRI).
10. Set SNMP Parameters (applicable if MultiVoipManager remote management software is used).
11. Set Regional Parameters (Phone Signaling Tones and Cadences).
12. Set Custom Tones and Cadences (optional).
13. Set SMTP Parameters (applicable if Log Reports are via Email).
14. Set Log Reporting Method (GUI, locally in MultiVOIP Configuration program; SNMP, remotely in MultiVoipManager program; or SMTP, via email).
15. Set Supplementary Services Parameters. The Supplementary Services screen allows voip deployment of features that are normally found in PBX or PSTN systems (e.g., call transfer and call waiting).
16. Set Baud Rate (of COM port connection to 'Command' PC).
17. View System Info screen and set updating interval (optional).
18. Save the MultiVOIP configuration.
19. Create a User Default Configuration (optional).

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

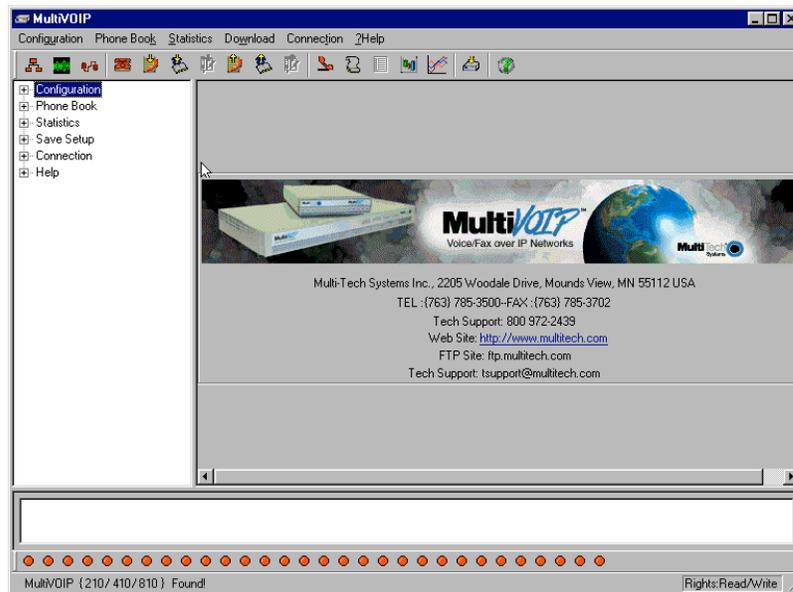
Local Configuration Procedure (Detailed)

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

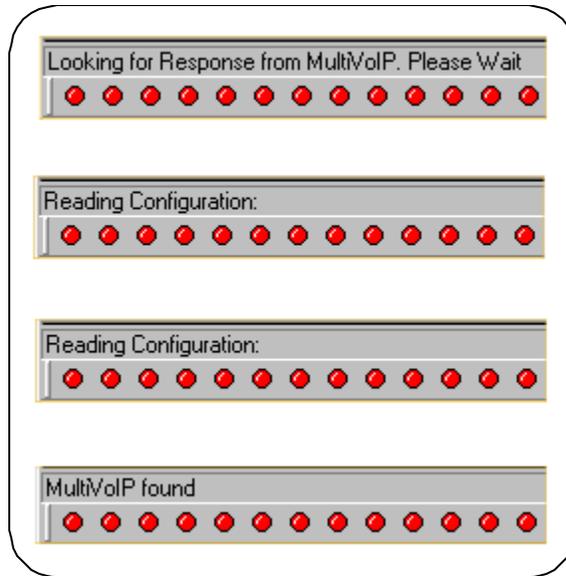
1. **Check Power and Cabling.** Be sure the MultiVOIP is turned on and connected to the computer via the MultiVOIP's Command Port (DB9 connector at computer's COM port; RJ45 connector at MultiVOIP).
2. **Start MultiVOIP Configuration Program.** Launch the MultiVOIP program from the Windows **Start** menu (from the folder location determined during installation).



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



In the lower left corner of the screen, the connection status of the MultiVOIP will be displayed. The messages in the lower left corner will change as detection occurs. The message “MultiVOIP Found” confirms that the MultiVOIP is in contact with the MultiVOIP configuration program. Skip to step 5.

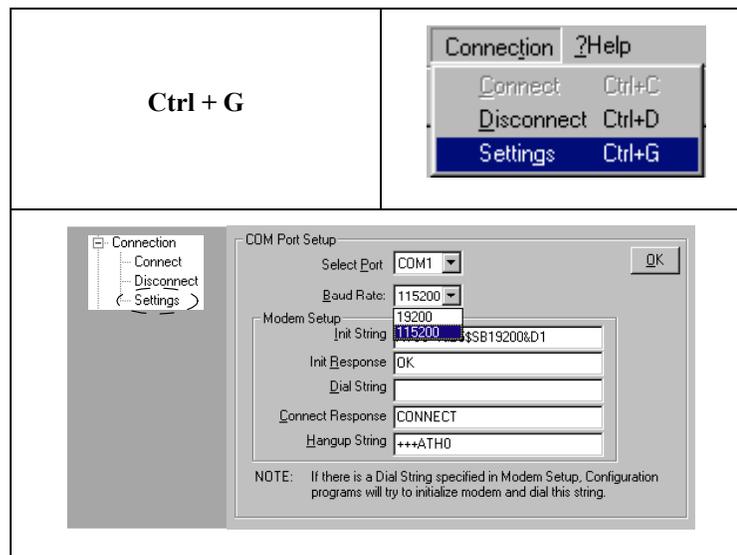


4. Solving Common Connection Problems. .

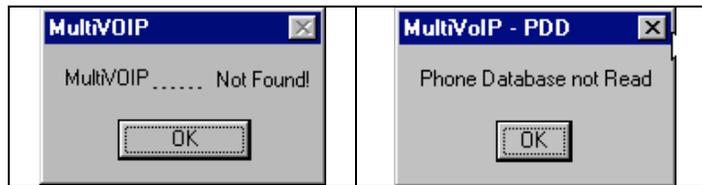
A. Fixing a COM Port Problem. If the MultiVOIP main screen appears but is grayed out and seems inaccessible, the COM port that was specified for its communication with the PC is unavailable and must be changed. An error message will appear.



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



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”).

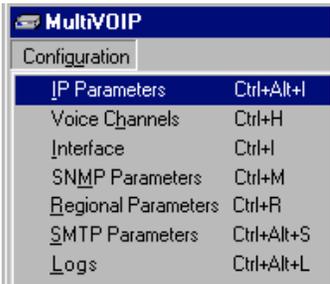
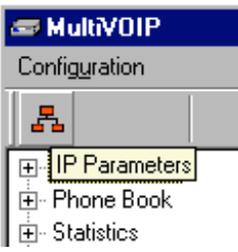
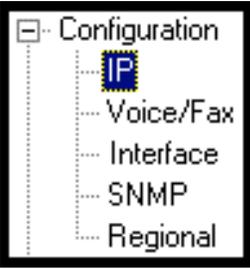


In this case, the MultiVOIP is simply disconnected from the network. For instructions on MultiVOIP cable connections, see the Cabling section of Chapter 3.

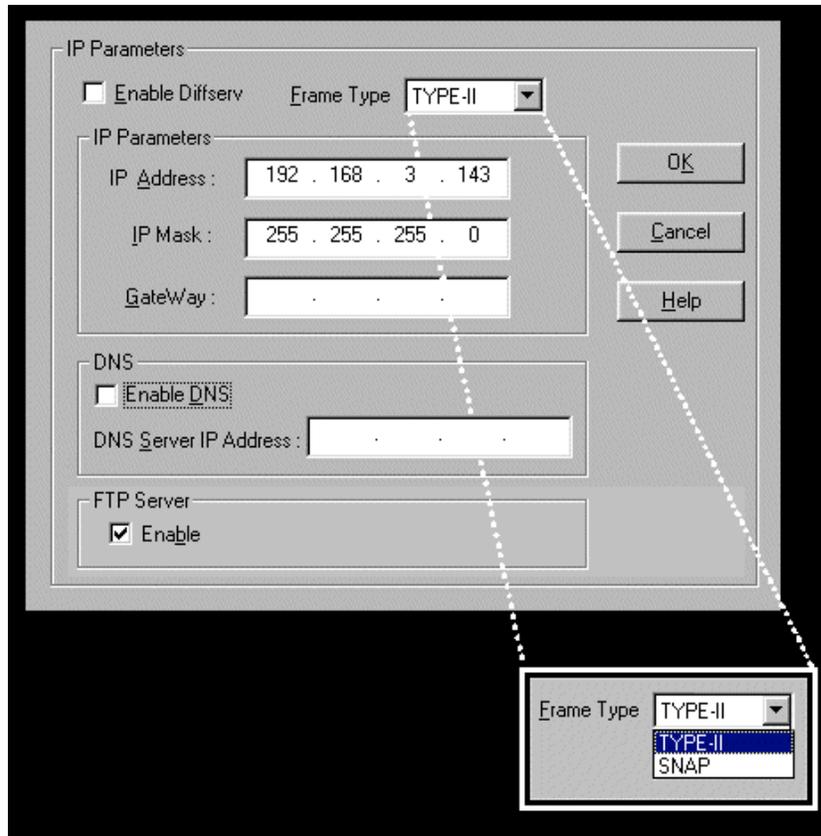
5. Configuration Parameter Groups: Getting Familiar, Learning About Access. The first part of configuration concerns IP parameters, Voice/FAX parameters, Telephony Interface parameters, SNMP parameters, Regional parameters, SMTP parameters, Supplementary Services parameters, Logs, and System Information.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

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



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

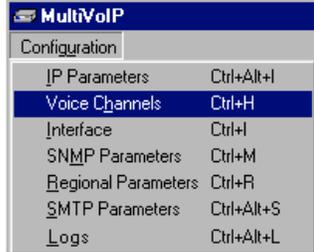
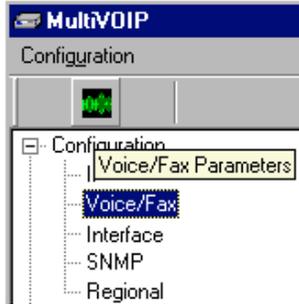
IP Parameter Definitions		
Field Name	Values	Description
Enable Diffserv	Y/N	Diffserv is used for QoS (quality of service). When enabled, the TOS (Type of Service) bits in the IP header are configured so that routers supporting Diffserv can give priority to the VOIP's IP packets. Disabled by default.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

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

- A. Set IP address of MultiVOIP unit using the MultiVOIP Configuration program (the Windows GUI).
- B. Save Setup in Windows GUI.
- C. Close Windows GUI.
- D. Install Java program from MultiVOIP product CD (on first use only).
- E. Open web browser.
- F. Browse to IP address of MultiVOIP unit.
- G. If username and password have been established, enter them when 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
<p>Ctrl + H</p>	

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

Voice/Fax Parameters

Select Channel Channel 1

Voice Gain
 Input 0 dB Output 0 dB

Dtmf
 Gain
 High -4 dB Low -7 dB

Duration 100 ms

DTMF : Out Of Band (Fixed Duration)

Coder
 Manual Automatic
 Selected Coder G.723.1 @ 6.3 kbps
 Max bandwidth 10 kbps

Auto Call
 Auto Call Enable
 Phone Number

Dynamic Jitter Buffer
 Minimum Jitter Value 150 ms
 Maximum Jitter Value 300 ms
 Optimization Factor 7

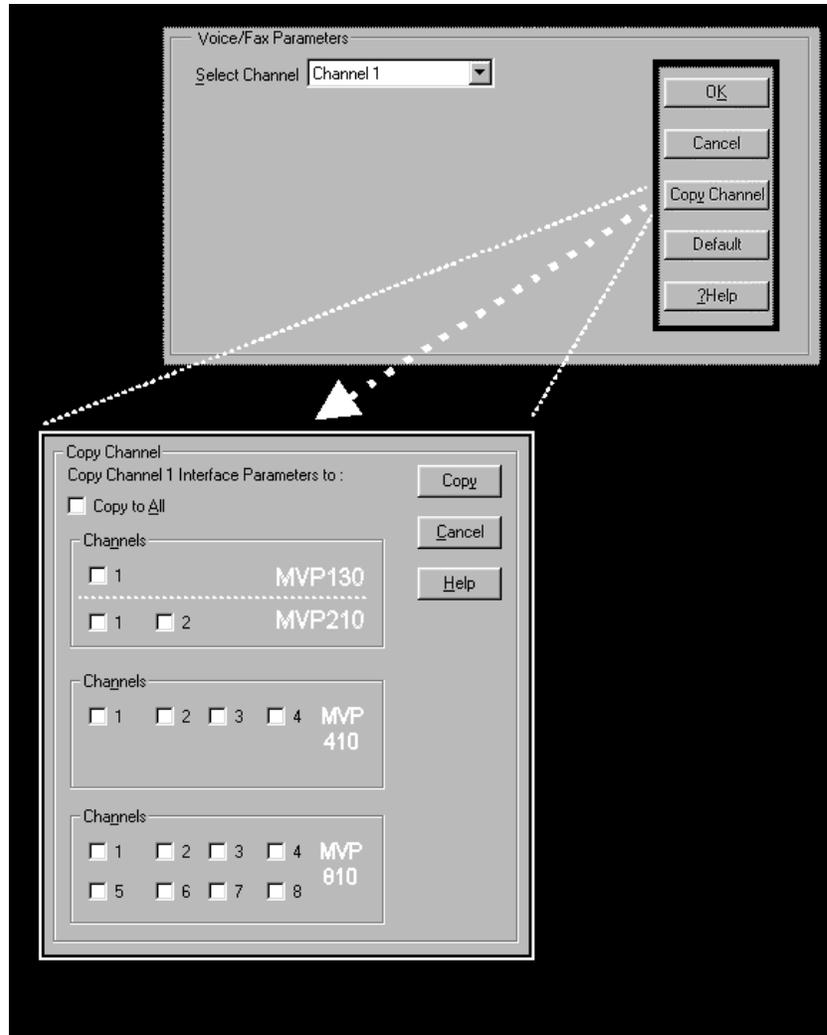
Automatic Disconnection
 Jitter Value 150 ms Consecutive Packets Lost 30
 Call Duration 180 seconds Network Disconnection 300 seconds

Fax
 Fax Enable
 Max Baud Rate 14400
 Fax Volume -9.5 dB
 Jitter Value 400 ms
 Mode FRF 11

Advanced Features
 Silence Compression
 Echo Cancellation
 Forward Error Correction

OK
 Cancel
 Copy Channel
 Default
 Help

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



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

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

Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
DTMF Parameters		
Duration (DTMF)	60 – 3000 ms	When DTMF: Out of Band is selected, this setting determines how long each DTMF digit ‘sounds’ or is held. Default = 100 ms. Not supported in 5.02c BRI software.
DTMF In/Out of Band	Out of Band, or Inband	When DTMF Out of Band is selected, the MultiVOIP detects DTMF tones at its input and regenerates them at its output. When DTMF Inband is selected, the DTMF digits are passed through the MultiVOIP unit as they are received. In 502c BRI software, “DTMF Out of Band” can be checked or unchecked.
FAX Parameters		
Fax Enable	Y/N	Enables or disables fax capability for a particular channel.
Max Baud Rate (Fax)	2400, 4800, 7200, 9600, 12000, 14400 bps	Set to match baud rate of fax machine connected to channel (see Fax machine’s user manual). Default = 14400 bps.
Fax Volume (Default = -9.5 dB)	-18.5 dB to -3.5 dB	Controls output level of fax tones. To be changed only under the direction of Multi-Tech’s Technical Support.
Jitter Value (Fax)	Default = 400 ms	Defines the inter-arrival packet deviation (in milliseconds) for the fax transmission. A higher value will increase the delay, allowing a higher percentage of packets to be reassembled. A lower value will decrease the delay allowing fewer packets to be reassembled.
Mode (Fax)	FRF 11; T.38 (T.38 not currently supported)	FRF11 is frame-relay FAX standard using these coders: G.711, G.728, G.729, G.723.1. T.38 is an ITU-T standard for storing and forwarding FAXes via email using X.25 packets. It uses T.30 fax standards and includes special provisions to preclude FAX timeouts during IP transmissions.

Voice/Fax Parameter Definitions (cont'd)		
Coder Parameters		
Coder	Manual or Auto-matic	Determines whether selection of coder is manual or automatic. When Automatic is selected, the local and remote voice channels will negotiate the voice coder to be used by selecting the highest bandwidth coder supported by both sides without exceeding the Max Bandwidth setting. G.723, G.729, or G.711 are negotiated.
Selected Coder	G.711 a/u law 64 kbps; G.726 , @ 16/24/32/40 kbps; G.727 , @ nine bps rates; G.723.1 @ 5.3 kbps, 6.3 kbps; G.729 , 8kbps; Net Coder @ 6.4, 7.2, 8, 8.8, 9.6 kbps	Select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling must have the same voice coder selected. Default = G.723.1 @ 6.3 kbps, as required for H.323. Here 64K of digital voice are compressed to 6.3K, allowing several simultaneous conversations over the same bandwidth that would otherwise carry only one. To make selections from the Selected Coder drop-down list, the Manual option must be enabled.
Max bandwidth (coder)	11 – 128 kbps	This drop-down list enables you to select the maximum bandwidth allowed for this channel. The Max Bandwidth drop-down list is enabled only if the Coder is set to Automatic. If coder is to be selected automatically (“Auto” setting), then enter a value for maximum bandwidth.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
Advanced Features		
Silence Compression	Y/N	Determines whether silence compression is enabled (checked) for this voice channel. With Silence Compression enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. Default = on.
Echo Cancellation	Y/N	Determines whether echo cancellation is enabled (checked) for this voice channel. Echo Cancellation removes echo and improves sound quality. Default = on.
Forward Error Correction	Y/N	Determines whether forward error correction is enabled (checked) for this voice channel. Forward Error Correction enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off
Auto Call Enable	Y/N	The Auto Call option enables the local MultiVOIP to call a remote MultiVOIP without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote MultiVOIP identified in the Phone Number box of this option.
Phone No. (Auto Call)	--	Phone number used for Auto Call function. A corresponding phone number must be listed in the Outbound Phonebook.

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

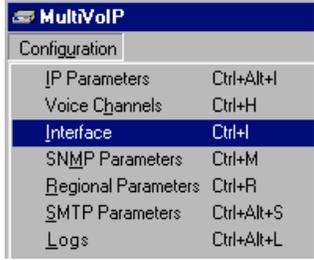
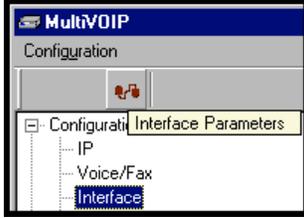
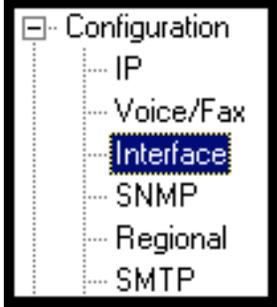
Voice/Fax Parameter Definitions (cont'd)		
Field Name	Values	Description
Dynamic Jitter		
Maximum Jitter Value	60 to 400 ms	The maximum dynamic jitter buffer of 400 milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 msec
Optimization Factor	0 to 12	The Optimization Factor determines how quickly the length of the Dynamic Jitter Buffer is changed based on actual jitter encountered on the network. Selecting the minimum value of 0 means low voice delay is desired, but increases the possibility of jitter-induced voice quality problems. Selecting the maximum value of 12 means highest voice quality under jitter conditions is desired at the cost of increased voice delay. Default = 7.

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

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

Accessing Telephony Interface Parameters	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + I</p>	

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

The screenshot shows a configuration window titled "Interface" with a dropdown menu set to "Channel 1". The window is divided into several sections:

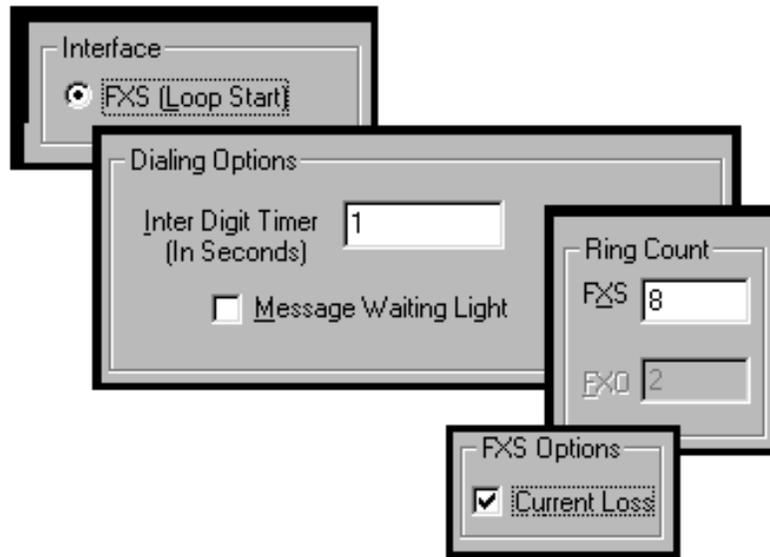
- Interface:** Radio buttons for "FXS (Loop Start)", "FXS (Ground Start)", "FXD", "E&M", and "Disable".
- Dialing Options:** Includes "Regeneration" with "Pulse" and "DTMF" options, "Inter Digit Timer (In Seconds)" set to 1, "Flash Hook Timer (in ms)" set to 600, "Message Waiting Light" checkbox, and "Inter Digit Regeneration Time" set to 100.
- E&M Options:** Includes "Signal" with "Dial Tone" and "Wink" options, "Wink Timer (in ms)" set to 250, "Type" dropdown set to "TYPE 1 2 Wire", and a "Pass Through" checkbox.
- FXD Disconnect On:** Includes checkboxes for "Current Loss" and "Tone Detection", "Silence Detection" dropdown set to "Two Way", "Disconnect Tone Sequence" with "'A'" and "None" options, "Silence Timer (In Seconds)" set to 15, and "Current Loss Detect Timer (in ms)" set to 500.
- Ring Count:** Includes "FXS" set to 8 and "FXD" set to 2.
- FXS Options:** Includes a "Current Loss" checkbox.
- Disconnect On Call Progress Tone:** Includes an "Enable" checkbox.

Buttons for "OK", "Cancel", "Copy Channel", "Default", and "?Help" are located on the right side of the window.

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

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

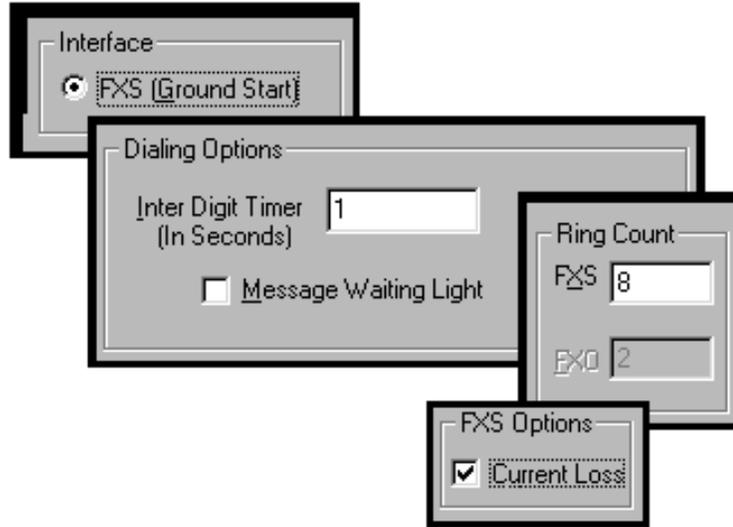
FXS Loop Start Parameters. The parameters applicable to FXS Loop Start are shown in the figure below and described in the table that follows.



FXS Loop Start Interface: Parameter Definitions		
Field Name	Values	Description
FXS Loop Start	Y/N	Enables FXS Loop Start interface type.
Inter Digit Timer	integer values in seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the phonebook for the number entered. Default = 2.

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

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



FXS Ground Start Interface: Parameter Definitions

Field Name	Values	Description
FXS Ground Start	Y/N	Enables FXS Loop Start interface type.
Inter Digit Timer	integer values in seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the phonebook for the number entered. Default = 2.

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

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

The image shows a configuration window for an FXO interface. The window is divided into several sections:

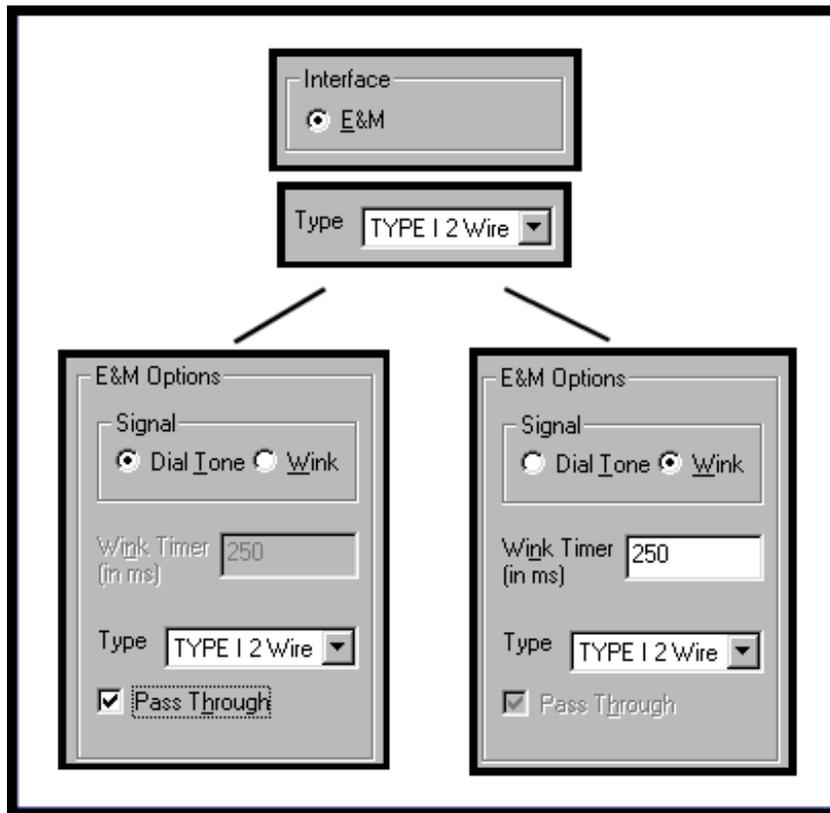
- Interface:** A radio button labeled "FXO" is selected.
- Dialing Options:**
 - Regeneration:** Two radio buttons, "Pulse" and "DTMF", are present. "DTMF" is selected.
 - Inter Digit Timer (In Seconds):** A text box containing the value "1".
 - Flash Hook Timer (in ms):** A text box containing the value "600".
 - Message Waiting Light:** An unchecked checkbox.
 - Inter Digit Regeneration Time:** A text box containing the value "100".
- FXO Disconnect On:**
 - Current Loss:** A checked checkbox.
 - Tone Detection:** A checked checkbox.
 - Silence Detection:** A dropdown menu with "Two Way" selected.
 - Disconnect Tone Sequence:** Two dropdown menus, the first containing "'A'" and the second containing "None", separated by a plus sign.
 - Silence Timer (In Seconds):** A text box containing the value "15".
 - Current Loss Detect Timer (in ms):** A text box containing the value "500".
- Ring Count:** Two text boxes, one labeled "FXS" containing "8" and one labeled "FXO" containing "2".
- Disconnect On Call Progress Tone:** An unchecked checkbox labeled "Enable".

FXO Interface: Parameter Definitions		
Field Name	Values	Description
Interface, FXO	Y/N	Enables FXO functionality
Dialing Options		
Regeneration	Pulse, DTMF	Determines whether digits generated and sent out will be pulse tones or DTMF.
Inter Digit Timer	integer values, in seconds	This is the length of time that the MultiVOIP will wait between digits. When the time expires, the MultiVOIP will look in the phonebook for the number entered. Default = 2.
Flash Hook Timer	integer values, in milliseconds	Length of flash hook that will be generated and sent out when the remote end initiates a flash hook and it is regenerated locally. Default = 600 ms.
Message Waiting Light	Y/N	Applicable only when MultiVOIP is used with Avaya Magix PBX units equipped with Merlin Messaging Centralized mail. When enabled, the Message Waiting Light feature allows the PBX to send mode-codes and message-waiting indications to another Avaya Magix PBX, which in turn will turn on the message waiting light on a phone station. It also allows Direct Inward Dialing, such that no additional dial tone is needed on voip call.

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

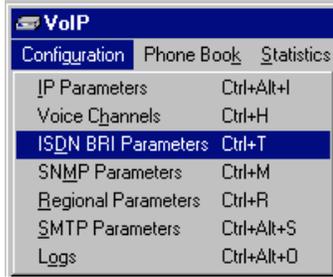
FXO Interface: Parameter Definitions (cont'd)																																					
Field Name	Values	Description																																			
FXO Disconnect On (cont'd)																																					
Disconnect Tone Sequence	1 st tone pair + 2 nd tone pair	These are DTMF tone pairs. Values for first tone pair are: *, #, 0, 1-9, and A-D. Values for second tone pair are: none, 0, 1-9, A-D, *, and #. The tone pairs 1-9, 0, *, and # are the standard DTMF pairs found on phone sets. The tone pairs A-D are "extended DTMF" tones, which are used for various PBX functions.																																			
	<table border="1" style="width: 100%; border-collapse: collapse; text-align: center;"> <thead> <tr> <th colspan="5">DTMF Tone Pairs</th> </tr> <tr> <th colspan="4"></th> <th>Low Tones</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>2</td> <td>3</td> <td>A</td> <td>697Hz</td> </tr> <tr> <td>4</td> <td>5</td> <td>6</td> <td>B</td> <td>770Hz</td> </tr> <tr> <td>7</td> <td>8</td> <td>9</td> <td>C</td> <td>852Hz</td> </tr> <tr> <td>*</td> <td>0</td> <td>#</td> <td>D</td> <td>941Hz</td> </tr> <tr> <td>High Tones</td> <td>1209Hz</td> <td>1336Hz</td> <td>1447Hz</td> <td>1633Hz</td> </tr> </tbody> </table>		DTMF Tone Pairs									Low Tones	1	2	3	A	697Hz	4	5	6	B	770Hz	7	8	9	C	852Hz	*	0	#	D	941Hz	High Tones	1209Hz	1336Hz	1447Hz	1633Hz
DTMF Tone Pairs																																					
				Low Tones																																	
1	2	3	A	697Hz																																	
4	5	6	B	770Hz																																	
7	8	9	C	852Hz																																	
*	0	#	D	941Hz																																	
High Tones	1209Hz	1336Hz	1447Hz	1633Hz																																	
Silence Detection	One-Way or Two-Way	Disconnection to be triggered by silence in one direction only or in both directions simultaneously.																																			
Silence Timer in seconds	integer value	Duration of silence required to trigger disconnection.																																			
Disconnect on Call Progress Tone	Y/N	Allows call on FXO port to be disconnected when a PBX issues a call-progress tone denoting that the phone station on the PBX that has been involved in the call has been hung up.																																			
Ring Count, FXO	integer value	Number of rings required before the MultiVOIP answers the incoming call.																																			

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

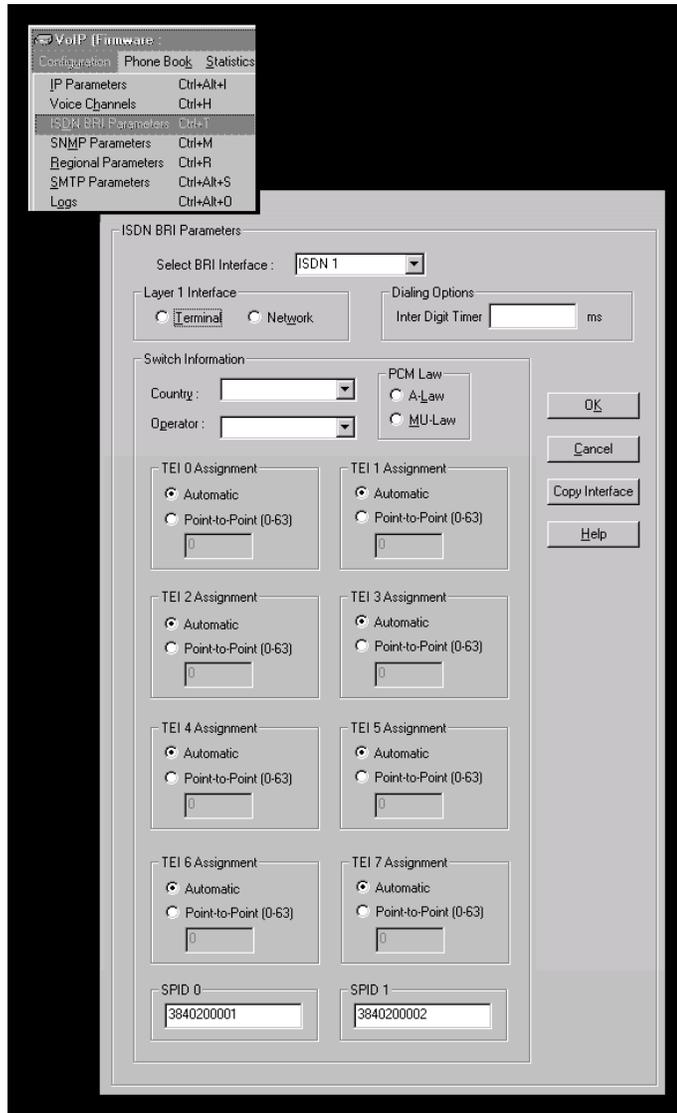


E&M Interface Parameter Definitions		
Field Name	Values	Description
Interface	E&M	enables E&M functionality
Type	Types 1-5. Each type can be 2-wire or 4-wire.	Refers to the type of E&M interface being used.
Signal	Dial Tone or Wink	When Dial Tone is selected, no wink is required on the E lead or M lead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

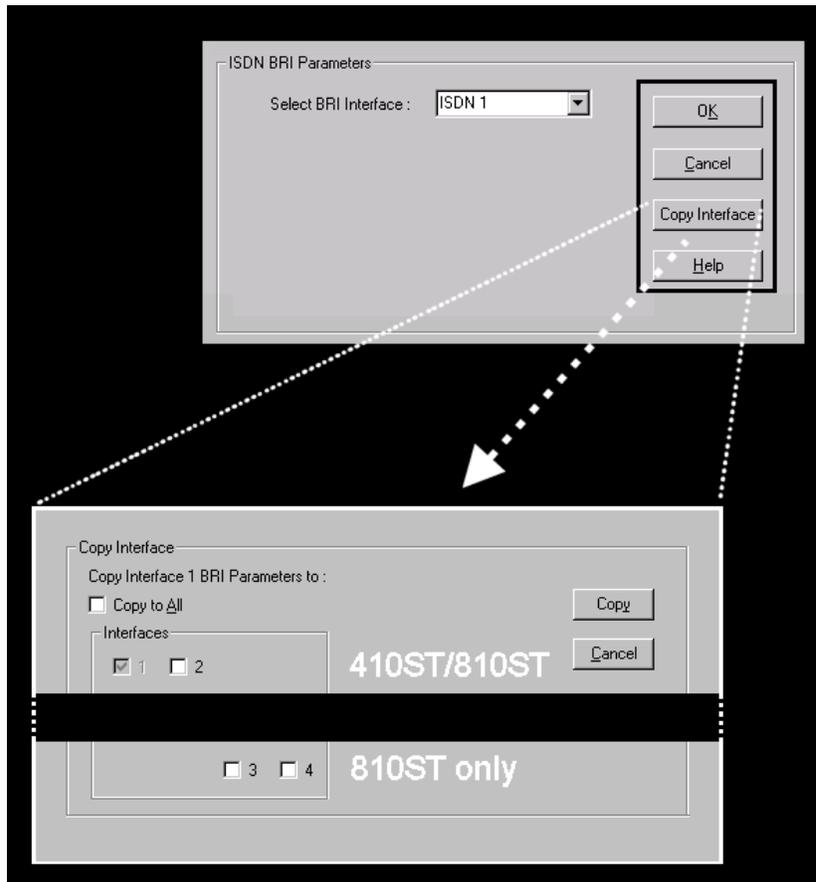
9b. (for ISDN-BRI MultiVOIP units). Set **ISDN Parameters**. This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

Accessing ISDN (BRI) Parameters	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + T</p>	

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



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



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

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

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

ISDN Parameters

Select BRI Interface :

Terminal Network

Country :

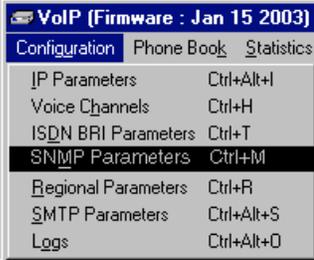
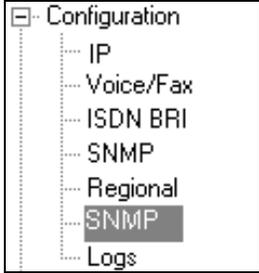
Operator :

PCM Law

A-Law MU-Law

Country :	Operator :
<input type="text"/>	<input type="text"/>
Australia	ETSI--A-law AUSTEL_1--A-law
Europe	ETSI--A-law ECMA_QSIG--A-law FT_VN6--A-law
France	FT_VN6--A-law
Hong Kong	HK_TEL A/mu, switch depndnt default = mu-law
Italy	ETSI--A-law
Japan	NTT--mu-law KDD--mu-law
Korea	KOREAN_OP A/mu, switch depndnt default = mu-law
USA	N_ISDN1--mu-law N_ISDN2--mu-law ATT_5E10--mu-law NT_DMS100--mu-law

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.

Accessing “SNMP Parameters”	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + M</p>	

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

SNMP Parameters

Enable SNMP Agent

Trap Manager

Address : 0 . 0 . 0 . 0

Community Name :

Port Number : 162

Community Name - 1 : public

Permissions : Read Only

Community Name - 1 : public

Permissions : Read/Write

Read Only

Read/Write

OK

Cancel

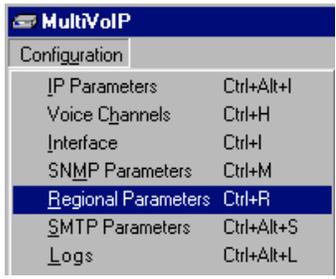
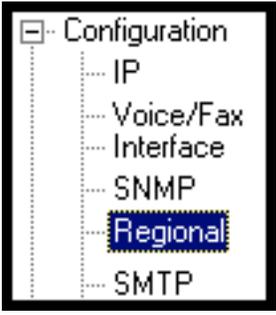
Help

The SNMP Parameter fields are described in the table below.

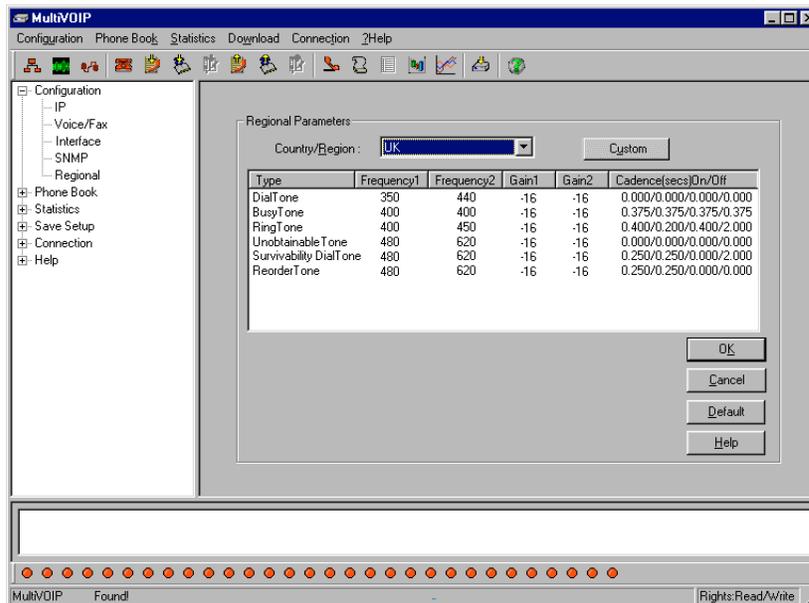
SNMP Parameter Definitions		
Field Name	Values	Description
Enable SNMP Agent	Y/N	Enables the SNMP code in the firmware of the MultiVOIP. This must be enabled for the MultiVOIP to communicate with and be controllable by the MultiVoipManager software. Default: disabled
Trap Manager Parameters		
Address	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
<p>Ctrl + R</p>	

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



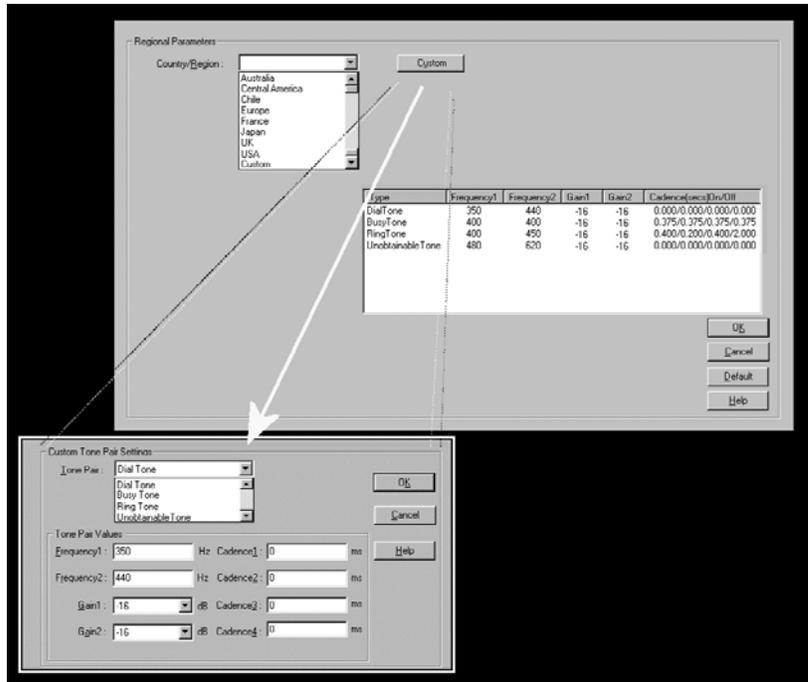
In each field, enter the values that fit your particular system.

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

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

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

12. **Set Custom Tones and Cadences** (optional). The **Regional Parameters** dialog box has a secondary dialog box that allows you to customize DTMF tone pairs to create unique ring-tones/dial-tones, busy-tones or “unobtainable” tones (fast busy signal) or “re-order” tones (telling the user that she must hang up an off-hook phone) or “survivability” tones (an indication of call-routing redundancy in MultiVantage systems only) for your system. This screen allows the user to specify tone-pair attributes that are not found in any of the standard national/regional telephony toning schemes. To access this customization feature, click on the **Custom** button on the **Regional Parameters** screen. (The “Custom” button is active only when “Custom” is selected in the **Country/Region** field.)

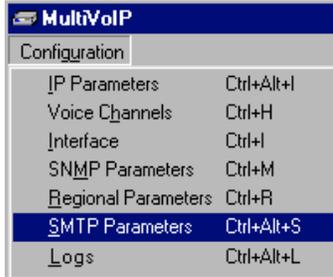
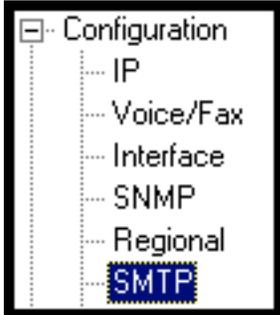


The **Custom Tone-Pair Settings** fields are described in the table below.

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

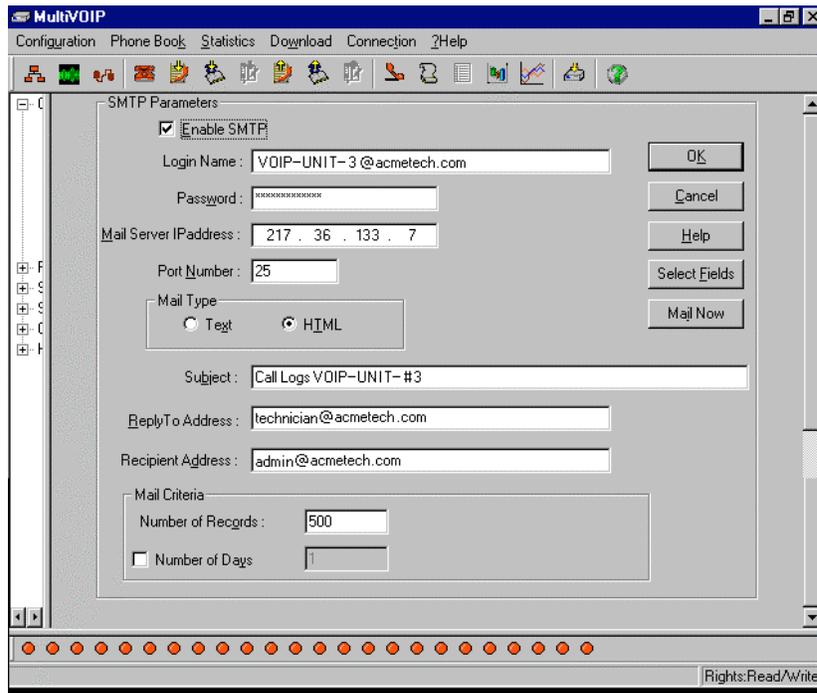
Custom Tone-Pair Settings Definitions		
Field Name	Values	Description
Cadence 1	integer time value in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, dial tone ("0" indicates continuous tone) survivability and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

13. **Set SMTP Parameters** (Log Reports by Email). The **SMTP Parameters** screen is applicable when the VOIP administrator has chosen to receive log reports by email (this is done by selecting the “SMTP” checkbox in the **Others** screen and selecting “Enable SMTP” in the **SMTP Parameters** screen.). The **SMTP Parameters** screen can be reached by pulldown menu, keyboard shortcut, or sidebar.

Accessing “SMTP Parameters”	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + Alt + S</p>	

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

The SMTP Parameters screen is shown below.

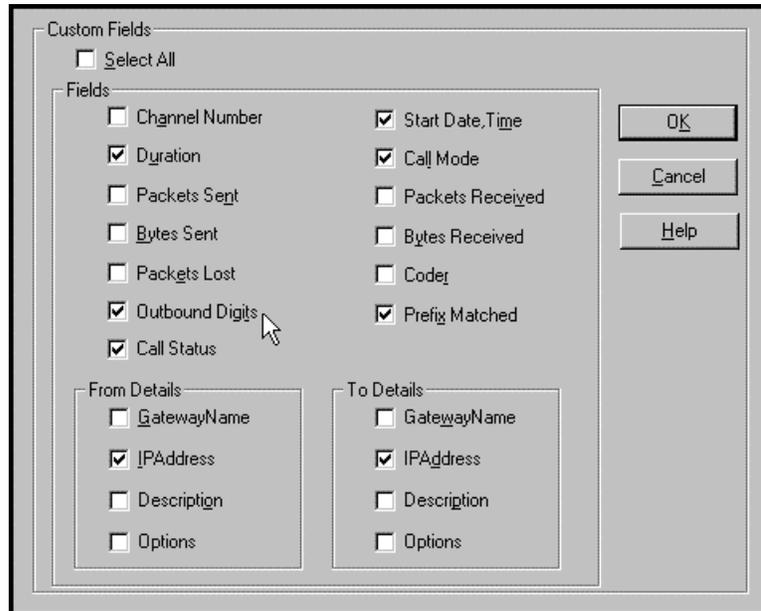


"SMTP Parameters" Definitions		
Field Name	Values	Description
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select "SMTP" in the Logs screen.
Login Name	alpha-numeric, per email domain	This is the User Name for the MultiVOIP unit's email account.
Password	alpha-numeric	Login password for MultiVOIP unit's email account.
Mail Server IP Address	n.n.n.n for n= 0 to 255	This is the mail server's IP address. This mail server must be accessible on the IP network to which the MultiVOIP is connected.
Port Number	25	25 is a standard port number for SMTP.

.....

“SMTP Parameters” Definitions (cont’d)		
Field Name	Values	Description
Mail Type	text or html	Mail type in which log reports will be sent.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP unit.
Reply-To Address	email address	User specified. This email address functions as a source email identifier for the MultiVOIP, which, of course, cannot usefully receive email messages. The Reply-To address provides a destination for returned messages indicating the status of messages sent by the MultiVOIP (esp. to indicate when log report email was undeliverable or when an error has occurred).
Recipient Address	email address	User specified. Email address at which VOIP administrator will receive log reports.
Mail Criteria		Criteria for sending log summary by email. The log summary email will be sent out either when the user-specified number of log messages has accumulated, or once every day or multiple days, <i>which ever comes first</i> .
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.

SMTP

SMTP Parameters

Enable SMTP

Login Name:

Password:

Mail Server IP Address:

Port Number:

Mail Type: Text HTML

Subject:

ReplyTo Address:

Recipient Address:

Mail Criteria:

Number of Records:

Number of Days:

Logs

Enable Console Messages

Turn Off Logs

Reporting Method:

GUI SMTP SNMP

Custom Fields

Select All

Fields:

Channel Number Start Date, Time

Duration Call Mode

Packets Sent Packets Received

Packets Lost Bytes Received

Outbound Digits Prefix Matched

Call Status

From Details: GatewayName IP Address Description Options

To Details: GatewayName IP Address Description Options

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

SMTP Parameters

To use the "SMTP Parameters" screen, SMTP must first be selected in the "Logs" screen.

Custom Fields

The secondary screen "Custom Fields" lets the user determine which technical details of each phone call to record in the logs.

SMTP Parameters

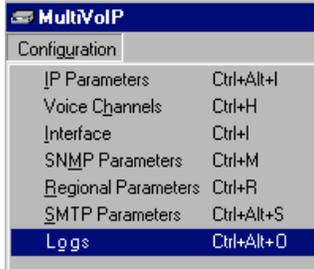
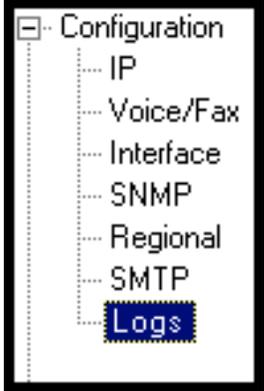
Email reports can be sent out periodically or when a certain number of phone calls have been logged.

Call Logs

The requested technical details will then appear in the log report that is automatically emailed to the VoIP administrator.

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
 <p>MultiVoIP Configuration IP Parameters Ctrl+Alt+I Voice Channels Ctrl+H Interface Ctrl+I SNMP Parameters Ctrl+M Regional Parameters Ctrl+R SMTP Parameters Ctrl+Alt+S Logs Ctrl+Alt+O</p>	
Shortcut	Sidebar
<p>Ctrl + Alt + O</p>	 <p>Configuration IP Voice/Fax Interface SNMP Regional SMTP Logs</p>

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).

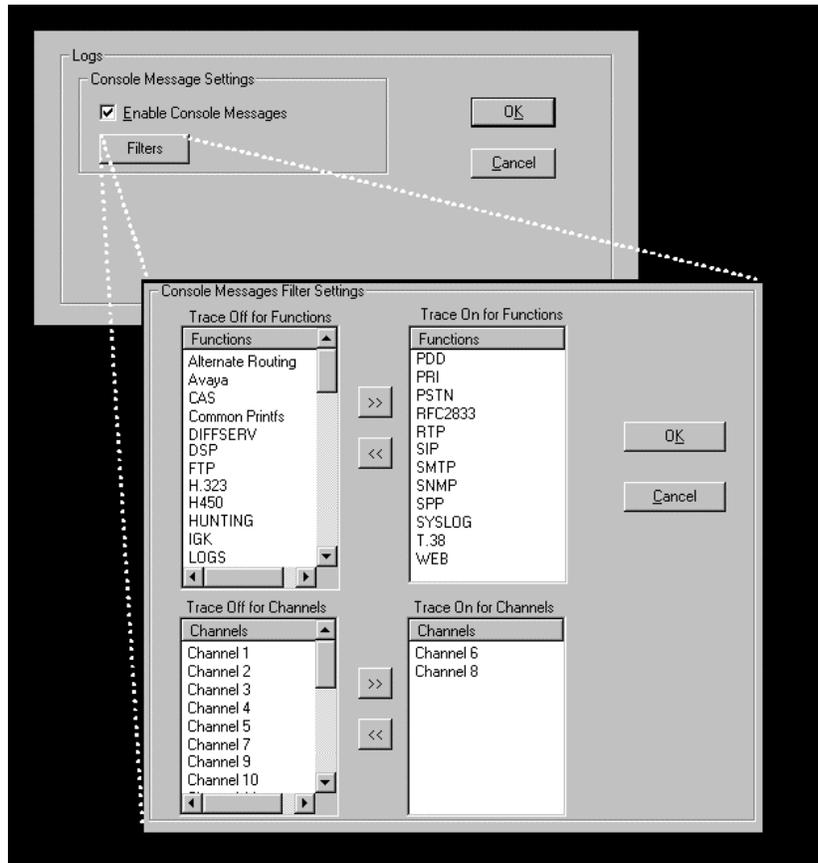
The screenshot shows a configuration window titled "Logs" with the following sections:

- Console Message Settings:**
 - Enable Console Messages
 - Filters
- Logs:**
 - Turn Off Logs
 - GUI
 - SMTP
 - SNMP
- SysLog Server:**
 - Enable
 - IP Address : []
 - Port : [514]
- Online Statistics Update Interval:** [5] Sec

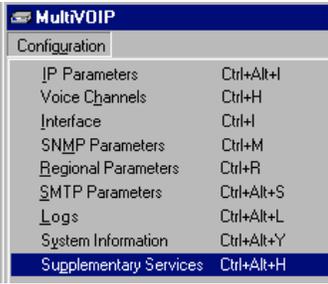
Buttons on the right side include OK, Cancel, and Help.

“Logs” Screen Definitions		
Field Name	Values	Description
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic terminal program like HyperTerminal™ or equivalent. Normally, this should be disabled because it uses MultiVOIP processing resources. Console messages are meant for tech support personnel.
Filters (button)		Click to access secondary screen on where console messages can be included/excluded by category and on a per-channel basis. (See the Console Messages Filter Settings screen on subsequent page.) Not supported in BRI 5.02c software.
Turn Off Logs	Y/N	Check to disable log-reporting function. Not supported in BRI 5.02c software.
Logs Buttons		Only one of these three log reporting methods, GUI, SMTP, or SNMP, may be chosen.
GUI	Y/N	User must view logs at the MultiVOIP configuration program.
SNMP	Y/N	Log messages will be delivered to the MultiVoipManager application program.
SMTP	Y/N	Log messages will be sent to user-specified email address.
SysLog Server Enable	Y/N	This box must be checked if logging is to be done in conjunction with a SysLog Server program. For more on SysLog Server, see <i>Operation & Maintenance</i> chapter. Not supported in BRI 5.02c software.
IP Address	n.n.n.n for n= 0-255	IP address of computer, connected to voip network, on which SysLog Server program is running. Not supported in BRI 5.02c software.
Port	514	Logical port for SysLog Server. 514 is commonly used. Not supported in BRI 5.02c software.
Online Statistics Updation Interval	integer	Set the interval (in seconds) at which logging information will be updated. Not supported in BRI 5.02c software.

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



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

Accessing “Supplementary Services” Parameters	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + Alt + H</p>	

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

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

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

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

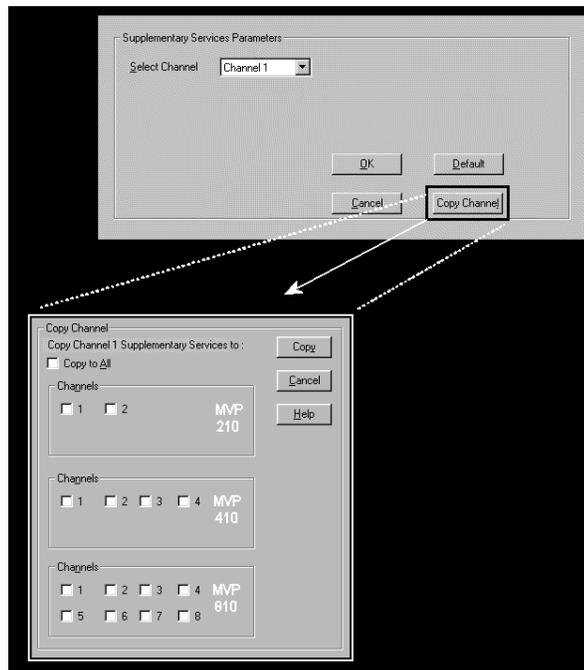
Call Hold. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function. Invoked by keypad sequence.

Call Waiting. Call Waiting notifies an engaged caller of an incoming call and allows them to receive a call from a third party while the party with whom they have been speaking is put on hold. Invoked by keypad sequence.

Call Name Identification. When enabled for a given voip unit (the 'home' voip), this feature gives notice to remote voips involved in calls. Notification goes to the remote voip administrator, not to individual phone stations. When the home voip is the caller, a plain English descriptor will be sent to the remote (callee) voip identifying

the channel over which the call is being originated (for example, “Calling Party - Omaha Sales Office Line 2”). If that voip channel is dedicated to a certain individual, the descriptor could say that, as well (for example “Calling Party - Harold Smith in Omaha”). When the home voip receives a call from any remote voip, the home voip sends a status message back to that caller. This message confirms that the home voip’s phone channel is either busy or ringing or that a connection has been made (for example, “Busy Party - Omaha Sales Office Line 2”). These messages appear in the **Statistics – Call Progress** screen of the remote voip.

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



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

Supplementary Services Parameter Definitions (Not supported in BRI 5.02c software.)		
Field Name	Values	Description
Select Channel	1-2 (210); 1-4 (410); 1-8 (810)	The channel to be configured is selected here.
Call Transfer Enable	Y/N	Select to enable the Call Transfer function in the voip unit. This is a “blind” transfer and the sequence of events is as follows: Callers A and B are having a conversation. Caller A wants to put B into contact with C. Caller A dials call transfer sequence. Caller A hears dial tone and dials number for caller C. Caller A gets disconnected while Caller B gets connected to caller C.
Transfer Sequence	any phone keypad character	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call transfer. The call-transfer sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). The sequences for call transfer, call hold, and call waiting can be from 1 to 4 digits in length consisting of any combination of digits 1234567890*#.

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Call Hold Enable	Y/N	Select to enable Call Hold function in voip unit. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function.
Hold Sequence	phone keypad characters	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call hold. The call-hold sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #).
Call Waiting Enable	Y/N	Select to enable Call Waiting function in voip unit.
Retrieve Sequence	phone keypad characters, two characters in length	The numbers and/or symbols that the caller must press on the phone keypad to initiate retrieval of a waiting call. The call-waiting retrieval sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). This is the phone keypad sequence that a user must press to retrieve a waiting call. Customize-able. Sequence should be distinct from sequence that might be used to retrieve a waiting call via the PBX or PSTN.

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Call Name Identification Enable		<p>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).</p> <p>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.</p> <p>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.)</p>

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Calling Party, Allowed Name Type (CNI)		<p>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.</p> <p>This field is applicable only when the 'home' voip unit is originating the call.</p> <p>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.</p> <p>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.</p>

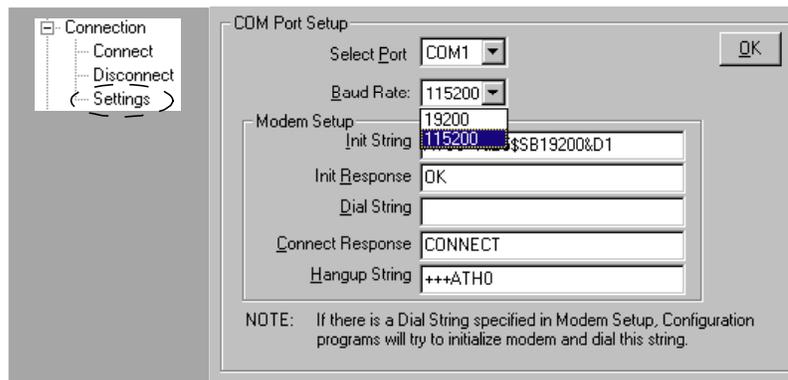
Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Alerting Party, Allowed Name Type (CNI)		<p>If the 'home' voip unit is receiving the call and Alerting Party is selected, then the identifier (from the Caller Id field) will tell the originating remote voip unit that the call is ringing.</p> <p>This field is applicable only when the 'home' voip unit is receiving the call.</p> <p>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.</p> <p>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.</p>

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Busy Party, Allowed Name Type (CNI)		<p>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.</p> <p>This field is applicable only when the 'home' voip unit is receiving the call.</p> <p>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.</p> <p>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.</p>

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Connected Party, Allowed Name Type (CNI)		<p>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.</p> <p>This field is applicable only when the 'home' voip unit is receiving the call.</p> <p>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.</p> <p>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.</p>

Supplementary Services Definitions (cont'd)		
Field Name	Values	Description
Caller ID		This is the identifier of a specific channel of the 'home' voip unit. The Caller Id field typically describes a person, office, or location, for example, "Harry Smith," or "Bursar's Office," or "Barnesville Factory."
Default	--	When this button is clicked, all Supplementary Service parameters are set to their default values.
Copy Channel	--	Copies the Supplementary Service attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.

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

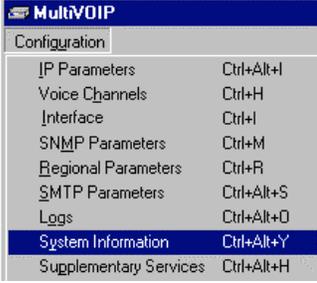
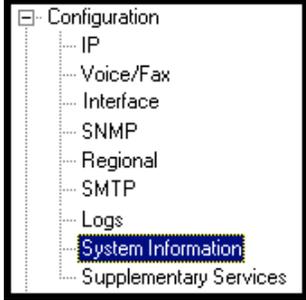


First, it is important to note that the default COM port established by the MultiVOIP program is COM1. ***Do not accept the default value until you have checked the COM port allocation on your PC.*** To do this, check for COM port assignments in the system resource dialog box(es) of your Windows operating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

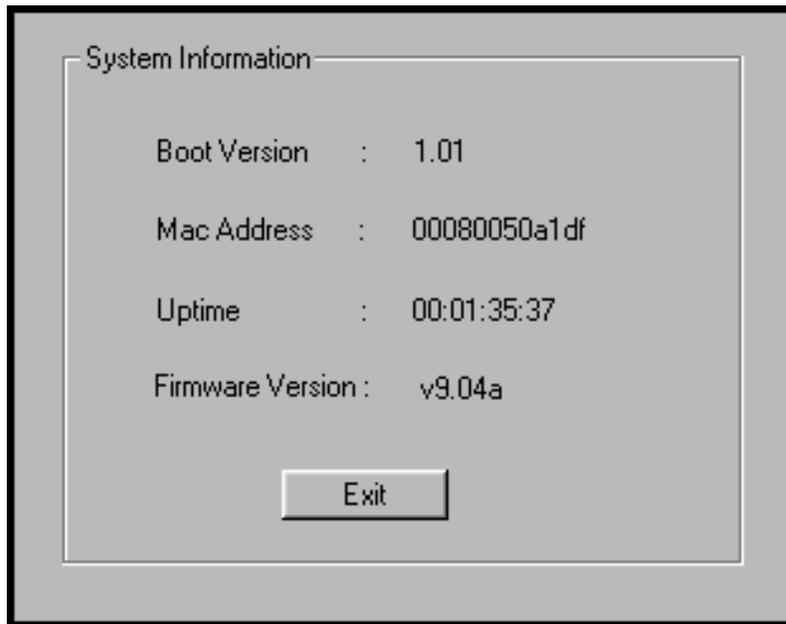
The default baud rate is 115,200 bps.

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

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

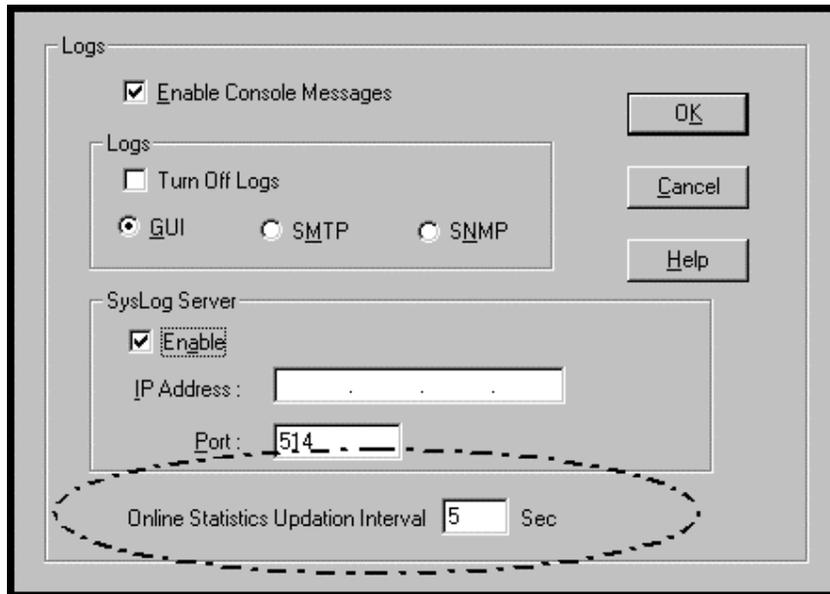
Accessing "System Information" Screen	
Pulldown	Icon
	
Shortcut	Sidebar
<p>Ctrl + Alt + Y</p>	

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

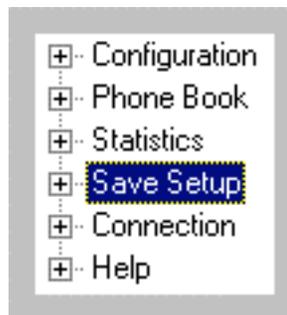


System Information Parameter Definitions		
Field Name	Values	Description
Boot Code Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Mac Address	alpha-numeric	Denotes the number assigned as the voip unit's unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.
Firmware Version	alpha-numeric	Indicates the version of the MultiVOIP firmware.

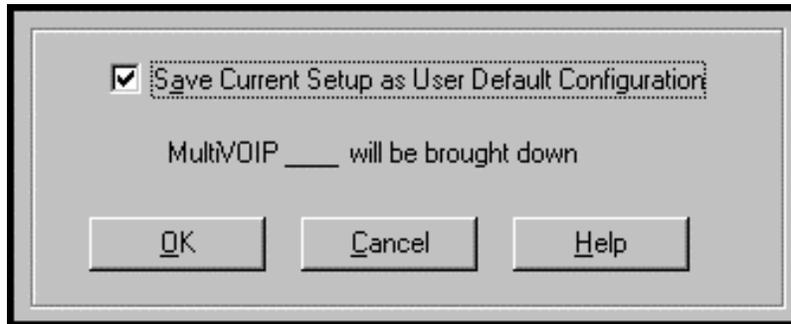
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



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



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



Chapter 7: T1 Phonebook Configuration

(North American Telephony Standards)

Configuring the MVP2400/2410 MultiVOIP Phonebooks

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

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

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

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

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

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

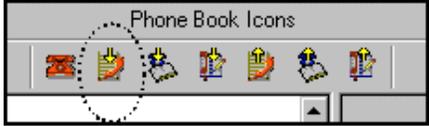
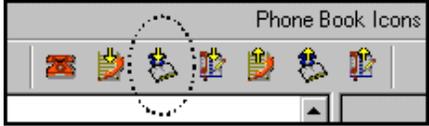
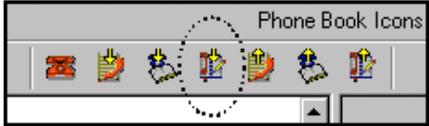
Consider two types of calls in the three-city system described above: (1) calls originating from the Miami office and terminating in the New York (Manhattan) office, and (2) calls originating from the Miami office and terminating in New York City but off the company's premises in an adjacent area code,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

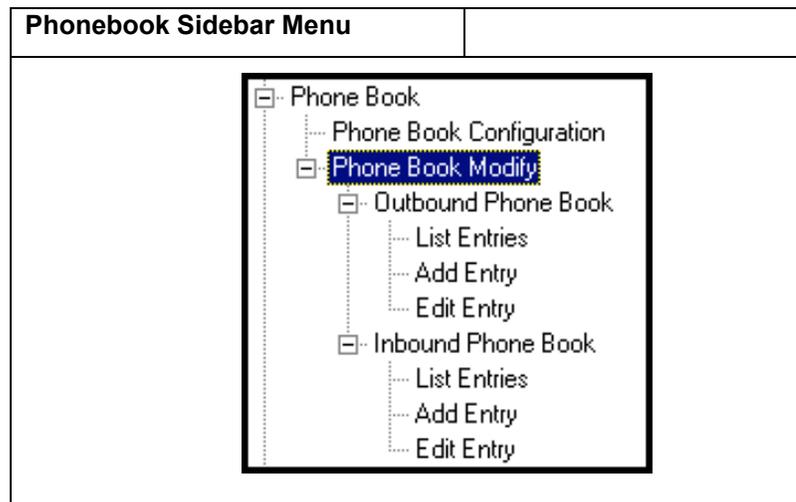
The first type of call requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound phonebook of the New York VOIP. These entries would allow the Miami caller to dial the New York office as if its phones were extensions on the Miami PBX.

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

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

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

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



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

Phone Book Configuration

Gateway Name :

Q.931 Parameters

Use Fast Start

Call Signaling Port :

Register with GateKeeper

Gatekeeper RAS Parameters

Gatekeeper/Clear Channel IP Address :

Port Number :

Gateway Prefix :

Gatekeeper Name :

Gateway H323 ID :

Enable SIP Proxy

SIP Proxy Parameters

Proxy Server IP Address :

Port Number :

UserName :

Password :

H323 Version 4 Options

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

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

SPP Protocol

Mode :

Direct
Client
Registrar

General Options

Port :

Retransmission (in ms) :

Max Retransmission :

Client Options

Registrar IP Address :

Registrar Port :

Registrar Options

Keep Alive (in sec) :

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

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

PhoneBook Configuration Parameter Definitions		
Field Name	Values	Description
Gateway Name	Y/N	This field allows you to specify a name for this MultiVOIP. When placing a call, this name is sent to the remote MultiVOIP for display in Call Progress listings, Logs, etc.
Q.931 Parameters		
Use Fast Start	Y/N	Enables the H.323 Fast Start procedure. May need to be enabled/disabled for compatibility with third-party VOIP gateways.
Call Signaling Port	port number	Default: 1720 (H.323)
GateKeeper RAS Parameters		
Gatekeeper / Clear Channel IP Address		IP address of the GateKeeper.
Port Number		Well-known port number for GateKeepers. Must match port number of GateKeeper, 1719.
Gateway Prefix		This number becomes registered with the GateKeeper. Call requests sent to the gatekeeper and preceded by this prefix will be routed to the VOIP gateway.
Gatekeeper Name	<i>alpha-numeric string</i>	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register.
Gateway H.323 ID		The H.323 ID is used to register this particular MultiVOIP with the GateKeeper.

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	Values	Description
SIP Proxy Parameters		
Enable Proxy	Y/N	Allows the MultiVOIP to work in conjunction with a proxy server.
Proxy Server IP Address	n.n.n.n where n=0-255	Network address of the proxy server that the voip is using.
Port Number		Logical port number for proxy communications.
User Name	Values: 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.	
Password	Values: alphanumeric Description: Password for proxy server function. See "User Name" description above.	

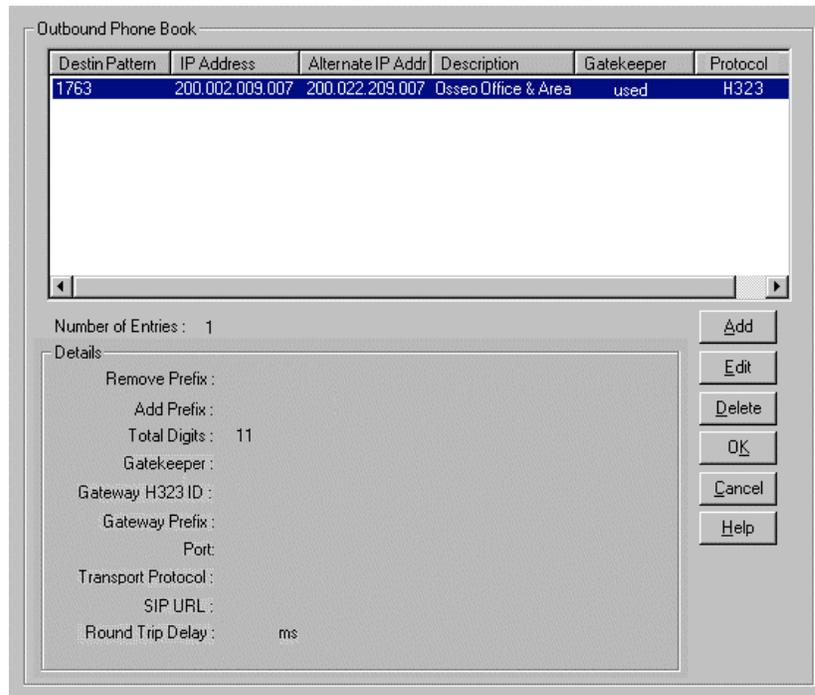
PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	Values	Description
H.323 Version 4 Parameters		
Q.931 Multiplexing (Mux)	Y/N	Signaling for multiple phone calls can be carried on a single port rather than opening a separate signaling port for each call. This conserves bandwidth resources.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

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

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	Values	Description
Single Port Protocol (SPP)		
Mode	Direct, Client, or Registrar	SPP voip systems can operate in two modes: in the direct mode , where all voip gateways have static IP addresses assigned to them; or in the registrar/client mode , where one voip gateway serves as registrar and all other gateways, being its clients, point to that registrar. The registrar assigns IP addresses dynamically.
General Options		
Port		The UDP port on which data transmission will occur. Each client voip has its own port. If two client voips are both behind the same firewall, then they must have different ports assigned to them. If there are two clients and each is behind a different firewall, then the clients could have different port numbers or the same port number. (Default port number = 10000.)
Re-transmission (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-transmission		Number of times the voip will retransmit a lost packet (if no acknowledgment has been received). (Default value = 3)

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	Values	Description
Single Port Protocol (SPP) [continued]		
Client Options		Client Option fields are active only in registrar/client mode and only for client voip units.
Registrar IP Address		This is the IP address of the registrar voip to which this client is assigned. (Default value = 0.0.0.0; effectively, there is no useful default value.)
Registrar Port		This is the port number of the registrar voip to which this client is assigned. (Default port number = 10000.)
Registrar Options		Registrar Option fields are active only in registrar/client mode and only for registrar voip units.
Keep Alive (in sec.)		Time-out duration before a registrar will unregister a client that does not send its "I'm here" signal. Client normally sends its "I'm here" signal every 20 seconds. Timeout default = 60 seconds.

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



Click **Add**.

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

The screenshot shows the MultiVOIP 2400 configuration interface. The left sidebar contains a tree view with the following items: Configuration, IP, Voice/Fax, T1/E1, SNMP, Phone Book, Phone Book Configuration, Phone Book Modify, Outbound Phone Book (selected), List Entries, Add Entry (highlighted), Edit Entry, Inbound Phone Book, Statistics, Save Setup, Connection, and Help. The main window title is 'MultiVOIP 2400' and the menu bar includes Configuration, Phone Book, Statistics, Download, Connection, and Help. The 'Add/Edit Outbound Phone Book' dialog is open, featuring the following fields and controls:

- Phone Number Details:**
 - Destination Pattern: [Text Field]
 - Total Digits: [0] [Spin Box]
 - Remove Prefix: [Text Field]
 - Add Prefix: [Text Field]
 - Buttons: OK, Cancel, Help
- IP Address:** [Text Field] [Advanced]
- Description:** [Text Field]
- Protocol Type:** Radio buttons for SIP, H.323 (selected), and SPP.
- H.323:**
 - Use Gatekeeper: [Checkbox]
 - Gateway H323 ID: [Text Field]
 - Gateway Prefig: [Text Field]
 - 931 Port Number: [1720] [Text Field]
- SIP:**
 - Use Proxy: [Checkbox]
 - Transport Protocol: Radio buttons for TCP (selected) and UDP.
 - SIP Port Number: [5060] [Text Field]
 - SIP URL: [Text Field]
- SPP Protocol:**
 - Use Registrar: [Checkbox]
 - Port Number: [10000] [Text Field]
 - Alternate Phone Number: [Text Field]
- MultiVoIP 110/120/200/400/800

Enter Outbound PhoneBook data for your MVP2400/2410. Note that the Advanced button gives access to the Alternate IP Routing feature, if needed. Alternate IP Routing can be implemented in a secondary screen (as described after the primary screen field definitions below).

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

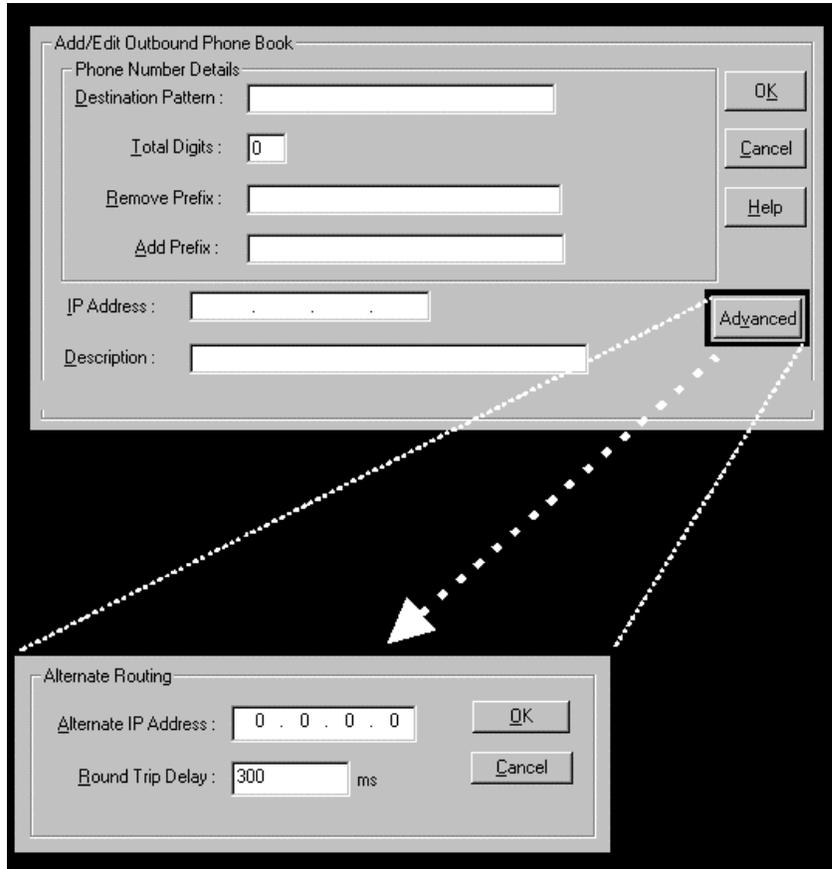
Add/Edit Outbound Phone Book: Field Definitions		
Field Name	Values	Description
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	number of digits the phone user must dial to reach specified destination
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination
Add Prefix	dialed digits	digits to be added before completing call to destination
IP Address	n.n.n.n for n = 0-255	the IP address to which the call will be directed if it begins with the destination pattern given
Description	alpha-numeric	Describes the facility or geographical location at which the call will be completed.
Protocol Type	SIP or H.323 or SPP	Indicates protocol to be used in outbound transmission. Single Port Protocol (SPP) is a non-standard protocol designed by Multi-Tech.

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

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	Values	Description
SIP Fields		
Use Proxy	Y/N	Select if proxy server is used.
Transport Protocol	TCP or UDP	Voip administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other *See RFC3087 ("Control of Service Context using SIP Request-URI," by the Network Working Group).	The SIP Port Number is a UDP logical port number. The voip will "listen" for SIP messages at this logical port. If SIP is used, 5060 is the default, standard, or "well known" port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	<i>sip.userphone@hostserver</i> , 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.

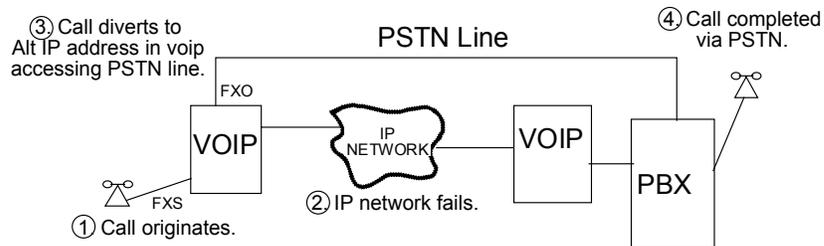
Add/Edit Outbound Phone Book: Field Def'ns (cont'd)		
Field Name	Values	Description
SPP Fields		
Use Registrar	Values: Y/N	Description: Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set in Phonebook Configuration screen) has a static IP address and all other voips (clients) point to the registrar's IP address as functionally their own. However, if your voip system overall is operating in "Registrar/Client" mode but you want to make an exception and use Direct mode for the destination pattern of this particular Add/Edit Phonebook entry, leave this checkbox unselected. Leave this checkbox unselected if your overall voip system is operating in the "Direct" SPP mode. In this mode, all voips in system are peers and each has its own static IP address.
Port Number	Values: numeric	Description: When operating in "Registrar/Client" mode, this is the port by which the gateway receives all SPP data and control messages from the registrar gateway. (This ability to receive all data and messages via one port allows the voip to operate behind a firewall with only one port open.) When operating in "Direct" mode, this is the Port by which peer voips receive data and messages.
Alternate Phone Number	numeric	Phone number associated with alternate IP routing.
MultiVOIP 110/120/200/400/800	Values: Y/N	Description: Select if any gateways of these model types are included in voip system and are operating in H.323 mode.
Advanced button	Values: N/A	Description: Gives access to secondary screen where an Alternate IP Route can be specified for backup or redundancy of signal paths. See discussion on next page. For SIP & H.323 operation only.

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



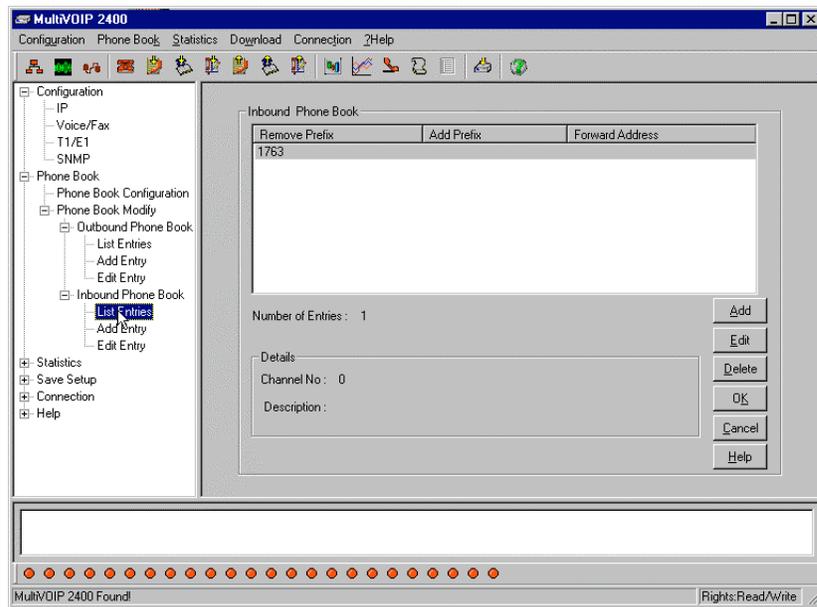
Alternate Routing Field Definitions		
Field Name	Values	Description
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when a data pathway is considered blocked. When the delay exceeds the threshold specified here, the data stream will be diverted to the alternate destination specified as the Alternate IP Address.

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



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

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



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

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

Add/Edit Inbound Phone Book: Field Definitions		
Field Name	Values	Description
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)
Channel Number	1-24, or "Hunting"	T1 channel number to which the call will be assigned as it enters the local telephony equipment (often a local PBX). "Hunting" directs the call to any available channel.

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	Values	Description
Description	--	Describes the facility or geographical location at which the call originated.
Call Forward Parameters		
Enable	Y/N	Click the check-box to enable the call-forwarding feature.
Forward Condition	Uncondit.; Busy No Resp.	Unconditional. When selected, all calls received will be forwarded. Busy. When selected, calls will be forwarded when station is busy. No Response. When selected, calls will be forwarded if called party does not answer after a specified number of rings, as specified in Ring Count field.
Forward Address/ Number	IP addr. or phone number	Phone number or IP address to which calls will be directed.
Ring Count	integer	When No Response is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.

6. When your Outbound and Inbound PhoneBook entries are completed, click on **Save Setup** in the sidebar menu to save your configuration.

You can change your configuration at any time as needed for your system.

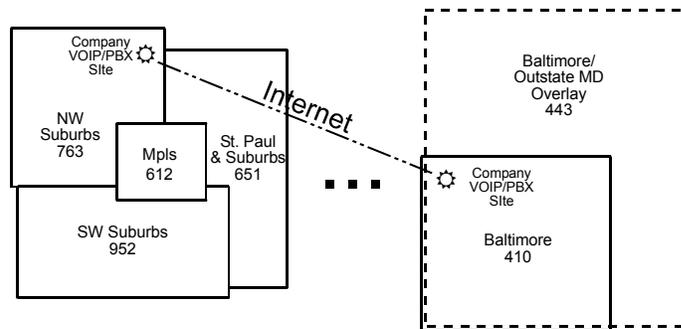
Remember that the initial MVP2400/2410 setup must be done locally using the MultiVOIP program. However, after the initial configuration is complete, all of the MVP2400/2410 units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVoipManager software program.

T1 Phonebook Examples

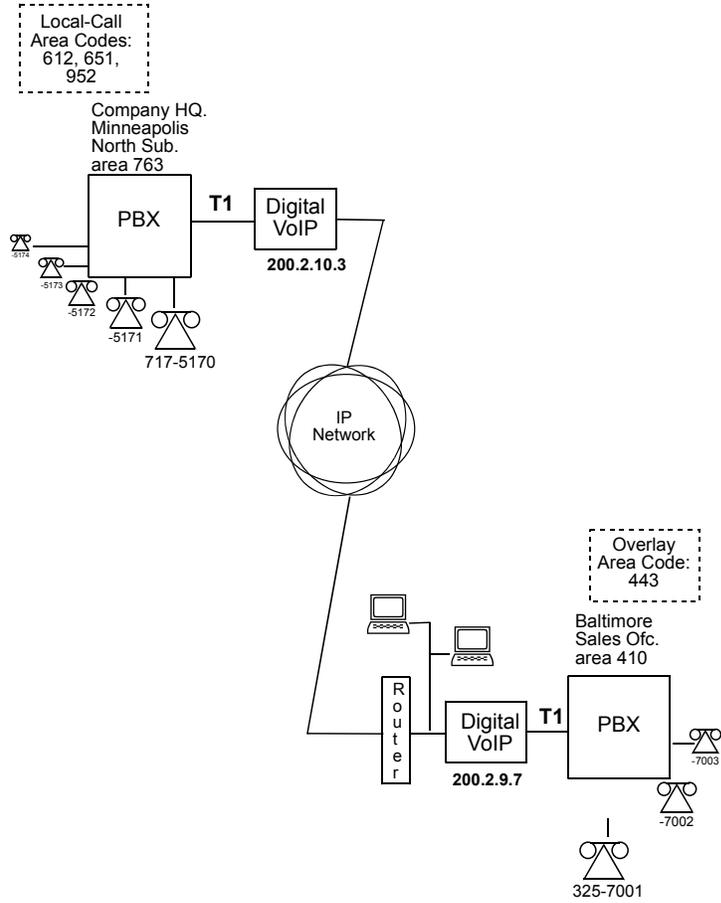
The following example demonstrates how Outbound and Inbound PhoneBook entries work in a situation of multiple area codes. Consider a company with offices in Minneapolis and Baltimore.

3 Sites, All-T1 Example

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



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



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

{ Baltimore voip unit }

Outbound PhoneBook

Dest Pattern	IP Address	Description
1612	200.002.010.003	Minneapolis
1651	200.002.010.003	St Paul
1763	200.002.010.003	Minneapolis, N Suburbs
1952	200.002.010.003	Minneapolis, S Suburbs

Number of Entries : 4

Details

H.323 ID :

Remove Prefix :

Add Prefix :

Total Digits : 11

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

{ Minneapolis voip unit }

Inbound PhoneBook

Rem Prefix	Add Prefix
1612	9,612
1651	9,651
1763	9,
17637175	5
1952	9,952

Number of Entries : 5

Details

Channel No : 0

Description : localcalls to Minneapolis (city)

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).

Rem Prefix	Add Prefix
1410	9
14103257	7
1443	9,443

Number of Entries : 3

Details

Channel No : 0

Description : Baltimore metro

Add Edit Delete Cancel

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”.

Outbound PhoneBook **{ Minneapolis voip unit }**

Dest Pattern	IP Address	Description
1410	200.002.009.007	Baltimore
1443	200.002.009.007	Baltimore overlay
7	200.002.009.007	Baltimore Office Extensions

Number of Entries : 3

Details

H.323 ID :

Remove Prefix :

Add Prefix :

Total Digits : 11

Add

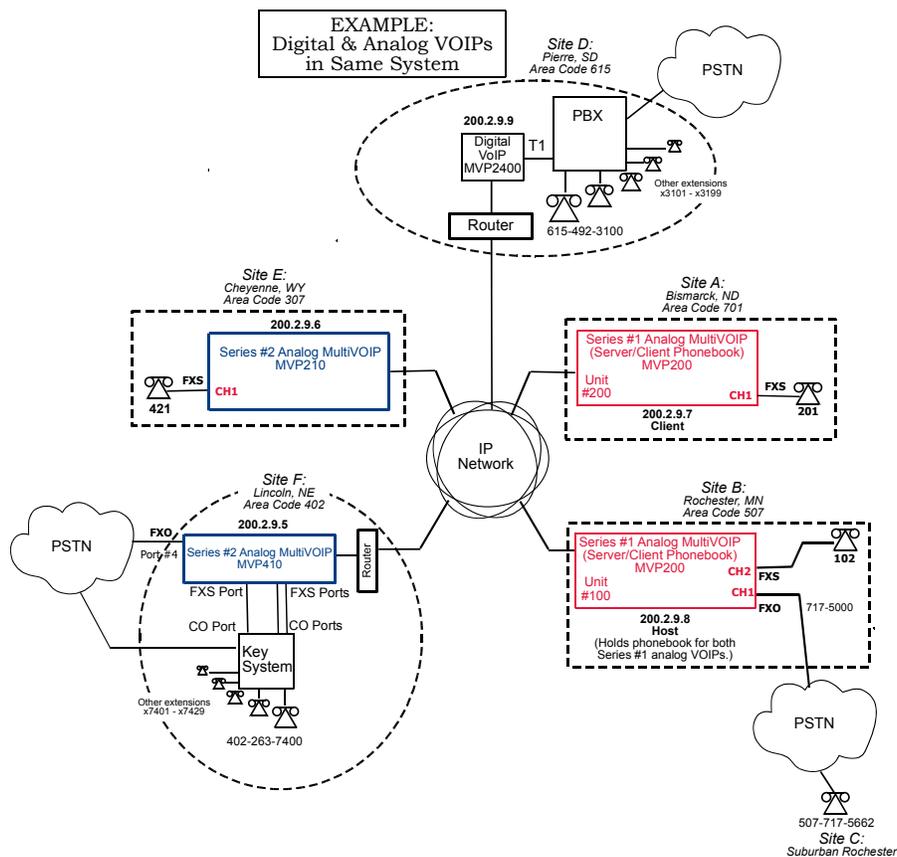
Edit

Delete

Cancel

Configuring Mixed Digital/Analog VOIP Systems

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



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

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

These seven phone books are shown below.

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

- Note 1. The “x” is a wildcard character.
- Note 2. By specifying “Channel 0,” we instruct the MVP2400/2410 to choose any available data channel to carry the call.
- Note 3. Note that Site F key system has only 30 extensions (x7400-7429). This destination pattern (140226374) actually directs calls to 402-263-74**30** through 402-263-74**99** into the key system, as well. This means that such calls, which belong on the PSTN, cannot be completed. In some cases, this might be inconsequential because an entire exchange (fully used or not) might have been reserved for the company or it might be unnecessary to reach those numbers. However, to specify only the 30 lines actually used by the key system, the destination pattern 140226374 would have to be replaced by three other destination patterns, namely 140226374**0**, 140226374**1**, and 140226374**2**. In this way, calls to 402-263-74**30** through 402-263-74**99** would be properly directed to the PSTN. In the Site D outbound phonebook, the 30 lines are defined exactly, that is, without making any adjacent phone numbers unreachable through the voip system.

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

Inbound Phonebook for MVP2400/2410 Digital VOIP (Site D)			
Remove Prefix	Add Prefix	Channel Number	Comment
1615	9, Note 4. Note 5.	0	Allows phone users at remote voip sites to call non-toll numbers within the Site D area code (615; Pierre, SD) over the VOIP network.
1615 49231	31	0	Allows voip calls directly to employees at Site D (at extensions x3101 to x3199).
<p>Note 4. "9" gives PBX station users access to outside line.</p> <p>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.</p>			

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

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

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

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

Call Completion Summaries

Site A calling Site C, Method 1

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

Site A calling Site C, Method 2

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

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

Site C calling Site A

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

Site D calling Site C

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

Site D calling Site F

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

A. The required entry in the Pierre Outbound Phonebook to facilitate origination of the call, would be 1402263742. The call would be directed to the Lincoln voip's IP address, 200.2.9.5.

(Generally on such a call, the caller would have to dial an initial "9." But typically the PBX would not pass the initial "9" to the voip. If the PBX *did* pass along that "9" however, its removal would have to be specified in the local Outbound Phonebook.)

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

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

Call Event Sequence

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

Site F calling Site D

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

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

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

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

Variations in PBX Characteristics

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

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

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

Chapter 8: E1 Phonebook Configuration

(European Telephony Standards)

MVP3010 Inbound and Outbound MultiVOIP Phonebooks

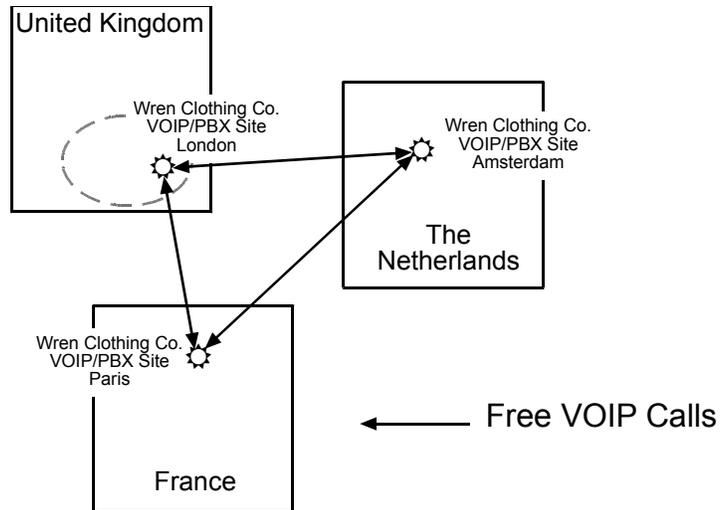
Important Definition:	The MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook describes the dialing sequences that can be used to call that MultiVOIP and how those calls will be directed.
-----------------------	--

When a VOIP serves a PBX system, the operation of the VOIP should be transparent to the telephone end user and savings in long-distance calling charges should be enjoyed. Use of the VOIP should not require the dialing of extra digits to reach users elsewhere on the VOIP network. On the contrary, VOIP service more commonly reduces dialed digits by allowing users (served by PBXs in facilities in distant cities) to dial their co-workers with 3-, 4-, or 5-digit extensions -- as if they were in the same facility. More importantly, the VOIP system should be configured to maximize savings in long-distance calling charges. To achieve both of these objectives, ease of use and maximized savings, the VOIP phonebooks must be set correctly.

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

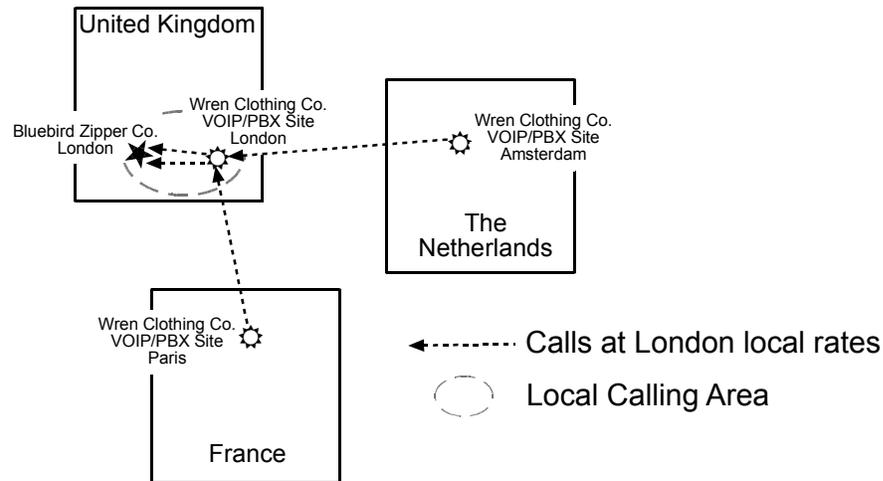
Free Calls: One VOIP Site to Another

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

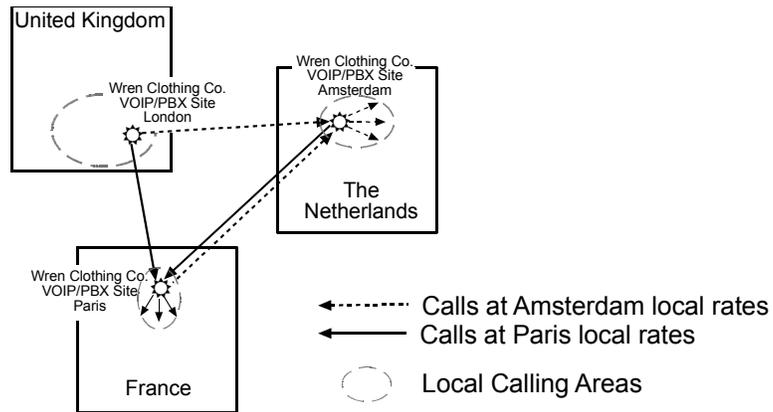


Local Rate Calls: Within Local Calling Area of Remote VOIP

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

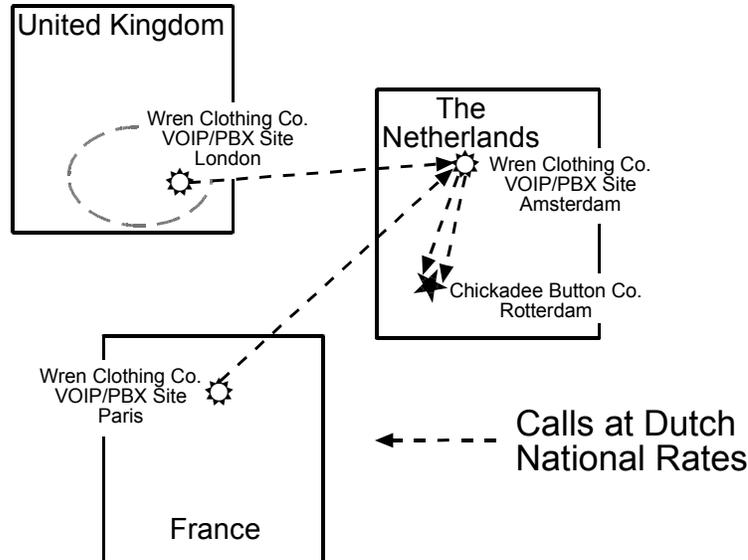


Similarly, the VOIP system allows Wren Clothing employees in London and Amsterdam to call anywhere in Paris at local rates; it allows Wren Clothing employees in Paris and London to call anywhere in Amsterdam at local rates.

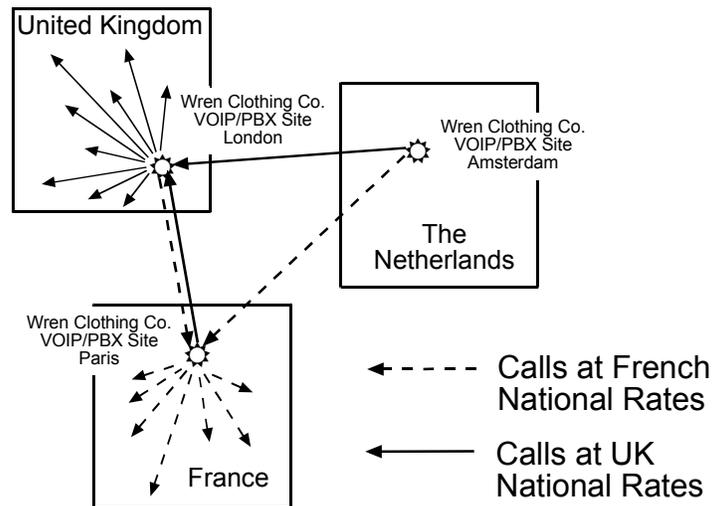


National Rate Calls: Within Nation of Remote VOIP Site

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



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



Inbound versus Outbound Phonebooks

To make the VOIP system transparent to phone users and to allow all possible free and reduced-rate calls, the VOIP administrator must configure the “Outbound” and “Inbound” phone-books of each VoIP in the system.

The “Outbound” phonebook for a particular VOIP unit describes the dialing sequences required for a call to originate locally (typically in a PBX in a particular facility) and reach any of its possible destinations at remote VOIP sites, including calls terminating at points beyond the remote VOIP site.

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

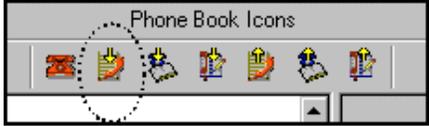
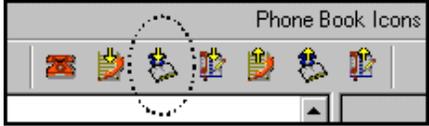
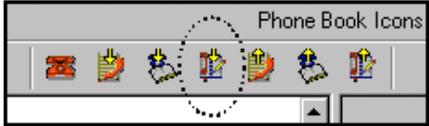
Briefly stated, *the MultiVOIP’s Outbound phonebook lists the phone stations it can call; its Inbound phonebook lists the dialing sequences that can be used to call that MultiVOIP.* (Of course, the phone numbers are not literally “listed” individually.) The phone stations that can originate or complete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

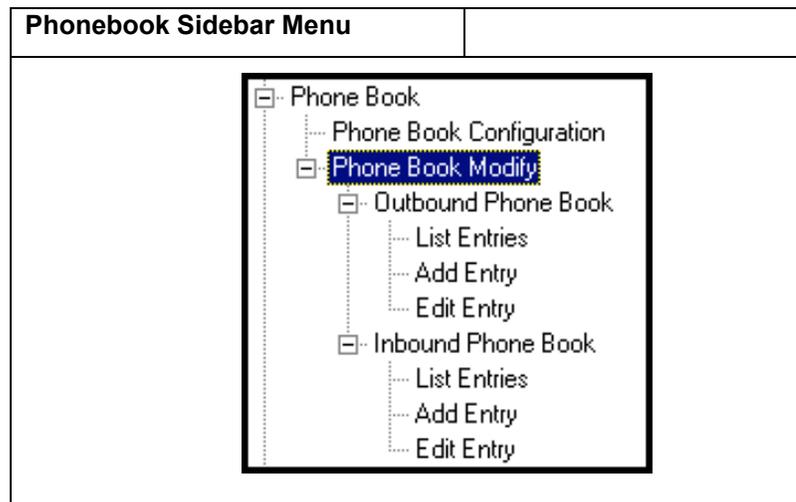
In order for any VOIP phone call to be made, there must be both an Inbound Phonebook entry and an Outbound Phonebook entry that describe the end-to-end connection. The phone station originating the call must be connected to

the VOIP system. The Outbound Phonebook for that VOIP unit must have a destination pattern entry that includes the 'called' phone (that is, the phone completing the call). The Inbound Phonebook of the VOIP where the call is completed must have a destination pattern entry that includes the digit sequence dialed by the originating phone station.

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

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

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



Phonebook Configuration Procedure

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

Phone Book Configuration

Gateway Name :

Q.931 Parameters

Use Fast Start

Call Signaling Port :

Register with GateKeeper

Gatekeeper RAS Parameters

Gatekeeper /Clear Channel IP Address :

Port Number :

Gatekeeper Prefix :

Gatekeeper Name :

Gateway H323 ID :

Enable SIP Proxy

SIP Proxy Parameters

Proxy Server IP Address :

Port Number :

UserName :

Password :

H323 Version 4 Options

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

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

SPP Protocol

Mode : (Dropdown menu: Direct, Client, Registrar)

General Options

Port :

Retransmission (in ms) :

Max Retransmission :

Client Options

Registrar IP Address :

Registrar Port :

Registrar Options

Keep Alive (in sec) :

OK

Cancel

Help

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

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

PhoneBook Configuration Parameter Definitions		
Field Name	Values	Description
Gateway Name	Y/N	This field allows you to specify a name for this MultiVOIP. When placing a call, this name is sent to the remote MVP3000 for display in Call Progress listings, Logs, etc.
Q.931 Parameters		
Use Fast Start	Y/N	Enables the H.323 Fast Start procedure. May need to be enabled/disabled for compatibility with third-party VOIP gateways.
Call Signaling Port	port number	Default: 1720 (H.323)
GateKeeper RAS Parameters		
Gatekeeper / Clear Channel IP Address		IP address of the GateKeeper.
Port Number		Well-known port number for GateKeepers. Must match port number of GateKeeper, 1719.
Gateway Prefix		This number becomes registered with the GateKeeper. Call requests sent to the gatekeeper and preceded by this prefix will be routed to the VOIP gateway.
Gatekeeper Name	<i>alpha-numeric string</i>	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register.
Gateway H.323 ID		The H.323 ID is used to register this particular MultiVOIP with the GateKeeper. H.323 ID is an alias entry sent to the GateKeeper, made of alphanumeric characters. For NetMeeting endpoints, numbers are preferred over letters. The H.323 ID identifies the IP calling sequence that the GateKeeper must 'dial' to contact the remote VOIP.

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

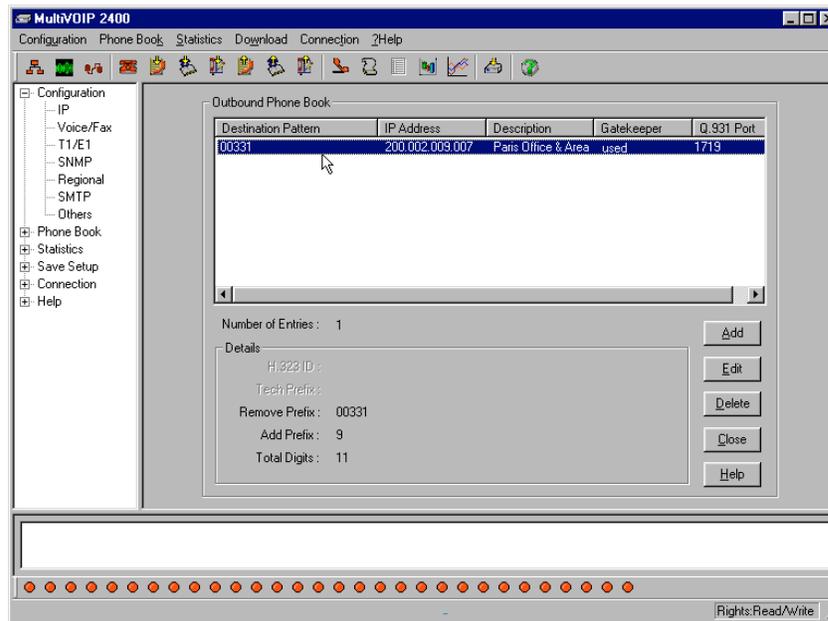
PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	Values	Description
H.323 Version 4 Parameters		
Q.931 Multiplexing (Mux)	Y/N	Signaling for multiple phone calls can be carried on a single port rather than opening a separate signaling port for each call. This conserves bandwidth resources.
H.245 Tunneling (Tun)	Y/N	H.245 messages are encapsulated within the Q.931 call-signaling channel. Among other things, the H.245 messages let the two endpoints tell each other what their technical capabilities are and determine who, during the call, will be the client and who the server. Tunneling is the process of transmitting these H.245 messages through the Q.931 channel. The same TCP/IP socket (or logical port) already being used for the Call Signaling Channel is then also used by the H.245 Control Channel. This encapsulation reduces the number of logical ports (sockets) needed and reduces call setup time.

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

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	Values	Description
Single Port Protocol (SPP)		
Mode	Direct, Client, or Registrar	SPP voip systems can operate in two modes: in the direct mode , where all voip gateways have static IP addresses assigned to them; or in the registrar/client mode , where one voip gateway serves as registrar and all other gateways, being its clients, point to that registrar. The registrar assigns IP addresses dynamically.
General Options		
Port		The UDP port on which data transmission will occur. Each client voip has its own port. If two client voips are both behind the same firewall, then they must have different ports assigned to them. If there are two clients and each is behind a different firewall, then the clients could have different port numbers or the same port number. (Default port number = 10000.)
Re-transmission (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-transmission		Number of times the voip will retransmit a lost packet (if no acknowledgment has been received). (Default value = 3)

PhoneBook Configuration Parameter Definitions (cont'd)		
Field Name	Values	Description
Single Port Protocol (SPP) [cont'd]		
Client Options		Client Option fields are active only in registrar/client mode and only for client voip units.
Registrar IP Address		This is the IP address of the registrar voip to which this client is assigned. (Default value = 0.0.0.0; effectively, there is no useful default value.)
Registrar Port		This is the port number of the registrar voip to which this client is assigned. (Default port number = 10000.)
Registrar Options		Registrar Option fields are active only in registrar/client mode and only for registrar voip units.
Keep Alive (in sec.)		Time-out duration before a registrar will unregister a client that does not send its "I'm here" signal. Client normally sends its "I'm here" signal every 20 seconds. Timeout default = 60 seconds.

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



Click **Add**.

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

The screenshot displays the MultiVOIP 3000 configuration software. The left sidebar shows a tree view with 'Phone Book' expanded to 'Outbound Phone Book', where 'Add Entry' is selected. The main window is titled 'Add/Edit Outbound Phone Book' and contains the following fields and options:

- Phone Number Details:**
 - Destination Pattern: 00334
 - Total Digits: 12
 - Remove Prefix: 00334
 - Add Prefix: 9
- Buttons: OK, Cancel, Help
- IP Address: 200 . 002 . 009 . 007
- Advanced button
- Description: Access to Lyon area
- Protocol Type: SIP, H.323, SPP
- H.323 section:
 - Use Gatekeeper
 - Gateway H323 ID: []
 - Gateway Prefig: []
 - Q.931 Port Number: 1720
- SIP section:
 - Use Proxy
 - Transport Protocol: TCP, UDP
 - SIP Port Number: 5060
 - SIP URL: []
- SPP Protocol section:
 - Use Registrar
 - Port Number: 10000
 - Alternate Phone Number: []
- Checkbox: MultiVoIP 110/120/200/400/800

Enter Outbound PhoneBook data for your MVP3010. Note that the Advanced button gives access to the Alternate IP Routing feature, if needed. Alternate IP Routing can be implemented in a secondary screen (as described after the primary screen field definitions below).

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

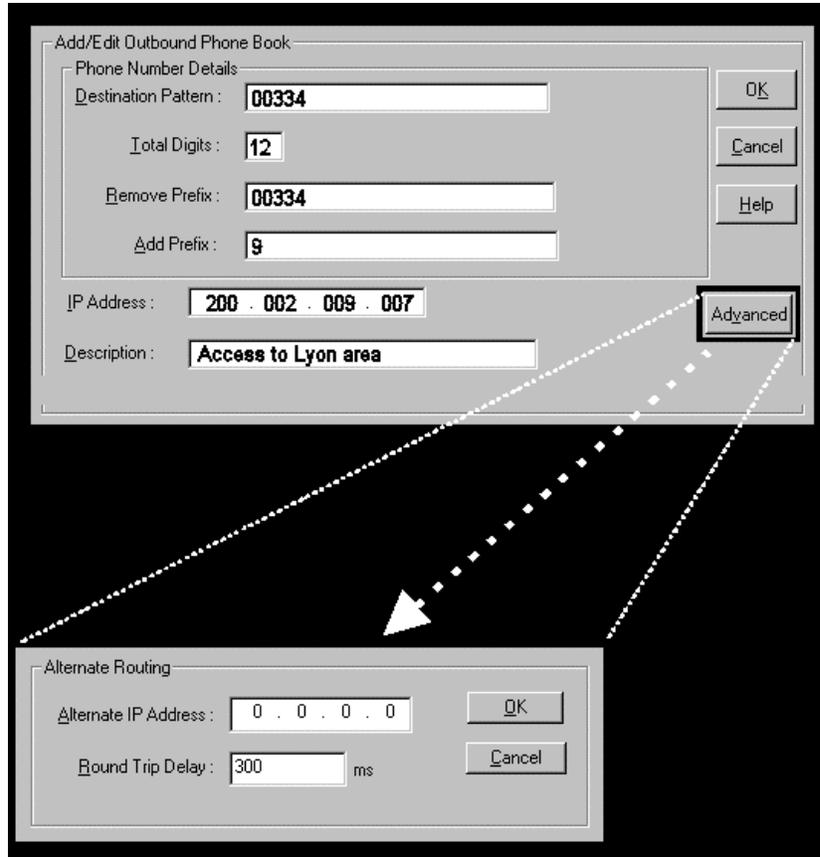
Add/Edit Outbound Phone Book: Field Definitions		
Field Name	Values	Description
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	number of digits the phone user must dial to reach specified destination
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination
Add Prefix	dialed digits	digits to be added before completing call to destination
IP Address	n.n.n.n for = 0-255	the IP address to which the call will be directed if it begins with the destination pattern given
Description	alpha-numeric	Describes the facility or geographical location at which the call will be completed.
Protocol Type	SIP, H.323, or SPP	Indicates protocol to be used in outbound transmission.

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

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	Values	Description
SIP Fields		
Use Proxy	Y/N	Select if proxy server is used.
Transport Protocol	TCP or UDP	Voip administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other *See RFC3087 ("Control of Service Context using SIP Request-URI," by the Network Working Group).	The SIP Port Number is a UDP logical port number. The voip will "listen" for SIP messages at this logical port. If SIP is used, 5060 is the default, standard, or "well known" port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	<i>sip.userphone@hostserver</i> , 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.

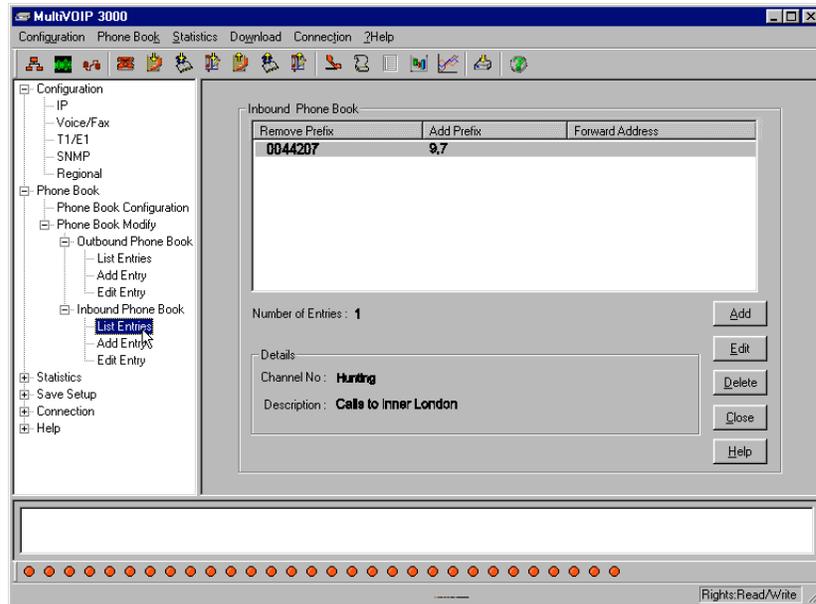
Add/Edit Outbound Phone Book: Field Def'ns (cont'd)		
Field Name	Values	Description
SPP Fields		
Use Registrar	Values: Y/N	Description: Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set in Phonebook Configuration screen) has a static IP address and all other voips (clients) point to the registrar's IP address as functionally their own. However, if your voip system overall is operating in "Registrar/Client" mode but you want to make an exception and use Direct mode for the destination pattern of this particular Add/Edit Phonebook entry, leave this checkbox unselected. Leave this checkbox unselected if your overall voip system is operating in the "Direct" SPP mode. In this mode, all voips in system are peers and each has its own static IP address.
Port Number	Values: numeric	Description: When operating in "Registrar/Client" mode, this is the port by which the gateway receives all SPP data and control messages from the registrar gateway. (This ability to receive all data and messages via one port allows the voip to operate behind a firewall with only one port open.) When operating in "Direct" mode, this is the Port by which peer voips receive data and messages.
Alternate Phone Number	numeric	Phone number associated with alternate IP routing.
MultiVOIP 110/120/200/400/800	Values: Y/N	Description: Select if any gateways of these model types are included in voip system and are operating in H.323 mode.
Advanced button	Values: N/A	Description: Gives access to secondary screen where an Alternate IP Route can be specified for backup or redundancy of signal paths. See discussion on next page. For SIP & H.323 operation only.

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



Alternate Routing Field Definitions		
Field Name	Values	Description
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when a data pathway is considered blocked. When the delay exceeds the threshold specified here, the data stream will be diverted to the alternate destination specified as the Alternate IP Address.

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



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

The screenshot shows the 'Add/Edit Inbound Phone Book' dialog box. It contains the following fields and controls:

- Remove Prefix:** Text input field containing '0044207'.
- Add Prefix:** Text input field containing '8,7'.
- Channel Number:** Dropdown menu set to 'Hunting'.
- Description:** Text input field containing 'Access to Inner London'.
- Call Forward:** A sub-section containing:
 - Enable**
 - Forward Condition:** Radio buttons for 'Unconditional' (selected), 'Busy', and 'No Response'.
 - Forward Address / Number:** Empty text input field.
 - Ring Count:** Text input field containing '0'.

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

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

Add/Edit Inbound Phone Book: Field Definitions		
Field Name	Values	Description
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)
Channel Number	1-30, or "Hunting"	E1 channel number to which the call will be assigned as it enters the local telephony equipment (often a local PBX). "Hunting" directs the call to any available channel.

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	Values	Description
Description	--	Describes the facility or geographical location at which the call originated.
Call Forward Parameters		
Enable	Y/N	Click the check-box to enable the call-forwarding feature.
Forward Condition	Uncondit.; Busy No Resp.	Unconditional. When selected, all calls received will be forwarded. Busy. When selected, calls will be forwarded when station is busy. No Response. When selected, calls will be forwarded if called party does not answer after a specified number of rings, as specified in Ring Count field.
Forward Address/ Number	IP addr. or phone number	Phone number or IP address to which calls will be directed.
Ring Count	integer	When No Response is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.

6. When your Outbound and Inbound PhoneBook entries are completed, click on **Save Setup** in the sidebar menu to save your configuration.

You can change your configuration at any time as needed for your system.

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

E1 Phonebook Examples

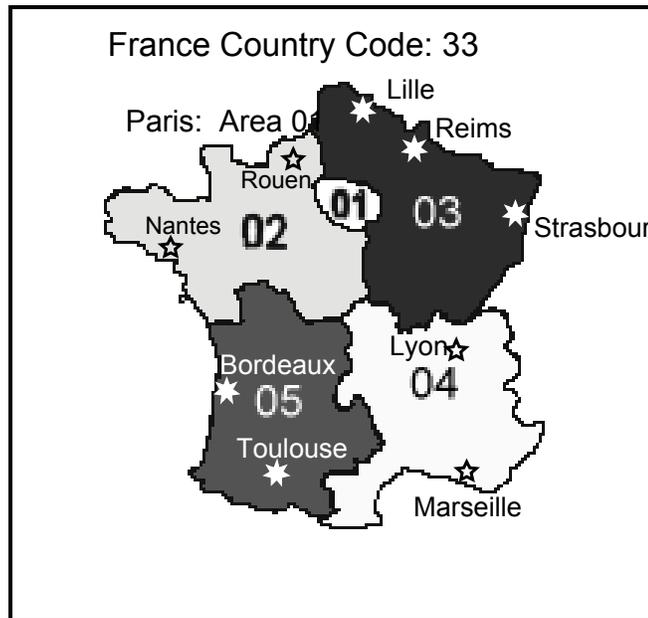
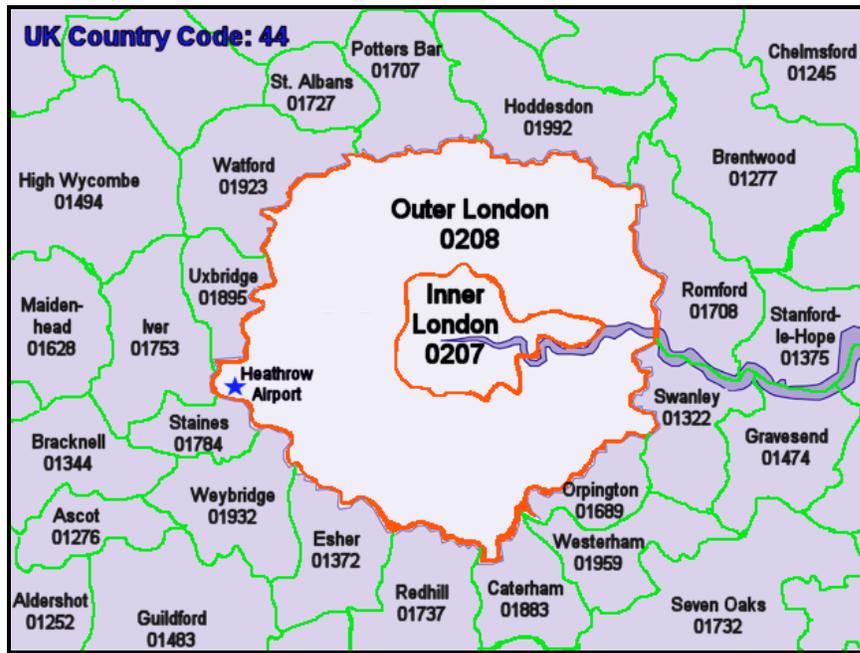
To demonstrate how Outbound and Inbound PhoneBook entries work in an international VOIP system, we will re-visit our previous example in greater detail. It's an international company with offices in London, Paris, and Amsterdam. In each office, a MVP3010 has been connected to the PBX system.

3 Sites, All-E1 Example

The VOIP system will have the following features:

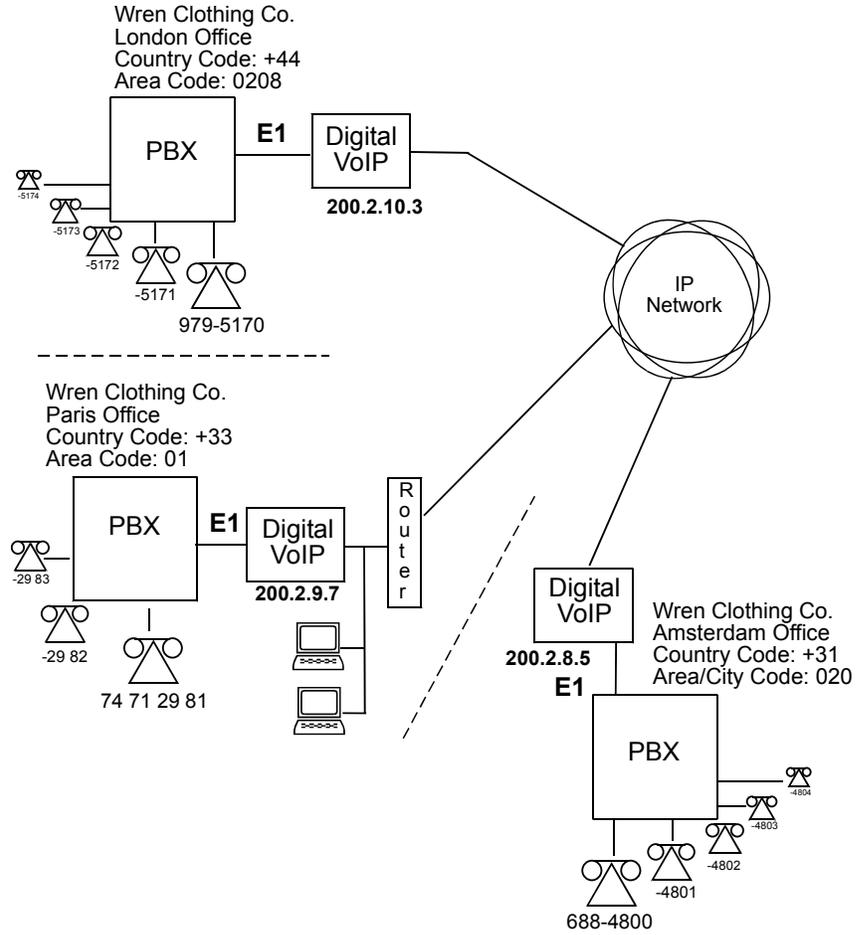
1. Employees in all cities will be able to call each other over the VOIP system using 4-digit extensions.
2. Calls to Outer London and Inner London, greater Amsterdam, and greater Paris will be accessible to all company offices as local calls.
3. Vendors in Guildford, Lyon, and Rotterdam can be contacted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

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





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

Outbound PhoneBook { London VOIP Unit }

Dest Pattern	IP Address	Description
00331	200.002.009.007	Paris
00334	200.002.009.007	Lyon
003120	200.002.008.005	Amsterdam
003110	200.002.008.005	Rotterdam
2	200.002.009.007	Paris (company office, empl. extensions)
4	200.002.008.005	Amsterdam (company office, employees)

Number of Entries : 6

Details

H.323 ID :

Remove Prefix :

Add Prefix :

Total Digits : 13

Add

Edit

Delete

Cancel

The Inbound PhoneBook for the London VOIP is shown below.

Inbound PhoneBook { London VOIP Unit }

Rem Prefix	Add Prefix
0044207	9,7
0044208	9,8
00441483	9,01483
00442089795	5
5	5

Number of Entries : 5

Details

Channel No : 0

Description : Inner London access for Paris & Amst employees

Add

Edit

Delete

Cancel

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.

{ Paris VOIP Unit }

Outbound PhoneBook

Dest Pattern	IP Address	Description
003120	200.002.008.005	Amsterdam
003110	200.002.008.005	Rotterdam
0044207	200.002.010.003	London (Inner)
0044208	200.002.010.003	London (Outer)
00441483	200.002.010.003	Guildford
5	200.002.010.003	London (company office, empl extensions)
4	200.002.008.005	Amsterdam (company office, employees)

Number of Entries : 7

Details

H.323 ID :

Remove Prefix :

Add Prefix :

Total Digits : 13

The Inbound PhoneBook for the Paris VOIP is shown below.

{ Paris VOIP Unit }

Inbound PhoneBook

Rem Prefix	Add Prefix
2	2
00331	9
00334	9,0

Number of Entries : 3

Details

Channel No : 0

Description : Access to Lyon for London & Amsterdam employees

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

{ Amsterdam VOIP Unit }

Outbound PhoneBook

Dest Pattern	IP Address	Description
0044208	200.002.010.003	London (outer)
0044207	200.002.010.003	London (inner)
00441483	200.002.010.003	Guildford
00331	200.002.009.007	Paris
00334	200.002.009.007	Lyon
5	200.002.010.003	London (company office, employ. ext.)
2	200.002.009.007	Paris (company office, employee ext.)

Number of Entries : 3

Details

H.323 ID :

Remove Prefix :

Add Prefix :

Total Digits : 14

The Inbound PhoneBook for the Amsterdam VOIP is shown below.

{ Amsterdam VOIP Unit }

Inbound PhoneBook

Rem Prefix	Add Prefix
4	4
003120	9
003110	9,010
0031206884	4

Number of Entries : 4

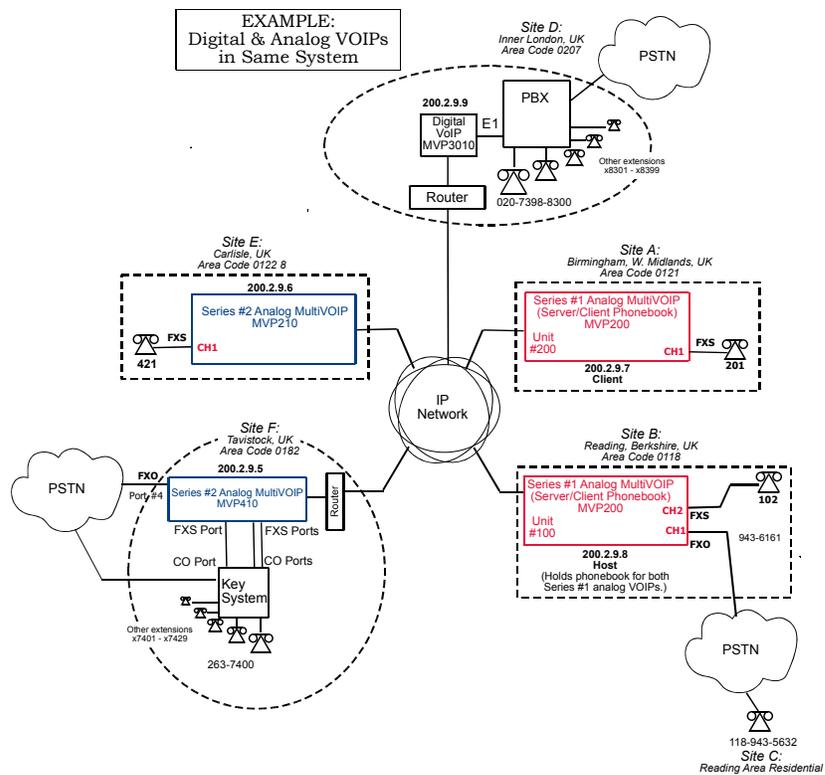
Details

Channel No : 0

Description : Access to Amsterdam office by London & Paris employees

Configuring Digital & Analog VOIPs in Same System

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



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

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

These seven phone books are shown below.

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

Phone Book for Analog VOIP Host Unit (Site B) (continued)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
0207 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to phone numbers in 0207 area code (Inner London) in which Site D is located.
0208 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to phone numbers in 0208 area code (Outer London) for which calls are local from Site D (Inner London).
<p>Note 1. The “x” is a wildcard character.</p> <p>Note 2. By specifying “Channel 0,” we instruct the MVP3010 to choose any available data channel to carry the call.</p> <p>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.</p>			

The Outbound PhoneBook of the MVP3010 is shown below.

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

The Inbound PhoneBook of the MVP3010 is shown below.

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

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

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

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

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

Call Completion Summaries

Site A calling Site C, Method 1

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

Site A calling Site C, Method 2

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

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

Site C calling Site A

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

Site D calling Site C

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

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

Site D calling Site F

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

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

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

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

Call Event Sequence

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

Site F calling Site D

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

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

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

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

Variations in PBX Characteristics

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

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

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

International Telephony Numbering Plan Resources

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

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

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

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

Chapter 9: Analog/BRI Phonebook Configuration

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

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

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

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

- Phonebook Starter Configuration

- Phonebook Tips

- Phonebook Example (One Common Situation)

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

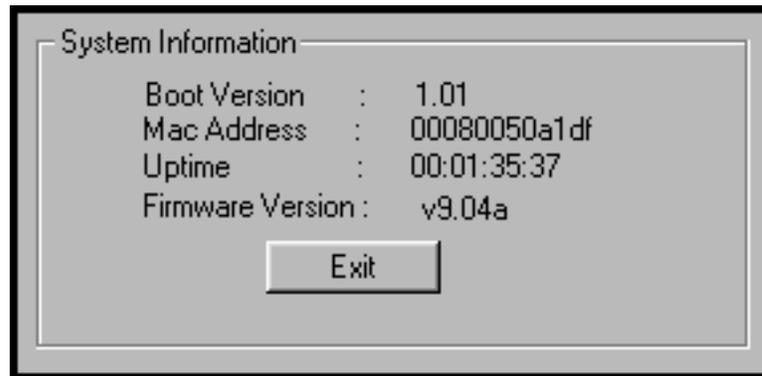
Chapter 10: Operation and Maintenance

Operation and Maintenance

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

System Information screen

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



System Information Parameter Definitions (not supported in BRI 5.02c software)		
Field Name	Values	Description
Boot Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Mac Address	alpha-numeric	Denotes the number assigned as the voip unit's unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.
Firmware Version	alpha-numeric	Indicates the version of the MultiVOIP firmware.

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

Logs

Enable Console Messages

OK

Logs

Turn Off Logs

GUI SMTP SNMP

Cancel

Help

SysLog Server

Enable

IP Address :

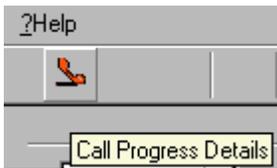
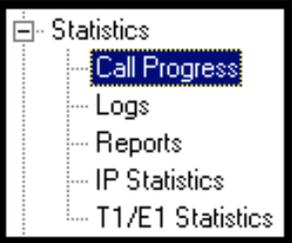
Port :

Online Statistics Update Interval Sec

Statistics Screens

Ongoing operation of the MultiVOIP, whether it is in a MultiVOIP/PBX setting or MultiVOIP/telco-office setting, can be monitored for performance using the Statistics functions of the MultiVOIP software.

About Call Progress

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

The Call Progress Details Screen

Call Progress Details

Channel:

Call Details

Duration: -
Mode: -
Voice Codec: -
Packets Sent: -
Packets Rcvd: -
Bytes Sent: -
Bytes Rcvd: -
Packets Lost: -
Outbound Digits: -
Prefix Matched: -

Disconnect
Exit
Help

From->To Details

From ----> To : - ---->
Gateway Name: -
IP Address:
Options: -

SC - Silence Compression FEC - Forward Error Correction

Supplementary Services Status

Call On Hold : - 193.100.099.202, Mpls, On Hold for 90 Seconds
Call Waiting : - 193.100.099.202, Mktgvoip3
Caller Id : - Calling Party - smithbob01

Call Status: On Hook
Call Control Status : - Tun, FS + Tun, AE, Mux

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.

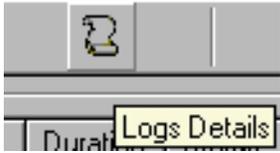
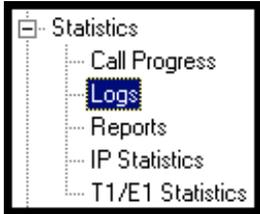
Call Progress Details: Field Definitions (cont'd)		
From – To Details		Description
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that handled this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address from which the call was received.
Options	SC, FEC	Displays VOIP transmission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

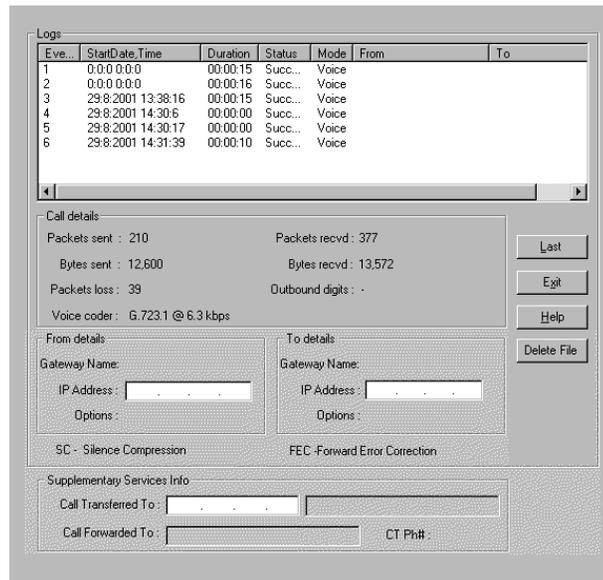
Call Progress Details: Field Definitions (cont'd)		
Field Name	Values	Description
Supplementary Services Status		
Caller ID	There are four values: “Calling Party + <i>identifier</i> ”; “Alerting Party + <i>identifier</i> ”; “Busy Party + <i>identifier</i> ”; and “Connected Party + <i>identifier</i> ”	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
<p>Alt + L</p>	

The Logs Screen



The Logs screen displays a table of call logs and detailed call information. The table has columns for Event, Start Date/Time, Duration, Status, Mode, From, and To. Below the table, there are sections for Call details, From details, To details, and Supplementary Services Info.

Event	Start Date/Time	Duration	Status	Mode	From	To
1	0:0:0:0:0	00:00:15	Succ...	Voice		
2	0:0:0:0:0	00:00:16	Succ...	Voice		
3	29-8-2001 13:38:16	00:00:15	Succ...	Voice		
4	29-8-2001 14:30:6	00:00:00	Succ...	Voice		
5	29-8-2001 14:30:17	00:00:00	Succ...	Voice		
6	29-8-2001 14:31:39	00:00:10	Succ...	Voice		

Call details

Packets sent : 210 Packets rcvcd : 377
 Bytes sent : 12,600 Bytes rcvcd : 13,572
 Packets loss : 39 Outbound digits : -
 Voice coder : G.723.1 @ 6.3 kbps

From details **To details**

Gateway Name: Gateway Name:
 IP Address: IP Address:
 Options: Options:

SC - Silence Compression FEC - Forward Error Correction

Supplementary Services Info

Call Transferred To: CT Ph#: Call Forwarded To: CT Ph#:

Logs Screen Details: Field Definitions		
Field Name	Values	Description
Event # column	1 or higher	All calls are assigned an event number in chronological order, with the most recent call having the highest event number.
Start Date,Time column	dd:mm:yyyy hh:mm:ss	The starting time of the call (event). The date is presented as a day expression of one or two digits, a month expression of one or two digits, and a four-digit year. This is followed by a time-of-day expression presented as a two-digit hour, a two-digit minute, and a two-digit seconds value. (statistics, logs) field
Duration column	hh:mm:ss	This describes how long the call (event) lasted in hours, minutes, and seconds.
Status column	success or failure	Displays the status of the call, i.e., whether the call was completed successfully or not.
Mode column	voice or FAX	Indicates whether the (event) being described was a voice call or a FAX call.
From column	gateway name	Displays the name of the voice gateway that originates the call.
To column	gateway name	Displays the name of the voice gateway that completes the call.
Special Buttons		
Last		Displays last log entry.
Delete File		Deletes selected log file.
Call Details		
Packets sent	integer value	The number of data packets sent over the IP network in the course of this call.
Bytes sent	integer value	The number of bytes of data sent over the IP network in the course of this call.

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

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

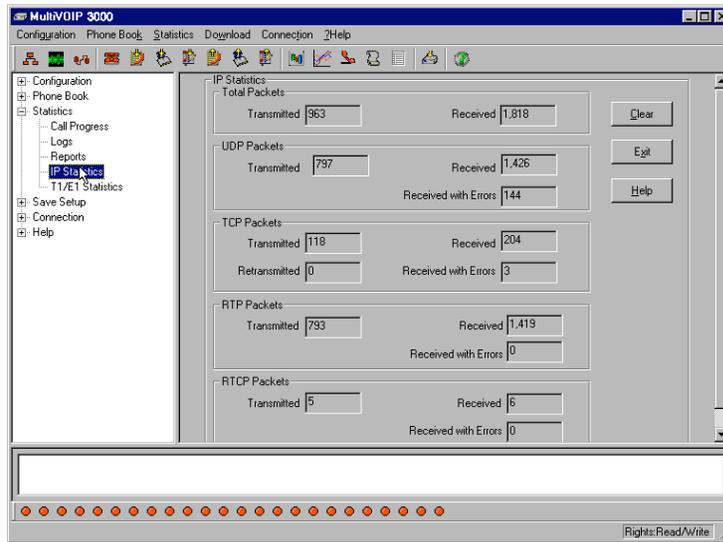
About Reports

This feature not implemented as of this writing.

About IP Statistics

Accessing IP Statistics	
Pulldown	Icon
<p>Statistics Call Progress Alt+A Logs Alt+L Reports Alt+R IP Statistics Alt+I T1/E1 Statistics Alt+T</p>	<p>Connection ?Help IP Statistics Details</p>
Shortcut	Sidebar
<p>Alt + I</p>	<p>Statistics Call Progress Logs Reports IP Statistics T1/E1 Statistics</p>

IP Statistics Screen



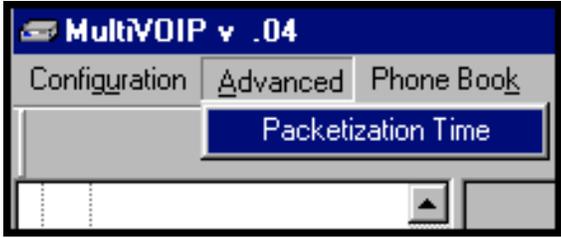
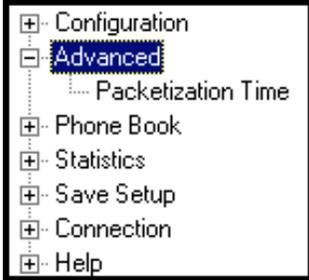
IP Statistics: Field Definitions		
Field Name	Values	Description
		<p>UDP versus TCP. (User Datagram Protocol versus Transmission Control Protocol). UDP provides unguaranteed, connectionless transmission of data across an IP network. By contrast, TCP provides reliable, connection-oriented transmission of data.</p> <p>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..</p> <p>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).</p>
"Clear" button	--	Clears packet tallies from memory.
Total Packets		Sum of data packets of all types.
Transmitted	integer value	Total number of packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Total number of packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

IP Statistics: Field Definitions (cont'd)		
Field Name	Values	Description
Total Packets (cont'd)		Sum of data packets of all types.
Received with Errors	integer value	Total number of error-laden packets received by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
UDP Packets		User Datagram Protocol packets.
Transmitted	integer value	Number of UDP packets transmitted by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of UDP packets received by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden UDP packets received by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
TCP Packets		Transmission Control Protocol packets.
Transmitted	integer value	Number of TCP packets transmitted by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of TCP packets received by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden TCP packets received by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.

IP Statistics: Field Definitions (cont'd)		
RTP Packets		Voice signals are transmitted in Realtime Transport Protocol packets. RTP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTP packets transmitted by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTP packets received by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTP packets received by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
RTCP Packets		Realtime Transport Control Protocol packets convey control information to assist in the transmission of RTP (voice) packets. RTCP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTCP packets transmitted by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTCP packets received by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTCP packets received by this VOIP gateway since the last “clearing” or resetting of the counter within the MultiVOIP software.

About Packetization Time

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

Accessing Packetization Time	
Pulldown	
	
Shortcut/Icon	Sidebar
none/none	

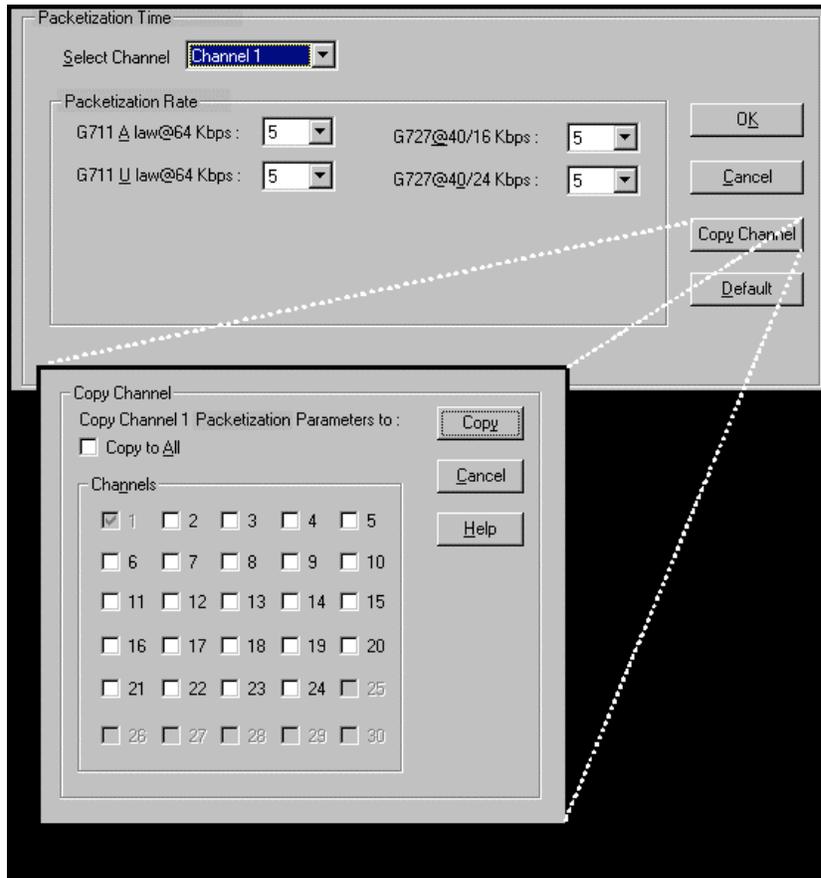
Packetization Time Screen

Packetization rates can be set separately for each channel.

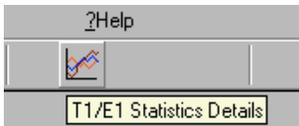
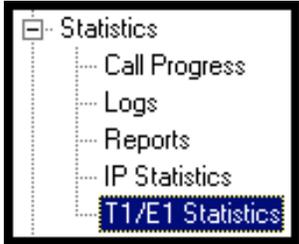
The table below presents the ranges and increments for packetization rates.

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
<p style="text-align: center;">Alt + T</p>	

The T1 and E1 Statistics screens are only accessible and applicable for the MVP2400, MVP2410, and MVP3010.

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

T1 Statistics Screen

T1 Statistics	
Red Alarm: <input type="text" value="0"/>	Yellow Alarm: <input type="text" value="0"/>
Blue Alarm: <input type="text" value="0"/>	Frame Search Restart Flag: <input type="text" value="0"/>
Loss of Frame Alignment: <input type="text" value="0"/>	Loss of MultiFrame Alignment: <input type="text" value="0"/>
Excessive Zeros: <input type="text" value="0"/>	Transmit Slip: <input type="text" value="0"/>
Status Freeze Signalling Active: <input type="text" value="0"/>	Pulse Density Violation: <input type="text" value="0"/>
Line Loopback Deactivation Signal: <input type="text" value="0"/>	Line Loopback Activation Signal: <input type="text" value="0"/>
Transmit Line Short: <input type="text" value="0"/>	Transmit Line Open: <input type="text" value="0"/>
Transmit Data Overflow: <input type="text" value="0"/>	Transmit Data Underrun: <input type="text" value="0"/>
Transmit Slip Positive: <input type="text" value="0"/>	Transmit Slip Negative: <input type="text" value="0"/>

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

Red Alarm: <input style="width: 80px;" type="text" value="145,388"/>	Yellow Alarm: <input style="width: 80px;" type="text" value="0"/>	<input type="button" value="Clear"/>
Blue Alarm: <input style="width: 80px;" type="text" value="0"/>	Status Freeze Signalling Active: <input style="width: 80px;" type="text" value="0"/>	<input type="button" value="Exit"/>
Loss of Frame Alignment: <input style="width: 80px;" type="text" value="145,388"/>	Loss of MultiFrame Alignment: <input style="width: 80px;" type="text" value="145,388"/>	<input type="button" value="Help"/>
Receive Timeslot 16 Remote Alarm: <input style="width: 80px;" type="text" value="0"/>	Receive Timeslot 16 Loss of Signal: <input style="width: 80px;" type="text" value="0"/>	
Receive Timeslot 16 Alarm Indication Signal: <input style="width: 80px;" type="text" value="0"/>	Receive Timeslot 16 Loss of Multiframe Alignment: <input style="width: 80px;" type="text" value="145,388"/>	
Transmit Line Short: <input style="width: 80px;" type="text" value="0"/>	Transmit Line Open: <input style="width: 80px;" type="text" value="0"/>	
Transmit Data Overflow: <input style="width: 80px;" type="text" value="0"/>	Transmit Data Underrun: <input style="width: 80px;" type="text" value="0"/>	
Transmit Slip Positive: <input style="width: 80px;" type="text" value="145,388"/>	Transmit Slip Negative: <input style="width: 80px;" type="text" value="145,388"/>	

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

Select BRI Interface :

Layer 1 Interface

Status: Loss Of Framing:

State: Loss Of Sync:

Switch Information

TEI Assignment

TEI 0:

TEI 1:

TEI 2:

TEI 3:

TEI 4:

TEI 5:

TEI 6:

TEI 7:

D-Channel Information

Tx Packets:

Rx Packets:

SPID 0

Status:

SPID 1

Status:

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 <i>number</i>)	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 <i>number</i>)	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

Description	IP Address	Port	Register Duration	Status

No of Entries :

Details

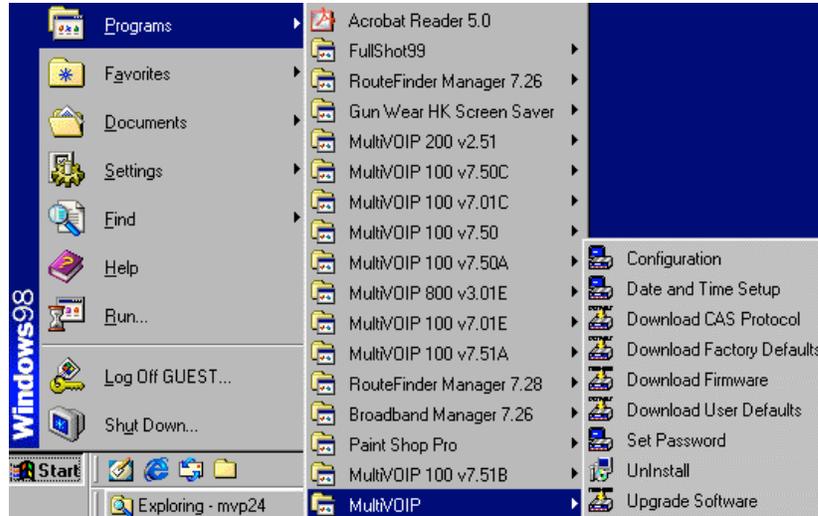
Count of Registered Numbers :

List of Registered Numbers :

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.
Details		
No. of Entries		The number of gateways currently registered to the Registrar. This includes all SPP clients registered and the Registrar itself.
Details		
Count of Registered Numbers		If a registered gateway is selected (by clicking on it in the screen), The "Count of Registered Numbers" will indicate the number of registered phone numbers for the selected gateway. When a client registers, all of its inbound phonebook's phone numbers become registered.
List of Registered Numbers		Lists all of the registered phone numbers for the selected gateway.

MultiVoip Program Menu Items

After the MultiVoip program is installed on the PC, it can be launched from the **Programs** group of the Windows **Start** menu (**Start** | **Programs** | **MultiVOIP** ____ | ...). In this section, we describe the software functions available on this menu.



Several basic software functions are accessible from the MultiVoip software menu, as shown below.

MultiVOIP Program Menu	
Menu Selection	Description
Configuration	Select this to enter the Configuration program where values for IP, telephony, and other parameters are set.
Date and Time Setup	Select this for access to set calendar/clock used for data logging.
Download CAS Protocol	Telephony CAS files are for Channel Associated Signaling. There are many CAS files, some labeled for specific functionality, others for countries or regions where certain telephony attributes are standard.

MultiVOIP Program Menu (cont'd)	
Menu Selection	Description
Download Factory Defaults	Select this to return the configuration parameters to the original factory values.
Download Firmware	Select this to download new versions of firmware as enhancements become available.
Download User Defaults	To be used after a full set of parameter values, values specified by the user, have been saved (using Save Setup). This command loads the saved user defaults into the MultiVOIP.
Set Password	Select this to create a password for access to the MultiVOIP software programs (Program group commands, Windows GUI, web browser GUI, & FTP server). Only the FTP Server function <i>requires</i> a password for access. The FTP Server function also requires that a username be established along with the password.
Uninstall	Select this to uninstall the MultiVOIP software (most, but not all components are removed from computer when this command is invoked).
Upgrade Software	Loads firmware (including H.323 stack) and factory default settings from the controller PC to the MultiVOIP unit.

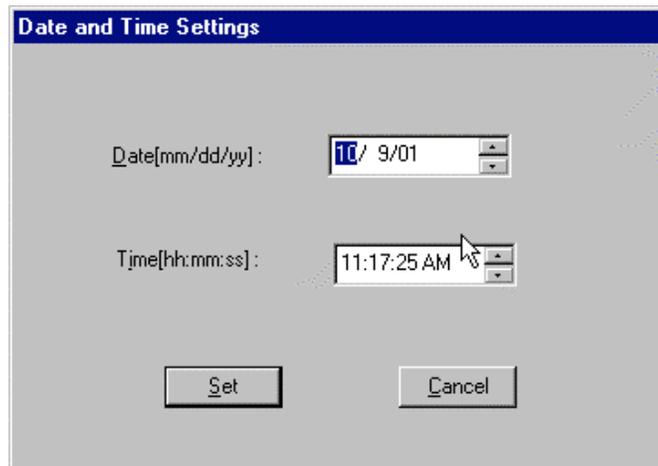
“Downloading” here refers to transferring program files from the PC to the nonvolatile “flash” memory of the MultiVOIP. Such transfers are made via the PC’s serial port. This can be understood as a “download” from the perspective of the MultiVOIP unit.

When new versions of the MultiVoip software become available, they will be posted on MultiTech’s web or FTP sites. Although transferring updated program files from the MultiTech web/FTP site to the user’s PC can generally be considered a download (from the perspective of the PC), this type of download cannot be initiated from the MultiVoip software’s Program menu command set.

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

Date and Time Setup

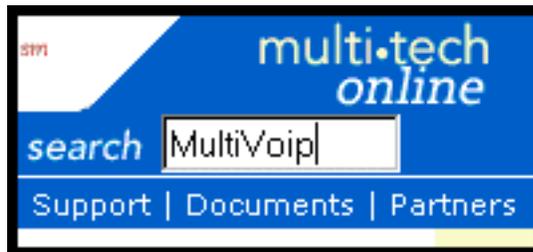
The dialog box below allows you to set the time and date indicators of the MultiVOIP system.



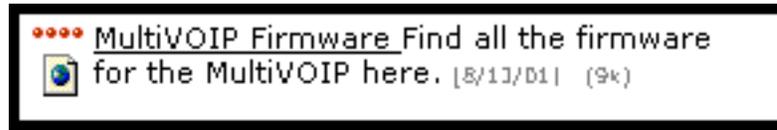
Obtaining Updated Firmware

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

Note that the structure of the MultiTech web/FTP site may change without notice. However, firmware updates can generally be found using standard web techniques. For example, you can access updated firmware by doing a search or by clicking on **Support**.



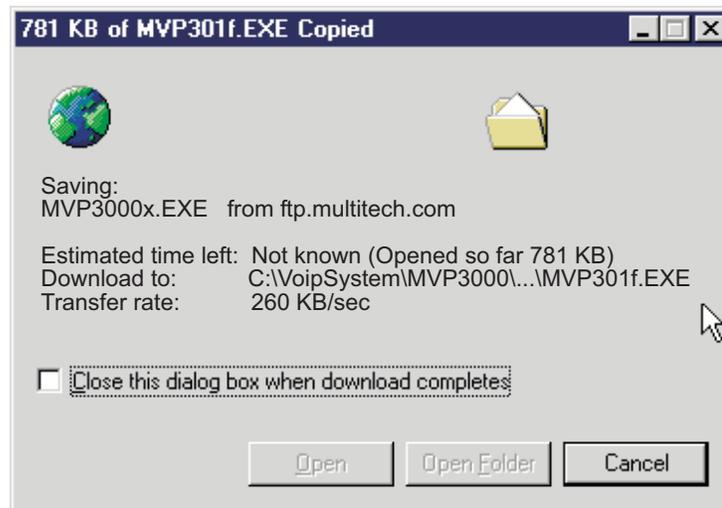
If you conduct a search, for example, on the word “MultiVoip,” you will be directed to a list of firmware that can be downloaded.



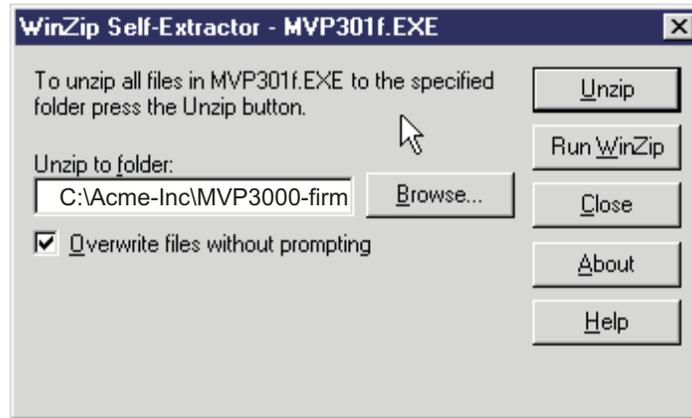
If you choose **Support**, you can select “MultiVoip” in the **Product Support** menu and then click on **Firmware** to find MultiVOIP resources.



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



Generally, the firmware file will be a self-extracting compressed file (with .zip extension), which must be expanded (decompressed, or “unzipped”) on the user’s PC in a user-specified directory.



Implementing a Software Upgrade

Beginning with the 4.03/6.03 software release, MultiVOIP software can be upgraded locally using a single command at the MultiVOIP Windows GUI, namely **Upgrade Software**. This command downloads firmware (including the H.323 stack), and factory default settings from the controller PC to the MultiVOIP unit.

When using the MultiVOIP Windows GUI, firmware and factory default settings can also be transferred from controller PC to MultiVOIP piecemeal using separate commands.

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

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

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

When upgrading firmware, the software commands “Download Firmware,” and “Download Factory Defaults” must be implemented in order, else the upgrade is incomplete.

Identifying Current Firmware Version

Before implementing a MultiVOIP firmware upgrade, be sure to verify the firmware version currently loaded on it. The firmware version appears in the MultiVoip Program menu. Go to **Start | Programs | MultiVOIP ____ x.xx**. The final expression, x.xx, is the firmware version number. In the illustration below, the firmware version is 4.00a, made for the E1 MultiVOIP (MVP3010).



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

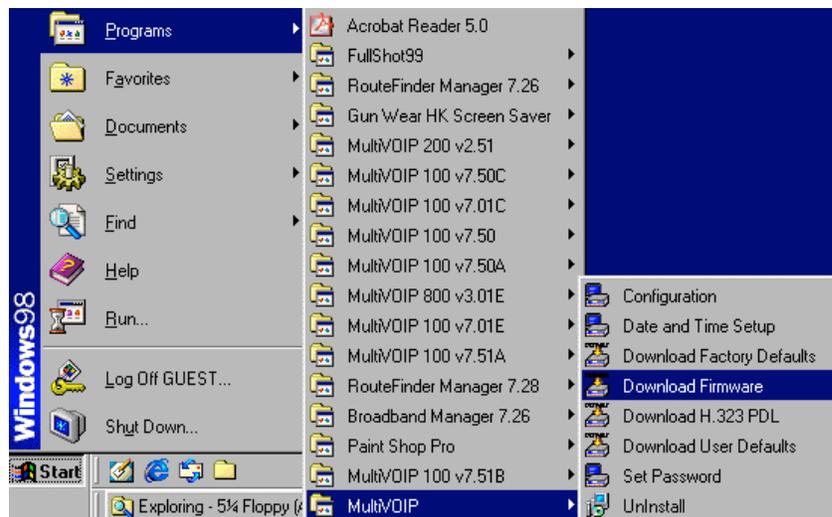
Download Firmware transfers the firmware (including the H.323 protocol stack) in the PC's MultiVOIP directory into the nonvolatile flash memory of the MultiVOIP.

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

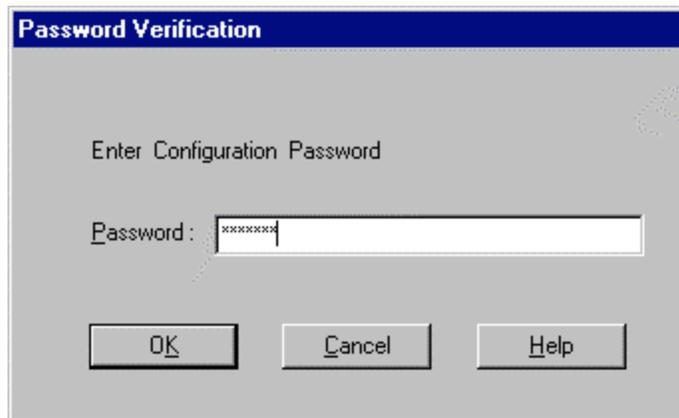
Upgrade Software implements both the **Download Firmware** command and the **Download Factory Defaults** command.

Downloading Firmware

1. The MultiVoip Configuration program must be off when invoking the **Download Firmware** command. If it is on, the command will not work.
2. To invoke the Download Factory Defaults command, go to **Start | Programs | MVP ___ x.xx | Download Firmware**.

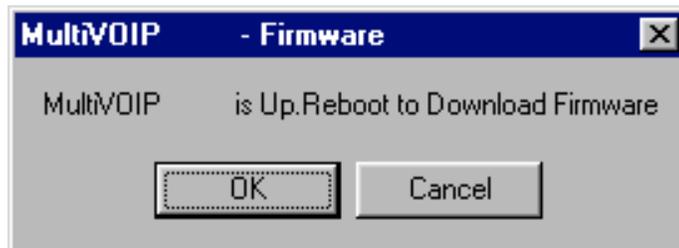


3. If a password has been established, the **Password Verification** screen will appear.



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

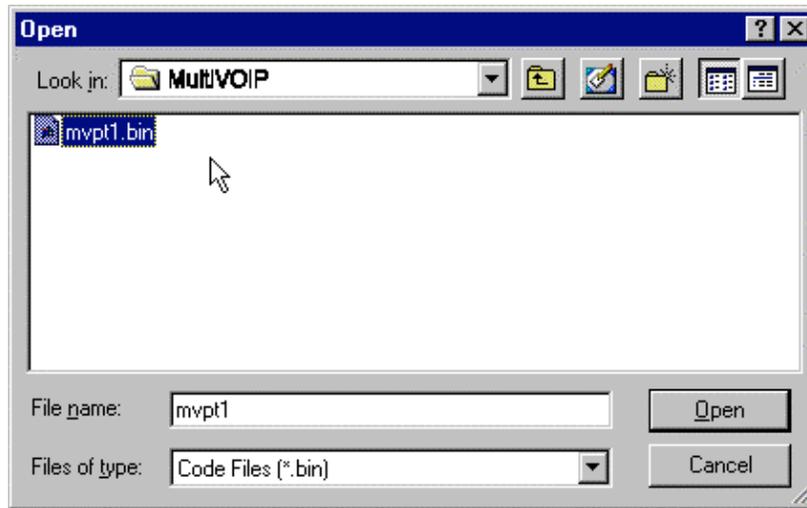
4. The **MultiVOIP ___ - Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"



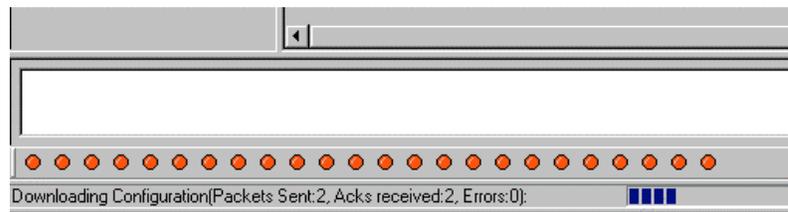
Click **OK** to download the firmware.

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

- The program will locate the firmware “.bin” file in the MultiVOIP directory. Highlight the correct (newest) “.bin” file and click **Open**.



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

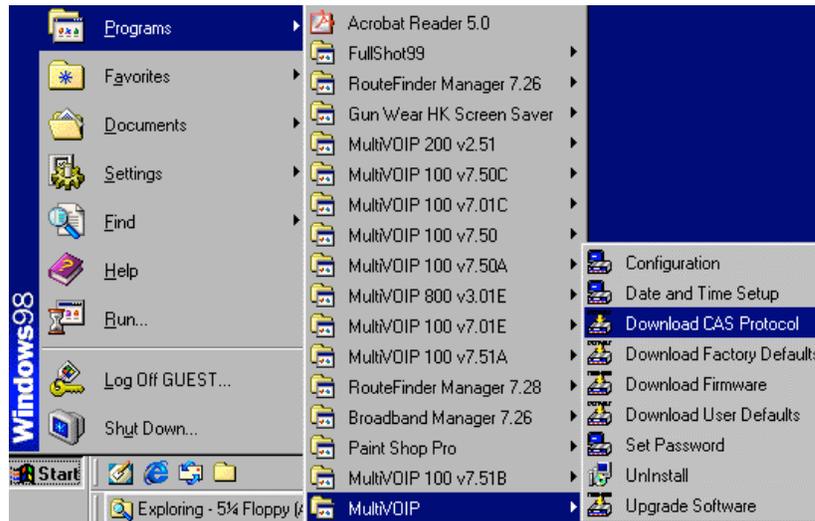


The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

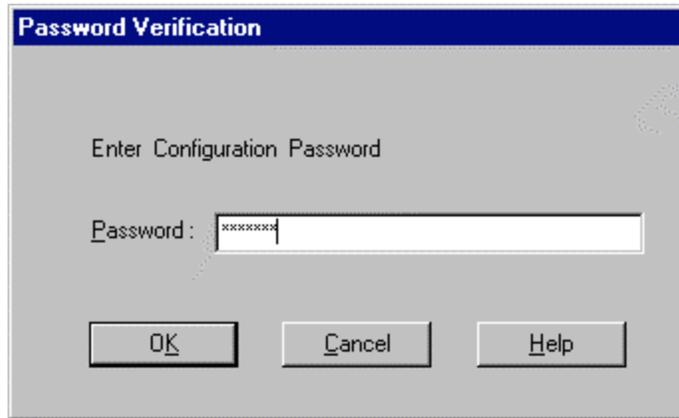
- The **Download Firmware** procedure is complete.

Downloading CAS Protocols

1. The MultiVoip Configuration program must be off when invoking the **Download CAS Protocol** command. If it is on, the command will not work.
2. To invoke the **Download H.323 PDL** command, go to **Start | Programs | MVP ___ x.xx | Download H.323 PDL**.

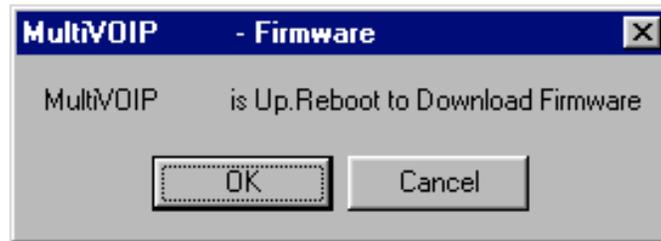


3. If a password has been established, the **Password Verification** screen will appear.



Type in password and click **OK**.

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



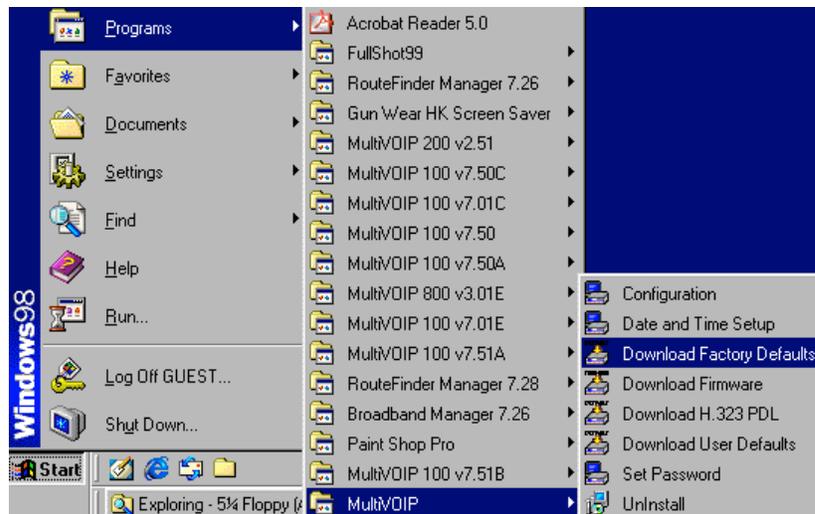
Click **OK** to download the CAS Protocol file(s) to the MultiVOIP.

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

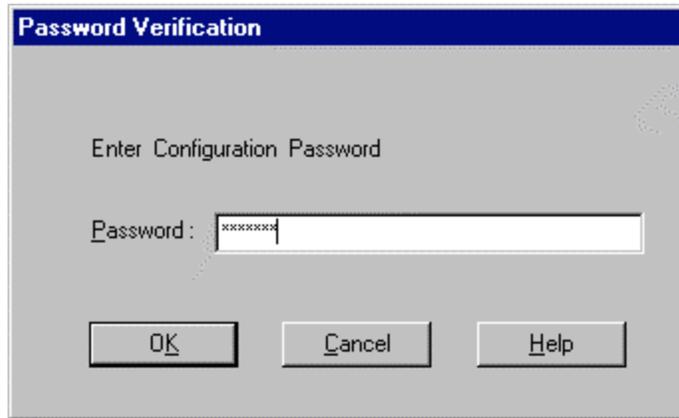
5. The program will locate the CAS protocol file in the MultiVOIP directory.
Highlight the correct (newest) file and click **Open**.
6. Progress bars will appear at the bottom of the screen during the file transfer.
The MultiVOIP's "Boot" LED will turn off at the end of the transfer.
7. The **Download CAS Protocol** procedure is complete.

Downloading Factory Defaults

1. The MultiVoip Configuration program must be off when invoking the **Download Factory Defaults** command. If it is on, the command will not work.
2. To invoke the **Download Factory Defaults** command, go to **Start | Programs | MVP ___ x.xx | Download Factory Defaults**.

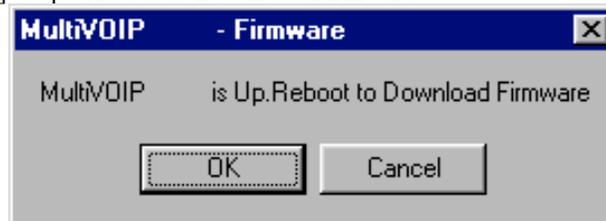


3. If a password has been established, the **Password Verification** screen will appear.



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

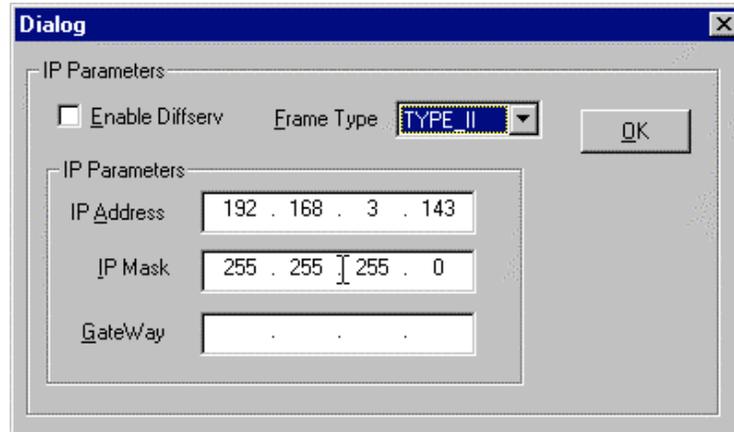
4. The **MVP ___ - Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"



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

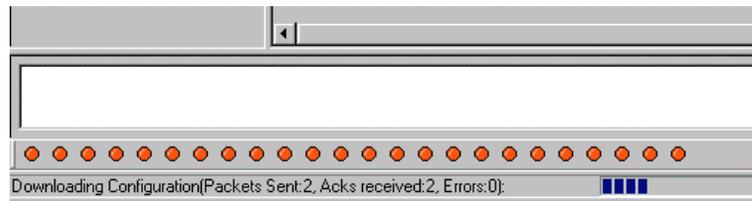
The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

- 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**.

- 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.

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

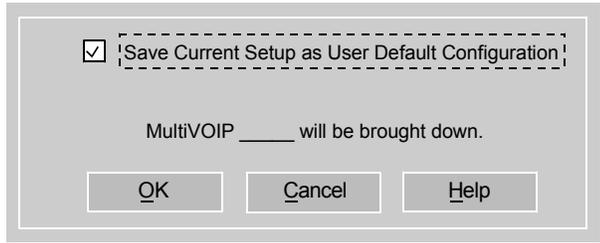
Setting and Downloading User Defaults

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

- 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.

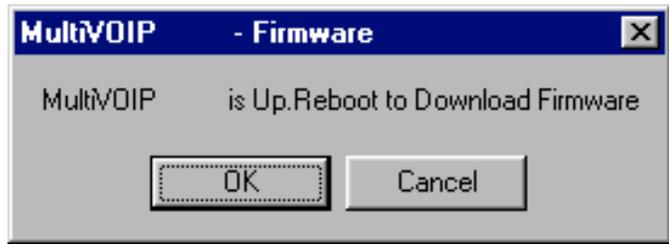


- Before the setup configuration is saved, you will be prompted to save the setup as the User Default Configuration. Select the checkbox and click **OK**.



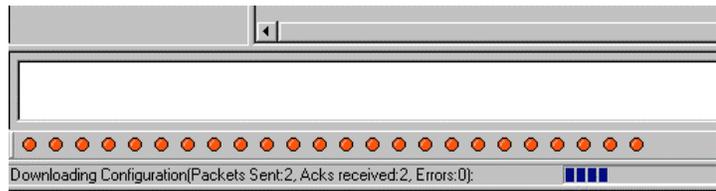
A user default file will be created.

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

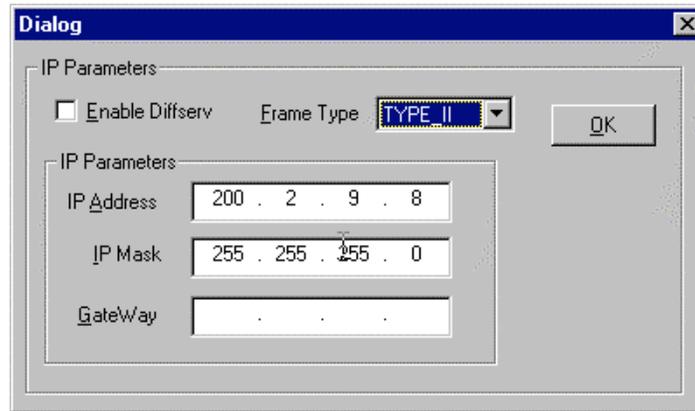


Click **OK** to download the factory defaults. The “Boot” LED on the MultiVOIP will light up and remain lit during the file transfer process.

- Progress bars will appear during the file transfer process.



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



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

Firmware

The Download IFM Firmware command applies only to the MVP210/410/810 and MVP210G/410G/810G models. This command transfers firmware to the telephony interface modules of each voice channel. These firmware modules handle the physical interface (FXS, FXO and E&M) to the attached analog telephony equipment.

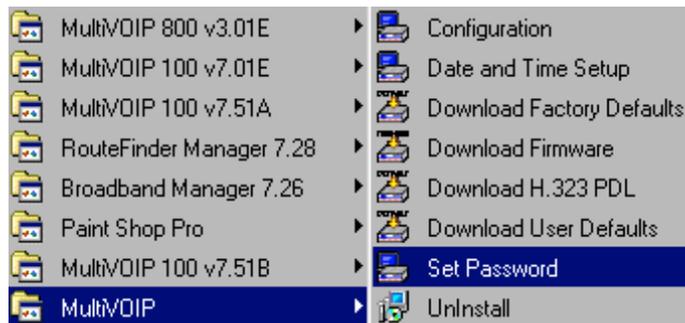


Setting a Password (Windows GUI)

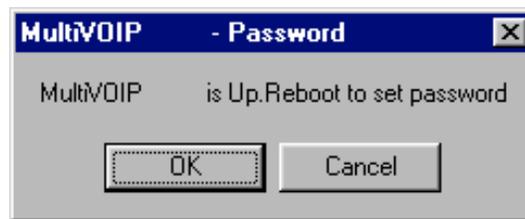
After a user name has been designated and a password has been set, that password is required to gain access to any functionality of the MultiVOIP software. Only one user name and password can be assigned to a voip unit. The user name will be required when communicating with the MultiVOIP via the web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

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

1. The MultiVoip configuration program must be off when invoking the **Set Password** command. If it is on, the command will not work.
2. To invoke the **Set Password** command, go to **Start | Programs | MVP _____ x.xx | Set Password**.



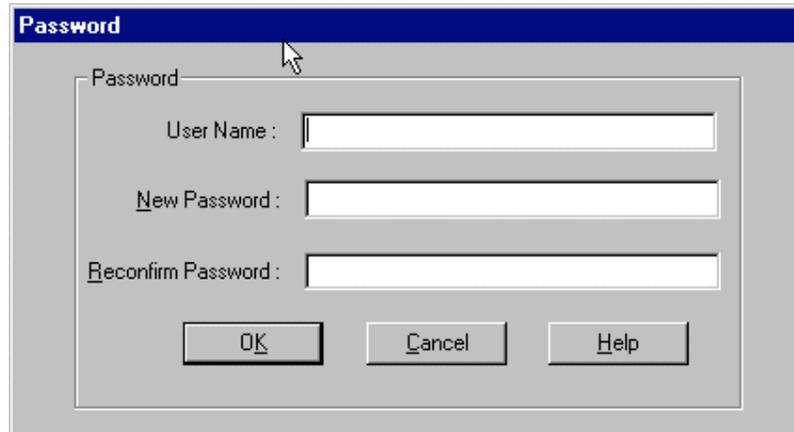
3. You will be prompted to confirm that you want to establish a password, which will entail rebooting the MultiVOIP (which is done automatically).



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

- 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**.

- 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.

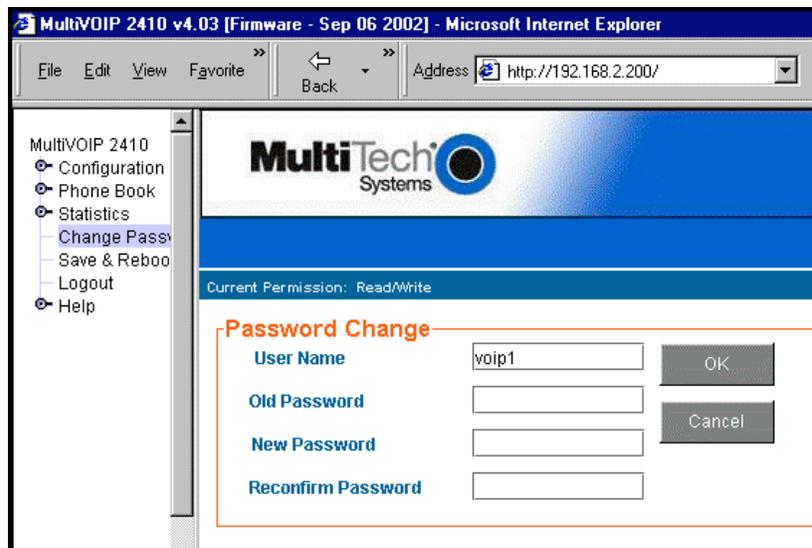
The MultiVOIP program will produce an error message if an invalid password is entered.



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.

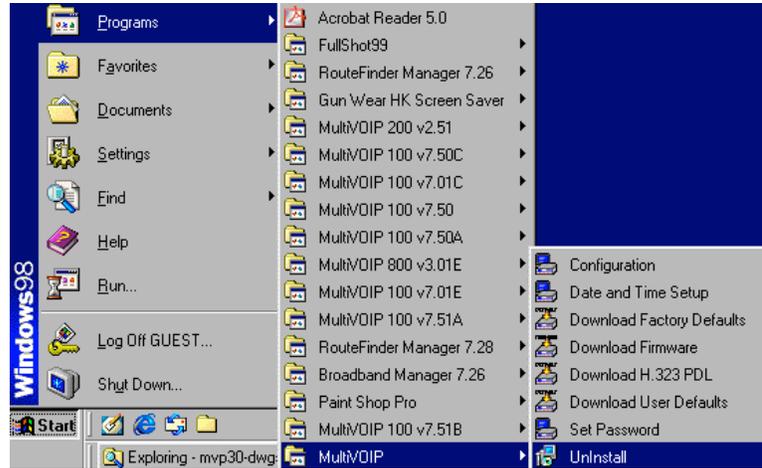


The screenshot shows a Microsoft Internet Explorer browser window titled "MultiVOIP 2410 v4.03 [Firmware - Sep 06 2002] - Microsoft Internet Explorer". The address bar shows "http://192.168.2.200/". The browser interface includes a menu bar (File, Edit, View, Favorite) and a navigation bar (Back). On the left, a navigation tree lists: MultiVOIP 2410, Configuration, Phone Book, Statistics, Change Password, Save & Reboot, Logout, and Help. The main content area features the MultiTech Systems logo and a blue header. Below the header, it displays "Current Permission: Read/Write". The "Password Change" section is highlighted with an orange border and contains the following fields and buttons:

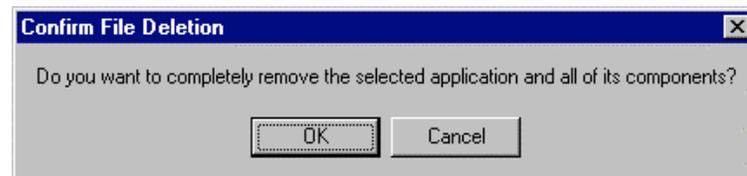
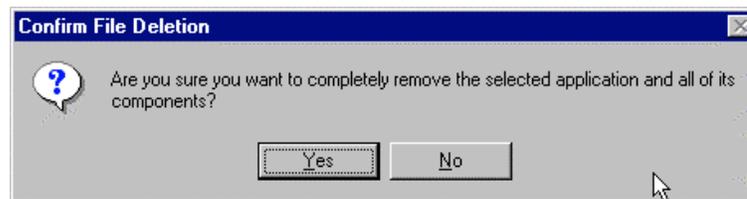
Password Change		
User Name	<input type="text" value="voip1"/>	OK
Old Password	<input type="text"/>	Cancel
New Password	<input type="text"/>	
Reconfirm Password	<input type="text"/>	

Un-Installing the MultiVOIP Software

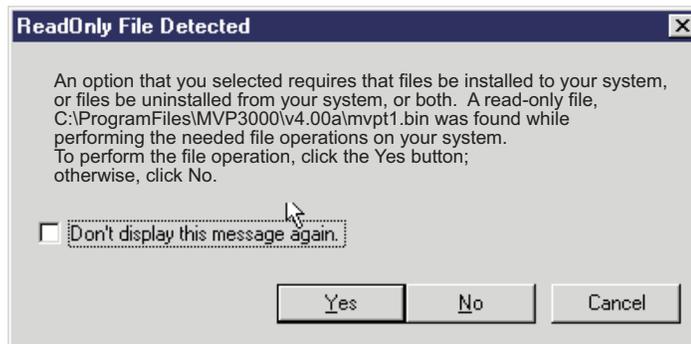
1. To un-install the MultiVOIP configuration software, go to **Start | Programs** and locate the MultiVOIP entry. Select **Uninstall MVP____ vx.xx** (versions may vary).



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



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



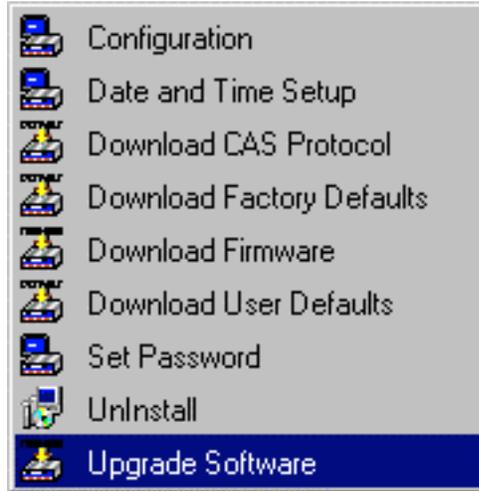
4. A completion screen will appear.



Click **Finish**.

Upgrading Software

As noted earlier (see the section *Implementing a Software Upgrade* above), the Upgrade Software command transfers, from the controller PC to the MultiVOIP unit, firmware (including the H323 stack) and factory default configuration settings. As such, **Upgrade Software** implements the functions of both **Download Firmware** and **Download Factory Defaults** in a single command.



FTP Server File Transfers (“Downloads”)

With the 4.03/6.03 software release, MultiTech has built an FTP server into the MultiVOIP unit. Therefore, file transfers from the controller PC to the voip unit can be done using an FTP client program or even using a browser (e.g., Internet Explorer or Netscape, used in conjunction with Windows Explorer).

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

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

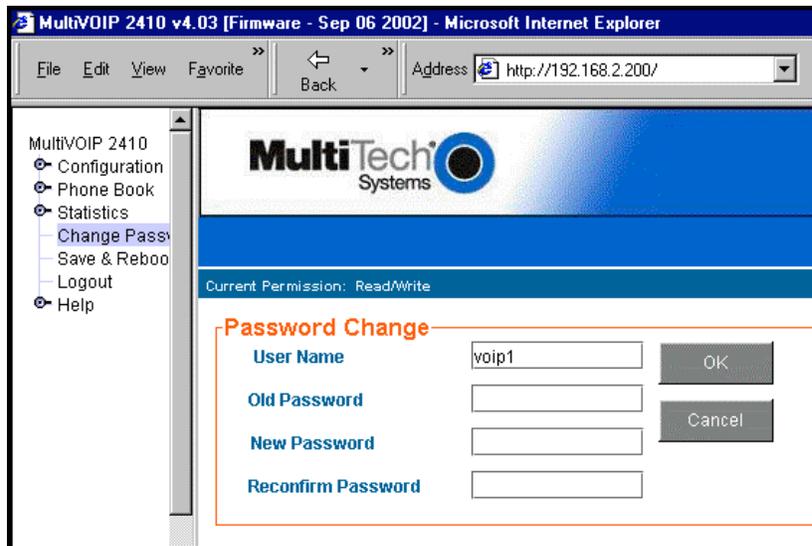
The phonebook downloading feature greatly reduces the data-entry required to establish inbound and outbound phonebooks for the voip units within a system. Although each MultiVOIP unit will require some unique phonebook entries, most will be common to the entire voip system. After the phonebooks for the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

To transfer files using the FTP server functionality in the MultiVOIP, follow these directions.

1. **Establish Network Connection and IP Addresses.** Both the controller PC and the MultiVOIP unit(s) must be connected to the same IP network. An IP address must be assigned for each.

IP Address of Control PC	_____ . _____ . _____ . _____
IP Address of voip unit #1	_____ . _____ . _____ . _____
⋮	⋮ ⋮ ⋮ ⋮
IP address of voip unit #n	_____ . _____ . _____ . _____

2. **Establish User Name and Password.** You must establish a user name and (optionally) a password for contacting the voip over the IP network. (When connection is made via a local serial connection between the PC and the voip unit, no user name is needed.)



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

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

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

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

IP Parameters

Enable Diffserv Frame Type: TYPE-II

IP Parameters

Enable DHCP

IP Address: 192 . 168 . 2 . 200

IP Mask: 255 . 255 . 255 . 0

Gateway: . . .

DNS

Enable DNS

DNS Server IP Address: . . .

FTP Server

Enable

OK

Cancel

Help

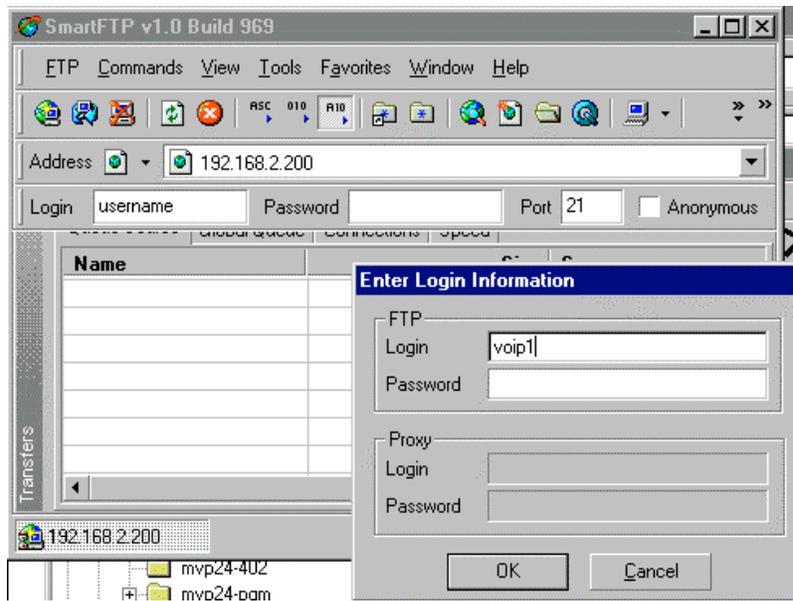
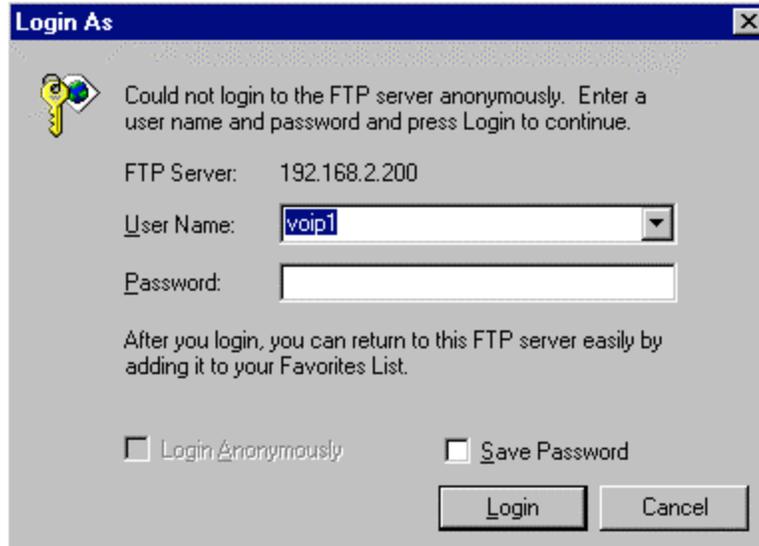
5. Identify Files to be Updated. Determine which files you want to update. Six types of files can be updated using the FTP feature. In some cases, the file to be transferred will have “Ftp” as the part of its filename just before the suffix (or extension). So, for example, the file “mvpt1Ftp.bin” can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

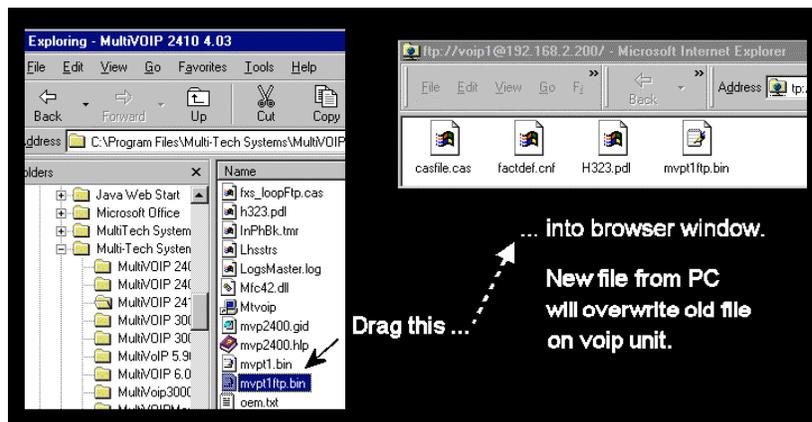


8. **Invoke Download.** Downloading can be done with a web browser or with an FTP client program.

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).

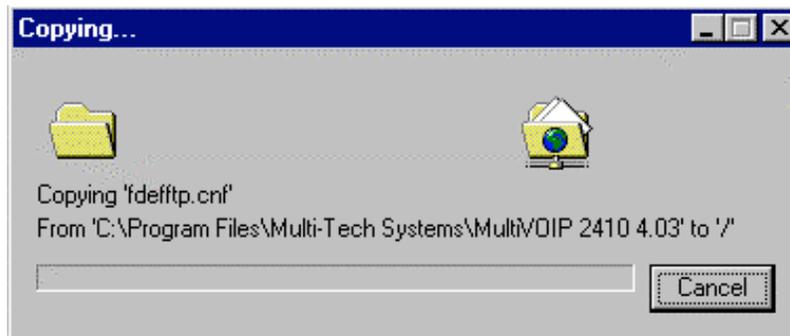
8A2. Drag-and-drop files from the local Windows browser (e.g., Windows Explorer) to the web browser.



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



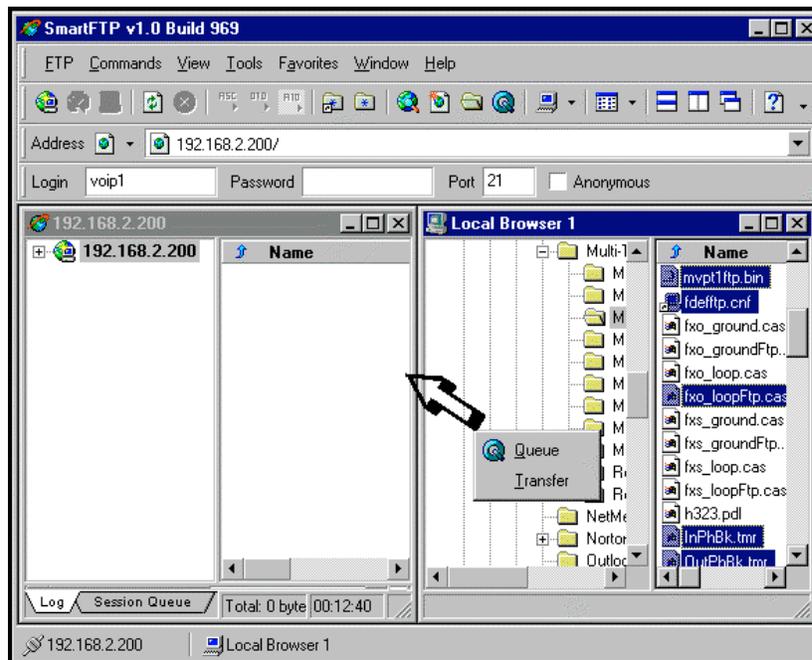
File transfer between PC and voip will look like transfer within voip directories.



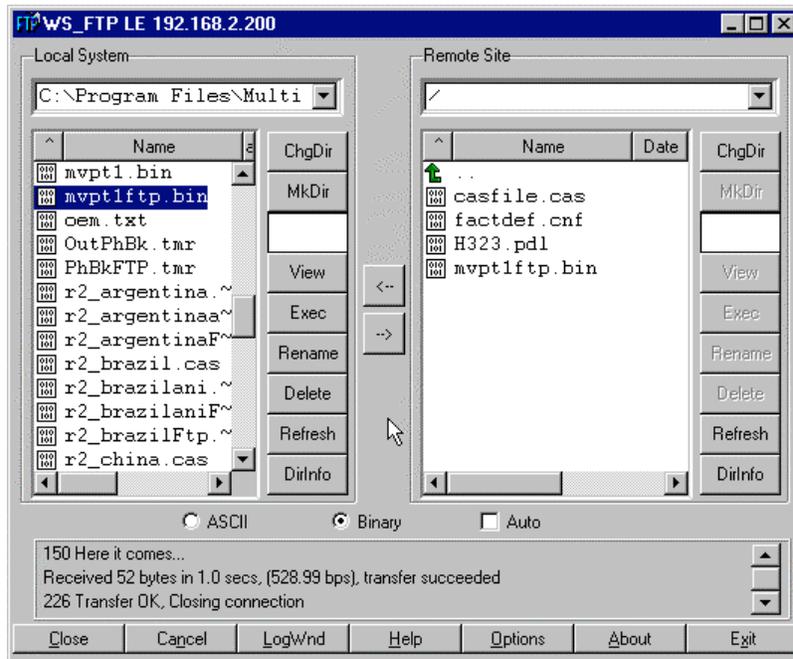
8B. Download with FTP Client Program.

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.

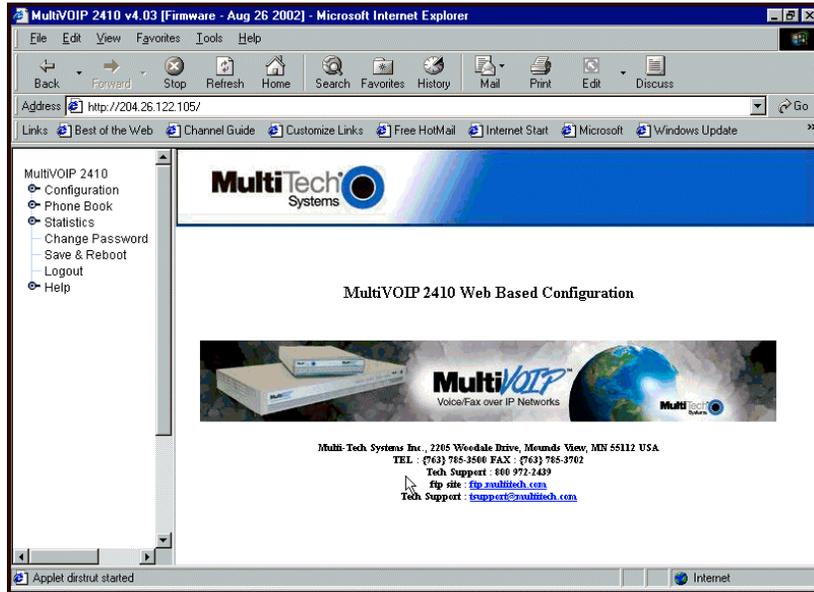


9. **Verify Transfer.** The files transferred will appear in the directory of the MultiVOIP.



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

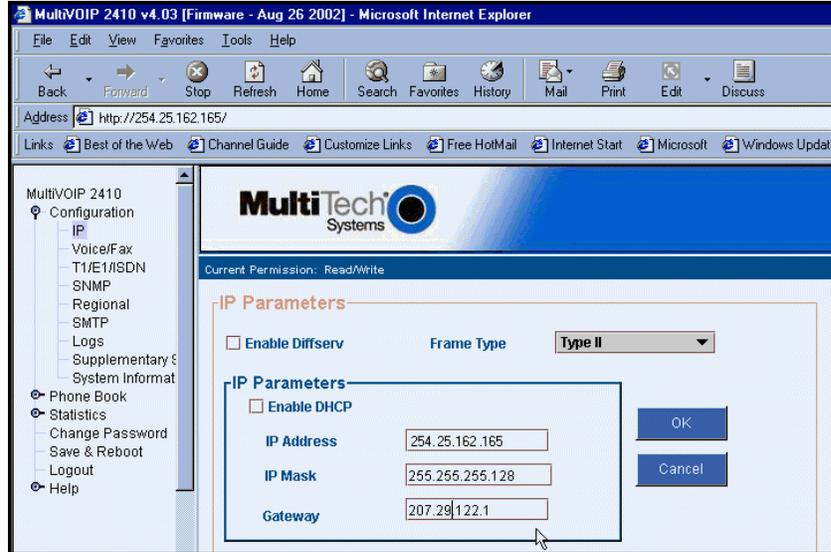


Beginning with the 4.03/6.03 software release, you can control the MultiVOIP unit with a graphic user interface (GUI) based on the common web browser platform. Qualifying browsers are InternetExplorer6 and Netscape6.

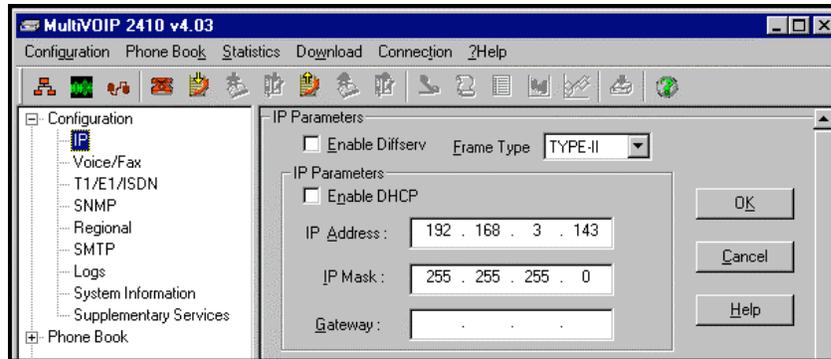
MultiVOIP Web Browser GUI Overview	
Function	Remote configuration and control of MultiVOIP units.
Configuration Prerequisite	Local Windows GUI must be used to assign IP address to MultiVOIP.
Browser Version Requirement	Internet Explorer 6.0 or higher; or Netscape 6.0 or higher
Java Requirement	Java Runtime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



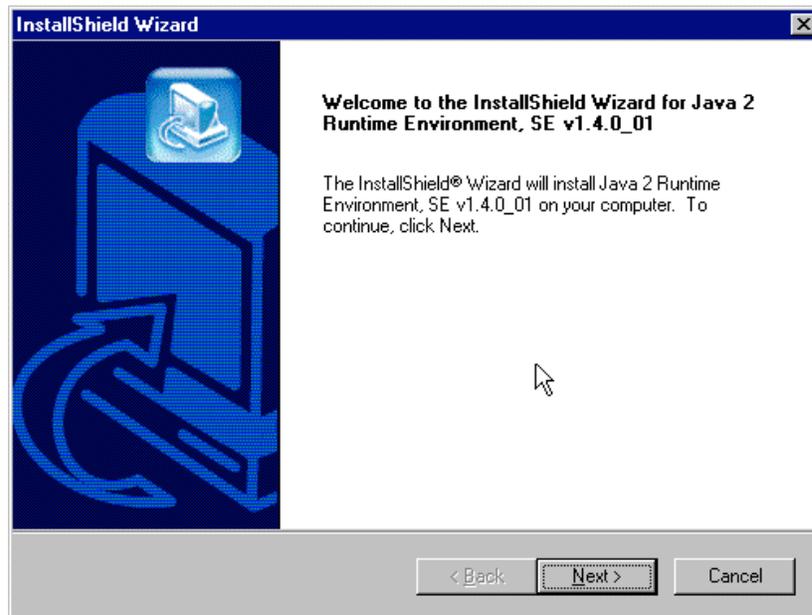
The web GUI, however, cannot perform logging in the same direct mode done in the Windows GUI. However, when the web GUI is used, logging can be done by email (SMTP).

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.

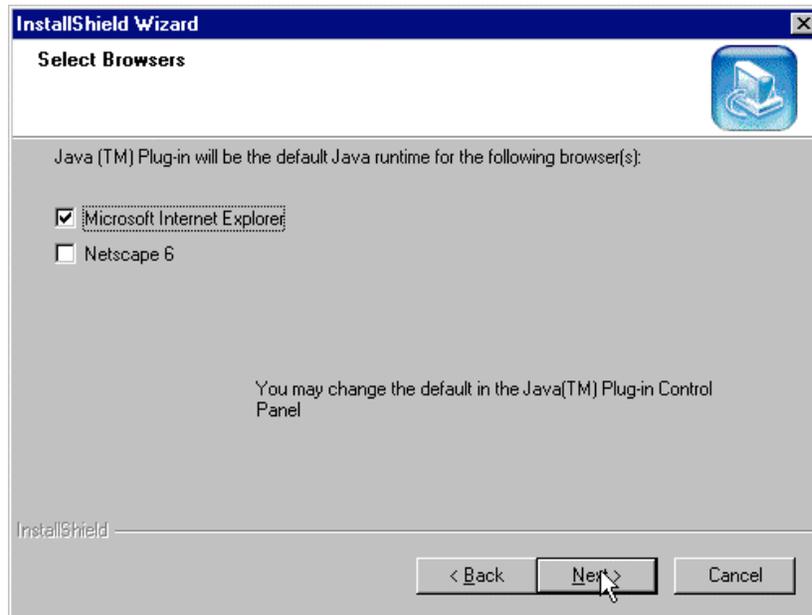
The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

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.

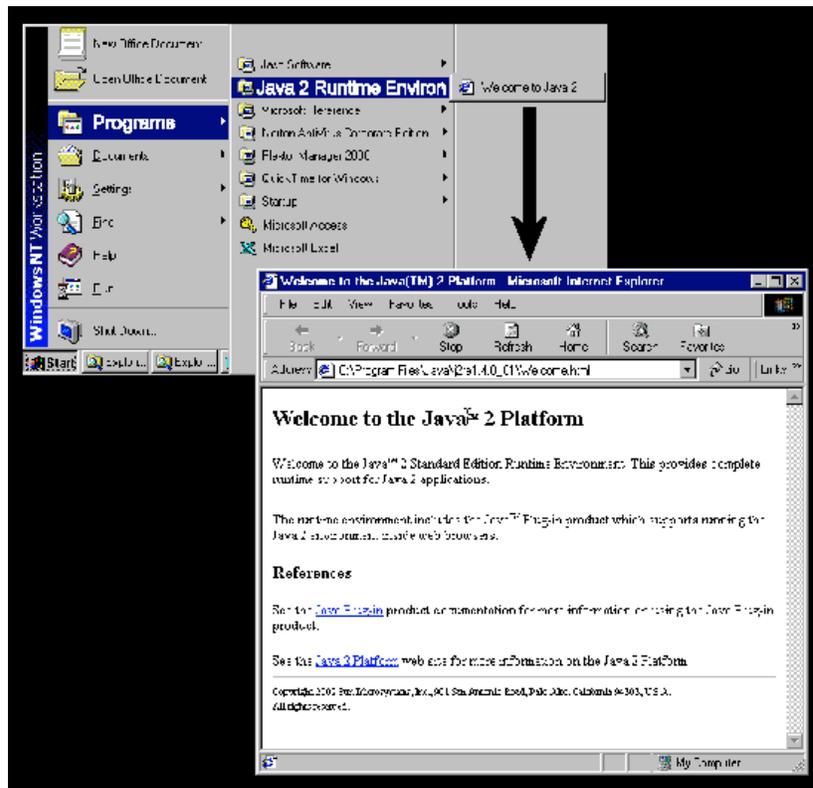
To install the Java program, go to the **Java** directory on the MultiVOIP product CD. Double-click on the EXE file to begin the installation. Follow the instructions on the Install Shield screens.



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

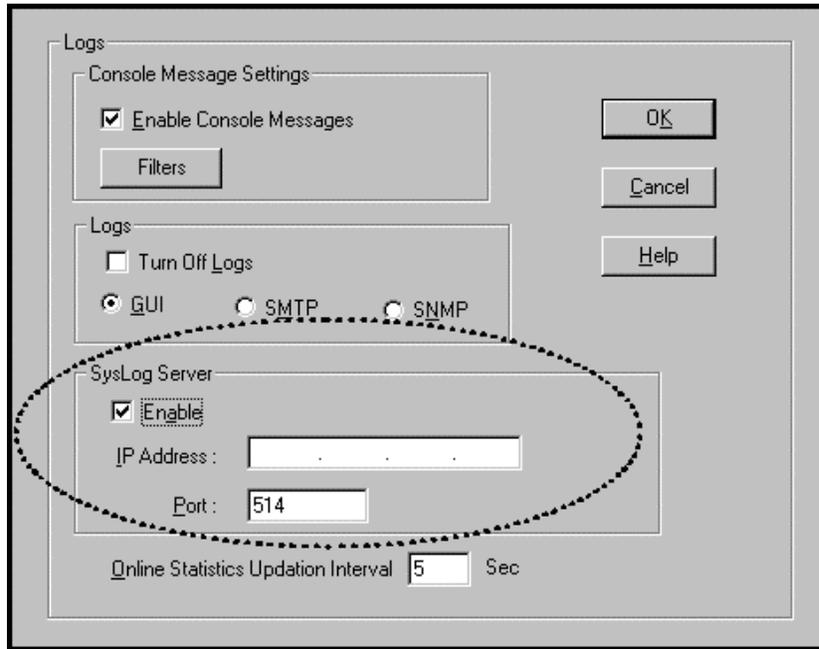
Beginning with the 4.03/6.03 software release, we have built SysLog server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.

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 screenshot displays the 'Logs' configuration window. It is divided into several sections:

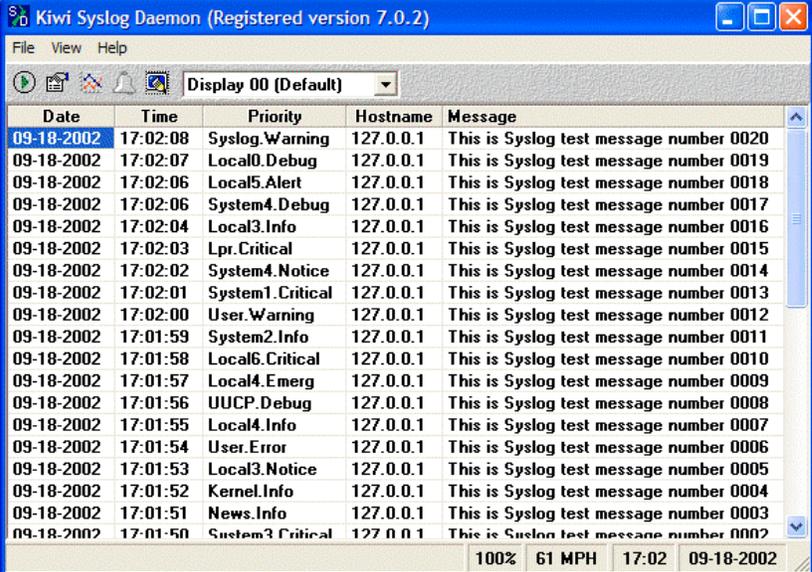
- Console Message Settings:** Contains a checked checkbox for 'Enable Console Messages' and a 'Filters' button.
- Logs:** Contains a 'Turn Off Logs' checkbox (unchecked) and three radio buttons: 'GUI' (selected), 'SMTP', and 'SNMP'.
- SysLog Server:** This section is circled with a dashed line. It includes a checked 'Enable' checkbox, an empty 'IP Address' text field, and a 'Port' field containing the value '514'.
- Online Statistics Update Interval:** A field containing the value '5' followed by the unit 'Sec'.

On the right side of the window, there are three buttons: 'OK', 'Cancel', and 'Help'.

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.



The screenshot shows the Kiwi Syslog Daemon application window. The title bar reads "Kiwi Syslog Daemon (Registered version 7.0.2)". The menu bar includes "File", "View", and "Help". Below the menu bar is a toolbar with several icons and a dropdown menu set to "Display 00 (Default)". The main area is a table with the following columns: Date, Time, Priority, Hostname, and Message. The table contains 20 rows of log entries, all from 09-18-2002. The messages are test messages with various priorities and hostnames. At the bottom of the window, there is a status bar showing "100%", "61 MPH", "17:02", and "09-18-2002".

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: Embedded Gatekeeper (for MVP-210G/410G/810G)

Introduction to Embedded Gatekeeper

This chapter describes how to configure and manage the MultiVOIP Gatekeeper software. The software comes pre-installed on the specially-equipped analog MultiVOIP units, MVP210G, MVP410G, and MVP810G. With gatekeeper functionality, network managers can define and control the flow of H.323 voice traffic across the IP network. In this chapter, we will present both a general description of how gatekeepers work and very specific information on how MultiTech's embedded gatekeeper units operate. In cases where the actual gatekeeper functionality implemented in the current software release differs from theoretically possible gatekeeper functionality, the differences will be noted (i.e., we describe some gatekeeper functionality that will only become available in a later software release and note all such cases).

A gatekeeper unit controls a "zone" on the IP network. (In fact, that is how a H.323 zone is defined; as the set of endpoints controlled by a gatekeeper.) One gatekeeper unit is needed to control a single zone. Therefore, when gatekeeper control is used, it's not necessary that all voip gateways within the system should be gatekeeper equipped – only one per zone is needed.

Network managers can configure, monitor, and manage the activity of registered network endpoints (including voip gateway units like the MVP210G/410G/810G). They can set policies and control bandwidth usage, thus customizing their network for better advantage.

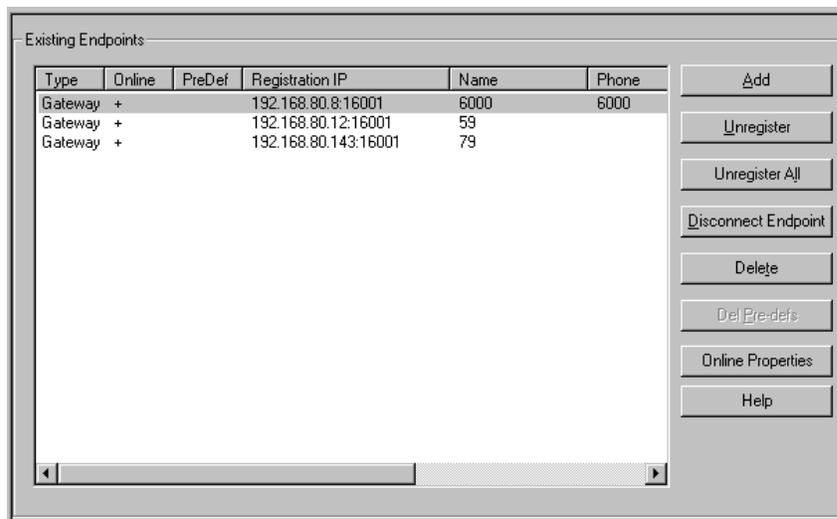
Gatekeeper facilitates interoperability between PBX dial plans and IP-based terminals. With it, call centers can route calls on the basis of need and implement other automatic call distribution features, as well.

Getting Started with the Gatekeeper-Equipped MultiVOIP

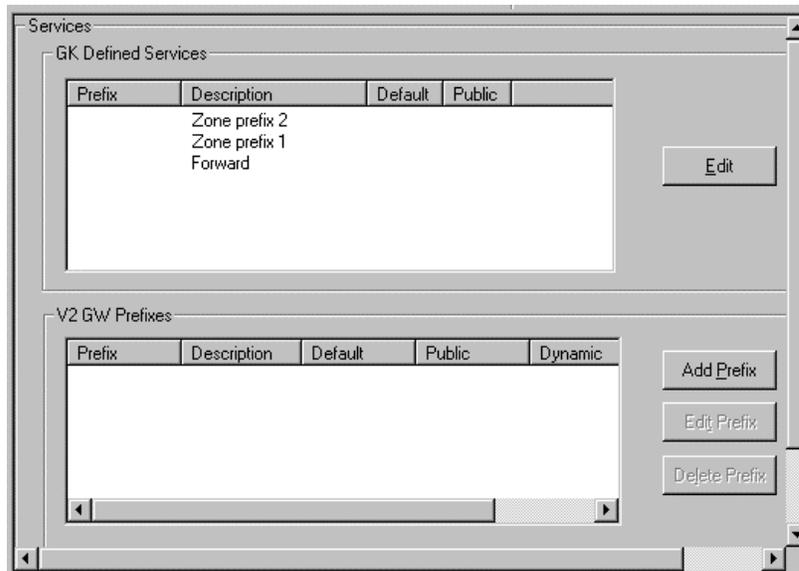
MultiVOIP units equipped with embedded gatekeeper functionality (MVP210G, MVP410G, or MVP810G) require configuration of their gatekeeper parameters before they can control a group of voip gateways. (This configuration is in addition to setting the technical parameters and phonebook parameters that are needed for the *gateway* functionality of these MultiVOIP units.)

Gatekeepers can be configured to enact a wide range of functionality, but they are primarily node points that direct and manage traffic to other endpoints. The essential question of “*whose messages go where?*” can be answered either by a gatekeeper that acts as a coordinating node or clearinghouse for the system or by phonebooks coordinated among the set of peer endpoints (gateways) that make up the system.

In its role as a node point, the gatekeeper directs call traffic between pairs of endpoints engaged in the call. To facilitate this node-point control, all endpoints (voip gateways) must be registered with the gatekeeper. This registration is done in the **Gatekeeper | Existing Endpoints** screen.



The basic function of directing calls to specified endpoints is done differently in gatekeeper-controlled systems than in systems controlled only by phonebooks. Phonebooks use “destination patterns” like area codes and local prefixes to route calls to specific endpoints. When gatekeepers perform this directive function, they do so by using “services,” which one configures in the **Gatekeeper | Services** screen.



Suppose a voip system consists of three endpoints in three different cities all having different area codes. If this voip system were controlled only by phonebooks, three different *destination patterns* (at least) would be needed; if controlled by a gatekeeper, three different *services* (at least) would be needed.

Matched Settings in Gatekeeper, Phonebook, & Tech Config Screens.

Generally, gatekeeper-equipped MultiVOIP units should be configured in this order:

1. Technical Configuration (setup for IP, voice/fax, telephony, etc.)
2. Phonebook Configuration (destination patterns, RAS settings, etc.)
3. Gatekeeper Configuration (listing endpoints, setting up services)

Also, generally, it's best to configure the gatekeeper-equipped MultiVOIP as fully as possible before configuring other gateways in the system. This is so because certain parameters that describe the gatekeeper unit must be entered the configuration screens of the ordinary voip gateway units.

Furthermore and very importantly, several settings needed in the **Gatekeeper | Existing Endpoints** screen and in the

Gatekeeper | Services screen must also be set in the Phonebook Configuration screen. In fact, if the ordered sequence above is followed (tech config, phonebook config, gatekeeper config), the software will automatically transfer several needed phonebook RAS parameters into the fields where they are required in the gatekeeper screens.

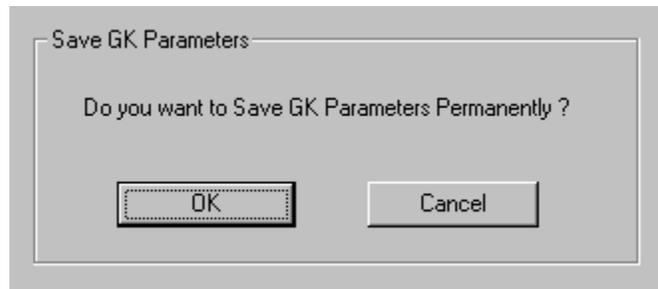
Full details on all of the gatekeeper configuration screens are presented in the “MultVOIP Gatekeeper Software Screens” section later in this chapter.

Saving the Gatekeeper Configuration. Just as you must save the technical configuration parameters and the phonebook configuration parameters, so also gatekeeper parameters must be saved in a separate step. In the sidebar menu, go to

Save Setup | Save GK Parameters.



A dialog box will appear to confirm that you want to invoke the ‘save’ function.



A second dialog box will appear to confirm that the save has been executed successfully.



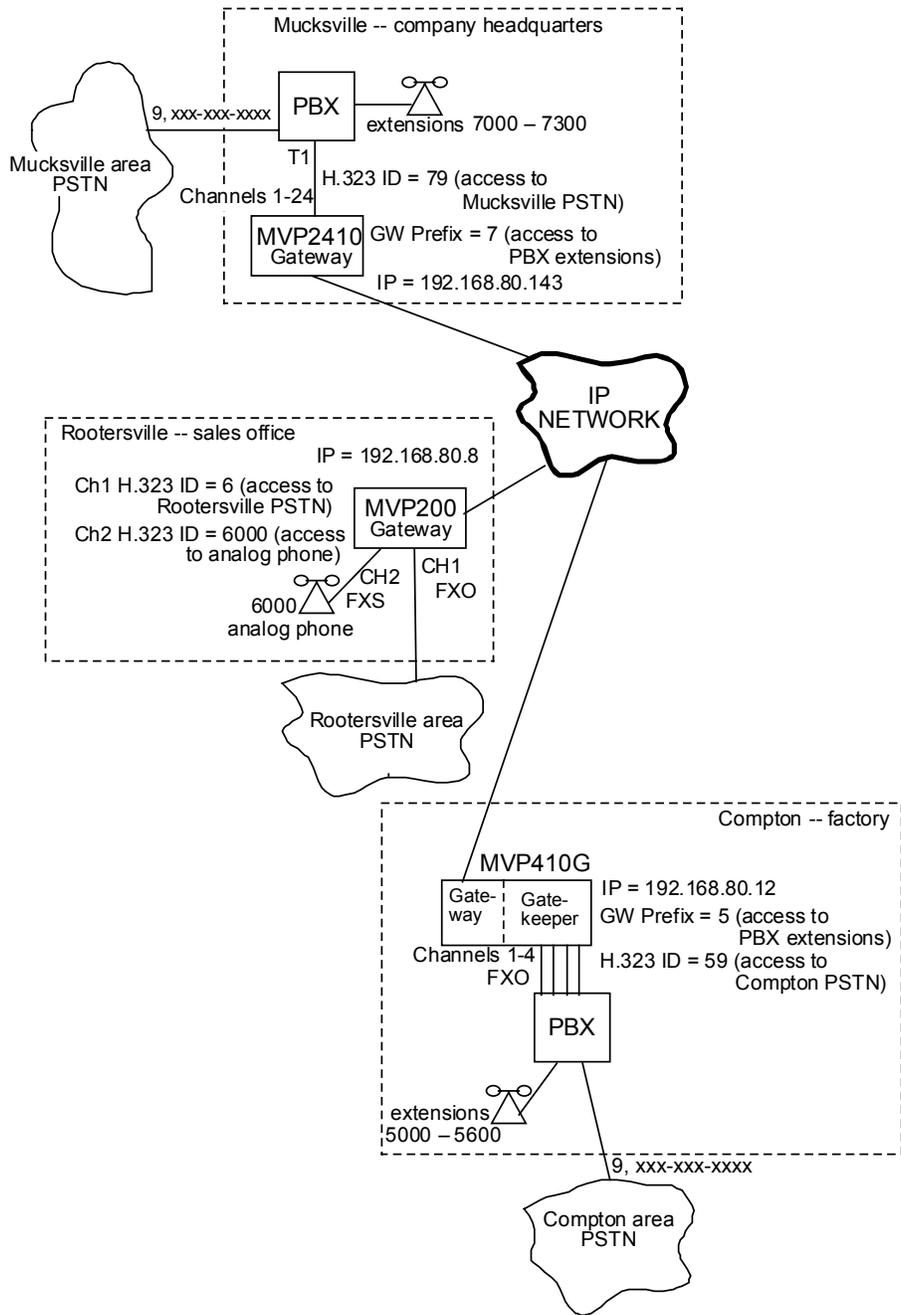
Embedded Gatekeeper System Example

The present example shows a voip system with three gateways, one of whose embedded gatekeeper functionality directs voip traffic in the system. The system design will give phone users at each office toll-free access to both the company employee phones (most are on PBXs) at the remote sites as well as the local PSTNs surrounding the remote sites.

The gatekeeper equipped MultiVOIP is an analog model (MVP410G) whose four channels are all connected (via FXO interface) to a PBX at a company's factory site in "Compton." The second gateway is a T1 digital voip gateway (MVP2410) connected to a PBX at the company's headquarters in "Mucksville." The third gateway, located in one of the company's small sales offices in "Rootersville," is a first-generation MultiTech gateway with two analog channels (MVP200), one serving an analog phone (via FXS interface) and the other giving access to its local area PSTN (via FXO interface).

To implement this configuration, we start with the gatekeeper-equipped MultiVOIP at the Compton site.

1. **MVP410G.** For the MVP410G at Compton, we need first to configure its phonebook with the gatekeeper configuration in mind. (We'll presume that its technical configuration has already been completed. Its IP address would have been set in the **Configuration | IP Parameters** screen and its four channels would have been set to "FXO" in its **Configuration | Interface** screen.)



The required MVP410G phonebook configuration is shown below.

“Compton” MVP410G Gateway Functions and Settings			
Function	PhBk Config Scn Settings ¹	Inbound PhoneBook Screen Settings	Phone User’s Actions
Put MVP410G gateway under gatekeeper control	Gatekeeper IP Address = 192.168.80.12	--	--
Give remote users access to Compton factory PBX extensions	Gateway Prefix = 5	Remove Prefix = 5; Add Prefix = 5	Dial 4 digits beginning with “5”
Give remote users access to Compton area PSTN	Gateway H.323 ID = 59	Remove Prefix = 59; Add Prefix = 9	Dial “59” plus Compton local number
		Outbound PhoneBook Screen Settings	
Get access to Mucksville office PBX extensions	--	Destination Pattern = 7 RemovePrefix = 7 Select “Use GateKeeper” Gateway H.323ID = none Gateway Prefix = 7	Dial 4 digits beginning with “7”
Get access to Mucksville area PSTN	--	Destination Pattern = 79 RemovePrefix = none Select “Use GateKeeper” Gateway H.323ID = 79 Gateway Prefix = none	Dial “79” plus Mucksville local number
Get access to Rootersville office phone	--	Destination Pattern = 6000 RemovePrefix = none Select “Use GateKeeper” Gateway H.323ID = 6000 Gateway Prefix = none	Dial 6000.
Get access to Rootersville area PSTN	--	Destination Pattern = 6 RemovePrefix = none Select “Use GateKeeper” Gateway H.323ID = 6 Gateway Prefix = none	Dial “6”; get second dial tone. Dial Hoot #.
1. “PhoneBook Configuration screen settings”			

2. **MVP410G.** We begin with the PhoneBook Configuration screen. Because the MVP410G serves as a gatekeeper for its own gateway, the Gatekeeper IP Address is the same as the gateway's regular IP address, as set in the IP Parameters screen.

Compton MVP410G MultiVOIP

The screenshot displays the 'Phone Book Configuration' window for the Compton MVP410G MultiVOIP. The window is titled 'Compton MVP410G MultiVOIP' and contains the following fields and options:

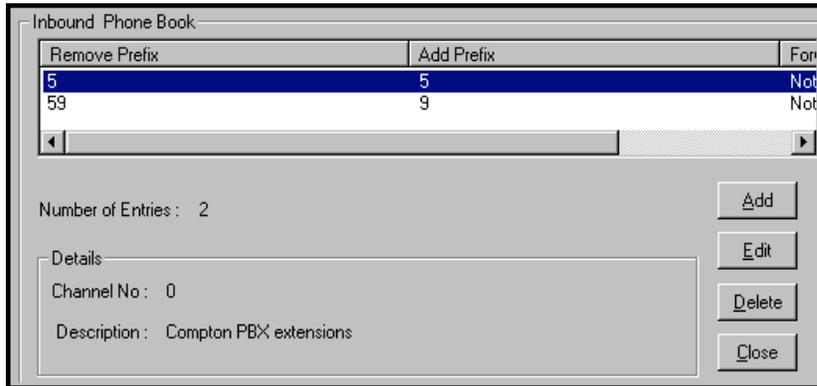
- Gateway Name:** Compton 410G #1
- Q.931 Parameters:**
 - Use Fast Start
 - Call Signaling Port: 1720
- Register with GateKeeper
- Gatekeeper RAS Parameters:**
 - Gatekeeper IP Address: 192 . 168 . 80 . 12
 - Port Number: 1719
 - Gateway Prefix: 5
 - Gatekeeper Name: MVP_IGK
 - Gateway H323 ID: 59
 - RAS TTL Value: 60 secs
- Enable SIP Proxy
- SIP Proxy Parameters:**
 - Proxy Server IP Address: 0 . 0 . 0 . 0

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

We have set the Gateway Prefix to 5 to give voip system phone users access to Compton office PBX extensions (this value will appear in the **Gateway | Services | V2 GW Prefixes** screen; see step 8). Because we have set the Gateway Prefix (to “5”) in the PhoneBook Configuration screen during the Phonebook Configuration process, it will automatically appear in the Gatekeeper GUI. We have set the Gateway H.323 ID to 59 to give voip system users access to the Compton area PSTN. The Gateway H.323 ID of 59 will need to be added manually to the **GateKeeper | Services** screen under “GK Defined Services.” The Gatekeeper Name can be customized for your needs. “MVP_IGK” is the default value.

3. **MVP410G.** The Inbound Phonebook of the MVP410G requires two entries, one for access to Compton PBX extensions, another for access to the Compton area PSTN.

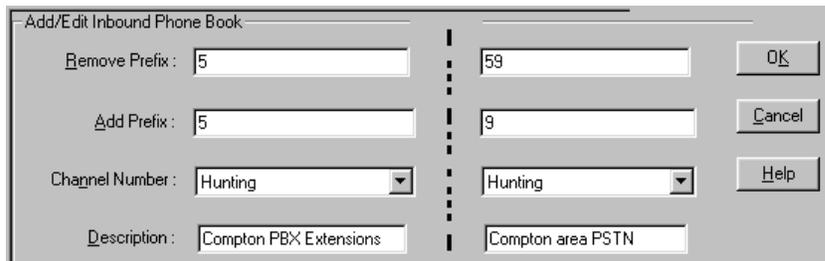
Compton MVP410G MultiVOIP



To create each of these entries, you must click on “Add” at the Inbound PhoneBook screen and enter the details for each entry in a separate Add/Edit Inbound PhoneBook screen, as shown below.

Compton MVP410G MultiVOIP: Adding Inbound Phonebook Entries

giving remote users access to local PBX ... and ... to the local area PSTN



4. **MVP410G**. The Outbound Phonebook of the MVP410G requires four entries.

Compton MVP410G MultiVOIP

Destination Pattern	IP Address	Protocol	Description	Alternate
6		H.323	Rootersville PSTN calls	
6000		H.323	Rootersville Analog phone	
7		H.323	Mucksville PBX extensions	
79		H.323	Mucksville area PSTN calls	

Number of Entries : 4

Details

Remove Prefix :
Add Prefix :
Gatekeeper : used
Gateway H.323 ID : 6
Gateway Prefix :
Q.931 Port : 1720
Transport Protocol :
SIP URL :
Round Trip Delay : 300 ms
Alternate Phone Number :

Add
Edit
Delete
Close
Help

Two outbound phonebook entries are for Rootersville, one describing access to its local PSTN and the other describing access to its office phone. To create each of these entries, you must click on “Add” at the Outbound PhoneBook screen and enter the details for each entry in a separate Add/Edit Outbound PhoneBook screen.

Compton MVP410G MultiVOIP: Adding Outbound Phonebook Entries
gaining access to a remote area PSTN ... and ... to a remote office phone

The screenshot displays the 'Add/Edit Outbound Phone Book' configuration window, which is split into two columns for creating two different entries. The left column is for 'Rootersville PSTN calls' and the right column is for 'Rootersville Analog phone'. Both columns have identical field structures:

- Phone Number Details:**
 - Destination Pattern: [6]
 - Total Digits: [0]
 - Remove Prefix: []
 - Add Prefix: []
- IP Address:** [0 . 0 . 0 . 0 . 0]
- Description:** [Rootersville PSTN calls]
- Protocol Type:**
 - [] SIP [x] H.323 [] SPP
- H.323:**
 - Use GateKeeper
 - Gateway H.323 ID: [6]
 - Gateway Prefix: []
 - Q.931 Port Number: [1720]

On the right side of the window, there are control buttons: OK, Cancel, Help, and an 'Advanced' button.

Another two outbound phonebook entries are for Mucksville for access to its PBX extensions and its local PSTN.

Compton MVP410G MultiVOIP: Adding Outbound Phonebook Entries
gaining access to a remote site PBX ... and ... to a remote area PSTN

The screenshot shows a configuration window titled "Add/Edit Outbound Phone Book" with two columns of settings. The left column is for "Mucksville PBX extensions" and the right column is for "Mucksville area PSTN calls".

Field	Mucksville PBX extensions	Mucksville area PSTN calls
Destination Pattern	7	79
Total Digits	0	0
Remove Prefix	7	
Add Prefix		
IP Address	0 . 0 . 0 . 0	0 . 0 . 0 . 0
Description	Mucksville PBX extensions	Mucksville area PSTN calls
Protocol Type	<input type="radio"/> SIP <input checked="" type="radio"/> H.323 <input type="radio"/> SPP	<input type="radio"/> SIP <input checked="" type="radio"/> H.323 <input type="radio"/> SPP
H.323 Use GateKeeper	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
Gateway H.323 ID		79
Gateway Prefig	7	
Q.931 Port Number	1720	1720

Buttons on the right side of the window include OK, Cancel, Help, and Advanced.

5. **MVP410G.** Save the MVP410G PhoneBook Configuration (the Save Setup command is in the sidebar menu) before proceeding to gatekeeper configuration. Click on **Save & Reboot** and then click **OK** on the screen that will appear directly thereafter.

6. **MVP140G Gatekeeper Function.** We will configure the gatekeeper function of the MVP410G at Compton as summarized in the table below. It is useful to begin the configuration process by listing the functionality that you want to implement in your system.

“Compton” Gatekeeper Functions & Settings			
Function	GK Services Screen Settings	GK General Settings Screen	Phone User's Actions
Activate gatekeeper function of MVP410G	--	Reg Pol. = All Endpts Accepts Calls Y GK Active Y	--
		GK Service Properties Screen Settings	
Access to Compton factory PBX extensions	V2 GW Prefix = TEL:5 As set in PhoneBook Configuration screen, Gateway Prefix field of Compton MVP410G voip.	“Allow as default to online endpoints” = Y “Allow as public for Out-of-Zone Endpoints” = Y	Dial 4 digits beginning with “5”
Access to Compton area PSTN	GK Defined Services Prefix = 59	“Allow as default to online endpoints” = Y	Dial “59” plus Compton local number
Access to Mucksville office PBX extensions	V2 GW Prefix = TEL:7 As set in PhoneBook Configuration screen, Gateway Prefix field of Mucksville MVP2410 voip.	“Allow as default to online endpoints” = Y “Allow as public for Out-of-Zone Endpoints” = Y	Dial 4 digits beginning with “7”
Access to Mucksville area PSTN	GK Defined Services Prefix = 79	“Allow as default to online endpoints” = Y	Dial “79” plus Mucksville local number
Access to Rootersville office phone	GK Defined Services Prefix = 6000	“Allow as default to online endpoints” = Y	Dial 6000.
Access to Rootersville area PSTN	GK Defined Services Prefix = 6	“Allow as default to online endpoints” = Y	Dial “6”. Dial local R’ville number.

7. **MVP410G.** Begin at the GK General Settings screen. The required settings are default values.

Compton MVP410G MultiVOIP Gatekeeper

GK General Settings

Registration Policy

No Endpoints

Predefined Endpoints

All Endpoints

Activity Configuration

Accepts Calls

GK Active

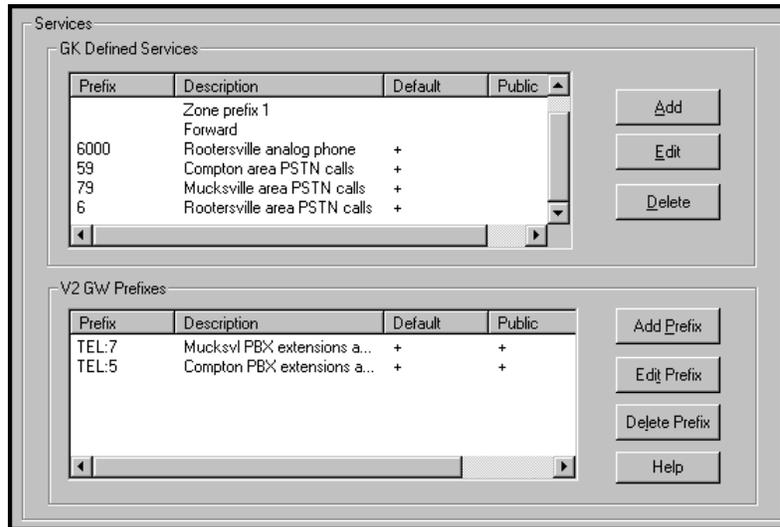
Debug Level: 10

Memory Settings

OK

8. MVP410G. Adding “services” and “prefixes” in the gatekeeper Services screen fulfills the same role as setting “destination patterns” in outbound phonebook screens. Even though they serve a function similar to destination patterns, the “service” and “prefix” gatekeeper entries do not eliminate the need for phonebook destination patterns; nor do phonebook destination patterns eliminate the need for gatekeeper services and prefixes. They all work together and all must be present for proper operation. (Note also that “Services” constitutes a wider category than we are discussing here. Generally, services can also be, essentially, features, like call forwarding.)

Compton MVP410G MultiVOIP Gatekeeper



To create each of the four required ‘GK-Defined-Services’, you must click on “Add” in the Gatekeeper **Services** screen and enter the details for each entry in a separate **Service Properties** screen, as shown below.

Compton MVP410G MultiVOIP Gatekeeper

The image displays four overlapping 'Service Properties' dialog boxes for the Compton MVP410G MultiVOIP Gatekeeper. Each dialog box contains the following fields and options:

- Prefix:** A text input field.
- Description:** A text input field.
- Allowed as default to Online Endpoints
- Allowed as public for Out-of-Zone Endpoints
- Buttons: OK, Cancel, and Help.

The four dialog boxes are stacked, showing the following details for each:

- Top dialog: Prefix: 6000, Description: Rootersville analog phone.
- Second dialog: Prefix: 59, Description: Compton area PSTN calls.
- Third dialog: Prefix: 79, Description: Mucksville area PSTN calls.
- Bottom dialog: Prefix: 6, Description: Rootersville area PSTN calls.

To give network-wide access to the Compton factory PBX extensions, the Gateway Prefix field of the MVP410G's **PhoneBook Configuration** screen has already been set to 5 (in step 2 above) and this setting appears automatically in the **V2 GW Prefix** screen. (There is no need to add this item manually in the V2 GW Prefixes screen.) Similarly, to give network-wide access to the Mucksville office PBX extensions, the Gateway Prefix of the Mucksville MVP2410's **PhoneBook Configuration** screen must be set to 7. When this setting has been made, and when that voip contacts the MVP410G gatekeeper unit, the setting will appear automatically in the **V2 GW Prefix** screen of the Compton MVP410G gatekeeper/gateway unit. (Again, there is no need to add this item manually in the **Services | V2 GW Prefixes** screen pane.) The **Service Properties** screens for these two V2 GW Prefixes are shown below.

Compton MVP410G MultiVOIP Gatekeeper

The image displays two overlapping 'Service Properties' dialog boxes. The top dialog box has the following fields and options:

- Prefix: TEL:7
- Description: Mucksvl PBX extensions access
- Allowed as default to Online Endpoints
- Allowed as public for Out-of-Zone Endpoints
- Buttons: OK, Cancel, Help

The bottom dialog box has the following fields and options:

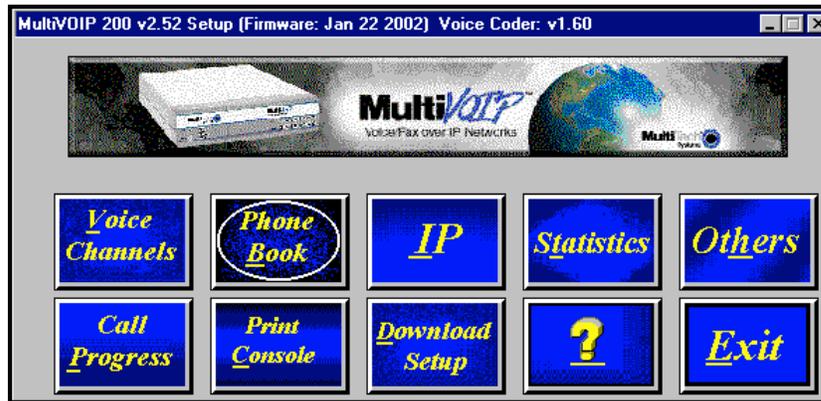
- Prefix: TEL:5
- Description: Compton PBX extensions acces
- Allowed as default to Online Endpoints
- Allowed as public for Out-of-Zone Endpoints
- Buttons: OK, Cancel, Help

9. **MVP410G.** Save the MVP410G gatekeeper configuration before configuring the other gateways in the system (the **Save Setup | Save GK Parameters** command is in the sidebar menu).
10. **MVP200.** A summary of the required MVP200 phonebook configuration is shown below. (We are presuming that the MVP200's IP address has been duly set in the **IP Parameters** screen and that its channels have been set in the **Voice Channels** screen as follows:
Ch1 = FXO; CH2 = FXS.) Again, it is useful to begin the configuration process by listing the system functionality that this particular voip unit will have to perform.

“Rootersville” MVP200 Gateway Functions & Settings			
Function	Phonebook Directory Data-Base screen settings	Add/Edit Phone-Book Entries screen settings	Phone User’s Actions
Put MVP200 gateway under gatekeeper control	Select “GateKeeper” radio button. RAS Parameters IP Address = 192.168.80.12;	IP Address = 192.168.80.8	--
Allow remote users access to Rootersville office phone	Phone Number = 6000 Destination Details = 6000	Phone Number = 6000 Ch2 H.323 ID = 6000	Dial “6000”
Allow remote users access to Rootersville area PSTN	Phone Number = 6 Destination Details = 6	Phone Number = 6 Ch1 H.323 ID = 6	Dial “6”. Dial local R’ville phone number.
Get access to Compton factory PBX extensions	<i>These functions are provided by gatekeeper within MVP410G.</i>		Dial 4 digits beginning with “5”
Get access to Compton area PSTN			Dial “59” plus Compton local number
Get access to Mucksville office PBX extensions			Dial 4 digits beginning with “7”
Get access to Mucksville area PSTN			Dial “79” plus Mucksville local number

11. MVP200. From the main MultiVOIP200 screen, select Phone Book.

Rootersville MVP200 MultiVOIP



12. **MVP200.** In the Phone Directory Database screen, click on the “Gatekeeper” radio button to put the MVP200 under the control of the MVP410G gatekeeper. Under “RAS Parameters” in the IP Address field, enter the IP address of the gatekeeper. In this case, since the MVP410G uses a single IP address for both its gateway and its gatekeeper functions, we simply use the MVP410G’s regular (and only) IP address (192.168.80.12). Then add the two required destination patterns: **6000** will direct calls to the analog phone in the Rootersville office; **6** will give remote users access to the Rootersville area PSTN (calls can be completed in a single dialing sequence).

Rootersville MVP200 MultiVOIP

MultiVOIP 200 - Phone Directory Database

Buttons: **Add (+)**, **Delete (-)**, **Edit**

Phone Number	Destination Details	Description
6	6	Rootersville PSTN calls
6000	6000	Rootersville analog phone

Number of Entries: 2

GateKeeper Proprietary PhoneBook

Q.931 Parameters:

Use Fast Start

Call Signalling Port: 1720

RAS Parameters:

IP Address: 192.168.80.12

Port Number: 1719

Buttons: **OK**, **Cancel**, **?**

MultiVOIP 200 - Add/Edit Phone Entry

Station Information:

Phone Number: 6

Description: Rootersville PSTN calls

Voice Channel: 1

Station Identification:

H323 ID: 6

IP Address: 192.168.80.8

Port: 1720

Buttons: **OK**, **Cancel**, **?**, **Copy From**

MultiVOIP 200 - Add/Edit Phone Entry

Station Information:

Phone Number: 6000

Description: Rootersville analog phone

Voice Channel: 2

Station Identification:

H323 ID: 6000

IP Address: 192.168.80.8

Port: 1720

Buttons: **OK**, **Cancel**, **?**, **Copy From**

13. **MVP200.** When you have completed the configuration, click **OK** on the Phonebook Directory Database screen. Then go to the MultiVOIP 200 main screen and click on **Download Setup** to save the configuration.

14. **MVP2410.** The required MVP2410 phonebook configuration is shown below. We are presuming here that technical configuration is already complete so that the MVP2410's IP address and other technical configuration parameters have already been duly set.

"Mucksville" MVP2410 Gateway Functions and Settings			
Function	PhBk Config Scn Settings ¹	Inbound PhoneBook Screen Settings	Phone User's Actions
Put MVP2410 under control of gatekeeper	Gatekeeper IP Address = 192.168.80.12	--	--
Give remote users access to Mucksville office PBX extensions	Gateway Prefix = 7	Remove Prefix = 7; Add Prefix = 7	Dial 4 digits beginning with "7"
Give remote users access to Mucksville area PSTN	Gateway H.323 ID = 79	Remove Prefix = 79; Add Prefix = 9	Dial "79" plus Mucksville local number
		Outbound PhoneBook Screen Settings ³	
Get access to Compton factory PBX extensions		Destination Pattern = 5 RemovePrefix = 5 Select "Use GateKeeper" Gateway H.323ID = none Gateway Prefix = 5	Dial 4 digits beginning with "5"
Get access to Compton area PSTN		Destination Pattern = 59 RemovePrefix = none Select "Use GateKeeper" Gateway H.323ID = 59 Gateway Prefix = none	Dial "59" plus Compton local number
Get access to Rootersville office phone	--	Destination Pattern = 6000 RemovePrefix = none Select "Use GateKeeper" Gateway H.323ID = 6000 Gateway Prefix = none	Dial 6000.
Get access to Rootersville area PSTN	--	Destination Pattern = 6 RemovePrefix = none Select "Use GateKeeper" Gateway H.323ID = 6 Gateway Prefix = none	Dial "6". Dial R'ville local phone number.
1. "PhoneBook Configuration screen settings"			

15. **MVP2410.** For the MVP2410 at Mucksville, we begin again with the PhoneBook Configuration screen. Because the MVP410G serves as a gatekeeper for the MVP2410, the MVP410G's IP address is the Gatekeeper IP Address for the MVP2410.

Mucksville MVP2410 MultiVOIP

Phone Book Configuration

Gateway Name :

Q.931 Parameters

Use Fast Start

Call Signaling Port :

Register with GateKeeper

Gatekeeper RAS Parameters

Gatekeeper IP Address :

Port Number :

Gateway Prefix :

Gatekeeper Name :

Gateway H323 ID :

RAS TTL Value : secs

Enable SIP Proxy

SIP Proxy Parameters

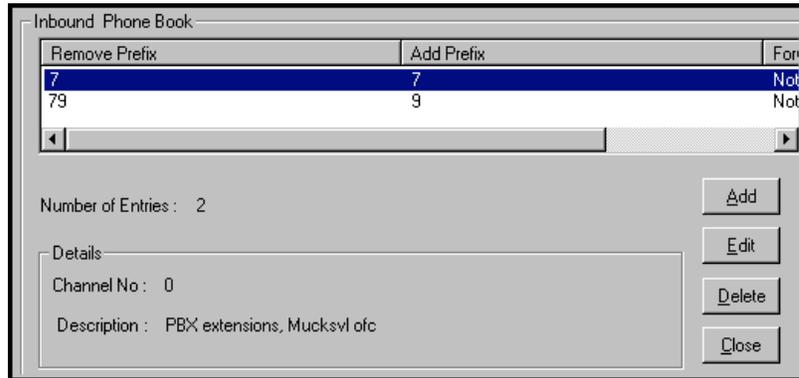
Proxy Server IP Address :

OK Cancel Help

We have set the Gateway Prefix to 7 to give voip system phone users access to Mucksville office PBX extensions. Because we have set the Gateway Prefix (to “7”) in the PhoneBook Configuration screen during the Phonebook Configuration process, it will automatically appear in the Gatekeeper GUI. We have set the Gateway H.323 ID to 79 to give voip system users access to the Mucksville area PSTN. The Gateway H.323 ID of 79 will need to be added manually to the **GateKeeper | Services** screen under “GK Defined Services.” The Gatekeeper Name can be customized for your needs. “MVP_IGK” is the default value.

16. **MVP2410.** The Inbound Phonebook of the MVP2410 requires two entries, one for access to Mucksville PBX extensions, another for access to the Mucksville area PSTN.

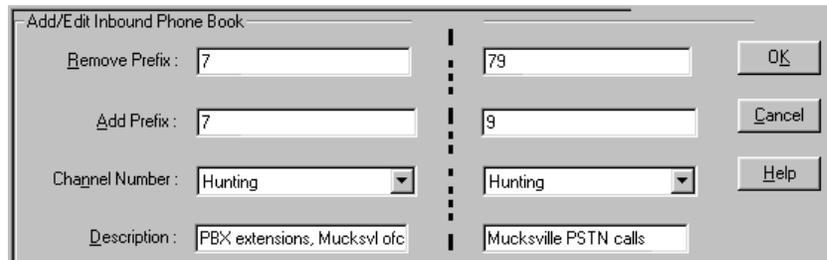
Mucksville MVP2410 MultiVOIP



To create each of these entries, you must click on “Add” at the Inbound PhoneBook screen and enter the details for each entry in a separate Add/Edit Inbound PhoneBook screen, as shown below.

Mucksville MVP2410 MultiVOIP: Adding Inbound Phonebook Entries

giving remote users access to local PBX ... and ... to the local area PSTN



17. **MVP2410.** The Outbound Phonebook of the MVP2410 requires four entries.

Mucksville MVP2410 MultiVOIP

Outbound Phone Book

Destination Pattern	IP Address	Alternate IP Address	Description
5			Compton PBX Extensions
59			Compton PSTN calls
6			Rootersville PSTN calls
6000			Rootersville Analog phone

Number of Entries : 4

Details

Remove Prefix : 5

Add Prefix :

Gatekeeper : used

Gateway H.323 ID :

Gateway Prefix : 5

Q.931 Port : 1720

Transport Protocol :

SIP URL :

Round Trip Delay : 300 ms

Add

Edit

Delete

Close

Help

Two outbound phonebook entries are to gain access to Compton’s PBX extensions and its local PSTN. To create each of these entries, you must click on “Add” at the Outbound PhoneBook screen and enter the details for each entry in a separate Add/Edit Outbound PhoneBook screen.

Mucksville MVP2410G MultiVOIP: Adding Outbound Phonebook Entries
gaining access to a remote site PBX ... and ... to a remote area PSTN

The screenshot displays the 'Add/Edit Outbound Phone Book' configuration window, which is split into two columns for creating two different entries. The left column is for 'Compton PBX Extensions' and the right column is for 'Compton PSTN calls'. Both columns have identical field structures:

- Phone Number Details:**
 - Destination Pattern: 5
 - Total Digits: 0
 - Remove Prefix: 5
 - Add Prefix: (empty)
- IP Address:** 0 . 0 . 0 . 0 . 0
- Description:** Compton PBX Extensions
- Protocol Type:**
 - SIP
 - H.323
 - SPP
- H.323 Settings:**
 - Use GateKeeper
 - Gateway H.323 ID: (empty)
 - Gateway Prefix: 5
 - Q.931 Port Number: 1720

On the right side of the window, there are control buttons: OK, Cancel, Help, and an 'Advanced' button.

Another two outbound phonebook entries are for Rootersville, one describing access to its local PSTN and the other describing access to its office phone.

Mucksville MVP2410 MultiVOIP: Adding Outbound Phonebook Entries
gaining access to a remote area PSTN ... and ... to a remote office phone

Add/Edit Outbound Phone Book	
Phone Number Details	
Destination Pattern : 6	6000
Total Digits : 0	0
Remove Prefix :	
Add Prefix :	
IP Address : 0 . 0 . 0 . 0 . 0	0 . 0 . 0 . 0 . 0
Description : Rootersville PSTN calls	Rootersville Analog phone
Protocol Type <input type="radio"/> SIP <input checked="" type="radio"/> H.323 <input type="radio"/> SPP	Protocol Type <input type="radio"/> SIP <input checked="" type="radio"/> H.323 <input type="radio"/> SPP
H.323 <input checked="" type="checkbox"/> Use GateKeeper	<input checked="" type="checkbox"/> Use GateKeeper
Gateway H.323 ID : 6	6000
Gateway Prefig :	
Q.931 Port Number : 1720	1720
	OK
	Cancel
	Help
	Advanced

18. **MVP2410.** Save the MVP2410 PhoneBook Configuration (the Save Setup command is in the sidebar menu).
19. **MVP410G.** The gatekeeper **Online Parameters** screen (go to **Gatekeeper | Endpoints** and click the “Online Parameters” button) for the Mucksville MVP2410 shows a useful summary of system capabilities and denotes those that have been enabled for the MVP2410 in particular.

**Mucksville MVP2410 MultiVOIP: its gatekeeper Online Parameters
(as seen in the Compton MVP410G’s MultiVOIP software display)**

“allowed” services are system-wide ... whereas ... “supported” services are those that are active in that particular voip endpoint

The screenshot shows the 'Online Parameters' configuration interface. At the top, there are fields for 'Online Properties for Endpoint' (set to 0) and 'Allowed Distance' (set to 0). Below this is the 'Endpoint type' dropdown menu, currently set to 'Gateway', and a 'Time To Live' field set to 20. The interface is divided into several sections:

- Names:** A list box containing the number '79'.
- Phones:** An empty list box.
- Other Aliases:** Fields for 'URL', 'Email', and 'Transport'.
- IP Addresses:** Fields for 'Registration IP' (192.168.80.143) and 'Port' (16001), and 'Call Signalling IP' (192.168.80.143) and 'Port' (1720).
- Party Number:** An empty field.
- Type:** An empty dropdown menu.
- GK Defined Services:** Two tables side-by-side.

Prefix	Description	Prefix	Description
6000	Rootersville Analog phor	79	Mucksville area PSTN
59	Compton area PSTN		
79	Mucksville area PSTN		
6	Rootersville area PSTN		
- V2 GW Prefixes:** Two tables side-by-side.

Prefix	Description	Pre...	Description
TEL:7	Mucksvl PBX extensions	TEL:7	Mucksvl PBX extensions
TEL:5	Compton PBX extensions		

The gatekeeper will route calls to an endpoint only if the service (dialing pattern) is supported by that endpoint. (Services may be “allowed” in the system but not “supported” by an endpoint.)

“GK Allowed Services” are the set of all services (roughly the equivalent of destination patterns in phonebooks) used in the voip system that the embedded gatekeeper is overseeing. “GK Supported Services” are all services (destination patterns) that direct calls to the MVP2410 gateway.

20. **Calls.** We will now consider examples of different types of voip calls that can be made within the system. We dial a sequence, complete the call, and then look at the **Call Progress** screen of the voip unit at which the call is completed.

21. MVP200. A call from the Rooterstown office to its local PSTN can be dialed 67637175592.

Rootersville MVP200 MultiVOIP

The screenshot displays the 'MultiVOIP 200 - Call Progress' window. At the top, there is a dropdown menu for 'Channel' set to 'Channel 1'. Below this, the 'Call Details' section lists various call parameters: Duration (00:00:05), Mode (Voice), Voice Coder (G.723.1 @ 6.3 kbps), Packets Sent (78), Packets Rcvd (83), Bytes Sent (4,392), Bytes Rcvd (3,796), Packets Lost (0), Outbound Digits (7637175592), Jitter (30), and Call Charges (\$ 0.00). To the right of these details are three buttons: 'Disconnect', a question mark icon, and 'Close'. Below the call details, the 'From' and 'To' information is shown: 'From: Analog phone on 200' and 'To: MVP200 PSTN call'. The 'From-->To Details' section provides a side-by-side comparison of parameters for both ends of the call, including Phone Number, IP Address, Interface, Firmware Version, and Options. At the bottom, there are legends for 'SC - Silence Compression' and 'FEC - Forward Error Correction', and a 'Status: Active' indicator.

From-->To Details	
Phone Number:	6000 6
IP Address:	192.168.80.8:2 192.168.80.8:1
Interface:	FXS Loop -----> FXO
Firmware Version:	MultiVolP v2.52 MultiVolP v2.52
Options:	SC SC

SC - Silence Compression FEC - Forward Error Correction
Status: Active

22. **MVP410G.** A call from the Rootersville analog phone to a PBX extension at the Compton office can be dialed 5592.

Compton MVP410G MultiVOIP

Call Progress Details
Call Progress Details for Channel 1

<p>Call Details</p> <p>Duration: 00:02:06 Mode: Voice Voice Coder: G.723.1 @ 6.3 kbps Packets Sent: 1,886 Packets Received: 1,786 Bytes Sent: 111,716 Bytes Received: 61,452 Packets Lost: 0 Outbound Digits: 5592 Prefix Matched: 5</p>	<p style="text-align: center;">Disconnect</p> <p style="text-align: center;">Exit</p> <p style="text-align: center;">Help</p>
---	---

From->To Details

From ----> To 6000	----> 5:1
Gateway Name:	Compton PBX extensions
IP Address: <input style="width: 100px;" type="text" value="192 . 168 . 80 . 8"/>	----> <input style="width: 100px;" type="text" value="192 . 168 . 80 . 12"/>
Options: SC	SC

SC - Silence Compression FEC - Forward Error Correction

23. **MVP410G.** A call from the Rootersville analog phone to a Compton area PSTN number can be dialed 59 7637172522.

Compton MVP410G MultiVOIP

Call Progress Details
Call Progress Details for Channel 1

Call Details

Duration: 00:01:00
Mode: Voice
Voice Coder: G.723.1 @ 6.3 kbps
Packets Sent: 680
Packets Received: 972
Bytes Sent: 40,152
Bytes Received: 41,432
Packets Lost: 0
Outbound Digits: 97637172522
Prefix Matched: 59

Disconnect

Exit

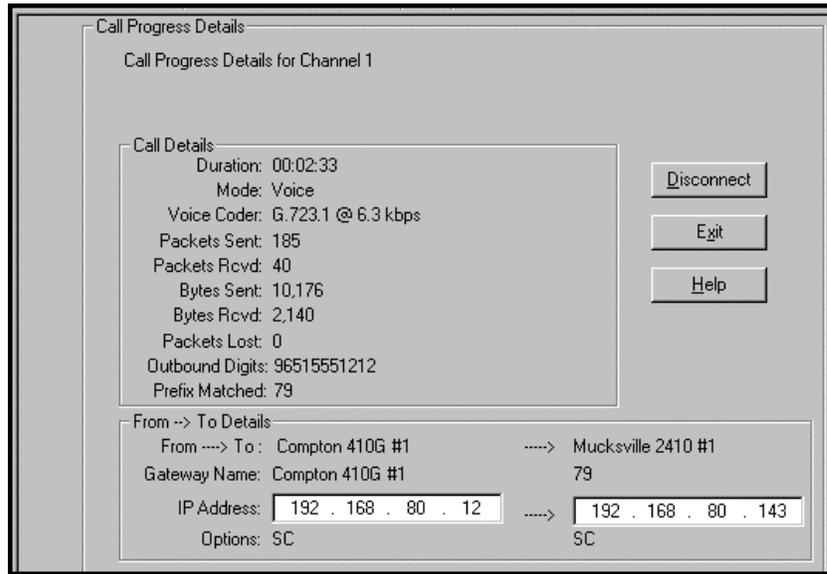
Help

From->To Details
From ----> To 6000 ----> 59:1
Gateway Name: Compton area PSTN calls
IP Address: 192 . 168 . 80 . 8 ----> 192 . 168 . 80 . 12
Options: SC SC

SC - Silence Compression FEC - Forward Error Correction

24. **MVP2410.** A call from a Compton PBX user to a Mucksville area PSTN number can be dialed 796515551212.

Mucksville MVP2410 MultiVOIP



End of Example.

Gatekeeper Basics

Introduction

Gatekeepers are optional within H.323 networks. However, when they are present, gateways (voip units) and other network endpoint devices (like terminals and Multipoint Control Units used in conferences) must use gatekeeper services. There are four functions that H.323 gatekeepers must provide to the network and many other functions, both standard and proprietary, that the gatekeeper *may* offer to network participants.

Mandatory Gatekeeper Functions

The mandatory gatekeeper functions are address translation, admission control, bandwidth control, and zone management.

Address Translation

The gatekeeper supports aliases, such as conventional E.164 phone numbers, for each endpoint registered within the zone. Users call each other within a zone by simply dialing a number or string of characters instead of an IP address. This function is particularly important when a phone on the circuit-switched network tries to call a phone connected to a gateway on an IP network.

Admission Control

The gatekeeper determines which network participants can and cannot make calls, according to established network permissions and rules. The gatekeeper controls admission using H.225 “RAS” messages (Registration, Admission, Status).

Bandwidth Control

With the MultiVOIP Gatekeeper, the network administrator can specify bandwidth limitations within a gatekeeper’s zone and can specify a bandwidth limit for gateway endpoints. The gatekeeper controls bandwidth using H.225 RAS messages. A gatekeeper may determine there is no bandwidth available for a call or no additional bandwidth available for an ongoing call requesting an increase. Dynamic (situation-dependent) changes in bandwidth allocation are typically called “bandwidth management,” which is considered an optional gatekeeper function.

Zone Management

Note. Zone Management and neighboring gatekeeper functionality are not included in the current software release. The discussion of this paragraph pertains primarily to the general theory of gatekeeper functionality. These functions are included in plans for subsequent software releases.

The gatekeeper allows or disallows call traffic between neighboring zones, depending upon established permissions. The zones themselves might be defined geographically (a company may have facilities in different cities, each being a separate network zone), by physical network connections (a range of IP addresses may comprise a zone, as may a subnet on a particular floor of a building), or by an organizational criterion (e.g., a large company might define separate network zones for engineering, manufacturing, marketing, and administration).

Optional Gatekeeper Functions

The MultiVOIP Gatekeeper supports the four main optional gatekeeper functions: call control signaling, call authorization, bandwidth management, and call management.

Call Control Signaling

The gatekeeper can, in “routed” mode, act as an intermediary for H.225 call-control signals between two endpoints participating in a call. In “direct” mode, this function is turned off and the endpoints exchange H.225 call-control messages directly.

Call Authorization

The gatekeeper can be programmed to restrict access (admission and registration) according to criteria set by the user.

Bandwidth Management

This is essentially dynamic bandwidth control (see “Bandwidth Control” section above).

Call Management

Note. Call Management functionality for re-routing calls is not included in the current software release. The discussion of this paragraph pertains primarily to the general theory of gatekeeper functionality. This function is included in plans for subsequent software releases.

The gatekeeper can keep a list of ongoing H.323 calls. This information allows the gatekeeper to re-route calls (where possible) to balance the traffic load on the networks.

Features

Ease of Use. The MultiVOIP Gatekeeper manages a zone, which is a collection of MultiVOIP gateways or other H.323 devices. Multiple gatekeepers can be configured to support several zones. For ease of use, the MultiVOIP Gatekeeper employs an intuitive graphical user interface. End-users can communicate using aliases (phone numbers). There's no need to remember complicated network addresses. Simple prefixes are used to access gatekeeper services such as call forwarding and out-of-zone dialing.

Capacities & Capabilities by Model. Within each zone, the MultiVOIP Gatekeeper supports a certain number of concurrent calls and registered endpoints. The capacities and capabilities of the various embedded gatekeeper voip units are described in the table below.

Model	Number of Simultaneous Calls Supported	Number of Registered Endpoints Supported	Protocols Supported
MVP210G	10	250	H.323 v4
MVP410G	20	250	H.323 v4
MVP810G	20 or 30	250	H.323 v4

•Ease of Control

With the MultiVOIP Gatekeeper, the network manager can determine the following settings:

•Network parameters

Maximum number of calls or registrations; maximum total bandwidth; upper bandwidth used per call; and frequency of sending information request (IRR) "keep alive" messages.

•Gatekeeper parameters

Gatekeeper registration policies; routing options; alias resolution policies; and endpoint permissions.

- **Gatekeeper services**

Built-in services such as call forward, zones and exit zone; and custom services.

The Gatekeeper Protocols

H.323 is an umbrella standard that consists of many subordinate protocols. Three protocols, Q.931, H.225, and H.245, are particularly relevant to gatekeepers.

The Q.931 protocol pertains to the setup and teardown of call connections between network endpoints.

The H.225 Call Signaling Protocol pertains to Registration, Admission, and Status (RAS). (Note that RAS in H.323 has nothing to do with the Remote Access Service that is used in ordinary TCP/IP networks.) H.323 RAS messages are concerned with general participation on the network (registration), specific involvement in particular calls between endpoints within and perhaps outside of the network zone (admission), and the status of endpoints (e.g., are they still “alive” or participating?).

H.245 is the conference control protocol. It pertains to negotiation between endpoints to establish a compatible set of media capabilities.

Because many user-settable parameters of the MultiTech gatekeeper software refer directly or indirectly to the H.225 protocol, we present a summary of common H.225 messages below.

Summary of H.323 RAS* Messages (Registration, Admission, & Status) of the H.225 Call Signaling Protocol	
In a gatekeeper-controlled H.323 network, when call is made, the RAS channel between gatekeeper and endpoint is the first logical channel opened.	
Admission Control Messages	<p>With an ARQ, an endpoint asks to participate in a phone call. The gatekeeper can either grant the request (by sending an ACF message) or deny the request (by sending an ARJ message). When admission is granted, the endpoints participating in the call can exchange (H.225) call signaling messages directly between themselves.</p> <p>When the call is done, each endpoint, in turn, requests disengagement (DRQ) and is granted disengagement (DCF) by the gatekeeper.</p>
ARQ	Admission Request.
ACF	Admission Confirmation.
ARJ	Admission Rejection.
DRQ	Disengagement Request.
DCF	Disengagement Confirmation.
Bandwidth Control Messages	<p>With a BRQ, an endpoint requests a certain amount of digital bandwidth for a call. If the gatekeeper grants the request, it returns a BCF message. If the gatekeeper denies the request, it returns a BRJ message, typically because all allocated data channels are in use. If a bandwidth request is rejected, it is possible for a call to be conducted</p>
BRQ	Bandwidth Request
BCF	Bandwidth Confirmation
BRJ	Bandwidth Rejection
* RAS in H.323 has nothing to do with the Remote Access Service that is used in ordinary TCP/IP networks.	

Summary of H.225 RAS Messages (cont'd)	
Address Translation Messages for Out-of-Zone Calling	<p>An LRQ is a request message between two H.323 gatekeepers to find the address of an H.323 endpoint. One gatekeeper is requesting the address translation services of the other. If the request is granted, an LCF message is returned.</p> <p>If the request is denied, an LRJ message is returned.</p>
LRQ	Location Request.
LCF	Location Confirmation.
LRJ	Location Request Rejection.
Registration Control Messages	<p>With an RRQ, an endpoint asks to be a participant in the network zone controlled by the gatekeeper. The gatekeeper can either grant the request (by sending an RCF message) or deny the request (by sending an RRJ message).</p> <p>If an endpoint's registration with the gatekeeper is temporary, its duration is specified in a TimeToLive field in the RCF message sent by the gatekeeper. After the registration duration has elapsed, the gatekeeper will send two IRQ messages (see "IRQ Interval" field in the Network Parameters screen) to see if the endpoint is still "alive." If the endpoint responds with an IRR, the registration will be extended. If not, the gatekeeper will send a URQ message to terminate the endpoint's registration. Thereafter, the endpoint must re-register with a full RRQ.</p>
RRQ	Registration Request.
RCF	Registration Confirmation.
RRJ	Registration Rejection.

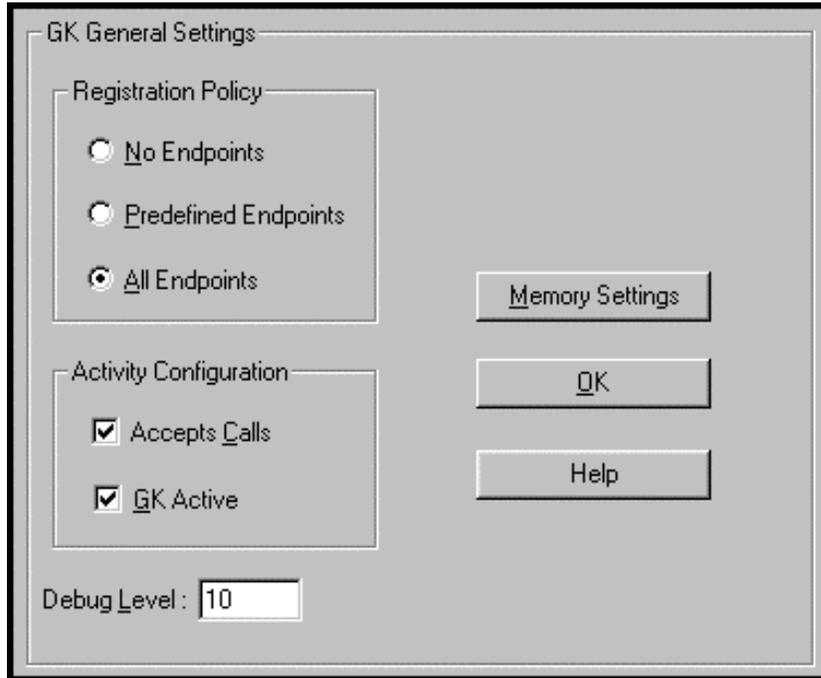
Summary of H.225 RAS Messages (cont'd)	
IRQ	Information Request
IRR	Extend Registration Request. (aka "keep-alive" request)
URQ	Unregister Request.
App URQ	When registration has timed out, the user application must decide how to respond.

MultiVOIP Gatekeeper Software Screens

Use the sidebar menu to access gatekeeper screens.

Accessing "Gatekeeper" Functions	
Pulldown	Icon
Sidebar	Sidebar with Submenus
	

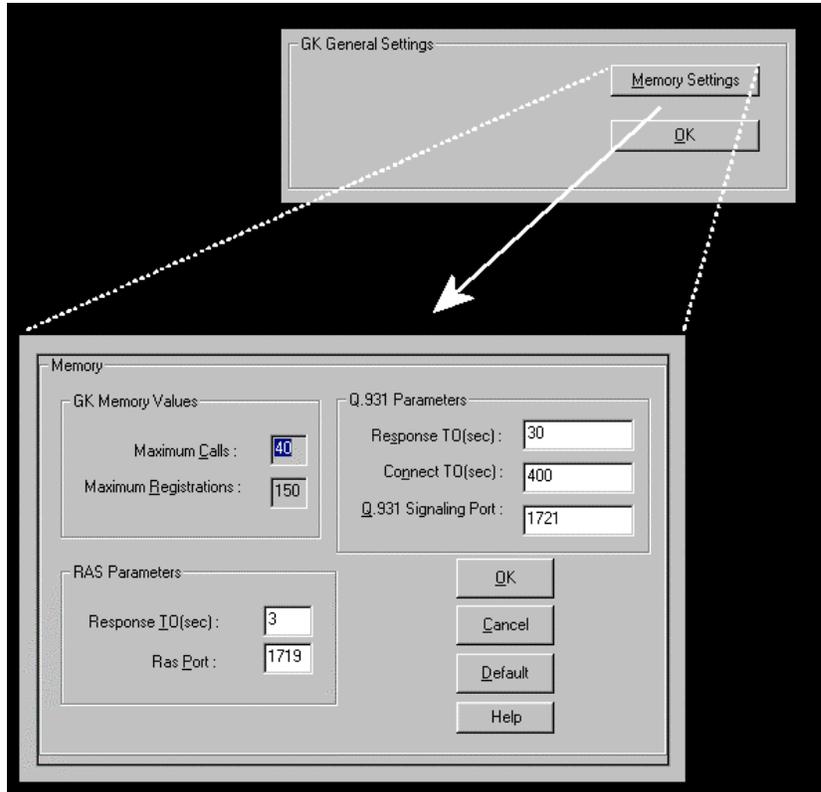
The fields in the main gatekeeper screen, the **GK General Settings** screen, are described in the table below.



GK General Settings Definitions		
Field Name	Values	Description
Registration Policy		
No Endpoints	Y/N	When selected, sets a policy whereby the Gatekeeper accepts no registrations.
Predefined Endpoints	Y/N	When selected, sets a strict zone policy, in which the Gatekeeper accepts only registrations that arrive from predefined endpoints. A strict zone policy controls network resources and services more tightly than an open zone policy.
All Endpoints	Y/N	When selected, sets an open zone policy, in which the Gatekeeper accepts any legal registration. Under this policy, the Gatekeeper can operate in “plug-and-play” mode.

GK General Settings Definitions (cont'd)		
Field Name	Values	Description
Activity Configuration		
Accepts Calls	Y/N	When checked, the voip unit will accept calls.
GK Active	Y/N	When checked, the voip unit's gatekeeper function is active.
Debug Level	0-100	The higher the value, the greater the details in Syslog or Console reports.
Buttons		
Memory Settings		Launches secondary screen on Memory issues. (See next table.)

Click on the **Memory Setting** button to access the **Memory** screen.



GK General Settings Definitions (cont'd)

Field Name	Values	Description
GK Memory Values		
Maximum Calls	10, 20, 30	The maximum number of concurrent calls. MVP210G support 10 calls; MVP410G supports 20 calls; MVP810G supports 30 calls.
Maximum Registrations	2 - 250	Maximum number of endpoints that can be registered on the gatekeeper-controlled network.

GK General Settings Definitions (cont'd)		
Field Name	Values	Description
RAS Parameters		In H.323, RAS parameters pertain to Registration, Admission, and Status in the H.225 Call Signaling Protocol.
Response TO		The timeout (in seconds) before re-transmission of a RAS message that had previously fetched no response.
RAS Port		The RAS port for gatekeeper communication with endpoints. Default value = 1719
Q.931 Parameters		In H.323, Q.931 parameters are those that pertain to the set-up and tear-down of connections between H.323 endpoints.
Response TO (sec)		The timeout (in seconds) waiting for the TCP reply.
Connect TO (sec)		The timeout (in seconds) waiting for the Connect message of a call.
Q.931 Signaling Port		Logical port through which Q.931 protocol messages are handled. Default value = 1721
Buttons		
Default		Invokes default values for all parameters on the GK General Settings screen.

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



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

Buttons on the right side of the table:

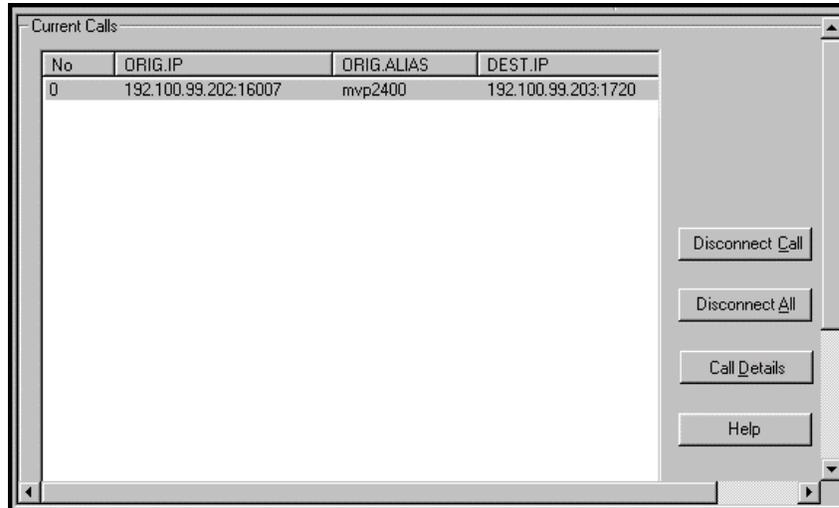
- Add
- Unregister
- Unregister All
- Disconnect Endpoint
- Delete
- DelPreDefs
- Online Properties
- Help

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

Existing Endpoints Parameter Definitions		
Field Name	Values	Description
Type	Gatekeeper, Gateway, MCU, Terminal, or Undefined.	The endpoint type . When an endpoint attempts to register with the Gatekeeper, the Gatekeeper compares the endpoint type with the predefined value. If the Gatekeeper detects a discrepancy, the registration is not accepted. If you are not sure of the endpoint type, select Undefined , which allows any endpoint of any type to register with the Gatekeeper. (Multipoint Control Units, MCUs, are used to facilitate conference calls.)
Online	+ or [blank]	When “+” appears, the endpoint’s registration is dynamic or “online.”
PreDef	+ or [blank]	When “+” appears, the endpoint’s registration is static or “predefined.”
Registration IP	n.n.n.n 0-255	The RAS address and RAS port of the endpoint.
Name		The H.323 ID alias of the endpoint.
Phone		The e164 alias number (conventional PSTN phone number)of the endpoint.
Other Aliases		Additional aliases for the endpoint: URL, e-mail address, transport address, party.address, or private network number (per ISO/IEC 11571). Alias addresses must be unique within a zone. Gatekeepers themselves cannot have aliases.
Msg	LRQ, RRQ, URQ, or AppURQ	The type of message sent by the endpoint when the mode for processing registration is manual. This can be an LRQ, RRQ, URQ, or AppURQ (which is a URQ sent by the Gatekeeper).).
TTL	seconds	The time remaining in seconds before the TimeToLive timer expires. If the endpoint fails to reregister within this time, the endpoint is unregistered.

Existing Endpoints Parameter Definitions (cont'd)		
Field Name	Values	Description
Command Buttons		
Add	--	Opens an empty Predefined Properties dialog box where you can predefine a new registration.
Unregister	--	Sends a URQ message to the selected endpoint, deleting the online (or dynamic) registration properties and unregistering the endpoint.
Unregister All	--	Sends a URQ to all the online endpoints in order to unregister them.
Disconnect Endpoint	--	Disconnects all calls with which the endpoint is involved.
Delete	--	Deletes the endpoint from the Gatekeeper database. A URQ will not be sent to the endpoint.
Del Pre-def	--	Deletes the predefined (static) properties of the endpoint.
Online Properties	--	Opens the Online properties screen or the selected endpoint whereupon are shown details of that endpoint's configuration.

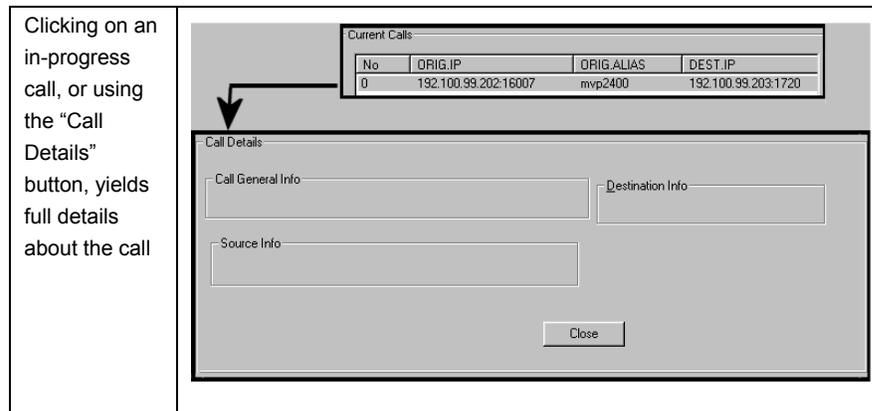
The fields of the **Current Calls** screen are described in the table below.



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

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

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



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

The screenshot displays a 'Call Details' window with a 'Call General Info' pane. The fields are as follows:

Field	Value
Call No	0
Cid Sum	1720
Call ID Sum	1721
Call Model	routed
Total B'W	10000
Conf. Goal	create
State	Bandwidth Change
Reason	B'W in BRQ

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

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

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

Call Details

Call General Info

Source Info

Destination Info

Names

1

2

3

Phone Numbers

1

2

3

Other Aliases

Email: Trans. Name:

URL:

Party Number: Type:

Call Rate:

Call Signalling IP

Port:

Bandwidth

Req. Bandwidth:

App. Bandwidth:

Additional Phone Numbers

1

2

3

Remote Extension

Phone:

Name:

Close

Call Details Field Definitions

Field Name	Values	Description
Destination Info fields		
Names		The H.323 alias name used to make the call.
Phone Numbers		The e164 alias phone number used to make the call.
Other Aliases: Email		An e-mail address used to make the call.
Other Aliases: Trans. Name		A transport name alias used to make the call, consisting of an IP address and port number.
Other Aliases: URL		A URL alias used to make the call.
Call Signaling IP		The call signaling transport address of the called endpoint.

Call Details Field Definitions (cont'd)		
Field Name	Values	Description
Destination Info fields		
Reg. Bandwidth		Requested Bandwidth. The bandwidth the called endpoint requested for the call, as it appears in the ARQ/BRQ messages.
App. Bandwidth		Approved Bandwidth. The bandwidth the Gatekeeper made available to the called endpoint for the call.
Additional Phone Numbers		These allow calling with more than one B-channel.
Remote Extension Phone		This is the phone number of the called endpoint on the remote LAN. It is used for calls between multiple gateways.
Remote Extension Name		This is the identifier (name) of the called endpoint on the remote LAN. It is used for calls between multiple gateways.

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



Network Parameter Definitions		
Field Name	Values	Description
Status Information		Use Update button to refresh the Status Information fields.
Ongoing Calls	number	The number of current calls with the Gatekeeper.
Currently Registered	number	The number of endpoints registered with the Gatekeeper.
Current BW Usage	number	The current bandwidth usage of the ongoing calls in Kbps.

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Configuration Options		
Alias Giving	Y/N	<p>When an endpoint sends an RRQ message, the Gatekeeper uses the additional aliases that were predefined for the endpoint as online aliases. This enables the Gatekeeper to assign terminal alias names through which the terminal can be accessed by others. The following are two examples of how this option can be used:</p> <ul style="list-style-type: none"> • Example of Alias Giving for a Terminal. To make a terminal accessible by dialing 100, add the alias 100 to the terminal's predefined information, and select the Alias Giving option. When the terminal sends an RRQ message, the 100 alias becomes a dynamic (online) alias, and all calls to 100 will be directed to the terminal. • Example of Alias Giving for Gateways. To make all Gateways supply Service 80, add Service 80 to the Service Table, add the 80 alias as predefined information to all registered gateways, and select the Alias Giving option. When the gateways register, they will support Service 80.
Pre-Granted ARQ		
PreGrant ALL	Y/N	Select to cause the Gatekeeper to send a pregrantedARQ permission in the RCF message for each endpoint that wishes to register. The pregranted ARQ permission is given to both makeCall and answerCall with routed mode. When an endpoint receives the permission, it may start the call with a Setup message or directly answer the call with a Connect message.

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Line Hunting Information		
Call to Out-of-Service Supplier	Y/N	“Y” enables the sending of RAI messages. In a normal scenario, the gatekeeper will hunt among all the available endpoints that have been registered using the same tech-prefix. Each endpoint can inform the gatekeeper about its resource availability using an RAI (Resource Available Indication) message. Upon receiving an RAI message from an endpoint, the gatekeeper would consider that endpoint as an Out-of-Service Supplier. The ‘Almost Out of Resources’ configuration would allow the gatekeeper to hunt such Out-of-Service Supplier endpoints for routing the calls.
Remove H.245 Addr in Call Hunt	Y/N	When selected, the gatekeeper will not convey in its outgoing setup message the H.245 address received in an incoming setup message. This prevents H.323 terminals from establishing a channel for a call only to refuse the call later.
Service Configurable Properties	Y/N	When “Y” is selected, the gatekeeper will perform a Priority Based Line Hunting among those destinations registered using the same tech-prefix.

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Call Proceeding		This parameter group pertains to the gatekeeper's handling of Q.931 "call-proceeding" messages.
Send Immediately	Y/N	Immediate return of call-proceeding message to originating endpoint. When selected, the gatekeeper will send the Q.931 call-proceeding message to the originating endpoint immediately after receiving that endpoint's call setup request.
With H.245 Addr	Y/N	When enabled, gatekeeper supplementary services will remove the H.245 address from the outgoing setup in order to prevent early H.245 establishment to the call's destination. This destination can be changed during Forward on Busy or during Forward on No Response (CFNR).
After Overlapped Sending	Y/N	Delayed return of call-proceeding message to originating endpoint. When selected (in routed mode), the gatekeeper will send a Q.931 call-proceeding message to the originating endpoint after it receives a return call-proceeding message back from the destination endpoint.

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Call Mode		
Direct Mode		Sets the call mode to direct. In this mode, terminals send ARQ messages to the Gatekeeper, but pass the call signaling and media control signaling directly between them.
Routed Mode		Sets the call mode to routed. In this mode, terminals pass admission requests and call signaling through the Gatekeeper. Media control information is sent directly between the terminals. Note: Though direct calls consume fewer Gatekeeper resources, call control is better for indirect (or routed) calls.
Configuration Parameters		
Max Number of Calls		The maximum number of concurrent calls allowed in the zone. This number can be increased up to 100, in increments of 20, by purchasing additional concurrent call licenses.
Max Total BW (KBps)		The amount of bandwidth in Kbps that call traffic can consume at any given time.

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Configuration Parameters		
Registration TO (hrs)		Registration Timeout. Sets the number of hours of inactivity after which the dynamic registration of a terminal expires. Only the dynamic (online) properties will be unregistered. If the endpoint is also static (predefined), the static properties remain valid.
IRQ Interval (sec)		<p>The interval, in seconds, between IRQ messages sent by the Gatekeeper. IRQ messages are sent to all online endpoints registered as dynamic in order to verify that the endpoints are online. The number you set determines the delay between two IRQ messages to the same endpoint. Choosing the desired delay should take into account the following factors:</p> <ul style="list-style-type: none"> • IRQ messages add to the traffic already present over the network, and the shorter the delay, the more IRQ messages are sent. However, the longer the delay, the longer it takes for the Gatekeeper to detect dynamic registrations that have ceased to be online. • The delay parameter relates to the interval between two IRQ messages per one endpoint, so the actual number of the IRQ messages the Gatekeeper creates during this interval should be multiplied by the number of endpoints registered dynamically. • To disable the IRQ polling, set this value to zero. • The effective IRQ interval cannot fall below three times the RAS timeout. • IRQ messages will not be sent at a rate exceeding 20 per second.

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Configuration Parameters		
Call IRQ Interval		<p>The interval, in seconds, between IRQ messages sent by the Gatekeeper to query the status of calls. IRQ messages are sent to all online endpoints registered as dynamic and having ongoing calls in order to verify that the calls are still ongoing. The number you set determines the delay between two IRQ messages to the same endpoint regarding the same call. Choosing the desired delay should take into account the following factors:</p> <p>IRQ messages add to the traffic already present over the network, and the shorter the delay, the more IRQ messages are sent. However, the longer the delay, the longer it takes for the Gatekeeper to detect calls that are stale.</p> <p>The delay parameter relates to the interval between two IRQ messages per one call, so the actual number of the IRQ messages the Gatekeeper creates during this interval should be multiplied by the number of ongoing calls registered dynamically.</p> <p>To disable the IRQ polling, set this value to zero.</p> <p>The effective IRQ interval cannot fall below three times the RAS timeout.</p> <p>IRQ messages will not be sent at a rate exceeding 20 per second.</p>

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Configuration Parameters		
Default Distance		<p>The “distance” (number device-to-device hops that a call must traverse between endpoints) allowed for endpoints which are only dynamically registered, such as an endpoint with no predefined values. This distance is compared to the distances of the neighbor gatekeepers and to the multicast distance in order to determine if an LRQ can be sent on behalf of the requesting endpoint.</p> <p>NOTE: The neighboring gatekeeper feature is not supported in the current software version.</p>
Out-of-Zone Distance		<p>The “distance” (number device-to-device hops that a call must traverse between endpoints) allowed for an out-of-zone endpoint that is making a call through the Gatekeeper. This distance is compared to the distances of the neighbor gatekeepers and to the multicast distance in order to see if an LRQ can be sent on behalf of the requesting endpoint.</p> <p>NOTE: The neighboring gatekeeper feature is not supported in the current software version.</p>

Network Parameter Definitions (cont'd)		
Field Name	Values	Description
Configuration Parameters		
Multicast Distance		The “distance” (number device-to-device hops that a call must traverse between endpoints) associated with sending an LRQ by multicast. NOTE: The neighboring gatekeeper feature is not supported in the current software version.
GK-ID		The name of the Gatekeeper. The terminals identify the Gatekeeper by this name during the discovery process. The Gatekeeper responds only to Discovery requests that either contain a matching Gatekeeper identifier or have no Gatekeeper identifier.
Update (button)	--	Click to update information in the “Status Information” fields of the Network Parameters screen.

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



Services Screen Definitions

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

Services Screen Definitions (cont'd)		
Field Name	Values	Description
GK Defined Services		
Public		For any GK-defined service being used, the user must select either "Default" or "Public." When Public is selected, the service is accessible to all endpoints that are not part of the zone.
V2 GW Prefixes		
		<p>H.323 Version 2 enables the gateway to specify prefixes that the user should dial before the WAN number in order to make a call using a certain medium. E.g., the user could dial the prefix 3 for voice calls or 77 for H.320 video calls. The prefixes are defined in the RRQ message at registration. Prefix can be any H.323 alias, including an H.323 ID & mail address.</p> <p>When a terminal places a LAN to WAN call, it should add one of the prefixes to the dialed number. The Gatekeeper identifies the prefix & routes the call to the appropriate gateway. If more than one gateway supplies the same prefix, line hunting is possible between the gateways.</p>
Prefix		Identifies the service. The prefix can be a numeric code, alphanumeric string, name, or phone number that the user dials. Per H.323 Vers. 2, prefixes can also be of URL and e-mail type. Also for H.323 Vers. 2, the type must precede the prefix. For example, TEL: 3 or NAME: John.
Description		A description of the service that is accessible by dialing the prefix.
Default		Select to make the service accessible to all endpoints that are not predefined in the zone.
Public		Select to make the service accessible to all endpoints that are not part of the zone.

Services Screen Definitions (cont'd)		
Field Name	Values	Description
V2 GW Prefixes		
Dynamic	Y/N	Indicates whether the service is static (essentially permanent) or timed & conditional (dynamic). This field indicates whether the service has been added manually (non-dynamically; field value =N) or dynamically (field value = Y) as part of registration from endpoints.
Buttons		These buttons allow you add, edit, or delete a selected service or prefix.

GK Defined Service Types

You can either define your own Gatekeeper services, or use any of the built-in services, which are predefined internally and supported by the Gatekeeper.

Example of a Gatekeeper Service

You can define a service named TECHSUPP and register five different terminals that provide technical support. Any call directed to TECHSUPP can connect to one of the five terminals.

To do so:

1. Add a service with a prefix TECHSUPP.
2. Make sure the terminals register with the additional alias TECHSUPP.
3. When a call for TECHSUPP arrives, the Gatekeeper automatically routes the call to one of terminals that provides the TECHSUPP Service.

Endpoints must be registered with the service name to receive calls for the service. This is achieved using one of the following methods:

- The endpoint is pre-configured using its own configuration. Then, using RAS messages, the endpoint is registered with a name or a phone number identical to the service prefix.
- The service prefix is predefined for the endpoint, using the configuration application of the Gatekeeper as an ID or phone number, and the **Alias Giving** option is activated. See the description of the **Alias Giving** option in the Network Parameters window section.

Built-in Gatekeeper-Defined Services

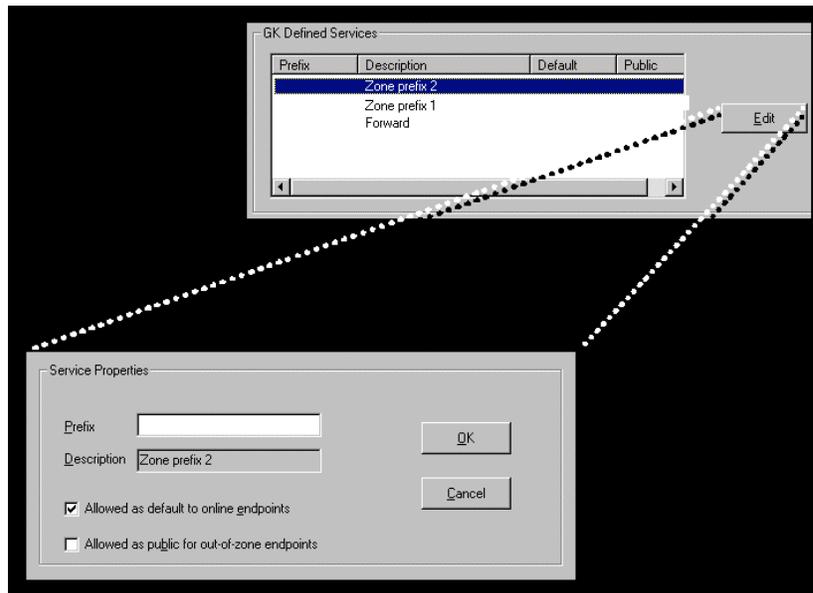
The current version of the Gatekeeper software supports the following services:

- Zone Prefix 1
- Zone Prefix 2
- Forward

Service Types: Zone Prefixes (1 and 2)

Note: This feature is for future use. Zone Prefix functionality is implemented in the current software release but it operates only in a context of neighboring gatekeeper functionality, which is not implemented in the current release. The discussion of this section pertains to a context in which neighboring gatekeeper functionality is implemented. Such functionality is included in plans for subsequent software releases.

MultiVOIP gatekeeper can operate in multiple zones. You can define one or two prefixes for a zone by entering the prefix for the services. The zone prefix functions in the same way as a telephone area code.



When one of the zone prefixes is defined, no calls from other zones can reach this zone, unless preceded by the prefix. If an endpoint in a zone dials a zone prefix before its number, and the Gatekeeper cannot resolve it in its zone, the Gatekeeper attempts to locate and route the call to a Neighbor Gatekeeper with the same prefix. For such calls, the Gatekeeper strips the zone prefix and then applies the destination location mechanism to route the call to its final destination.

You can use the zone prefix to devise a dialing plan in a multi-zone environment. If zone prefixes are not defined, the zone accepts the following calls:

- Calls prefixed to a service defined in the zone and allowed as default.
- Calls to on-line terminals in the zone.
- Calls to terminals marked as Forward in the zone.

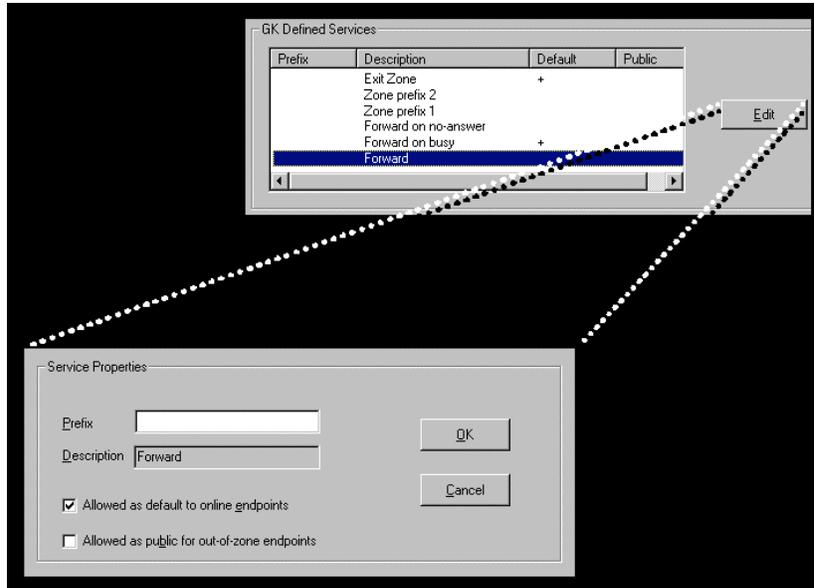
Example of comparing Zone prefix use when using Zone prefixes

- Zone A has a 01 prefix. In this zone, the phone number of user A1 is 123 and the phone number of user A2 is 456. The Gateway service has a prefix of 8.
- Zone B has a 02 prefix. In this zone, the phone number of user B1 is 123 and the phone number of user B2 is 456. The Gateway number is 555444 and the Gateway service has a prefix of 9.
- A1 calls A2 by dialing 456.
- A1 calls using zone A Gateway 8555444.
- A1 calls B1 by dialing 02123.

Note: The call is completed only if the Gateway service is allowed as default in Zone B.

Service Types: Forward

This call-forwarding feature is non-contingent, i.e., it forwards all calls for a selected station to another destination.



Gatekeeper Log Data Data Files

The embedded gatekeeper does not create files for its log data. For debugging or other purposes, such log data can be viewed/printed using a SysLog application program or HyperTerminal.

Gatekeeper Software User License Agreement

The MultiVOIP Gatekeeper software is licensed by Multi-Tech Systems, Inc., to the original end-user purchaser of the product, hereafter referred to as "Licensee." The License includes the distribution disc, other accompanying programs, and the documentation. The MultiVOIP Gatekeeper software, hereafter referred to as "Software," consists of the computer program files included on the original distribution disc.

Licensee agrees that by purchase and/or use of the Software, he hereby accepts and agrees to the terms of this License Agreement.

In consideration of mutual covenants contained herein, and other good and valuable considerations, the receipt and sufficiency of which is acknowledged, Multi-Tech Systems, Inc. does hereby grant to the Licensee a non-transferable and non-exclusive license to use the Software and accompanying documentation on the following conditions and terms: The software is furnished to the Licensee for execution and use on a single computer system only and may be copied (with the inclusion of the Multi-Tech Systems, Inc. copyright notice) only for use on that computer system. The Licensee hereby agrees not to provide or otherwise make available any portion of this software in any form to any third party without the prior express written approval of Multi-Tech Systems, Inc.

Licensee is hereby informed that this Software contains confidential proprietary and valuable trade secrets developed by or licensed to Multi-Tech Systems, Inc. and agrees that sole ownership shall remain with Multi-Tech Systems, Inc.

The Software is copyrighted. Except as provided herein, the Software and documentation supplied under this agreement may not be copied, reproduced, published, licensed, sub-licensed, distributed, transferred, or made available in any form, in whole or in part, to others, without expressed written permission of Multi-Tech Systems, Inc. Copies of the Software may be made to replace worn or deteriorated copies for archival or backup procedures.

Licensee agrees to implement sufficient security measures to protect Multi-Tech Systems, Inc. proprietary interests and not to allow the use, copying or transfer by any means, other than in accordance with this agreement. Licensee agrees that any breach of this agreement will be damaging to Multi-Tech Systems, Inc.

Licensee agrees that all warranties, implied or otherwise, with regard to this Software, including all warranties of merchantability and fitness for any particular purpose are expressly waived, and no liability shall extend to any damages, including consequential damages, whether known to Multi-Tech Systems, Inc. It is hereby expressly agreed that Licensee's remedy is limited to replacement or refund of the license fee, at the option of Multi-Tech Systems, Inc., for defective distribution media. There is no warranty for misused materials.

This package contains a compact disc. Neither this software nor the accompanying documentation may be modified or translated without the written permission of Multi-Tech Systems, Inc.

This agreement shall be governed by the laws of the State of Minnesota. The terms and conditions of this agreement shall prevail regardless of the terms of any other submitted by the Licensee. This agreement supersedes any proposal or prior agreement. Licensee further agrees that this License Agreement is the complete and exclusive statement of Agreement, oral, written, or any other communications between Multi-Tech Systems, Inc. and Licensee relating to the subject matter of this agreement. This agreement is not assignable without written permission of an authorized agent of Multi-Tech Systems, Inc.

Chapter 12 Warranty, Service, and Tech Support

Limited Warranty

Multi-Tech Systems, Inc. (“MTS”) warrants that its products will be free from defects in material or workmanship for a period of two years from the date of purchase, or if proof of purchase is not provided, two years from date of shipment. MTS MAKES NO OTHER WARRANTY, EXPRESSED OR IMPLIED, AND ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE HEREBY DISCLAIMED. This warranty does not apply to any products which have been damaged by lightning storms, water, or power surges or which have been neglected, altered, abused, used for a purpose other than the one for which they were manufactured, repaired by the customer or any party without MTS’s written authorization, or used in any manner inconsistent with MTS’s instructions.

MTS’s entire obligation under this warranty shall be limited (at MTS’s option) to repair or replacement of any products which prove to be defective within the warranty period, or, at MTS’s option, issuance of a refund of the purchase price. Defective products must be returned by Customer to MTS’s factory—transportation prepaid.

MTS WILL NOT BE LIABLE FOR CONSEQUENTIAL DAMAGES AND UNDER NO CIRCUMSTANCES WILL ITS LIABILITY EXCEED THE PURCHASE PRICE FOR DEFECTIVE PRODUCTS.

Repair Procedures for U.S. and Canadian Customers

In the event that service is required, products may be shipped, freight prepaid, to our Mounds View, Minnesota factory:

Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, MN 55112
Attn: Repairs, Serial # _____

A Returned Materials Authorization (RMA) is not required. Return shipping charges (surface) will be paid by MTS.

Please include, inside the shipping box, a description of the problem, a return shipping address (it must be a street address, not a P.O. Box number), your telephone number, and if the product is out of warranty, a check or purchase order for repair charges.

For out-of-warranty repair charges, go to www.multitech.com/documents/warranties

Extended two-year overnight replacement service agreements are available for selected products. Please call MTS at (888) 288-5470, extension 5308, or visit our web site at www.multitech.com/programs/orc for details on rates and coverages.

Please direct your questions regarding technical matters, product configuration, verification that the product is defective, etc., to our Technical Support department at (800) 972-2439 or email tsupport@multitech.com. Please direct your questions regarding repair expediting, receiving, shipping, billing, etc., to our Repair Accounting department at (800) 328-9717 or (763) 717-5631, or email mtsrepair@multitech.com.

Repairs for damages caused by lightning storms, water, power surges, incorrect installation, physical abuse, or used-caused damages are billed on a time-plus-materials basis.

Technical Support

Multi-Tech Systems has an excellent staff of technical support personnel available to help you get the most out of your Multi-Tech product. If you have any questions about the operation of this unit, or experience difficulty during installation you can contact Tech Support via the following:

Contacting Technical Support

Country	By E-mail	By telephone
France	support@multitech.fr	(33) 1-64 61 09 81
India	support@multitechindia.com	(91) 124-340778
U.K.	support@multitech.co.uk	(44) 118 959 7774
U.S. & Canada	tsupport@multitech.com	(800) 972-2439
Rest of World	support@multitech.com	(763) 785-3500

Internet: http://www.multitech.com/_forms/email_tech_support.htm

Please have your product information available, including model and serial number.

Chapter 13: Regulatory Information



EMC, Safety, and R&TTE Directive Compliance

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

Council Directive 89/336/EEC of 3 May 1989 on the approximation of the laws of Member States relating to electromagnetic compatibility,

and

Council Directive 73/23/EEC of 19 February 1973 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits,

and

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

FCC Declaration

NOTE: This equipment has been tested and found to comply with the limits for a **Class A** digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy, and if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

This device complies with Part 15 of the FCC rules.

Operation is subject to the following two conditions:

- (1) This device may not cause harmful interference.
- (2) This device must accept any interference that may cause undesired operation.

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

Industry Canada

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

Cet appareil numérique de la classe A respecte toutes les exigences du Règlement 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 from the network until the problem is resolved.
7. No repairs are to be made by you. Repairs are to be made only by Multi-Tech Systems or its licensees. Unauthorized repairs void registration and warranty.
8. Manufacturer: Multi-Tech Systems, Inc.
Trade name: MultiVOIP
Model number: MVP2400
FCC registration number: US: AU7DDNAN46050

Modular jack (USOC):	RJ-48C
Service center in USA:	Multi-Tech Systems, Inc. 2205 Woodale Drive Mounds View, MN 55112 Tel: (763) 785-3500 FAX: (763) 785-9874

Canadian Limitations Notice

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

Appendix A: Expansion Card Installation (MVP24-48 & MVP30-60)

Installation

Both the MVP2410 and the MVP3010 use the same mechanical chassis. This chassis accommodates a second MultiVOIP circuit card or motherboard module. The add-on module for the MVP2410 is the MVP24-48 product; the add-on module for the MVP3010 is the MVP30-60 product. The MVP2410G will not accept an expansion card because its second card slot is occupied by gatekeeper circuitry.

To install an expansion card into an MVP2410 or MVP3010, you must:

1. Power down and unplug the MVP2410/3010 unit.
2. Using a Phillips or star-bit screwdriver, remove the blank plate at the rear of the MVP2410/3010 chassis (see Figure A-1). Save the screw.

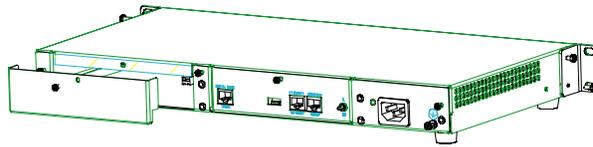


Figure A-1: Remove Plate Covering Expansion Slot

3. A power cable for the expansion card (+5V) is already present within the MVP2410/3010 unit. This power cable has a two-pin “molex” connector. When the rear cover plate has been removed, the cable is accessible from the rear at the right side of the expansion slot. Locate this connector within the MVP2410/3010. See Figure A-2.

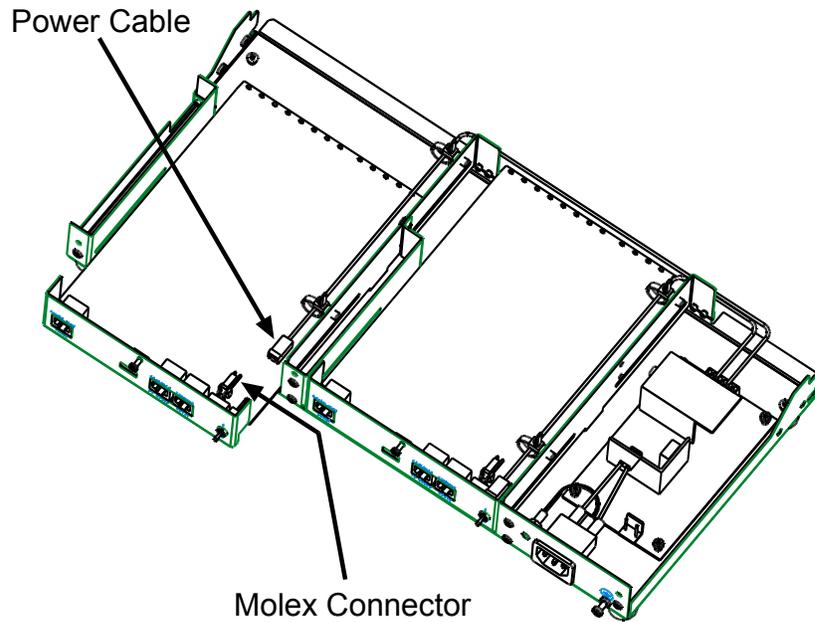


Figure A-2: MVP2410/3010 Chassis (top/rear view)

4. While keeping the power cable out of the way, fit the MVP24-48 or MVP30-60 card into the grooves of the expansion slot. Push it in far enough to allow connection of the power cable to the receptacle on the vertical plate of the expansion card. (See Figure A-2.) Connect the power cable.
5. Push the expansion card fully into the chassis. See Figure A-3.

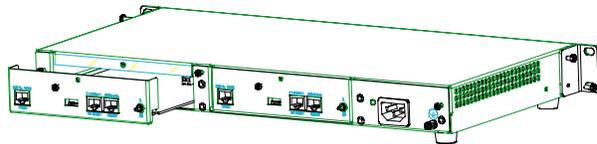


Figure A-3: Sliding Expansion Card into Chassis

Secure the vertical plate of the expansion card to the chassis with a screw.

Operation

The MVP2410/3010 front panel has two sets of identical LEDs. In the MVP2410/3010 without an expansion card, only the left-hand set of LEDs is functional. However, when the MultiVOIP unit has been upgraded with an MVP24-48 or MVP30-60 expansion card, the right-hand set of LEDs will also become active.

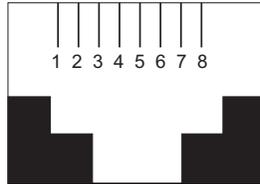
Remember that the expansion card must be configured as though it were simply another complete MultiVOIP unit: it requires its own T1/E1 line; it requires its own connection to a computer running the MultiVOIP configuration software. All of the procedures and operations that apply to the original motherboard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

Appendix B: Cable Pinouts

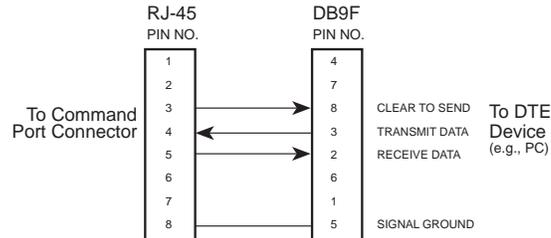
Appendix B: Cable Pinouts

Command Cable

RJ-45 Connector



End-to-End Pin Info



RJ-45 connector plugs into Command Port of MultiVOIP.

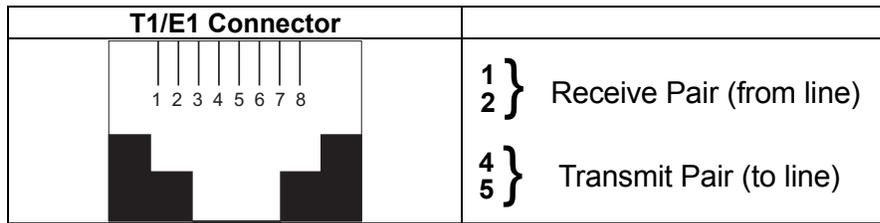
DB-9 connector plugs into serial port of command PC (which runs MultiVOIP configuration software).

Ethernet Connector

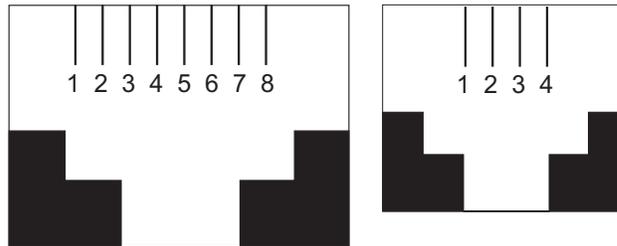
The functions of the individual conductors of the MultiVOIP's Ethernet port are shown on a pin-by-pin basis below.

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

T1/E1 Connector



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)

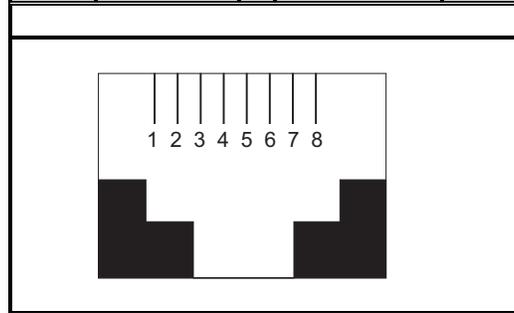
Pin Functions (FXS/FXO Interface)			
FXS Pin	Description	FXO Pin	Description
2	N/C	2	N/C
3	Ring	3	Tip
4	Tip	4	Ring
5	N/C	5	N/C

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.

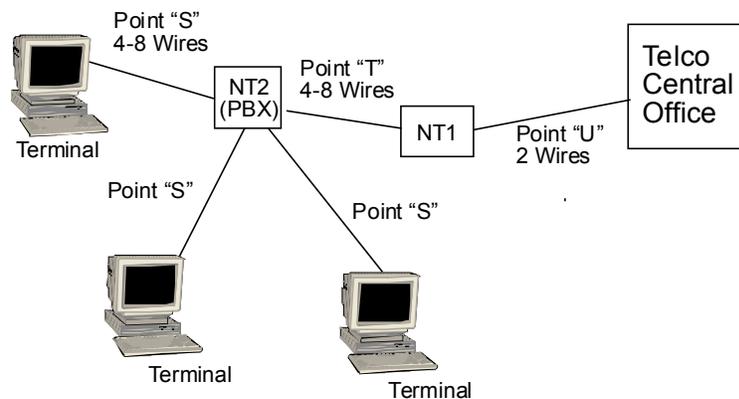
An ISDN Basic Rate (BRI) U-Loop consists of two conductors from the telco central office to the customer premises. The equipment on both sides of the U-loop accommodates the extensive length of the U-loop and the noisy environment in which it may operate. At the customer premises, the U-loop is terminated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

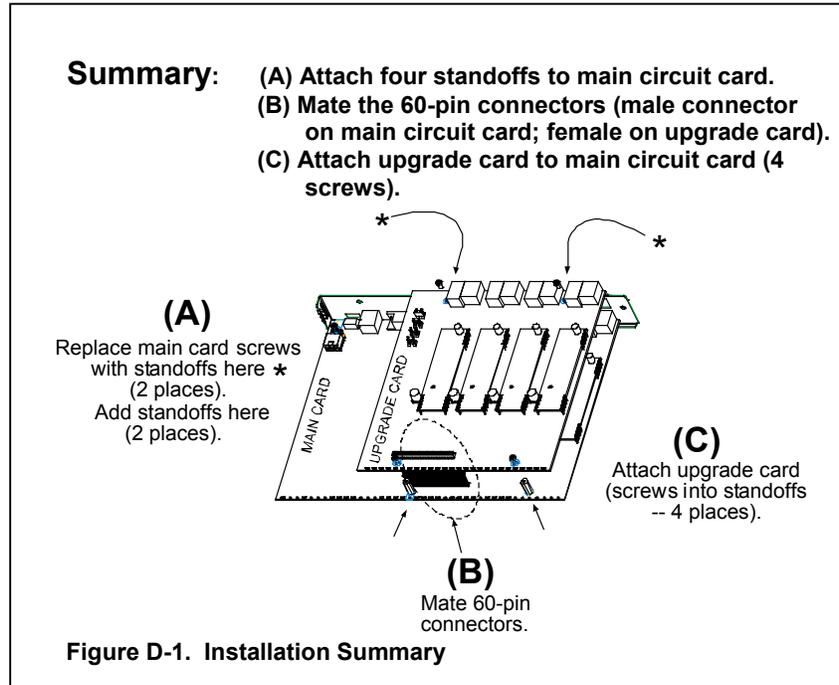
Well-Known Port Numbers

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.



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.

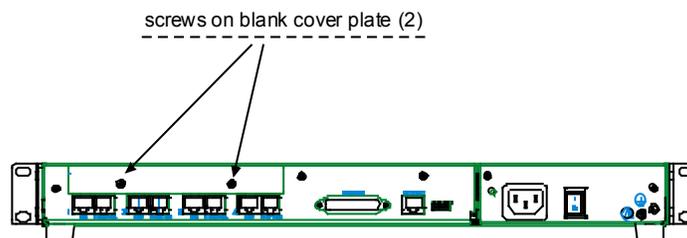


Figure D-2: Removing screws from blank cover plate

- 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.

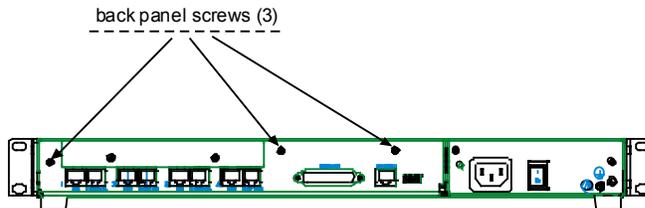


Figure D-3: Removing screws from back panel

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

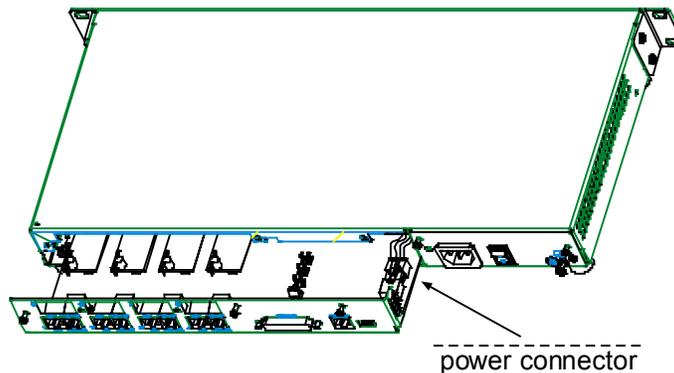


Figure D-4: Accessing power connector

- Unplug the power connector from the main circuit board.
- Slide the main circuit board completely out of the chassis and place on a non-conductive, static-safe tabletop surface.
- Remove mounting hardware (2 screws, 2 nuts, and 4 standoffs) from its package.

8. On the phone-jack side of the circuit card, three screws attach the circuit card to the back panel. Two of these screws are adjacent to the four phone-jack pairs. Remove these two screws.

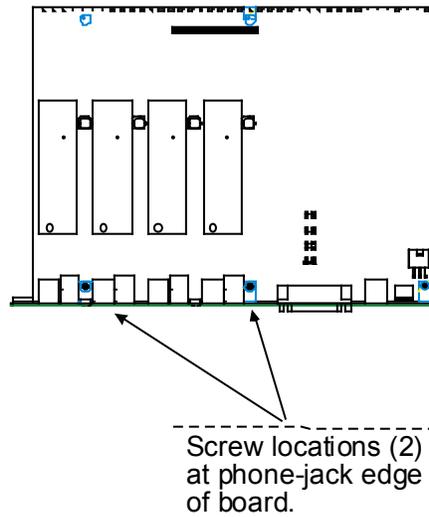


Figure D-5: Screws to be removed and replaced with standoffs (phone-jack edge of board; top view)

9. Replace these two screws with standoffs.
10. There are two copper-plated holes at the LED edge of the circuit card. Place a nut beneath each hole (lockwasher side should be in contact with board) and attach a standoff to each location).

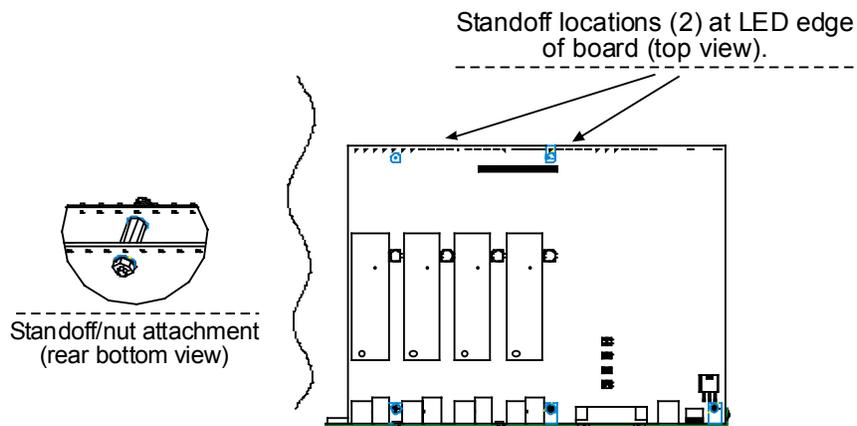


Figure D-6: Standoffs at LED edge of board (top view)

11. Locate the male 60-pin vertical connector near the LED edge of the main circuit card. Check that pins are straight and evenly spaced. If not, then correct for straightness and spacing. Locate the 60-pin female connector on the upgrade circuit card.
12. Set the upgrade circuit card on top of the main circuit card. Align the upgrade card's 4 pairs of phone-jacks with the 4 pairs of holes in the backplane of the main card. Slide the phone jacks into the holes.
13. Mate the upgrade card's 60-pin female connector with the main card's 60-pin male connector.

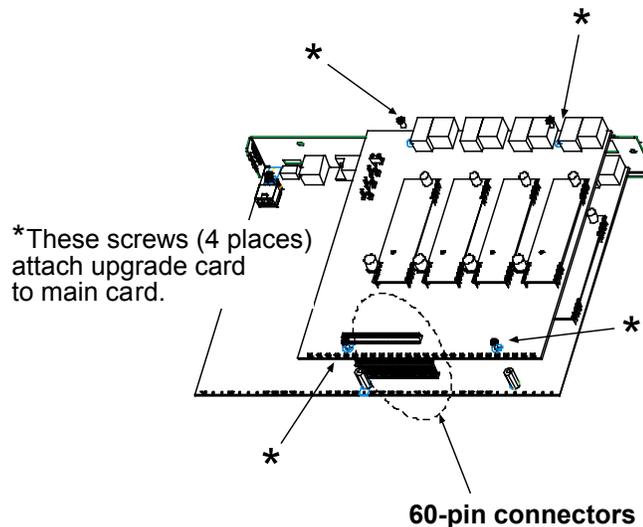


Figure D-7. Attaching upgrade card to main circuit card (secure 4 Phillips screws; mate 60-pin connectors)

14. There are four copper-plated attachment holes, two each at the front and rear edges of the upgrade card. Attach the upgrade card to the main card using 4 Phillips screws. The upgrade card should now be firmly attached to the main card.
15. Slide the main circuit card back into the chassis far enough to allow re-connection of power cable.
16. Re-connect power cable.
17. Slide the main circuit card fully into the chassis.
18. Re-attach the backplane of the main circuit card to the chassis with 3 screws.

Appendix E: Call States & Reasons for Embedded Gatekeepers

Call States and Call Reasons

MultiVOIP units with embedded gatekeeper functionality track call states and the reasons for those states. We present here a complete listing of these call states and call reasons. These relate to the **Call Details** screen, which is a secondary screen that can be launched from the **Calls** (“Current Calls”) screen of the embedded gatekeeper software.

Possible Call States of which the Embedded Gatekeeper Software can be notified

No	State	Description
1	Wait Orig Admission	Needs application approval for sending an ACF to the origin.
2	Wait NW Setup	Waits for the Setup message to arrive after sending an ACF back to the origin.
3	Wait Dest Admission	Needs application approval for sending an ACF to the Destination.
4	Wait NW Connect	Waits for the Connect message to arrive after sending an ACF back to the destination.
5	Wait Dest Connect	Needs application approval for Connecting the destination to the origin.
6	Connect Sent To Orig	The Gatekeeper passed the Connect message of the destination back to the origin.
7	Setup Arrived	A Setup message is received from the network.
8	Wait Orig Offering	Needs application approval before sending a Setup message from the originator of the call to the destination.
9	Wait LRQ	Needs application approval to do an LRQ for the call.
10	Sending LRQ	A notification is given for each outgoing LRQ.

Call States Listing (cont'd)		
No	State	Description
11	LRQ Sent	An LRQ was sent on the network. Waiting for a reply.
12	received LCF	An LCF was received. The application should decide whether or not to accept it.
13	Setup Sent To Dest	The Gatekeeper sends the Setup message to the Destination.
14	Call To Forward Service	A call is to the forward service and hence will be disconnected.
15	Dial Tone	A Setup message was sent.
16	Proceeding	Waiting for the end user's phone to ring.
17	Setup Ack	A Notification given on a SetupAck message arrived from the destination of a call.
18	Dest Alert	The end user's phone is ringing.
19	Disconnected	A connected call was disconnected.
20	Call Cannot Complete	The destination did not connect. Waiting for application instruction whether to disconnect or perform address translation again after the application sets new addresses.
21	Connected	The call connected.
22	Address Resolution	The application may replace call addresses. The various reasons for this state are mentioned in the reason table. Each time the address is changed by the Gatekeeper (such as stripping a zone prefix or translating an alias to IP address), the application is notified with the suitable reason.
23	Address Resolution Done	The application may review the final destination. This can be sent with two reasons: 1)AddressFound 2)NeedLRQ. The application needs to approve the final result or reroute the call
24	Admission Reject	Lets the application know about the reject.
25	Setup Reject	Lets the application know about the reject.
26	Orig Admission Reject	Lets the application know about the reject.
27	Dest Admission Reject	Lets the application know about the reject.

Call States Listing (cont'd)		
No	State	Description
28	GK Disconnected Call	Lets the application know about a call that the Gatekeeper disconnected.
29	Wait Line Hunting	Line Hunting failed on one line. Line Hunting can still continue after application approval.
30	DRJ Sent	Lets the application know when sending a DRJ.
31	DCF Sent	Lets the application know when sending a DCF.
32	ARJ Sent	Lets the application know when sending an ARJ.
33	GK Initiated DRQ	Lets the application know when the Gatekeeper initiated a DRQ.
34	Bandwidth Change	Notification of a change of the call bandwidth.
35	Idle	The call was terminated. Waiting for the application to release the handle.
36	Unknown	State unknown.

Call Reasons sent to Embedded Gatekeeper Software with respect to a Call State.

No	State	Description
1	Undefined	No reason.
2	Resource Unavailable	The call was rejected because of a lack of Gatekeeper resources.
3	Invalid Endpoint	The ARQ/DRQ was rejected because no valid endpoint was identified.
4	Route Call To GK	The destination ARQ was rejected because no Setup message preceded it.
5	Lines Busy	The call cannot be completed because Line Hunting failed.
6	Destination Out Of Service	The call cannot be completed because the destination cannot be reached.
7	Destination Busy	The call cannot be completed because destination is busy.
8	No Answer at Destination	The call cannot be completed because the user at the destination did not answer in the given time.
9	Destination Rejected the Call	The call cannot be completed because the party at the destination rejected the call.
10	Origin Disconnected	A connected call was disconnected because of the origin. The reason for state Disconnected.
11	Destination Disconnected	A connected call was disconnected because of the destination. The reason for state Disconnected.
12	New Admission from Origin	The reason for address resolution because of a new admission.
13	New Setup from Origin	The reason for address resolution because of a new Setup.
14	Origin Setup	The reason for wait offering when the Setup is not the first message in call. (An ARQ was received.)
15	Destination Info In LRQ	An LCF arrived with no CallSignal Address but with a new destinationInfo alias. The Gatekeeper sent an Address Resolution state with this reason in order to translate the new found alias to a valid IP address.

Call Reasons Listing (cont'd)		
No	State	Description
16	No Change. Service Prohibited	The reason for address resolution. The required service is not allowed for the endpoint.
17	Zone Prefix Removed	The reason for address resolution after the zone prefix was removed.
18	Exit Zone Prefix Removed	The reason for address resolution after the exit zone prefix was removed.
19	Ip Address Set	The reason for address resolution after the IP address was found from the aliases.
20	Address Forwarded	The reason for address resolution after finding that the call should be forwarded.
21	Address Found	The reason for state AddressResolutionDone.
22	Need to Send LRQ	The reason for state AddressResolutionDone.
23	Failure in App. Event Handler	The call cannot be completed because of a failure in the application event handler. (For example, the return value < 0.)
24	Internal Failure	The call cannot be completed because of an internal error.
25	Service Not Allowed	The call cannot be completed because a required service is not allowed.
26	Exit Zone Not Allowed	The call cannot be completed because it was dialed without an exit zone prefix, or the exiting zone is not allowed for call.
27	No Destination in Call	The call cannot be completed because it was dialed without a destination.
28	Cannot Send LRQ	The call cannot be completed because an LRQ cannot be sent.
29	Address Not Found after LRQ	The call cannot be completed because an LCF was not accepted for the LRQ.
30	Call Not Register	The reason for sending a DRJ.
31	Origin Connected First	The reason for a Connect message that arrives without first asking the application. This happens when the origin is already connected when the destination connects, which is an error.
32	DCF to Origin	A DCF was sent to the origin.
33	DCF to Dest	A DCF was sent to the destination.
34	App. Disconnected Destination	An application initiated disconnect of the destination (associated with the Call Cannot Complete state or with GK Disconnect Call state.)

Call Reasons Listing (cont'd)		
No	State	Description
35	App. Timeout	The call was disconnected because of a timeout on waiting for an application reply.
36	call cannot complete-missing line hunting addresses	The call cannot be completed because no application Line Hunting addresses were supplied when the application Line Hunting mode was on.
37	Additional Address Complete	The Additional Address information exchange has been completed.
38	Additional Address	The Additional Address procedure (digit collection) is in progress.
39	GK Connect Call	The Gatekeeper has connected to the call as the destination, forming a one-legged call. This reason accompanies the Wait Dest Connect state when the application replies to Setup Arrived with the Send Connect To Orig reply.
40	GK Initiated Call	This reason accompanies the Address Resolution and Connected states to indicate a one-legged call initiated from the Gatekeeper by the application.
41	Unknown	Reason unknown

Index

INDEX

A

- abbreviated dialing, inter-office
 - E1 322
 - T1 279
- Accepts Calls option (Gatekeeper General Settings screen) 489
- access codes, PBX 69
- access codes, types
 - PBX 74
 - PSTN 74
 - special 74
- access digits, PBX69. *See* phonebook digits, types used
- access to network
 - analog 243
 - T1/E1 161
- access to remote PSTN
 - E1 19
 - T1 12
- accessing Statistics, Logs screen . 382
- accessing Call Details (gatekeeper) screen 496
- accessing Call Progress (Statistics) screen 376
- accessing configuration parameter groups
 - analog 208
 - T1/E1 131
- accessing Current Calls (gatekeeper) screen 495
- accessing Endpoints (gatekeeper) screen 492
- accessing GK (gatekeeper) General Settings screen 487
- accessing interface parameters 223
- accessing IP Parameters screen
 - analog 209
 - T1/E1 132
- accessing IP Statistics screen 386
- accessing Logs (Statistics) screen . 382
- accessing logs screen
 - analog 257
 - T1/E1 175
- accessing Network Parameters (gatekeeper) screen 503
- accessing Regional Parameters
 - analog 244
 - T1/E1 162
- accessing Registered Gateway Details (Statistics) screen 406
- accessing Registered Gateway Details screen 405, 406
- accessing RTP Parameters screen . 390
- accessing Services (gatekeeper) screen 512
- accessing SMTP parameters
 - analog 251
 - T1/E1 169
- accessing SNMP parameters
 - analog 241
 - T1/E1 159
- accessing Supplementary Services screen
 - analog 261
 - T1/E1 179
- accessing System Information screen
 - analog 273
 - T1/E1 191
- accessing T1 Statistics screen 393
- accessing T1/E1/ISDN Parameters screen 146
- accessing Voice/FAX Parameters screen 136, 213
- ACF Admission Confirmation messages (gatekeeper, H.225) ... 485
- Add endpoints command (gatekeeper) 494
- Add Inbound Phonebook Entry icons
 - E1 328
 - T1 280
- Add Outbound Phonebook Entry icon
 - E1 328

- T1..... 280
- Add Prefix (inbound) field
 - E1..... 347
 - T1..... 299
- Add Prefix (outbound) field
 - E1..... 341
 - T1..... 292
- Add/Edit Inbound Phonebook field definitions
 - E1..... 347, 348
 - T1..... 299, 300
- Add/Edit Inbound Phonebook screen
 - E1..... 347
 - T1..... 299
- Add/Edit Inbound Phonebook screen fields (E1)
 - Add Prefix..... 347
 - Channel Number..... 347
 - Description (callee location)..... 348
 - Enable (Call Forwarding) 348
 - Forward Address/Number..... 348
 - Forward Condition..... 348
 - Remove Prefix 347
 - Ring Count..... 348
- Add/Edit Inbound Phonebook screen fields (T1)
 - Add Prefix..... 299
 - Channel Number..... 299
 - Description (callee location)..... 300
 - Enable (Call Forwarding) 300
 - Forward Address/Number..... 300
 - Forward Condition..... 300
 - Remove Prefix 299
 - Ring Count..... 300
- Add/Edit Outbound Phonebook field definitions
 - E1..... 341, 342, 343, 344
 - T1..... 292, 293, 294, 295
- Add/Edit Outbound Phonebook fields (E1)
 - Add Prefix..... 341
 - Advanced button..... 343
 - Description..... 341
 - destination pattern..... 341
 - Gateway Prefix 342
 - H.323 ID 342
 - IP Address..... 341
 - Protocol Type 341
- Q.931 Port Number..... 342
- Remove Prefix 341
- SIP Port Number 343
- SIP URL..... 343
- Total Digits 341
- Transport Protocol (SIP) 343
- Use Gatekeeper 342, 344
- Use Proxy (SIP) 343
- Add/Edit Outbound Phonebook fields (T1)
 - Add Prefix..... 292
 - Advanced button 294
 - Description..... 292
 - destination pattern..... 292
 - Gateway Prefix..... 293
 - H.323 ID 293
 - IP Address..... 292
 - Protocol Type..... 292
 - Q.931 Port Number 293
 - Remove Prefix 292
 - SIP Port Number 294
 - SIP URL..... 294
 - Total Digits 292
 - Transport Protocol (SIP) 294
 - Use Gatekeeper 293, 295
 - Use Proxy (SIP) 294
- Add/Edit Outbound Phonebook screen
 - E1 340
 - T1 291
- Add/Edit Outbound Phonebook SPP Fields
 - E1 344
 - T1 295
- Additional Phone Numbers
 - gatekeeper field (Call Details, Destination Info) 502
- add-on module (analog, 4-to-8 channel), installation 544
- add-on module (T1/E1) operation 534
- add-on module (T1/E1), installation 532
- Address (SNMP) field
 - analog..... 243
 - T1/E1 161
- address translation (gatekeeper).... 481
- address translation messages (gatekeeper H.225)

LCF.....	486	Allowed Name Types, Call Name ID (T1/E1)	
LRJ.....	486	Alerting Party.....	186
LRQ.....	486	Busy Party.....	187
admission control (gatekeeper).....	481	Calling Party.....	185
admission control messages (gatekeeper, H.225)		Connected Party.....	188
ACF.....	485	Alternate IP Address field	
ARJ.....	485	E1.....	346
ARQ.....	485	T1.....	297
DCF.....	485	Alternate IP Routing	
DRQ.....	485	E1.....	340
Advanced button, Outbound		T1.....	291
Phonebook		Alternate Phone Number, SPP (Add/Edit Outbound Phonebook)	
E1.....	344	E1.....	344
T1.....	295	T1.....	295
Advanced Features field group		Alternate Routing	
analog.....	219	PSTN failover feature, and.....	297
T1/E1.....	142	Alternate Routing field definitions	
After Overlapped Sending option (gatekeeper, Network Parameters)		E1.....	346
.....	506	T1.....	297
airflow.....	95	Alternate Routing field definitions (E1)	
Alerting Party		Alternate IP Address.....	346
Supplementary Services (analog)		Round Trip Delay.....	346
.....	268, 269, 270	Alternate Routing field definitions (T1)	
Supplementary Services (T1/E1)		Alternate IP Address.....	297
.....	186, 187, 188	Round Trip Delay.....	297
Alias Giving field (gatekeeper,		analog phonebook.....	372
Network Parameters).....	504	using T1 & E1 examples for.....	372
alias giving, description.....	504	analog phonebook examples.....	196
Alias Giving, example.....	516	analog telephony interface parameters	
alias giving, examples.....	504	200
aliases.....	500, 501	Annex E field	
aliases, other (gatekeeper).....	493	E1.....	336
All endpoints option (Gatekeeper		T1.....	287
General Settings screen).....	488	area codes.....	73
Allowed Name Type (analog)		ARJ Admission Rejection messages (gatekeeper, H.225).....	485
Alerting Party.....	268, 269, 270	ARQ Admission Request messages (gatekeeper, H.225).....	485
Calling Party.....	267	Auto Call Enable field	
Allowed Name Type (T1/E1)		analog.....	219
Alerting Party.....	186, 187, 188	T1/E1.....	142
Calling Party.....	185	Auto Disconnect field group	
Allowed Name Types, Call Name ID (analog)		analog.....	222
Alerting Party.....	268	T1/E1.....	145
Busy Party.....	269		
Calling Party.....	267		
Connected Party.....	270		

- Automatic Disconnection field
 analog..... 222
 T1/E1 145
- Avaya Magix PBX (FXO)
 and Message Waiting Light 230
- Avaya Magix PBX (FXS Ground
 Start)
 and Message Waiting Light 228
- Avaya Magix PBX (FXS Loop Start)
 and Message Waiting Light 226
- B**
- bandwidth 500, 507
 coder (analog) 218
 coder (T1/E1)..... 141
- bandwidth control (gatekeeper) 481
- bandwidth control messages
 (gatekeeper, H.225)
 BCF 485
 BRJ 485
 BRQ 485
- bandwidth management
 with gatekeeper 481
- bandwidth management (gatekeeper)
 483
- bandwidth management (versus
 control)..... 482
- bandwidth, requested/approved 502
- battery caution 88
- baud rate, default (MultiVOIP
 software connection)
 T1/E1 190
 analog..... 272
- baud rate, fax
 analog..... 217
 T1/E1 140
- baud rate, setting
 analog..... 272
 T1/E1 190
- BCF Bandwidth Confirmation
 messages (gatekeeper, H.225).... 485
- Bipolar Violation (E1 stats) field.. 400
- Bipolar Violation (T1 stats) field.. 397
- Blue Alarm (E1 stats) field 398
- Blue Alarm (T1 stats) field 395
- Boot Code Version
 System Info (analog)..... 274
- Boot LED
 analog models 33, 34
 BRI models 40
 MVP-210x..... 106
 MVP-410/810 100
 MVP-410ST/810ST 104
 on MVP-2400..... 99
 on MVP-2410/3010..... 98
- Boot Version
 System Info (T1/E1)..... 192, 374
- booting time
 analog..... 33, 34
 BRI..... 40
 E1 25
 T1 18
- box contents
 verifying 89
- BRI connector pinout 539
- BRI interface types
 ST and U 540
- BRJ Bandwidth Rejection messages
 (gatekeeper, H.225)..... 485
- BRQ Bandwidth Request messages
 (gatekeeper, H.225)..... 485
- busy tone, custom
 analog..... 249
 T1/E1 166, 167
- busy-tones
 analog..... 248
 T1/E1 166
- Bytes Received (call progress) field
 378
- Bytes Received (SMTP logs) field
 analog..... 254
 T1/E1 172
- Bytes received (statistics, logs) field
 384
- Bytes Sent (call progress) field 378
- Bytes Sent (SMTP logs) field
 analog..... 254
 T1/E1 172
- Bytes sent (statistics, logs) field.... 383
- C**
- cable length, maximum span
 E1 154
 T1 149
- cabling diagram, quick
 analog models 53, 54, 56, 57

BRI models	55	Call Progress Details (statistics)	
E1 models	53	field	381
MVP130	57	Call Control Status (call progress)	
MVP210	56	field	381
MVP2400	56	Call Details (gatekeeper) screen....	498
MVP2410	53	Call Details (gatekeeper) screen,	
MVP3010	53	accessing	496
MVP-410/410G	54	Call Details button (gatekeeper	
MVP-410ST/810ST	55	Current Calls screen).....	496
MVP-810/810G	54	Call Details gatekeeper (Destination	
T1 models	53, 56	Info) screen fields	
cabling problem, fixing		Additional Phone Numbers	502
analog models	208	App. Bandwidth	502
T1/E1 models	131	Call Signalling IP	501
cabling procedure		Names	501
MVP130	107	Other Aliases	
MVP210x	105	Email	501
MVP2400	98	Trans. Name	501
MVP2410	97	URL	501
MVP3010	97	Phone Numbers	501
MVP410	99	Remote Extension Name.....	502
MVP-410ST	101	Remote Extension Phone	502
MVP810	99	Req. Bandwidth.....	502
MVP-810ST	101	Call Details gatekeeper (Source Info)	
Cadence 1 (custom) field		screen fields	
analog.....	250	App. Bandwidth	500
T1/E1	168	Call Signalling IP	500
Cadence 2 (custom) field		Names	500
analog.....	250	Other Aliases	
T1/E1	168	Email	500
Cadence 3 (custom) field		Trans. Name	500
analog.....	250	URL	500
T1/E1	168	Phone Numbers	500
Cadence 4 (custom) field		Req. Bandwidth.....	500
analog.....	250	Call Details gatekeeper screen fields	
T1/E1	168	Call ID Sum	498
Cadence field		Call Model	498
analog.....	247	Call No.	498
T1/E1	165	Cid Sum	498
cadences, custom		Conf. (conference)Goal.....	499
T1.E1	168, 250	Reason.....	499
T1/E1	166	State	499
cadences, signaling		Total BW.....	499
analog.....	244	Call Duration field	
T1/E1	162	analog.....	222
call authorization (gatekeeper).....	482	T1/E1	145
call control signalling (gatekeeper)482		Call Forward Parameters (inbound	
Call Control Status		phonebook)	

- E1 348
- T1 300
- Call Forwarded To
 - logs (statistics) field 385
- Call Hold
 - ANALOG 30
 - BRI 39
 - E1 24
 - T1 17
- Call Hold (analog) 262
- Call Hold (T1/E1) 180
- Call Hold Enable
 - analog 265
 - T1/E1 183
- Call ID Sum gatekeeper field (Call
 - Details) 498
- call IRQ interval 509
- Call IRQ Interval field (gatekeeper,
 - Network Parameters) 509
- call management (gatekeeper) 483
- Call Mode (SMTP logs) field
 - analog 254
 - T1/E1 172
- Call Mode field (gatekeeper, Network
 - Parameters) 507
- Call Models gatekeeper field (Call
 - Details) 498
- call modes 507
- Call Name Identification
 - ANALOG 30
 - BRI 39
 - E1 24
 - T1 17
- Call Name Identification (analog)
 - Alerting Party 268, 269, 270
 - Calling Party 267
- Call Name Identification (T1/E1)
 - Alerting Party 186, 187, 188
 - Calling Party 185
- Call Name Identification (analog) 262
- Call Name Identification (T1/E1) 180
- Call Number gatekeeper field (Call
 - Details) 498
- Call On Hold
 - Call Progress Details (statistics)
 - field 378, 380
- Call on Hold (call progress) field 380
- Call Proceeding field (gatekeeper,
 - Network Parameters) 506
- Call Progress (Statistics) 376
- Call Progress Details (statistics)
 - screen field
 - Call On Hold 378
 - Call Waiting 378
 - Caller ID 378
 - Call On Hold 380
 - Call Waiting 380
 - Caller ID 381
- Call Progress Details (statistics)
 - screen fields
 - Channel 378
 - Duration 378
 - Mode 378
 - Voice Coder 378
 - Packets Sent 378
 - Packets Received 378
 - Bytes Sent 378
 - Bytes Received 378
 - Packets Lost 378
 - Outbound Digits 378
 - Prefix Matched 378
 - Gateway Name 379
 - IP Address 379
 - Options 379
 - Silence Compression 379
 - Forward Error Correction 379
 - Status 381
 - Call Control Status 381
- call reasons (call details) listing 549
- call setup 484
- Call Signalling Port field
 - E1 333
 - T1 284
- call states (call details) listing 549
- Call Status (SMTP logs) field
 - analog 255
 - T1/E1 173
- call tear-down 484
- Call to Out-of-Service Supplier field
 - (gatekeeper, Network Parameters)
 - 505
- Call Transfer
 - ANALOG 30
 - BRI 39
 - E1 24

- T1 17
- Call Transfer (analog) 262
- Call Transfer (T1/E1) 180
- Call Transfer Enable
 - analog 264
 - T1/E1 182
- Call Transferred To
 - logs (statistics) field 385
- Call Waiting
 - ANALOG 30
 - BRI 39
 - Call Progress Details (statistics)
 - field 378, 380
 - E1 24
 - T1 17
- Call Waiting (analog) 262
- Call Waiting (call progress) field 380
- Call Waiting (T1/E1) 180
- Call Waiting Enable
 - analog 265
 - T1/E1 183
- Caller ID
 - Call Progress Details (statistics)
 - field 378, 381
 - Caller ID (analog) 262
 - Caller ID (call progress) field 381
 - Caller ID (Supplementary Services)
 - field
 - analog 271
 - T1/E1 189
 - Caller ID (T1/E1) 181
- Caller Name Identification Enable
 - analog 266
 - T1/E1 184
- calling area codes 73
- Calling Party
 - Supplementary Services (analog)
 - 267
 - Supplementary Services (T1/E1)
 - 185
- Canadian Class A requirements 529
- Canadian Limitations Notice
 - (regulatory) 530
- CAS Protocol field
 - E1 155
 - T1 150
- CAS Protocols, downloading 417
- CAS vs. CCS
 - T1 150, 155
- CCS vs. CAS
 - T1 150, 155
- CD
 - MultiVOIP 45
- Channel (call progress) field 378
- channel capacity 10
 - analog 26
 - BRI 35
 - E1 19
 - T1 12
- Channel Number (inbound) field
 - E1 347
 - T1 299
- Channel Number (SMTP logs) field
 - analog 254
 - T1/E1 172
- channel tracing on/off (logging)
 - analog 260
 - T1/E1 178
- Cid Sum gatekeeper field (Call
 - Details) 498
- city codes 73
- Clear (button), ISDN BRI Statistics
 - screen 404
- Clear (IP Statistics) button 387
- Client Options fields
 - E1 338
 - T1 289
- Clocking field
 - E1 157
 - T1 152
- coder (analog)
 - bandwidth, max 218
 - G.711 218
 - G.723.1 218
 - G.726 218
 - G.727 218
 - G.729 218
 - Net Coder 218
- Coder (SMTP logs) field
 - analog 254
 - T1/E1 172
- coder (T1/E1)
 - bandwidth, max 141
 - G.711 141
 - G.723.1 141
 - G.726 141

- G.727 141
- G.729 141
- Net Coder..... 141
- Coder field
 - analog..... 218
 - T1/E1 141
- coder options
 - packetization rates and..... 390
- Coder Parameters field group
 - analog..... 218
 - T1/E1 141
- coder types (voice/fax, RTP packetization)
 - T1/E1 391
- COL LED
 - analog models 33
 - BRI models 40
- COM port
 - on command PC..... 114
- COM port (analog models)
 - conflict, resolving 207
 - error message 207
- COM port (T1/E1 models)
 - conflict, resolving 130
 - error message 130
- COM port allocation
 - analog..... 272
 - T1/E1 190
- COM port assignments
 - analog..... 272
 - T1/E1 190
- COM port conflict
 - error message 114
- COM Port Setup screen 114
- COM Port Setup screen (analog models) 207
- COM Port Setup screen (T1/E1 models) 130
- comma
 - meaning/use in phonebook 75
- comma use
 - and second dial tone..... 75
- command cable pinout..... 536
- command PC
 - COM port assignment (detailed)114
 - COM port requirement..... 52
 - demands upon 52
 - non-dedicated use 52
 - operating system 52
 - settings 52
 - specifications..... 52
- Command PC
 - COM port requirement..... 41
 - non-dedicated use of 41
 - operating system 41
- community (voip) defined
 - analog..... 243
 - T1/E1 161
- Community Name 1 (SNMP) field
 - analog..... 243
 - T1/E1 161
- compatibility, Fast Start
 - E1 333
 - T1 284
- compatibility, H.450 with H.323, not with SIP
 - analog..... 27, 261
 - BRI..... 36
 - E1 20
 - T1 13
 - T1/E1 179
- compression standard
 - E1 157
 - T1 152
- compression, silence
 - analog..... 219
 - T1/E1 142
- Compression, Silence (SMTP logs)
 - analog..... 255
 - T1/E1 173
- computer requirements..... 41
- concurrent calls
 - maximum number 507
- concurrent calls supported, embedded gatekeeper 490
- Conf. (conference) Goal gatekeeper field (Call Details)..... 499
- conference media compatibility
 - H.225 and..... 484
- configuration of voip (analog)
 - local versus remote..... 197
- configuration of voip (T1/E1)
 - local versus remote..... 120, 121
- Configuration option (MultiVOIP program menu)..... 407

- Configuration Options gatekeeper field (Network Parameters)..... 504
- Configuration Parameter Groups, accessing
 - analog..... 208
 - T1/E1 131
- Configuration Parameters fields (gatekeeper, Network Parameters) 507, 508, 509, 510
- configuration procedure, local
 - detailed, analog..... 204
 - detailed, T1/E1..... 127
 - summary, analog..... 203
 - summary, T1/E1 126
- configuration, local
 - analog/BRI..... 199
 - T1/E1 122
- configuration, phonebook
 - E1 327
 - starter 66
 - T1 279
- configuration, saving
 - analog..... 275
 - T1/E1 193
 - user 422
- configuration, starter
 - phone/IP..... 59
- configuration, user default
 - analog..... 276
 - T1/E1 194
- Configuring MultiVOIP phonebooks, general
 - E1 321
 - T1 278
- confirming connectivity 84
- conflicts
 - COM port..... 114
- Connect TO (time-out) field (gatekeeper Memory screen) 491
- Connection Problems, Solving
 - analog..... 207
 - T1/E1 130
- connectivity
 - confirmation of 84
 - confirming with remote voip 51, 66
 - pinging and 85
- connectivity test..... 81
- Consecutive Packets Lost field
 - analog..... 222
 - T1/E1 145
- Console Message Settings, Filters for
 - analog..... 260
 - T1/E1 178
- console messages 61, 81, 83, 84
- console messages, enabling
 - analog..... 258
 - T1/E1 176
- console parameters tracked
 - analog..... 260
 - T1/E1 178
- contacting technical support..... 526
- coordinated phonebook entries
 - E1 327
 - T1 279
- Copy Channel command
 - analog..... 215
 - T1/E1 138
- Copy Channel field
 - analog..... 216
 - T1/E1 139
- Copy Channel, Supplementary Services command
 - analog..... 263
 - T1/E1 181
- Copy Channel, Supplementary Services field
 - analog..... 271
 - T1/E1 189
- Copy Interface command
 - BRI..... 237
- Count of Registered Numbers field (Registered Gateway Details) ... 406
- country
 - ISDN type and..... 158
 - switch type and ISDN 158
- Country (ISDN) field
 - E1/ISDN..... 156
- country codes 73
- Country definitions
 - ISDN-BRI 240
- Country field
 - ISDN-BRI 238
- Country field (ISDN)
 - T1/ISDN..... 151
- Country/Region (tone schemes) field
 - analog..... 246

T1/E1	164	Options.....	255
CRC and ESF frame format (T1)..	149	Options.....	255
CRC Check field		Description (callee).....	255
T1	149	Description (caller)	255
Creating a User Default Configuration		Duration	254
analog.....	276	From Gateway Number.....	255
T1/E1	194	From IP Address	255
CT Ph#		Outbound Digits.....	255
logs (statistics) field.....	385	Packets Lost	254
Current Bandwidth Usage gatekeeper		Packets Received	254
field (Network Parameters).....	503	Packets Sent	254
Current Calls (gatekeeper) fields		Prefix Matched.....	255
Call Details (button)	496	Select All.....	254
DEST IP.....	496	Start Date, Time	254
Disconnect All (button)	496	To Gateway Number.....	255
Disconnect Call (button).....	496	To IP Address	255
No (number)	496	Custom Fields, SMTP log email	
ORIG ALIAS.....	496	(T1/E1)	
ORIG IP.....	496	Bytes Received.....	172
Current Calls (gatekeeper) screen		Bytes Sent	172
accessing.....	495	Call Mode.....	172
Current Loss (FXO disconnect		Call Status	173
criteria) field	231	Channel Number	172
Current Loss field		Coder.....	172
FXS Ground Start	228	Options.....	173
FXS Loop Start.....	226	Options.....	173
Currently Registered gatekeeper field		Description (callee).....	173
(Network Parameters).....	503	Description (caller)	173
Custom (tones, Regional)field		Duration	172
analog.....	247	From Gateway Number.....	173
T1/E1	165	From IP Address	173
custom cadences		Outbound Digits.....	173
analog.....	250	Packets Lost	172
T1/E1	168	Packets Received	172
custom DTMF		Packets Sent	172
analog.....	249	Prefix Matched.....	173
T1/E1	166, 167	Select All.....	172
Custom Fields (SMTP) definitions		Start Date, Time	172
analog.....	254, 255	To Gateway Number.....	173
T1/E1	172, 173	To IP Address	173
Custom Fields, SMTP log email		Custom Tone-Pair Settings (analog)	
(analog)		fields	
Bytes Received	254	Cadence 1	250
Bytes Sent	254	Cadence 2	250
Call Mode	254	Cadence 3	250
Call Status.....	255	Cadence 4.....	250
Channel Number.....	254	Custom Tone-Pair Settings (T1/E1)	
Coder	254	fields	

- Cadence 1 168
 - Cadence 2 168
 - Cadence 3 168
 - Cadence 4 168
 - Custom Tone-Pair Settings definitions
 - analog..... 249, 250
 - T1/E1 167, 168
 - Custom Tone-Pair Settings fields
 - (analog)
 - Frequency 1 249
 - Frequency 2 249
 - Gain 1 249
 - Gain 2 249
 - Tone Pair..... 249
 - Custom Tone-Pair Settings fields (T1/E1)
 - Frequency 1 167
 - Frequency 2 167
 - Gain 1 167
 - Gain 2 167
 - Tone Pair..... 167
 - custom tones, setting
 - T1/E1 166
 - customized log email
 - analog..... 254, 255
 - T1/E1 172, 173
- D**
- D Channel Information fields (ISDN
 - BRI Statistics)..... 403
 - data capacity 10
 - analog..... 26
 - BRI 35
 - E1 19
 - T1 12
 - data compression
 - analog..... 27
 - BRI 36
 - E1 20
 - T1 13
 - Date & Time Setup (program menu option), command 409
 - Date and Time Setup option (MultiVOIP program menu) 407
 - DCF Disengagement Confirmation
 - messages (gatekeeper, H.225).... 485
 - Debug Level (Gatekeeper General Settings screen)..... 489
 - debugging messages
 - analog..... 259
 - T1/E1 176
 - Default (Supplementary Services) field
 - analog..... 271
 - T1/E1 189
 - Default (Voice/FAX) field
 - analog..... 216
 - T1/E1 139
 - default baud rate (MultiVOIP software connection)
 - analog..... 272
 - T1/E1 190
 - Default button (gatekeeper Memory screen)..... 491
 - default configuration, user
 - analog..... 276
 - T1/E1 194
 - default distance 510
 - Default Distance field (gatekeeper, Network Parameters)..... 510
 - Default gatekeeper field (Services, GK Defined)..... 513
 - Default gatekeeper field (Services, V2 GW Prefixes) 514
 - default values, software..... 419
 - defined services..... 516
 - delay, packets
 - analog..... 220
 - T1/E1 143
 - delay, versus voice quality
 - analog..... 221
 - T1/E1 144
 - Delete endpoints command
 - gatekeeper 494
 - Delete File button
 - Logs (Statistics) screen 383
 - Delete Predefined endpoints
 - command (Del Pre-def)
 - gatekeeper 494
 - Description (callee location)
 - E1 348
 - T1 300
 - Description (callee, outbound phonebook)
 - E1 341
 - T1 292

- Description field (Registered Gateway Details)..... 406
- Description gatekeeper field (Services, GK Defined)..... 513
- Description gatekeeper field (Services, V2 GW Prefixes)..... 514
- Description, From Details (SMTP logs) field
 analog..... 255
 T1/E1 173
- Description, To Details (SMTP logs) field
 analog..... 255
 T1/E1 173
- DEST IP field (gatekeeper Current Calls screen) 496
- Destination Pattern (outbound) field
 E1 341
 T1 292
- destination patterns
 digits used 73
 tips about..... 73
- destination patterns, discussion
 E1 326
 T1 278
- dial tone, custom
 analog..... 249
 T1/E1 166, 167
- dial tone, second
 and comma use 75
 pausing for 75
- Dialing Options (FXO) fields 230, 231
- Dialing Options field
 ISDN-BRI 238
- dialing patterns
 digits used 73
 inbound/outbound matching 75
 tips about..... 73
- dial-tones
 analog..... 248
 T1/E1 166
- DID
 and FXO interface..... 230
 FXS Ground Start 228
 FXS Loop Start 226
- digits in phonebook
 specialized codes 74
 types..... 73
- dimensions
 analog models 44
 E1 models..... 43
 T1 models..... 42
- direct call mode 507
- Direct Inward Dialing
 FXS Ground Start 226, 228
- direct mode (call control signalling) 482
- Direct Mode option (gatekeeper, Network Parameters)..... 507
- direct-mode calls 498
- Disabled Interface option 224
- Disconnect All button (gatekeeper Current Calls screen)..... 496
- Disconnect Call button (gatekeeper Current Calls screen)..... 496
- Disconnect endpoints command gatekeeper 494
- Disconnect on Call Progress Tone (FXO) field..... 232
- Disconnect Tone Sequence (FXO) field 232
- disconnection criteria, FXO .. 231, 232
- distances 510
- distances in networks 510
- DNS Server IP Address
 analog..... 211
 T1/E1 134
- Download CAS Protocol (program menu option) , command 417
- Download CAS Protocol option (MultiVOIP program menu) 407
- Download Factory Defaults (program menu option) , command 419
- Download Factory Defaults option (MultiVOIP program menu) 408
- Download Firmware (program menu option), command 413, 414
- Download Firmware option description (MultiVOIP program menu) 408
- Download User Defaults (program menu option) , command 421
- Download User Defaults option description (MultiVOIP program menu) 408

- downloading firmware, machine perspective 408, 431
 - downloading user defaults 421
 - downloads vs. uploads (FTP)..... 431
 - dropping digits, in phonebook 74
 - DRQ Disengagement Request messages (gatekeeper, H.225).... 485
 - DTMF
 - extended..... 232
 - standard..... 232
 - DTMF frequency chart 232
 - DTMF Gain (High Tones) field
 - analog..... 216
 - T1/E1 139
 - DTMF Gain (Low Tones) field
 - analog..... 216
 - T1/E1 139
 - DTMF Gain field
 - analog..... 216
 - T1/E1 139
 - DTMF In/Out of Band field
 - analog..... 217
 - T1/E1 140
 - DTMF inband
 - analog..... 217
 - T1/E1 140
 - DTMF out of band
 - analog..... 217
 - T1/E1 140
 - DTMF Parameters
 - T1/E1 139
 - DTMF, custom tone pairs
 - analog..... 249
 - T1/E1 166, 167
 - Duration (call progress) field..... 378
 - Duration (DTMF) field
 - analog..... 217
 - T1/E1 140
 - Duration (SMTP logs) field
 - analog..... 254
 - T1/E1 172
 - Duration (statistics, logs) field..... 383
 - dynamic endpoint registration (with gatekeeper)..... 508
 - Dynamic gatekeeper field (Services, V2 GW Prefixes) 515
 - Dynamic Jitter Buffer field
 - analog..... 220
 - T1/E1 143
 - Dynamic Jitter field group
 - analog..... 220
 - T1/E1 143
 - Dynamic Jitter fields
 - analog..... 221
 - T1/E1 144
- E**
- E&M interface (MVP210)
 - matching telco trunk line..... 106
 - uses of..... 106
 - E&M interface (MVP-410/810)
 - matching telco trunk line..... 100
 - uses of..... 100
 - E&M Interface Parameter fields
 - Interface 234
 - Pass Through..... 234
 - Signal 234
 - Type 234
 - Wink Timer 234
 - E&M Parameter definitions 234
 - E&M Parameters..... 233
 - E.164 phone numbers..... 481
 - E1 Parameter definitions 154, 155, 157
 - Clocking..... 157
 - Line Build-Out..... 157
 - Line Coding 157
 - PCM Law 157
 - Pulse Shape Level 157
 - E1 Parameter fields
 - CAS Protocol 155
 - CRC Check 155
 - Frame Format..... 155
 - Long-Haul Mode..... 155
 - E1 Parameters screen 153
 - E1 Statistics field definitions 398, 399, 400
 - E1 Statistics fields
 - Bipolar Variation 400
 - Blue Alarm..... 398
 - Excessive Zeroes..... 400
 - Loss of Frame Alignment..... 398
 - Loss of MultiFrame Alignment. 399
 - Receive Slip 400
 - Receive Timeslot 16 Alarm Indication Signal 399

Receive Timeslot 16 Loss of MultiFrame Alignment	400	subject line	61
Receive Timeslot 16 Loss of Signal	399	T1/E1	169
Red Alarm.....	398	email logs, illustration	
Status Freeze Signalling Active	399	analog.....	256
Transmit Data Overflow	399	T1/E1	174
Transmit Data Underrun	400	embedded gatekeeper capacities & capabilities	483
Transmit Line Open.....	400	EMC, Safety, R&TTE Directive Compliance	528
Transmit Line Short.....	399	emergency phone numbers caution about.....	75
Transmit Slip	400	Enable (Call Fwdg)	
Transmit Slip Negative.....	400	E1	348
Transmit Slip Positive.....	399	T1	300
Yellow Alarm	399	Enable Call Hold	
E1 telephony parameters.....	124	analog.....	265
E1/ISDN Parameter definitions ...	156	T1/E1	183
E1/ISDN Parameter fields		Enable Call Transfer	
Country	156	analog.....	264
Enable ISDN-PRI	156	T1/E1	182
Operator	156	Enable Call Waiting	
Terminal Network.....	156	analog.....	265
e164 aliases.....	501	T1/E1	183
Echo Cancellation field		Enable Caller Name Identification	
analog.....	219	analog.....	266
T1/E1	142	T1/E1	184
echo, removing		Enable Console Messages field	
analog.....	219	analog.....	259
T1/E1	142	T1/E1	176
Edit selected Inbound Phonebook		Enable Diffserv field	
Entry icon		analog.....	211
E1	328	T1/E1	134
T1	280	Enable DNS field	
Edit selected Outbound Phonebook		analog.....	211
Entry icon		T1/E1	134
E1	328	Enable ISDN-PRI field	
T1	280	E1/ISDN.....	156
email account for voip unit		T1/ISDN.....	151
analog.....	252	Enable Proxy field	
T1/E1	170	E1	334
email address for voip		T1	285
analog.....	202, 251	Enable SMTP field	
quick	51	analog.....	252
T1/E1	125, 169	T1/E1	170
email log reports		Enable SNMP Agent.....	159, 241
analog.....	251	Enable SNMP Agent field	
quick	60	analog.....	243
recipient	61	T1/E1	161
reply-to address.....	61		

- enabling SMTP
 - analog..... 251
 - T1/E1 169
 - enabling web browser GUI
 - analog..... 59, 212
 - T1/E1 135
 - endpoint types, gatekeeper..... 493
 - Error Correction (SMTP logs)
 - analog..... 255
 - T1/E1 173
 - error correction, forward
 - analog..... 219
 - T1/E1 142
 - error message
 - COM port conflict..... 114
 - COM port conflict (analog models)
 - 207
 - error message (analog models)
 - MultiVOIP Not Found..... 208
 - Phone Database Not Read..... 208
 - error message (T1/E1 models)
 - MultiVOIP Not Found..... 131
 - Phone Database Not Read..... 131
 - ESF and CRC frame format (T1).. 149
 - ethernet cable pinout..... 536
 - Ethernet interface
 - analog..... 26
 - BRI 35
 - Ethernet LEDs (analog)
 - COL 33
 - LNK 33
 - RCV 33
 - XMT 33
 - Ethernet LEDs (BRI)
 - COL 40
 - LNK..... 40
 - RCV 40
 - XMT 40
 - European Community Directives.. 528
 - Event # (statistics, logs) field..... 383
 - Excessive Zeroes (E1 stats) field .. 400
 - Excessive Zeroes (T1 stats) field .. 395
 - exchanges, phone
 - dedicated..... 74
 - institutional 74
 - local 74
 - non-local 74
 - organizational 74
 - Existing Endpoints (gatekeeper) fields
 - Msg 493
 - Existing Endpoints (gatekeeper) screen
 - accessing 492
 - Existing Endpoints (gatekeeper) screen commands
 - Add..... 494
 - Del Pre-def..... 494
 - Delete 494
 - Disconnect..... 494
 - Unregister..... 494
 - Unregister All..... 494
 - Existing Endpoints screen fields
 - Msg 493
 - Name 493
 - Online..... 493
 - Other Aliases..... 493
 - Phone 493
 - PreDef 493
 - Registration IP 493
 - TTL (TimeToLive timer) 493
 - Type 493
 - expansion card (analog, 4-to-8 channel) installation 544
 - expansion card (T1/E1) installation 532
 - expansion card (T1/E1)operation.. 534
- F**
- factory default software settings ... 419
 - factory defaults, downloading 419
 - factory repair for customers U.S. & Canada 524
 - failover (PSTN)
 - analog models 27
 - BRI models 36
 - E1 models..... 20
 - T1 models..... 13
 - failover (PSTN) feature..... 297
 - FAQ for MultiVOIPs 11
 - fast busy (unobtainable) tones
 - analog..... 166, 248
 - Fast Connect*See* Fast Start. *See* Fast Start
 - E1 336
 - T1 287
 - Fast Start compatibility

E1.....	333	forgotten password.....	424, 427
T1.....	284	Forward Address/Number	
Fast Start plus H.245 Tunneling field		E1.....	348
E1.....	336	T1.....	300
T1.....	287	Forward Condition (Call Fwdg)	
fax baud rate, default		E1.....	348
analog.....	217	T1.....	300
T1/E1.....	140	Forward Error Correction (call	
Fax Enable field		progress) field.....	379
analog.....	217	Forward Error Correction (SMTP	
T1/E1.....	140	logs)	
fax machine		analog.....	255
connecting to analog voip		T1/E1.....	173
(MVP130).....	107	Forward Error Correction field	
connecting to analog voip		analog.....	219
(MVP210).....	106	T1/E1.....	142
connecting to analog voip (MVP-		forward on busy	
410/810).....	100	T1.....	300, 348
FAX Parameters		Forward upon No Response	
analog.....	217	E1.....	348
T1/E1.....	140	T1.....	300
fax tones, output level		Forward, gatekeeper defined service	
analog.....	217	519
T1/E1.....	140	Frame Format field	
Fax Volume field		E1.....	154
analog.....	217	T1.....	149
T1/E1.....	140	frame relay, and fax	
FCC Declaration.....	528	analog.....	217
FCC Part 68 Telecom rules.....	529	T1/E1.....	140
FCC registration number.....	530	Frame Search Restart Flag (T1 stats)	
FCC rules, Part 15.....	528	field.....	396
features.....	483	Frame Type field	
Filters (Console Message Settings)		analog.....	211
analog.....	260	T1/E1.....	134
T1/E1.....	178	free calls	
Filters button (Console Message		E1.....	322
Settings)		T1.....	278
analog.....	259	frequencies, touch tone.....	232
T1/E1.....	177	Frequency 1 (custom tone) field	
firmware upgrade, implementing..	413	analog.....	249
Firmware Version		T1/E1.....	167
(analog).....	274	Frequency 1 (tone pair scheme)	
Firmware Version (System Info)		analog.....	246
T1/E1.....	192	T1/E1.....	164
firmware version, identifying.....	413	Frequency 2 (custom tone) field	
firmware, downloading.....	414	analog.....	249
firmware, obtaining updated.....	409	T1/E1.....	167
Flash Hook Timer field.....	230	Frequency 2 (tone pair scheme)	

- analog..... 246
- T1/E1 164
- frequency, power
 - analog models 44
 - E1 models 43
 - T1 models 42
- FRF11
 - analog..... 217
 - T1/E1 140
- From (gateway, statistics, logs) field
 - 383
- front panel
 - analog models 33
 - BRI models 40
 - E1 25
 - MVP2400..... 17
 - MVP2410..... 17
 - MVP3010..... 25
 - T1 17
- FTP client program 431
- FTP client program, obtaining 433
- FTP client programs
 - graphic vs. textual orientation... 440
- FTP file transfers
 - using FTP client program 433
 - using web browser 433
- FTP Server Enable field
 - analog..... 211
 - T1/E1 134
- FTP Server function
 - as added feature 431
 - enabling 433
- FTP Server, contacting 435
- FTP Server, invoking
 - download/transfer
 - using FTP client program 439
 - using web browser 437
- FTP Server, logging in..... 436
- FTP Server, logging out..... 440
- FTP transfers
 - file types 431, 434
 - phonebooks 431
 - server location..... 431
- function tracing on/off (logging)
 - analog..... 260
 - T1/E1 178
- FXO Current Detect Timer field... 231
- FXO Disconnect On fields.... 231, 232
- FXO disconnection criteria 231
- FXO disconnection, triggering of 231, 232
- FXO interface (MVP130)
 - uses of 107
- FXO interface (MVP210)
 - uses of 106
- FXO Interface Parameter definitions
 - 230, 231
- FXO Interface Parameter Definitions
 - 232
- FXO Interface Parameter fields
 - Disconnect on Call Progress Tone
 - 232
 - Disconnect Tone Sequence 232
 - Ring Count 232
 - Silence Detection 232
 - Silence Timer 232
- FXO interface(MVP-410/810)
 - uses of 100
- FXO Parameter fields
 - Current Loss 231
 - Flash Hook 231
 - FXO Current Detect Timer 231
 - Inter Digit Regeneration Timer 231
 - Inter Digit Timer (dialing) 230
 - Message Waiting Light 231
 - Regeneration (dialing)..... 230
 - Tone Detection 231
- FXO Parameters 229
- FXS Ground Start Interface parameter
 - definitions 227
- FXS Ground Start Parameter fields
 - Inter Digit Timer 227
 - Message Waiting Light 227
- FXS Ground Start Parameters 227
- FXS interface(MVP130)
 - uses of 107
- FXS interface(MVP210)
 - uses of 106
- FXS interface(MVP-410/810)
 - uses of 100
- FXS Loop Start Interface parameter
 - definitions 225
- FXS Loop Start Parameter fields
 - Current Loss 226
 - Inter Digit Timer 225
 - Message Waiting Light 225

Ring Count.....	226	Trans. Name" field (Call Details, Destination Info)	501
FXS Loop Start Parameters	225	Trans. Name" field (Call Details, Source Info)	500
FXS/FXO connector		gatekeeper "Registration TO (time- out)" field (Network Parameters)	508
MVP130.....	107	gatekeeper "Remove H.245 Addr in Call Hunt" field (Network Parameters)	505
MVP-210	105	gatekeeper "With H.245 Addr" option (Network Parameters, Call Proceeding)	506
MVP-410/810	100	Gatekeeper / Clear Channel IP Address (Gatekeeper RAS) field	
G		E1	333
G711 coders (RTP packetization, voice/fax)		T1	284
T1/E1	391	gatekeeper Add-endpoints command	494
G723 coders (RTP packetization, voice/fax)		gatekeeper Additional Phone Numbers field (Call Details)	502
T1/E1	391	gatekeeper address translation messages (H.225)	
G726 coders (RTP packetization, voice/fax)		LCF (Location Confirmation) ...	486
T1/E1	391	LRQ (Location Rejection)	486
G727 coders (RTP packetization, voice/fax)		LRQ (Location Request)	486
T1/E1	391	gatekeeper admission control messages (H.225)	
G729 coders (RTP packetization, voice/fax)		ACF (Admission Confirmation)	485
T1/E1	391	ARJ (Admission Rejection)	485
Gain 1 (custom tone) field		ARQ (Admission Request)	485
analog.....	249	DCF (Disengagement Confirmation).....	485
T1/E1	167	DRQ (Disengagement Request)	485
Gain 1 (tone pair scheme)		gatekeeper Alias Giving field (Network Parameters)	504
analog.....	246	gatekeeper App. Bandwidth field (Call Details, Destination Info) .	502
T1/E1	164	gatekeeper App. bandwidth field (Call Details, Source Info)	500
Gain 2 (custom tone) field		gatekeeper bandwidth control messages (H.225)	
analog.....	249	BCF (Bandwidth Confirmation)	485
T1/E1	167	BRJ (Bandwidth Rejection)	485
Gain 2 (tone pair scheme)		BRQ (Bandwidth Request)	485
analog.....	246	gatekeeper bandwidth management	481
T1/E1	164	Gatekeeper Basics	481
gatekeeper			
registration with	492		
gatekeeper	491, 500, 501		
gatekeeper "After Overlapped Sending" option (Network Parameters, Call Proceeding)	506		
gatekeeper "Max Total BW" field (Network Parameters)	507		
gatekeeper "Other Aliases Email" field (Call Details, Destination Info)	501		
Email" field (Call Details, Source Info)	500		

- gatekeeper Call Details button
(Current Calls) 496
- gatekeeper Call ID Sum field (Call
Details)..... 498
- gatekeeper Call IRQ Interval field
(Network Parameters)..... 509
- gatekeeper Call Mode fields (Network
Parameters) 507
- gatekeeper Call Model field (Call
Details)..... 498
- gatekeeper Call No. field (Call
Details)..... 498
- gatekeeper Call Proceeding fields
(Network Parameters)..... 506
- gatekeeper Call Signalling IP field
(Call Details, Destination Info) . 501
- gatekeeper Call Signalling IP field
(Call Details, Source Info)..... 500
- gatekeeper Cid Sum field (Call
Details)..... 498
- gatekeeper Conf. Goal field (Call
Details)..... 499
- gatekeeper Configuration Options
field..... 504
- gatekeeper Configuration Options
field (Network Parameters)..... 504
- gatekeeper Configuration Parameters
fields (Network Parameters)507,
508, 509, 510
- gatekeeper Connect TO field (GK
General Settings, Q.931
Parameters) 491
- gatekeeper Current Bandwidth Usage
field..... 503
- gatekeeper Current Bandwidth Usage
field (Network Parameters)..... 503
- gatekeeper Currently Registered field
..... 503
- gatekeeper Currently Registered field
(Network Parameters)..... 503
- gatekeeper Default Distance field
(Network Parameters)..... 510
- gatekeeper Default field (Services,
GK Defined) 513
- gatekeeper Default field (Services, V2
GW Prefixes) 514
- gatekeeper defined services, built-in
Forward..... 519
- Zone Prefixes 1 and 2..... 517
- gatekeeper Delete-endpoints
command..... 494
- gatekeeper Delete-predefined-
endpoints command 494
- gatekeeper Description field
(Services, GK Defined)..... 513
- gatekeeper Description field
(Services, V2 GW Prefixes)..... 514
- gatekeeper DEST IP field (Current
Calls)..... 496
- gatekeeper Direct Mode option
(Network Parameters, Call Mode)
..... 507
- gatekeeper Disconnect All button
(Current Calls) 496
- gatekeeper Disconnect Call button
(Current Calls) 496
- gatekeeper Disconnect-endpoints
command..... 494
- gatekeeper Dynamic field (Services,
V2 GW Prefixes)..... 515
- gatekeeper endpoint types 493
- gatekeeper Endpoints fields
Msg 493
- gatekeeper functionality 450
- gatekeeper functions
optional 482
- gatekeeper functions, mandatory ..481
- gatekeeper GK Defined Services
fields..... 514
- gatekeeper GK-ID field (Network
Parameters) 511
- gatekeeper interaction
analog models 27
- BRI models 36
- E1 models..... 20, 21
- T1 models..... 13, 14
- gatekeeper IRQ Interval field
(Network Parameters) 508
- gatekeeper Line Hunting Information
fields (Network Parameters) 505
- gatekeeper Max Number of Calls field
(Network Parameters) 507
- gatekeeper Maximum Calls field (GK
General Settings, Memory) 490

- gatekeeper Maximum Registrations field (GK General Settings, Memory) 490
- gatekeeper Multicast Distance field (Network Parameters) 511
- Gatekeeper Name (Gatekeeper RAS) field
E1 333
T1 284
- gatekeeper Name field (Existing Endpoints) 493
- gatekeeper Names field (Call Details, Destination Info) 501
- gatekeeper Names field (Call Details, Source Info) 500
- gatekeeper No. (number) field (Current Calls) 496
- gatekeeper Ongoing Calls field 503
- gatekeeper Ongoing Calls field (Network Parameters) 503
- gatekeeper Online field (Existing Endpoints) 493
- gatekeeper ORIG ALIAS field (Current Calls) 496
- gatekeeper ORIG IP field (Current Calls) 496
- gatekeeper Other Aliases field (Existing Endpoints) 493
- gatekeeper Out-of-Zone Distance field (Network Parameters) 510
- gatekeeper Phone field (Existing Endpoints) 493
- gatekeeper Phone Numbers field (Call Details, Destination Info) 501
- gatekeeper Phone Numbers field (Call Details, Source Info) 500
- gatekeeper PreDef field (Existing Endpoints) 493
- gatekeeper Prefix field (Services, GK Defined) 513
- gatekeeper Prefix field (Services, V2 GW Prefixes) 514
- gatekeeper PreGrant All field (Network Parameters) 504
- gatekeeper protocols 484
- gatekeeper Public field (Services, GK Defined) 514
- gatekeeper Public field (Services, V2 GW Prefixes) 514
- GateKeeper RAS Parameters
T1 284
- gatekeeper RAS Port field (GK General Settings, RAS Parameters) 491
- gatekeeper Reason field (Call Details) 499
- gatekeeper registration capacity 483
- gatekeeper registration control messages (H.225)
IRQ (Information Request) 487
IRR (Extend Registration Request) 487
RCF (Registration Confirmation) 486
RRJ (Registration Rejection) 486
RRQ (Registration Request) 486
URQ (Unregister Request) 487
- gatekeeper Registration IP field (Existing Endpoints) 493
- gatekeeper Remote Extension Name field (Call Details) 502
- gatekeeper Remote Extension Phone field (Call Details) 502
- gatekeeper Req. bandwidth field (Call Details, Destination Info) 502
- gatekeeper Req. bandwidth field (Call Details, Source Info) 500
- gatekeeper Response TO field (GK General Settings, Q.931 Parameters) 491
- gatekeeper Response TO field (GK General Settings, RAS Parameters) 491
- gatekeeper Routed Mode option (Network Parameters, Call Mode) 507
- gatekeeper Send Immediately option (Network Parameters, Call Proceeding) 506
- gatekeeper service (user defined), example 516
- gatekeeper Service Configurable Properties field (Network Parameters) 505
- gatekeeper software license 521

gatekeeper State field (Call Details)	E1	333
.....	T1	284
gatekeeper Status Information fields	Gateway Prefix (outbound	
.....	phonebook) field	
gatekeeper Status Information fields	E1	342
(Network Parameters)	T1	293
gatekeeper Time-To-Live (TTL) timer	gateway-supported services	514
field	General Options fields	
gatekeeper Total BW field (Call	E1	337
Details)	T1	288
gatekeeper Type field (Existing	GK (gatekeeper) General Settings	
Endpoints)	fields	488, 489, 490, 491
gatekeeper Unregister-All-endpoints	GK (gatekeeper) General Settings	
command	screen	488
gatekeeper Unregister-endpoints	GK (gatekeeper) General Settings	
command	screen fields	
gatekeeper V2 GW Prefixes fields	Activity Configuration	489
gatekeeper, embedded	Debug Level	489
gatekeeper, example system	Memory Settings (button)	489
gatekeeper, registration with	Registration Policy	488
gatekeeper-defined services, built-in	GK Active option (Gatekeeper	
Zone Prefix 1	General Settings screen)	489
Gateway (IP Parameters) field	GK Defined Service Types	516
analog	GK Defined Services field	
T1/E1	(gatekeeper, Services)	514
Gateway H.323 ID (Gatekeeper RAS)	GK identifier	511
field	GK-ID field (gatekeeper, Network	
E1	Parameters)	511
T1	grounding	
Gateway Name (call progress) field	in rack installations	95
.....	MVP210	106
Gateway Name (callee, statistics,	MVP410	100
logs) field	MVP410ST	104
Gateway Name (caller, statistics, logs)	MVP810	100
field	MVP810ST	104
Gateway Name field	grounding screw, diagrams	
E1	(MVP-2410/3010)	53
T1	(MVP-410/410G/810/810G)	54
Gateway Number, From Details	(MVP-410ST/810ST)	55
(SMTP logs) field	GUI (log reporting type) button	
analog	analog	259
T1/E1	T1/E1	177
Gateway Number, To Details (SMTP		
logs) field	H	
analog	H.225 protocol and gatekeeper	484
T1/E1	H.225 RAS messages	481
Gateway Prefix (Gatekeeper RAS)	H.245	
field		

- conference media compatibility and 484
 - H.245 Tunneling field
 - E1 335
 - T1 286
 - H.320 514
 - H.323
 - compatibility (analog models) 26
 - compatibility (BRI models) 36
 - compatibility (E1 models) 20
 - compatibility (T1 models) 13
 - aliases 500, 501, 514
 - Annex E field
 - E1 336
 - T1 287
 - coder
 - analog 218
 - T1/E1 141
 - fields (Outbound Phonebook)
 - E1 342
 - T1 293
 - gatekeeper protocols 484
 - ID (Outbound Phonebook) field
 - T1 293, 342
 - version 4 features
 - analog 27
 - BRI 36
 - E1 20
 - T1 13
 - Version 4 Parameters
 - E1 335, 336
 - T1 286, 287
 - H.450 features, incompatible with SIP
 - analog 27, 261
 - BRI 36
 - E1 20
 - T1 13
 - T1/E1 179
 - H.450 functionality
 - logs for 385
 - H.450 standard
 - ANALOG 30
 - BRI 39
 - E1 24
 - T1 17
 - Hold Sequence
 - analog 265
 - T1/E1 183
 - Hold Sequence (analog) 262
 - Hold Sequence (T1/E1) 180
 - hookup
 - MVP130 57
 - MVP210 56
 - MVP2400 56
 - MVP2410 53
 - MVP3010 53
 - MVP-410/410G 54
 - MVP-410ST/810ST 55
 - MVP-810/810G 54
 - HyperTerminal program
 - and connectivity testing 82
- I**
- IANA 542
 - icon
 - variable version 11, 111
 - icons, phonebook
 - E1 328
 - T1 280
 - identifying current firmware version
 - 413
 - implementing firmware upgrade ... 413
 - in band, DTMF
 - analog 217
 - T1/E1 140
 - inbound phonebook
 - example 76
 - Inbound Phonebook Entries List icon
 - E1 328
 - T1 280
 - Inbound Phonebook entries, list
 - E1 346
 - T1 298
 - inbound phonebook example
 - quick 70
 - inbound vs. outbound phonebooks
 - E1 326
 - T1 278
 - Industry Canada requirements 529
 - info sources
 - analog telephony details 49, 200
 - BRI telephony details 50
 - E1 details 49
 - E1 telephony details 124
 - IP details 48

IP details (analog system)	199	BRI models	35
IP details (T1/E1 system).....	122	E1 models.....	19
ISDN-BRI telephony details.....	201	T1 models.....	12
SMTP details	51	installation, quick	
T1 details	48	log reports by email.....	51
T1 telephony details.....	123	voip email account	51
voip email account.....	51	installing Java vis-a-vis web GUI .	443
info sources (analog models)		integrated phone/data networks.....	321
SMTP details	202	Inter Digit Regeneration Time field	
voip email account.....	202	231
info sources (T1/E1 models)		Inter Digit Timer (dialing) field	
SMTP details	125	FXO	230
voip email account.....	125	FXS Ground Start	227
Input Gain field		FXS Loop Start	225
analog.....	216	Interface (telephony) Disabled.....	224
T1/E1	139	Interface field (E&M)	234
installation		interface parameters, accessing.....	223
airflow.....	95	interface parameters, setting.....	223
analog prerequisites	199, 200	interface types, BRI	
BRI prerequisites	50	ST and U	540
E1 prerequisites	49, 124	interfaces	
expansion card (analog, 4-to-8		analog telephony	54
channel)	544	BRI telephony	55
expansion card (T1/E1).....	532	inter-office dialing	
full summary.....	47	E1	322
in a nutshell.....	45	T1	279
in rack	94	inter-operation (analog)	
IP prerequisites	48	with T1/E1 voips.....	26
ISDN-BRI prerequisites.....	201	inter-operation (BRI)	
log reports by email (analog		with T1/E1/BRI voips	35
models)	202	inter-operation with phone system	
log reports by email (T1/E1		analog models	26
models)	125	BRI models	35
software (detailed).....	109	E1 models.....	19
T1 prerequisites	48, 123	T1 models.....	12
T1/E1 prerequisites.....	122	IP Address (call progress) field.....	379
upgrade card (analog, 4-to-8		IP Address (callee, statistics, logs)	
channel)	544	field	384
upgrade card (T1/E1).....	532	IP Address (caller, statistics, logs)	
voip email account(analog models)		field	384
.....	202	IP Address (outbound phonebook)	
voip email account(T1/E1 models)		E1	341
.....	125	T1	292
installation preparations (optional)		IP Address field	
log reports by email	51	analog.....	211
voip email account.....	51	T1/E1	134
installation, mechanical		IP Address field (Registered Gateway	
analog models	26	Details).....	406

- IP Address, From Details (SMTP logs) field
 analog..... 255
 T1/E1 173
- IP address, SysLog Server
 analog..... 259
 T1/E1 177
- IP Address, To Details (SMTP logs) field
 analog..... 255
 T1/E1 173
- IP Mask field
 analog..... 211
 T1/E1 134
- IP parameter definitions
 analog..... 211
 T1/E1 134
- IP Parameter fields (analog)
 DNS Server IP Address 211
 Enable Diffserv 211
 Enable DNS 211
 Frame Type 211
 FTP Server Enable 211
 Gateway 211
 IP Address 211
 IP Mask 211
- IP Parameter fields (T1/E1)
 DNS Server IP Address 134
 Enable Diffserv field 134
 Enable DNS 134
 Frame Type 134
 FTP Server Enable 134
 Gateway 134
 IP Address 134
 IP Mask 134
- IP Parameters screen, accessing
 analog..... 209
 T1/E1 132
- IP startup configuration 59
- IP Statistics field definitions . 386, 388
- IP Statistics fields
 Clear..... 386
 Received (RTCP Packets)..... 389
 Received (RTP Packets) 389
 Received (TCP Packets) 388
 Received (Total Packets) 386
 Received (UDP Packets)..... 388
- Received with errors (RTCP Packets)..... 389
- Received with errors (RTP Packets) 389
- Received with errors (TCP Packets) 388
- Received with errors (Total Packets) 388
- Received with errors (UDP Packets) 388
- Transmitted (RTCP Packets)..... 389
- Transmitted (RTP Packets) 389
- Transmitted (TCP Packets) 388
- Transmitted (Total Packets) 386
- Transmitted (UDP Packets)..... 388
- IP Statistics function 386
- IRQ Information Request messages (gatekeeper, H.225)..... 487
- IRQ interval 508
- IRQ Interval field (gatekeeper, Network Parameters)..... 508
- IRQ polling 509
- IRR Extend Registration Request messages (gatekeeper, H.225).... 487
- ISDN BRI Interface screen fields
 Status, Layer 1 Interface 402
 Status, SPID0 403
 Status, SPID1 404
- ISDN BRI Parameters
 TEI n Assignment 239
- ISDN BRI Parameters fields
 A-Law 239
 Country 238
 Dialing Options 238
 Inter Digit Timer 238
 Layer 1 Interface 238
 MU-Law 239
 Operator 238
 PCM Law 239
 Select BRI Interface 238
 SPID 0 239
 SPID 1 239
 Switch Information 238
- ISDN BRI Statistics screen fields
 Clear (button) 404
 D Channel Information (field group)..... 403
 Layer 1 Interface (field group).. 402

- Loss of Framing..... 402
 - Loss of Sync 402
 - Rx Packets 403
 - Select BRI Interface..... 402
 - SPIDO 403
 - SPID1 404
 - State 402
 - Switch Information (field group)
 - 403
 - Tx Packets..... 403
 - ISDN parameters, setting..... 158
 - ISDN-BRI operating modes
 - MVP-410ST/810ST)..... 103
 - ISDN-BRI Parameter definitions.. 238
 - ISDN-BRI telephony interfaces
 - uses of..... 103
 - ISDN-BRI telephony parameters.. 201
 - ISDN-PRI
 - types supported 158
 - ISDN-PRI implementations..... 158
- J**
- Java
 - installing 443
 - web GUI and..... 443
 - jitter buffer
 - analog..... 220
 - T1/E1 143
 - Jitter Value (Fax) field
 - analog..... 217
 - T1/E1 140
 - Jitter Value field
 - analog..... 222
 - T1/E1 145
 - jitter, dynamic
 - analog..... 220
 - T1/E1 143
- K**
- Keep Alive field
 - E1 338
 - T1 289
 - key system
 - connecting to analog voip
 - (MVP130)..... 107
 - connecting to analog voip
 - (MVP210)..... 106
 - connecting to analog voip (MVP-
 - 410/810)..... 100
 - Knowledge Base (online, for
 - MultiVOIPs) 11
- L**
- lab voip network
 - use in setup..... 75
 - Last button
 - Logs (Statistics) screen 383
 - Layer 1 Interface
 - ISDN-BRI 238
 - Layer 1 Interface fields (ISDN BRI
 - Statistics)..... 402
 - LCF Location Confirmation messages
 - (gatekeeper, H.225)..... 486
 - LED definitions
 - analog models 33
 - BRI models 40
 - E1 25
 - MVP2400 17
 - MVP2410 17
 - MVP3010 25
 - T1 17
 - LED definitions (analog)
 - Boot..... 33, 34
 - COL 33, 34
 - Ethernet 33, 34
 - LNK 33
 - Power 33, 34
 - RCV (channel) 33, 34
 - RCV (Ethernet) 33
 - RSG 33, 34
 - XMT (channel)..... 33, 34
 - XMT (Ethernet) 33
 - XSG 33, 34
 - LED definitions (BRI)
 - Boot..... 40
 - COL 40
 - Ethernet 40
 - LNK 40
 - Power 40
 - RCV (channel) 40
 - RCV (Ethernet) 40
 - XMT (channel)..... 40
 - XMT (Ethernet) 40
 - LED definitions (E1)
 - Boot..... 25

COL	25	E1	157
E1	25	T1	152
IC	25	Line Coding field	
LC	25	E1	157
LNK	25	T1	152
LS	25	Line Hunting Information field	
ONL	25	(gatekeeper, Network Parameters)	
Power	25	505
PRI	25	Line Loopback Activation Signal (T1	
RCV	25	stats) field.....	396
XMT	25	Line Loopback Deactivation Signal	
LED definitions (T1)		(T1 stats) field.....	395
Boot	18	List of Registered Numbers field	
COL	18	(Registered Gateway Details) ...	406
IC	18	lithium battery caution	88
LC	18	LNK LED	
LNK	18	analog models	33
LS	18	BRI models	40
ONL	18	load balancing (gatekeeper)	483
Power	18	loading of weight in rack	95
PRI	18	local configuration	
RCV	18	analog/BRI	199
T1	18	T1/E1	122
XMT	18	local configuration procedure	
LED indicators		detailed, analog	204
E1	24	detailed, T1/E1	127
T1	17	summary, analog	203
LED indicators (analog)		summary, T1/E1	126
channel operation.....	31	local exchange numbers	74
general operation	31	local voip configuration (analog) ..	197
LED indicators (BRI)		local voip configuration (T1/E1)...	120
channel operation.....	39	local Windows GUI vs. web GUI	
general operation	39	comparison.....	442
LED indicators, active		local-rate access (E1)	
analog.....	31	to remote PSTN.....	19
E1	24	local-rate calls to remote voip sites	
T1	17	E1	323
LED sets (T1/E1), left and right ...	534	log report email, customizing	
LED types		analog.....	254, 255
analog models	31	T1/E1	172, 173
BRI models	39	log report email, triggering	
license, gatekeeper software	521	analog.....	253
lifting		T1/E1	171
precaution about.....	88	log reporting method, setting	
limitations notice (regulatory),		analog.....	257
Canadian	530	T1/E1	175
limited warranty	524	log reports	
Line Build Out field		analog models	202

T1/E1 models.....	125	Logs (Statistics) screen	
log reports & SMTP		Delete File button.....	383
analog.....	251	Last button	383
T1/E1	169	logs and web browser GUI	
log reports and SMTP		analog.....	258
quick	60	T1/E1	176
log reports by email		logs by email, illustration	
analog.....	251	analog.....	256
quick	60	T1/E1	174
T1/E1	169	Logs screen definitions	
log reports, quick	51	analog.....	258
logging options		T1/E1	176
analog.....	258	Logs screen field definitions	
T1/E1	176	analog.....	259
logging update interval		T1/E1	177
analog.....	258	Logs screen parameters (analog)	
T1/E1	176	Enable Console Messages	259
logging, web GUI and.....	442	Filters	259
Login Name (SMTP) field		GUI	259
analog.....	252	IP Address (SysLog Server).....	259
T1/E1	170	Online Statistics Updation Interval	
Logs (Statistics) fields		259
Bytes received.....	384	Port (SysLog Server).....	259
Bytes Sent	383	SMTP	259
Call Forwarded to	385	SNMP.....	259
Call Transferred to	385	SysLog Server Enable.....	259
CT Ph#.....	385	Turn Off Logs	259
Duration.....	383	Logs screen parameters (T1/E1)	
Event #.....	383	Console Message Settings.....	177
From (gateway).....	383	Enable Console Messages	176
Gateway Name (callee).....	384	Filters	177
Gateway Name (caller).....	384	GUI	177
H.450 functionality	385	IP Address (SysLog Server).....	177
IP Address (callee).....	384	Online Statistics Updation Interval	
IP Address (caller).....	384	177
Mode.....	383	Port (SysLog Server).....	177
Options (caller).....	384	SMTP	177
Options callee	384	SNMP.....	177
Outbound digits	384	SysLog Server Enable.....	177
Packets Lost.....	384	Turn Off Logs	177
Packets received.....	384	logs screen, accessing	
Packets Sent.....	383	analog.....	257
Start Date, Time.....	383	T1/E1	175
Status	383	long distance call savings	
Supplementary Services info	385	T1	278
To (gateway).....	383	long-distance call savings	
Voice coder.....	384	E1	321
Logs (Statistics) function.....	382	Long-Haul Mode field	

- E1 154
- T1 149
- Loss of Frame Alignment (E1 stats) field 398
- Loss of Frame Alignment (T1 stats) field 395
- Loss Of Framing field (ISDN BRI Statistics, Layer 1 Interface) 402
- Loss of MultiFrame Alignment (E1 stats) field 399
- Loss of MultiFrame Alignment (T1 stats) field 396
- Loss of Sync field (ISDN BRI Parameters, Layer 1 Interface) .. 402
- lost packets, consecutive
 - analog 222
 - T1/E1 145
- lost password 424, 427
- LRJ Location Request Rejection messages (gatekeeper, H.225).... 486
- LRQ Location Request messages (gatekeeper, H.225) 486, 493
- M**
- Mac Address
 - System Info (analog)..... 274
 - System Info (T1/E1) 192, 374
- mail criteria (SMTP), records
 - analog 253
 - T1/E1 171
- Mail Server IP Address (SMTP) field
 - analog 252
 - T1/E1 170
- Mail Type (SMTP logs) field
 - analog 253
 - T1/E1 171
- mains frequency
 - analog models 44
 - E1 models 43
 - T1 models 42
- management (E1 models)
 - local 21
 - remote (SNMP)..... 21
 - remote (web browser GUI) 21
- management of voips, remote
 - analog 241
 - T1/E1 159
- Max bandwidth (coder)
 - analog 218
 - T1/E1 141
- Max Baud Rate field
 - analog 217
 - T1/E1 140
- Max Number of Calls field (gatekeeper, Network Parameters) 507
- Max Retransmission (SPP, General Options) field
 - E1 337
 - T1 288
- Max Total BW field (gatekeeper, Network Parameters)..... 507
- maximum cable span
 - E1 154
 - T1 149
- Maximum Calls field (Gatekeeper General Settings, Memory) 490
- Maximum Jitter Value field
 - analog 221
 - T1/E1 144
- maximum number of concurrent calls 507
- Maximum Registrations field (Gatekeeper General Settings, Memory) 490
- Memory (Gatekeeper General Settings) screen fields
 - GK Memory Values 490
 - Maximum Calls 490
 - Maximum Registrations 490
 - Q.931 Parameters 491
 - RAS Parameters 491
 - RAS Port 491
 - Response TO (time-out, RAS) .. 491
- Memory (Gatekeeper General Settings) secondary screen 490
- Memory Settings button (Gatekeeper General Settings screen)..... 489
- Message Waiting Light (FXO) and Avaya Magix PBX 230
- and DID 230
- Message Waiting Light (FXS Ground Start) and Avaya Magix PBX 228
- and DID 228

Message Waiting Light (FXS Loop Start) and Avaya Magix PBX and DID	226	E1 models	21
Message Waiting Light field		T1 models	14
FXO	230	MultiVOIP FAQ (on MTS web site)	11
FXS Ground Start	228	MultiVOIP Program Menu items	407
FXS Loop Start	226	MultiVOIP Program Menu options	
Minimum Jitter Value field		Configuration	407
analog	220	Date & Time Setup	407
T1/E1	143	Download CAS Protocol	407
Mode (call progress) field	378	Download Factory Defaults	408
Mode (Fax) field		Download Firmware	408
analog	217	Set Password	408
T1/E1	140	Uninstall	408
Mode (SPP) field		Upgrade Software	408
E1	337	MultiVOIP program menu, option descriptions	407, 408
T1	288	MultiVOIP software	
Mode (statistics, logs) field	383	installing	109
model descriptions		location of files	112
E1	19	program icon location	113
modem relay		uninstalling	116, 428
analog	221	MultiVOIP software (analog)	
T1/E1	144	moving around in	208
modem traffic on voip network		MultiVOIP software (T1/E1)	
analog	221	moving around in	131
T1/E1	144	MultiVoipManager	11
mounting		analog	197
analog models	26	T1/E1	121
BRI models	35	MultiVoipManager software	
E1 models	19	E1 models	21
T1 models	12	T1 models	14
mounting in rack	94	MVP130	
procedure for	96	cabling procedure	107
safety	88, 95	Introduction	26
mounting options	10	unpacking	93
multicast		MVP210	
distance	511	grounding	106
Multicast Distance field (gatekeeper, Network Parameters)	511	MVP210x	
Multiplexed UDP field		cabling procedure	105
E1	336	unpacking	92
T1	287	MVP2400	
MultiVOIP 110/120/200/400/800 field (Outbound Phonebook)		cabling procedure	98
E1	344	unpacking	90
T1	295	MVP2410	
MultiVOIP configuration software	58	cabling procedure	97
		unpacking	89
		MVP3010	
		cabling procedure	97

- unpacking..... 89
- MVP410
 - cabling procedure..... 99
 - grounding..... 100
- MVP410ST
 - grounding..... 104
- MVP-410ST
 - cabling procedure..... 101
- MVP410x
 - unpacking..... 91
- MVP810
 - cabling procedure..... 99
 - grounding..... 100
- MVP810ST
 - grounding..... 104
- MVP-810ST
 - cabling procedure..... 101
- MVP810x
 - unpacking..... 91
- N**
- Name field (gatekeeper)..... 493
- Names gatekeeper field (Call Details, Destination Info)..... 501
- Names gatekeeper field (Call Details, Source Info)..... 500
- national-rate calls to foreign voip sites
 - E1..... 325
- neighbor gatekeepers..... 510
- neighboring zones
 - gatekeeper..... 482
- Netcoder coders (RTP packetization, voice/fax)
 - T1/E1..... 391
- network access
 - analog..... 243
 - T1/E1..... 161
- Network Disconnection field
 - analog..... 222
 - T1/E1..... 145
- Network Parameters (gatekeeper)
 - screen
 - accessing..... 503
 - Update button..... 511
- Network Parameters (gatekeeper)
 - screen fields
 - After Overlapped Sending (Call Proceeding option)..... 506
- Call IRQ Interval..... 509
- Call Mode..... 507
- Call Proceeding..... 506
- Call to Out-of-Service Supplier 505
- Configuration Parameters 507, 508, 509, 510
- Default Distance..... 510
- Direct Mode (Call Mode option)
 - 507
- GK-ID..... 511
- IRQ Interval..... 508
- Line Hunting Information..... 505
- Max Number of Calls..... 507
- Max Total BW (Kbps)..... 507
- Multicast Distance..... 511
- Out-of-Zone Distance..... 510
- Registration TO (time-out)..... 508
- Routed Mode (Call Mode option)
 - 507
- Send Immediately (Call Proceeding option)..... 506
- Service Configurable Properties (Line Hunting Information)... 505
- With H.245 Addr (Call Proceeding option)..... 506
- Network Parameters (gatekeeper)
 - screen fields:..... 505
- Network Parameters gatekeeper
 - screen fields
 - Alias Giving..... 504
 - Current BW Usage..... 503
 - Currently Registered..... 503
 - Ongoing Calls..... 503
 - PreGrant All..... 504
- network/terminal settings, voip and PBX
 - E1/ISDN..... 156
 - ISDN-BRI..... 238
 - T1/ISDN..... 151
- No (number) field (gatekeeper
 - Current Calls screen)..... 496
- No endpoints option (Gatekeeper
 - General Settings screen)..... 488
- No. of Entries field (Registered
 - Gateway Details)..... 406
- NT1 device
 - when required for MVP410ST.. 102
 - when required for MVP810ST.. 102

- NT1 device, use of
 BRI voip units..... 50, 102
- Number of Days (email log criteria)
 analog..... 253
 T1/E1 171
- Number of Records (email log
 criteria)
 analog..... 253
 T1/E1 171
- numbering plan resources 369
- O**
- obtaining updated firmware 409
- official phone numbers
 caution about..... 75
- Ongoing Calls gatekeeper field
 (Network Parameters)..... 503
- Online field (gatekeeper) 493
- Online Statistics Updation Interval
 field (Logs)
 analog..... 259
 T1/E1 177
- operating system 41
- operating temperature 95
- operating voltage
 analog models 44
 T1 models 42, 43
- operation
 expansion card (T1/E1)..... 534
- Operator (ISDN) field
 E1/ISDN 156
 T1/ISDN 151
- Operator definitions
 ISDN-BRI..... 240
- Operator field
 ISDN-BRI..... 238
- Optimization Factor field
 analog..... 221
 T1/E1 144
- Options (call progress) field 379
- Options (callee, statistics, logs) field
 384
- Options, From Details (SMTP logs)
 field
 analog..... 255
 T1/E1 173
- Options, To Details (SMTP logs) field
 analog..... 255
- T1/E1 173
- ORIG ALIAS field (gatekeeper
 Current Calls screen)..... 496
- ORIG IP field (gatekeeper Current
 Calls screen)..... 496
- Other Aliases
 Email gatekeeper field (Call
 Details, Destination Info) 501
 Email gatekeeper field (Call
 Details, Source Info) 500
- Other Aliases field (gatekeeper).... 493
- out of band, DTMF
 analog..... 217
 T1/E1 140
- Outbound Digits (call progress) field
 378
- Outbound Digits (SMTP logs) field
 analog..... 255
 T1/E1 173
- Outbound digits (statistics, logs) field
 384
- outbound phonebook
 example 76
- Outbound Phonebook Entries List
 icon
 E1 328
 T1 280
- Outbound Phonebook entries, list
 E1 339
 T1 290
- outbound phonebook example
 quick..... 66
- outbound vs. inbound phonebooks
 E1 326
 T1 278
- out-of-zone distance 510
- Out-of-Zone Distance field
 (gatekeeper, Network Parameters)
 510
- Output Gain field
 analog..... 216
 T1/E1 139
- output level, fax tones
 analog..... 217
 T1/E1 140
- outside line, access to..... 74, 76

P

- packetization (RTP), ranges & increments
 - T1/E1 391
- packetization rates
 - coder options and 390
- Packets Lost (call progress) field.. 378
- Packets Lost (SMTP logs) field
 - analog..... 254
 - T1/E1 172
- Packets lost (statistics, logs) field . 384
- Packets Received (call progress) field
 - 378
- Packets Received (SMTP logs) field
 - analog..... 254
 - T1/E1 172
- Packets received (statistics, logs) field
 - 384
- Packets Sent (call progress) field.. 378
- Packets Sent (SMTP logs) field
 - analog..... 254
 - T1/E1 172
- Packets sent (statistics, logs) field 383
- packets, consecutive lost
 - analog..... 222
 - T1/E1 145
- Parallel H.245 field
 - E1 336
 - T1 287
- parameters tracked by console
 - analog..... 260
 - T1/E1 178
- Pass Through (E&M) field 234
- Password (proxy server) field
 - E1 334
 - T1 285
- Password (SMTP) field
 - analog..... 252
 - T1/E1 170
- password, lost/forgotten..... 424, 427
- password, setting..... 424
- web browser GUI..... 427
- patents..... 2
- patterns, destination
 - tips about..... 73
- PBX characteristics, variations in
 - E1 368
 - T1 319
- PBX interaction
 - analog models 26
 - BRI models 35
 - E1 models..... 19
 - T1 models..... 12
- PC, command
 - COM port assignment (detailed)114
 - COM port requirement..... 52
 - demands upon 52
 - non-dedicated use..... 52
 - operating system 52
 - settings 52
 - specifications..... 52
- PCM Law field
 - E1 157
 - ISDN-BRI 239
 - T1 152
- Permissions (SNMP) field
 - analog..... 243
 - T1/E1 161
- personnel requirement
 - for rack installation 95
 - to lift during installation..... 96
 - to lift unit during installation..... 88
- phone exchanges
 - dedicated 74
 - institutional 74
 - local..... 74
 - non-local 74
 - organizational..... 74
- Phone field (gatekeeper) 493
- Phone Number (Auto Call) field
 - analog..... 219
- Phone Number (Auto Call)field
 - T1/E1 142
- Phone Numbers gatekeeper field (Call Details, Destination Info)..... 501
- Phone Numbers gatekeeper field (Call Details, Source Info) 500
- Phone Signaling Tones & Cadences
 - analog..... 244
 - T1/E1 162
- phone startup configuration 59
- phone switch types
 - ISDN implementations in..... 158
- phone/IP details
 - importance of writing down 47

- importance of writing down
 - (analog)..... 199
- importance of writing down
 - (T1/E1)..... 122
- phonebook
 - FTP remote file transfers 431
- phonebook configuration
 - starter 66
- phonebook configuration (analog) 196, 372
- phonebook configuration (remote) 431
- phonebook configuration (T1/E1). 120
- Phonebook Configuration icon
 - E1 328
 - T1 280
- Phonebook Configuration Parameter definitions
 - E1 333, 334, 335, 336
 - T1 284, 285, 286, 287
- Phonebook Configuration procedure
 - T1 279
- Phonebook Configuration Procedure
 - E1 327
- Phonebook Configuration screen
 - E1 330
 - T1 279
- Phonebook Configuration screen (E1)
 - Mode (SPP Protocol) 337
- Phonebook Configuration screen (T1)
 - Mode (SPP Protocol) 288
- Phonebook Configuration screen fields (E1)
 - Annex E (H.323, UDP multiplexing)..... 336
 - Call Signalling Port..... 333
 - Client Options 337
 - Enable Proxy 334
 - Gatekeeper Name 333
 - Gatekeeper/Clear Channel IP Address 333
 - Gatekeeper/Clear-Channel IP Address 333
 - Gateway H.323 ID 333
 - Gateway Name 333
 - Gateway Prefix 333
 - General Options 337
 - H.245 Tunneling 335
 - Keep Alive 337
- Max Retransmission (SPP, General Options)..... 337
- Parallel H.245 (Tunneling with Fast Start)..... 336
- Port (SPP, General Options) 337
- Port Number (Gatekeeper) 333
- Port Number (proxy server) 334
- Proxy Server IP Address 334
- Q.931 Multiplexing 335
- Register with GateKeeper 333
- Registrar IP Address 337
- Registrar Options 337
- Registrar Port 337
- Retransmission (SPP, General Options)..... 337
- Use Fast Start 333
- User Name (proxy server) 334
- Phonebook configuration screen fields (T1)
 - Password (proxy server)..... 285
- Phonebook Configuration screen fields (T1)
 - Annex E (H.323, UDP multiplexing)..... 287
 - Call Signalling Port..... 284
 - Client Options 288
 - Enable Proxy 285
 - Gatekeeper Name 284
 - Gatekeeper/Clear Channel IP Address 284
 - Gateway H.323 ID 284
 - Gateway Name 284
 - Gateway Prefix 284
 - General Options 288
 - H.245 Tunneling 286
 - Keep Alive 288
 - Max Retransmission (SPP, General Options)..... 288
 - Parallel H.245 (Tunneling with Fast Start)..... 287
 - Password (proxy server)..... 334
 - Port (SPP, General Options) 288
 - Port Number (Gatekeeper) 284
 - Port Number (proxy server) 285
 - Proxy Server IP Address 285
 - Q.931 Multiplexing 286
 - Register with GateKeeper 284
 - Registrar IP Address 288

- Registrar Options 288
- Registrar Port 288
- Retransmission (SPP, General Options) 288
- Use Fast Start 284
- User Name (proxy server)..... 285
- phonebook destination patterns 73
- phonebook dialing patterns 73
- phonebook digits
 - dropping 74
 - leading 74
 - non-PSTN type 74
 - specialized codes 74
 - types used 73
- phonebook entries, coordinating
 - E1 327
 - T1 279
- phonebook examples
 - analog 196
 - mixed digital/analog 76
- phonebook icons
 - E1 328
 - T1 280
- phonebook objectives & considerations
 - E1 326
- phonebook sidebar menu
 - E1 329
 - T1 281
- phonebook tips 73
- phonebook worksheet 79, 80
- phonebook, analog voips 372
- phonebook, inbound
 - example 76
 - example, quick 70
- phonebook, outbound
 - example 76
 - example, quick 66
- phonebooks, inbound vs. outbound
 - E1 326
 - T1 278
- phonebooks, objectives & considerations
 - T1 278
- Phonebooks, objectives & considerations
 - E1 321
- phonebooks, sample 78
- pinging and connectivity 85
- pinout
 - BRI connector 539
 - command cable 536
 - ethernet cable 536
 - T1/E1 connector 537
 - Voice/FAX connector 537
- polling, IRQ 509
- Port (SPP, General Options) field
 - E1 337
 - T1 288
- Port field (Registered Gateway Details) 406
- Port field, SysLog Server
 - analog 259
 - T1/E1 177
- Port Number (Gatekeeper RAS) field
 - E1 333
 - T1 284
- Port Number (proxy server)
 - E1 334
- Port Number (proxy server) field
 - T1 285
- Port Number (SMTP) field
 - analog 252
 - T1/E1 170
- port number (SNMP) field
 - analog 243
 - T1/E1 161
- Port Number field, SPP (Outbound Phonebook)
 - E1 344
 - T1 295
- power consumption
 - analog models 44
 - E1 models 43
 - T1 models 42
- power frequency
 - analog models 44
 - E1 models 43
 - T1 models 42
- Power LED
 - analog models 33, 34
 - BRI models 40
- powering of ISDN-BRI phones
 - MVP-410ST/810ST 103
- PreDef field (gatekeeper) 493

- Predefined endpoints option
(Gatekeeper General Settings
screen)..... 488
- Prefix gatekeeper field (Services, GK
Defined)..... 513
- Prefix gatekeeper field (Services, V2
GW Prefixes) 514
- Prefix Matched (call progress) field
..... 378
- Prefix Matched (SMTP logs) field
analog..... 255
T1/E1 173
- prefixes 514
- PreGrant All field (gatekeeper,
Network Parameters) 504
- pregrantedARQ permissions..... 504
- prerequisites
for technical configuration (analog)
..... 199
for technical configuration (T1/E1)
..... 122
- prerequisites for installation
BRI info 50
E1 info 49
IP info 48
T1 info 48
- PRI
ISDN implementations 158
- product CD 45
use in software installation . 58, 109
- Product CD
E1 models 21
T1 models 14
- product family..... 10, 11
- product groups 9
- Program Menu items..... 407
- Protocol Type (outbound phonebook)
E1 341
T1 292
- protocols, gatekeeper 484
- Proxy Server IP Address
E1 334
- Proxy Server IP Address field
T1 285
- PSTN failover feature
Alternate Routing, and..... 297
analog models 27
BRI models 36
- E1 models.....20
T1 models.....13
- Public gatekeeper field (Services, GK
Defined) 514
- Public gatekeeper field (Services, V2
GW Prefixes) 514
- Pulse Density Violation (T1 stats)
field 396
- Pulse Shape Level field
E1 157
T1 152
- Q**
- Q.931 Multiplexing field
E1 335
T1 286
- Q.931 Parameters
T1 284
- Q.931 Parameters fields
Connect TO (time-out)..... 491
Q.931 Signaling Port..... 491
Response TO (time-out)..... 491
- Q.931 Port Number (outbound
phonebook) field
E1 342
T1 293
- Q.931 Signaling Port field (gatekeeper
Memory screen) 491
- quality-of-service
analog.....27
BRI.....36
E1 20
T1 13
- R**
- rack mounting
grounding 95
safety 88, 95
- rack mounting instructions..... 94
- rack mounting procedure 96
- rack, equipment
weight capacity of 95
- rack-mountable voip models 88
- RAS (H.323) vs. TCP/IP RAS 484
- RAS Parameters fields (gatekeeper
Memory screen) 491
- RAS Port field (gatekeeper Memory
screen)..... 491

- RCF messages..... 504
- RCF Registration Confirmation
messages (gatekeeper, H.225).... 486
- RCV (channel) LED
analog models 33, 34
BRI models 40
- RCV (Ethernet) LED
analog models 33
BRI models 40
- Reason gatekeeper field (Call Details)
..... 499
- Receive Slip (E1 Stats) field 400
- Receive Slip (T1 Stats) field 397
- Receive Timeslot 16 Alarm Indication
Signal (E1 stats) field..... 399
- Receive Timeslot 16 Loss of
MultiFrame Alignment (E1 stats)
field..... 400
- Receive Timeslot 16 Loss of Signal
(E1 stats) field..... 399
- Received (RTCP Packets, IP Stats)
field..... 389
- Received (RTP Packets, IP Stats) field
..... 389
- Received (TCP Packets, IP Stats) field
..... 388
- Received (Total Packets, IP Stats)
field..... 387
- Received (UDP Packets, IP Stats)
field..... 388
- Received with Errors (RTCP Packets,
IP Stats) field 389
- Received with Errors (RTP Packets,
IP Stats) field 389
- Received with Errors (TCP Packets,
IP Stats) field 388
- Received with Errors (Total Packets,
IP Stats) field 388
- Received with Errors (UDP Packets,
IP Stats) field 388
- Recipient Address (email logs) field
T1/E1 171
- Recipient Address (email logs)field
analog..... 253
- recovering voice packets
analog..... 219
T1/E1 142
- Red Alarm (E1 stats) field 398
- Red Alarm (T1 stats) field 395
- Regeneration (dialing, FXO) field 230
- Regional Parameter definitions
analog..... 246, 247
T1/E1 164, 165
- Regional Parameter fields (analog)
Cadence..... 247
Custom (tones)..... 247
Pulse Generation Ratio..... 247
- Regional Parameter fields (T1/E1)
Cadence..... 165
Country/Region (tone schemes) 164
Custom (tones)..... 165
Frequency 1 164
Frequency 2 164
Gain 1 164
Gain 2 164
type (of tone)..... 164
- regional parameters, setting
analog..... 244
T1/E1 162
- Register Duration field (Registered
Gateway Details)..... 406
- Registered Gateway Details
(Statistics) screen, accessing 406
- Registered Gateway Details
'Statistics' function 405, 406
- Registered Gateway Details screen
- Registered Gateway Details screen
fields
Description..... 406
IP Address 406
No. of Entries 406
Port..... 406
Register Duration 406
Status..... 406
- Registered Gateway Details screen
fields: 406
- Registrar IP Address field
E1 338
T1 289
- Registrar Options fields
E1 338
T1 289
- Registrar Port field
E1 338
T1 289
- registration

timeout.....	508	re-order tone, custom	
registration (with gatekeeper)		T1/E1	166
description	492	repair procedures for customers U.S.	
registration control messages		& Canada	524
(gatekeeper, H.225)		Reply-To Address (email logs) field	
IRQ	487	T1/E1	171
IRR	487	Reply-To Address (email logs)field	
RCF.....	486	analog.....	253
RRJ.....	486	Reports function.....	385
RRQ.....	486	Resolutions (MultiVOIP	
URQ.....	487	troubleshooting).....	11
Registration IP field (gatekeeper) .	493	Response TO field (gatekeeper	
registration of endpoints with		Memory screen)	
gatekeeper		Q.931 Parameters	491
dynamic	508	RAS Parameters	491
Registration Policy field (Gatekeeper		Retransmission (SPP, General	
General Settings screen)	488	Options) field	
Registration TO (time-out) field		E1	337
(gatekeeper, Network Parameters)		T1	288
.....	508	Retrieve Sequence	
registration with gatekeeper.....	493	analog.....	265
remote control/configuration		T1/E1	183
web GUI and.....	443	Retrieve Sequence (analog)	262
Remote Extension Name gatekeeper		Retrieve Sequence (T1/E1)	180
field (Call Details, Destination		RFC768.....	542
Info)	502	RFC793	542
Remote Extension Phone gatekeeper		ring cadences, custom	
field (Call Details, Destination		analog.....	250
Info)	502	T1/E1	166, 168
remote phonebook configuration ..	431	Ring Count (FXO) field	232
remote voip		Ring Count field	
using to confirm configuration51,		FXS Ground Start	228
66		FXS Loop Start	226
remote voip configuration (analog)		Ring Count forwarding condition	
.....	197	E1	348
remote voip configuration (T1/E1)120		T1	300
Remote Voip Management		ring tone, custom	
analog.....	241	analog.....	249
T1/E1	159	T1/E1	166, 167
Remove H.245 Addr in Call Hunt		ring-tones	
field (gatekeeper, Network		analog.....	248
Parameters)	505	T1/E1	166
Remove Prefix (inbound) field		Round Trip Delay field	
E1	347	E1	346
T1.....	299	T1	297
Remove Prefix (outbound) field		routed call mode.....	507
E1	341	routed mode (call control signalling)	
T1.....	292	482

- Routed Mode option (gatekeeper, Network Parameters) 507
- routed-mode calls..... 498
- RRJ Registration Rejection messages (gatekeeper, H.225) 486
- RRQ messages 504
- RRQ Registration Request messages (gatekeeper, H.225) 486, 493
- RSG LED
analog models 33, 34
- RTP packetization, ranges & increments..... 391
- RTP Parameters screen 391
- Rx Packets field (ISDN BRI Statistics, D-Channel Information) 403
- S**
- Safety Recommendations for Rack Installations..... 95
- safety warnings 88
- Safety Warnings Telecom** 88
- sample phonebooks..... 78
- Save Setup command
analog..... 275
T1/E1 193
- saving configuration
analog..... 275
T1/E1 193
user 422
- Saving the MultiVOIP Configuration
analog..... 275
T1/E1 193
- savings on toll calls
E1 321
T1 278
- scale-ability
E1 19
T1 12
- second dial tone
and comma use 75
- Select All (SMTP logs) field
analog..... 254
T1/E1 172
- Select BRI Interface field 402
- Select BRI Interface ISDN-BRI field
BRI 238
- Select Channel field
analog..... 216
T1/E1 139
- Select Channel, Supplementary Services field
analog..... 264
T1/E1 182
- Selected Coder field
analog..... 218
T1/E1 141
- Send Immediately option (gatekeeper, Network Parameters)..... 506
- Service Configurable Properties field (gatekeeper, Network Parameters) 505
- Services (gatekeeper) screen 513
- Services (gatekeeper) screen fields
Default (GK Defined Services) .513
Default (Services, V2 GW Prefixes) 514
Description (GK Defined Services) 513
Description (Services, V2 GW Prefixes) 514
Dynamic (Services, V2 GW Prefixes) 515
Prefix (GK Defined Services) ... 513
Prefix (Services, V2 GW Prefixes) 514
Public (GK Defined Services)... 514
Public (Services, V2 GW Prefixes) 514
- Services (gatekeeper) screen, accessing 512
- Services screen fields
GK Defined Services 514
V2 GW Prefixes..... 514
- Set Baud Rate
analog..... 272
T1/E1 190
- Set Custom Tones & Cadences
T1/E1 166
- Set ISDN Parameters 158
- Set Log Reporting Method
analog..... 257
T1/E1 175
- Set Password (program menu option) , command..... 424

Set Password (web browser GUI), command	427	analog telephony (MVP-410/810)	100
Set Password option description (MultiVOIP program menu)	408	Silence Compression (call progress) field	379
Set Regional Parameters		Silence Compression (SMTP logs) analog	255
analog	244	T1/E1	173
T1/E1	162	Silence Compression field analog	219
Set SMTP Parameters		T1/E1	142
analog	251	Silence Detection (FXO) field	232
T1/E1	169	Silence Timer (FXO) field	232
Set SNMP Parameters		simulated voip network use in startup	75
analog	241	Single-Port Protocol, general description	
T1/E1	159	analog	27
Set Supplementary Services Parameters		BRI	36
analog	261	E1	20
T1/E1	179	T1	13
Set T1/E1/ISDN Parameters	146	SIP	
Set Telephony Interface Parameters	223	compatibility	
Set Voice/FAX Parameters		analog models	27
analog	213	BRI models	36
T1/E1	136	E1 models	20
setting IP parameters		T1 models	13
analog	209	SIP Fields (Outbound Phonebook)	
T1/E1	132	E1	343
setting password	424	T1	294
web browser GUI	427	SIP incompatibility with H.450 Supplementary Services	
setting RTP Parameters	391	analog	27, 261
setting user defaults	421	BRI	36
setup, saving		E1	20
analog	275	T1	13
T1/E1	193	T1/E1	179
user	422	SIP Port Number field	
setup, saving user values	421	E1	343
Signal (type, E&M) field	234	T1	294
signaling cadences		SIP port number, standard	
analog	244	E1	343
T1/E1	162	T1	294
signaling parameters (analog telephony)	223	SIP Proxy Parameters	
signaling tones		E1	334
analog	244	T1	285
T1/E1	162	SIP URL field	
signaling types		E1	343
analog telephony	54	T1	294
analog telephony (MVP130)	107		
analog telephony (MVP210)	106		

SMTP	
quick setup.....	60
SMTP (log reporting type) button	
analog.....	259
T1/E1	177
SMTP logs by email, illustration	
analog.....	256
T1/E1	174
SMTP Parameters definitions	
analog.....	252
T1/E1	170
SMTP Parameters fields (analog)	
Mail Server IP Address.....	252
Mail Type	253
Number of Days.....	253
Number of Records.....	253
Port Number	252
Recipient Address	253
Reply-To Address	253
Subject	253
SMTP Parameters fields (T1/E1)	
Enable SMTP	170
Login Name	170
Mail Server IP Address.....	170
Mail Type	171
Number of Days.....	171
Number of Records.....	171
Password.....	170
Port Number	170
Recipient Address	171
Reply-To Address	171
Subject	171
SMTP parameters, accessing	
analog.....	251
T1/E1	169
SMTP parameters,setting	
analog.....	251
T1/E1	169
SMTP port, standard	
analog.....	252
T1/E1	170
SMTP prerequisites	
analog models	202
quick	51
T1/E1 models.....	125
SMTP, enabling	
analog.....	251
T1/E1	169
SNMP (log reporting type) button	
analog.....	259
T1/E1	177
SNMP agent program	
analog.....	197
T1/E1	121
SNMP agent, enabling	
analog.....	241
T1/E1	159
SNMP Parameter Definitions	
T1/E1	161
SNMP Parameter fields (analog)	
Address	243
Community Name (2)	243
Community Name 1	243
Enable SNMP Agent.....	243
Permissions (1).....	243
Permissions (2).....	243
Port Number.....	243
SNMP Parameter fields (T1/E1)	
Address	161
Community Name (2)	161
Community Name 1	161
Enable SNMP Agent.....	161
Permissions (1).....	161
Permissions (2).....	161
Port Number.....	161
SNMP Parameters, setting	
analog.....	241
T1/E1	159
software	
control.....	58
uninstalling (detailed)	116
updates (analog).....	197
updates (T1/E1).....	121
software (MultiVOIP)	
uninstalling.....	428
software configuration	
summary.....	109
software installation	
detailed.....	109
quick.....	58
software license, gatekeeper.....	521
software loading	109
software loading, quick	58
software version numbers	111
software, MultiVOIP (analog)	
screen-surfing in.....	208

- software, MultiVOIP (T1/E1)
 - moving around in 131
 - screen-surfing in 131
- software, MultiVOIP(analog)
 - moving around in 208
- software, on command PC 58
- Solving Common Connection Problems
 - analog 207
 - T1/E1 130
- sound quality, improving
 - analog 219
 - T1/E1 142
- specialized codes, in dialing 74
- specifications
 - E1 models 43
 - T1 models 42
- SPID 0
 - ISDN-BRI 239
- SPID 1
 - ISDN-BRI 239
- SPID0 field (ISDN BRI Statistics, Switch Information) 403
- SPID1 field (ISDN BRI Statistics, Switch Information) 404
- SPP Fields (Outbound Phonebook)
 - E1 344
 - T1 295
- SPP Fields (Phonebook Configuration screen)
 - T1 288
- SPP Fields (PhoneBook Configuration screen)
 - E1 338
- SPP, general description
 - analog 27
 - BRI 36
 - E1 20
 - T1 13
- SPP, strengths & compatibilities of
 - analog 27
 - BRI 36
 - E1 20
 - T1 13
- ST interface (ISDN-BRI)
 - description 540
- Start Date, Time (SMTP logs) field
 - analog 254
 - T1/E1 172
- Start Date,Time (statistics, logs) field 383
- starter configuration
 - inbound phonebook 70
 - outbound phonebook 66
 - phone/IP 59
- startup tasks 47
- State field (ISDN BRI Statistics, Layer 1 Interface) 402
- State gatekeeper field (Call Details) 499
- Options (caller 384
- Status (call progress) field 381
- Status (statistics, logs) field 383
- Status field (ISDN BRI Statistics, Layer 1 Interface) 402
- Status field (ISDN BRI Statistics, SPID0) 403
- Status field (ISDN BRI Statistics, SPID1) 404
- Status field (Registered Gateway Details) 406
- Status Freeze Signalling Active (E1 stats) field 399
- Status Freeze Signalling Active (T1 stats) field 395
- Status Information gatekeeper fields (Network Parameters) 503
- Subject (email logs) field
 - analog 253
 - T1/E1 171
- supervisory signaling
 - analog telephony 54
- supervisory signaling (analog) 224
- supervisory signaling parameters (analog telephony) 223
- supervisory signaling types
 - MVP130 107
 - MVP210 106
 - MVP-410/810 100
- Supplementary (Telephony) Services
 - ANALOG 30
 - BRI 39
 - E1 24
 - T1 17
- Supplementary Services (analog)
 - Alerting Party 268, 269, 270

- Call Hold..... 262
- Call Hold Enable..... 265
- Call Name Identification..... 262
- Call Transfer 262
- Call Waiting..... 262
- Call Waiting Enable..... 265
- Caller Name Identification Enable
..... 266
- Calling Party 267
- Enable Call Hold..... 265
- Enable Call Transfer 264
- Enable Call Waiting..... 265
- Enable Caller Name Identification
..... 266
- Hold Sequence 265
- Retrieve Sequence 265
- Select Channel 264
- Transfer Sequence 264
- Supplementary Services (T1/E1)
 - Alerting Party..... 186, 187, 188
 - Call Hold..... 180
 - Call Hold Enable..... 183
 - Call Name Identification..... 181
 - Call Transfer 180
 - Call Transfer Enable 182, 264
 - Call Waiting..... 180
 - Call Waiting Enable..... 183
 - Caller Name Identification Enable
..... 184
 - Calling Party 185
 - Enable Call Hold..... 183
 - Enable Call Transfer 182
 - Enable Call Waiting..... 183
 - Enable Caller Name Identification
..... 184
 - Hold Sequence 183
 - Retrieve Sequence 183
 - Select Channel 182
 - Transfer Sequence 182
- Supplementary Services Info
 - logs for..... 385
- Supplementary Services Parameter
 - buttons (analog)
 - Copy Channel 271
 - Default 271
- Supplementary Services Parameter
 - buttons (T1/E1)
 - Copy Channel 189
- Default 189
- Supplementary Services Parameter
 - Definitions
 - analog 264, 265, 266, 267, 268, 269,
270, 271
 - T1/E1 182, 183, 184, 185, 186, 187,
188, 189
- Supplementary Services Parameter
 - fields (analog)
 - Call Transfer Enable 264
 - Call Waiting Enable 265
 - Hold Sequence 265
 - Retrieve Sequence 265
 - Transfer Sequence 264
- Supplementary Services Parameter
 - fields (analog)
 - Alerting Party..... 268
 - Allowed Name Types 267, 268,
269, 270
 - Busy Party..... 269
 - Call Hold Enable..... 265
 - Call Name Identification Enable 266
 - Caller ID 271
 - Calling Party 267
 - Connected Party 270
 - Select Channel 264
- Supplementary Services Parameter
 - fields (T1/E1)
 - Call Transfer Enable 182
 - Call Waiting Enable 183
 - Hold Sequence 183
 - Retrieve Sequence..... 183
 - Transfer Sequence..... 182
- Supplementary Services Parameter
 - fields (T1/E1)
 - Alerting Party..... 186
 - Allowed Name Types 185, 186,
187, 188
 - Busy Party..... 187
 - Call Hold Enable..... 183
 - Call Name Identification Enable 184
 - Caller ID 189
 - Calling Party 185
 - Connected Party 188
 - Select Channel 182
- Supplementary Services Parameters
 - screen, accessing
 - analog..... 261

- T1/E1 179
- Supplementary Services parameters,
 setting
 analog..... 261
- T1/E1 179
- Supplementary Services, incompatible
 with SIP
 analog..... 27, 261
- BRI 36
- E1 20
- T1 13
- T1/E1 179
- support, technical..... 526
- Switch Information fields (ISDN BRI
 Statistics) 403
- switch types (phone) and ISDN-PRI
 158
- SysLog client
 ANALOG 29
- BRI 38
- E1 23
- T1 16
- SysLog client programs
 availability 446
- features & presentation types.... 448
- SysLog functionality
 ANALOG 29
- BRI 38
- E1 23
- T1 16
- SysLog server
 ANALOG 29
- BRI 38
- E1 23
- T1 16
- SysLog Server Enable field
 analog..... 259
- T1/E1 177
- SysLog Server function
 as added feature 446
- capabilities of..... 448
- enabling 447
- location of..... 446
- SysLog Server IP Address field
 analog..... 259
- T1/E1 177
- SysLog Server, enabling
 analog..... 258
- T1/E1 176
- System Information screen
 for op & maint..... 374
- System Information screen, accessing
 analog..... 273
- T1/E1 191
- System Information update interval,
 setting
 analog..... 273
- for op & maint..... 375
- T1/E1 191
- T**
- T1 model descriptions..... 12
- T1 Parameter definitions 149, 150, 152
- Clocking..... 152
- Line Build-Out..... 152
- Line Coding 152
- PCM Law 152
- Pulse Shape Level 152
- Yellow Alarm Format 152
- T1 Parameter fields
 CAS Protocol 150
- CRC Check 149
- Frame Format..... 149
- Long-Haul Mode..... 149
- T1/E1/ISDN 149
- T1 Parameters screen 148
- T1 Statistics field definitions 396, 397
- T1 Statistics fields
 Bipolar Violation 397
- Frame Search Restart Flag 396
- Line Loopback Activation Signal
 396
- Loss of MultiFrame Alignment. 396
- Pulse Density Violation 396
- Receive Slip 397
- Transmit Data Underrun 396
- Transmit Line Open 396
- Transmit Slip..... 396
- Transmit Slip Negative 396
- Yellow Alarm..... 396
- T1 telephony parameters..... 123
- T1/E1 connector pinout..... 537
- T1/E1 Statistics function..... 393
- T1/E1/ISDN field
 E1 154
- T1 149

- T1/E1/ISDN Parameters screen,
 - accessing 146
- T1/E1/ISDN parameters, setting... 146
- T1/ISDN Parameter definitions ... 151
- T1/ISDN Parameter fields
 - Country 151
 - Enable ISDN-PRI 151
 - Operator 151
 - Terminal Network 151
- table-top voip models 88
- TCP/UDP compared
 - E1 343
 - IP Statistics context 387
 - T1 294
- technical configuration
 - startup 59
- technical configuration (analog)
 - prerequisites to 199
 - summary 196
- technical configuration (T1/E1)
 - prerequisites to 122
 - summary 120
- technical configuration procedure
 - detailed, analog 204
 - detailed, T1/E1 127
 - summary, analog 203
 - summary, T1/E1 126
- technical support 526
- TEI Assignment fields (ISDN BRI
 - Statistics, Switch Information) . 403
- TEI n Assignment
 - ISDN-BRI 239
- TEIn fields (ISDN BRI Statistics,
 - Switch Information) 403
- telco authorities and ISDN 158
- telecom safety warnings** 88
- telephony interface parameters,
 - setting 223
- telephony interfaces
 - uses of 100, 106, 107
- telephony interfaces, analog 54
- telephony interfaces, BRI 55
- telephony signaling cadences
 - analog 244
 - T1/E1 162
- telephony signaling tones
 - analog 244
 - T1/E1 162
- telephony startup configuration 59
- telephony toning schemes
 - analog 248
 - T1/E1 166
- temperature
 - operating 95
- terminal mode (ISDN-BRI) & D-
 - channel support
 - MVP-410ST/810ST 102
- Terminal Network field
 - E1/ISDN 156
 - T1/ISDN 151
- terminal/network settings, voip and
 - PBX
 - E1/ISDN 156
 - ISDN-BRI 238
 - T1/ISDN 151
- Time To Live (TTL) timer field,
 - gatekeeper 493
- TimeToLive (gatekeeper, RCF
 - message)
 - details about 486
- tips, phonebook 73
- To (gateway, statistics, logs) field. 383
- toll-call savings
 - E1 321
 - T1 278
- toll-free access (T1)
 - to remote PSTN 12
 - within voip network 12
- toll-free access (within voip network)
 - E1 19
 - T1 12
- Tone Detection (FXO disconnect
 - criteria) field 231
- Tone Pair (custom) field
 - analog 249
 - T1/E1 167
- tone pairs, custom
 - T1/E1 166
- tones, signaling
 - analog 244
 - T1/E1 162
- Total BW gatekeeper field (Call
 - Details) 499
- Total Digits (outbound) field
 - E1 341
 - T1 292

- touch tone frequencies 232
 - trace on/off (logging)
 - analog..... 260
 - T1/E1 178
 - Transfer Sequence
 - analog..... 264
 - T1/E1 182
 - Transfer Sequence (analog) 262
 - Transfer Sequence (T1/E1)..... 180
 - Transmit Data Overflow (E1 stats)
 - field..... 399
 - Transmit Data Overflow (T1 stats)
 - field..... 395
 - Transmit Data Underrun (E1 stats)
 - field..... 400
 - Transmit Data Underrun (T1 stats)
 - field..... 396
 - Transmit Line Open (E1 stats) field
 - 400
 - Transmit Line Open (T1 stats) field
 - 396
 - Transmit Line Short (E1 stats) field
 - 399
 - Transmit Line Short (T1 stats) field
 - 395
 - Transmit Slip (E1 stats) field 400
 - Transmit Slip (T1 stats) field 396
 - Transmit Slip Negative (E1 stats) field
 - 400
 - Transmit Slip Negative (T1 stats) field
 - 396
 - Transmit Slip Positive (E1 stats) field
 - 399
 - Transmit Slip Positive (T1 stats) field
 - 395
 - Transmitted (RTCP Packets, IP Stats)
 - field..... 389
 - Transmitted (RTP Packets, IP Stats)
 - field..... 389
 - Transmitted (TCP Packets, IP Stats)
 - field..... 388
 - Transmitted (Total Packets, IP Stats)
 - field..... 387
 - Transmitted (UDP Packets, IP Stats)
 - field..... 388
 - transport name alias 500, 501
 - Transport Protocol (SIP) field
 - E1 343
 - T1 294
 - trap manager parameters (SNMP)
 - T1/E1 161
 - triggering log report email
 - analog..... 253
 - T1/E1 171
 - troubleshooting 85
 - Troubleshooting Resolutions for
 - MultiVOIPs..... 11
 - TTL (gatekeeper) 493
 - Turn Off Logs field
 - analog..... 259
 - T1/E1 177
 - Tx Packets field (ISDN BRI Statistics, D-Channel Information)..... 403
 - Type (E&M type) field 234
 - Type (of tone) field
 - analog..... 246
 - T1/E1 164
 - Type field (gatekeeper) 493
- U**
- U interface (ISDN-BRI)
 - description..... 540
 - UDP multiplexed (H.323 Annex E)
 - field
 - E1 336
 - T1 287
 - UDP/TCP compared
 - E1 343
 - IP Statistics context..... 387
 - T1 294
 - unconditional forwarding
 - E1 348
 - T1 300
 - Uninstall (program menu option) ,
 - command..... 428
 - Uninstall option description
 - (MultiVOIP program menu) 408
 - uninstalling MultiVOIP software 116, 428
 - unobtainable tone, custom
 - analog..... 249
 - T1/E1 166, 167
 - unobtainable tones
 - analog..... 166, 248
 - unpacking
 - MVP130..... 93

- MVP210x..... 92
MVP2410..... 89, 90
MVP3010..... 89
MVP410x..... 91
MVP810x..... 91
Unregister All endpoints command
 Gatekeeper..... 494
Unregister endpoints command
 Gatekeeper..... 494
Up Time
 System Info (analog)..... 274
 System Info (T1/E1) 192, 374
Update button (gatekeeper Network
 Parameters) 511
update interval (logging)
 analog..... 258
 T1/E1 176
updated firmware, obtaining 409
upgrade
 E1 19
 T1 12
upgrade card (analog, 4-to-8 channel)
 installation 544
upgrade card (T1/E1) installation . 532
Upgrade Software option description
 MultiVOIP program menu 408
upgrade, firmware..... 413
uploads vs. downloads (FTP)..... 431
URQ Unregister Request messages
 (gatekeeper, H.225) 487, 493
Use Fast Start (Q.931) field
 E1 333
 T1 284
Use Gatekeeper (Outbound
 Phonebook) field
 E1 342
 T1 293
Use Proxy (SIP) field
 E1 343
 T1 294
Use Registrar field (Outbound
 Phonebook)
 E1 344
 T1 295
user default configuration, creating
 analog..... 276
 T1/E1 194
user defaults, downloading 421
user defaults, setting..... 421
user name
 Windows GUI 424
User Name (proxy server) field
 E1 334
 T1 285
user values (software), saving 421
- V**
- V2 GW Prefixes field (gatekeeper,
 Services)..... 514
variations in PBX characteristics
 E1 368
 T1 319
version numbers 11
version numbers (software)..... 111
version, firmware 413
Voice Coder (call progress) field .. 378
Voice coder (statistics, logs) field .384
voice delay
 analog..... 220, 221
 T1/E1 143, 144
Voice Gain field
 analog..... 216
 T1/E1 139
voice packets (analog)
 recovering lost/corrupted 219
voice packets (T1/E1)
 recovering lost/corrupted 142
voice packets, consecutive lost
 analog..... 222
 T1/E1 145
voice packets, delayed
 analog..... 220, 221
 T1/E1 143, 144
voice packets, re-assembling
 analog..... 217
voice packets, re-assembly
 T1/E1 140
voice quality, improving
 analog..... 219
 T1/E1 142
voice quality, versus delay
 analog..... 221
 T1/E1 144
Voice/FAX connector pinout 537
Voice/FAX Parameter definitions
 analog..... 221, 222

- T1/E1 144, 145
- Voice/FAX Parameter Definitions
 - analog..... 216, 217, 218, 219, 220
 - T1/E1139, 140, 141, 142, 143
- Voice/FAX Parameter fields (analog)
 - Auto Call Enable.....219
 - Automatic Disconnection 222
 - Call Duration 222
 - Consecutive Packets Lost 222
 - Copy Channel 216
 - Default 216
 - DTMF Gain 216
 - DTMF Gain (High Tones) 216
 - DTMF Gain (Low Tones)..... 216
 - DTMF In/Out of Band 216
 - Duration (DTMF) 216
 - Dynamic Jitter Buffer 220
 - Echo Cancellation 219
 - Fax Enable 217
 - Fax Volume 217
 - Forward Error Correction 219
 - Input Gain 216
 - Jitter Value..... 222
 - Jitter Value (Fax) 217
 - Max Baud Rate (Fax)..... 217
 - Maximum Jitter Value 221
 - Minimum Jitter Value 220
 - Mode (Fax) 217
 - Network Disconnection 222
 - Optimization Factor 221
 - Output Gain 216
 - Phone Number (Auto Call) 219
 - Select Channel 216
 - Silence Compression 219
 - Voice Gain 216
- Voice/FAX Parameter fields (T1/E1)
 - Auto Call Enable..... 142
 - Automatic Disconnection 145
 - Call Duration 145
 - Consecutive Packets Lost 145
 - Copy Channel 139
 - Default 139
 - DTMF Gain 139
 - DTMF Gain (High Tones) 139
 - DTMF Gain (Low Tones)..... 139
 - DTMF In/Out of Band 139
 - Duration (DTMF) 139
 - Dynamic Jitter Buffer 143
 - Echo Cancellation 142
 - Fax Enable 140
 - Fax Volume..... 140
 - Forward Error Correction..... 142
 - Input Gain 139
 - Jitter Value 145
 - Jitter Value (Fax) 140
 - Max Baud Rate 140
 - Maximum Jitter Value 144
 - Minimum Jitter Value 143
 - Mode (Fax)..... 140
 - Network Disconnection..... 145
 - Optimization Factor 144
 - Output Gain..... 139
 - Phone Number (Auto Call) 142
 - Select Channel 139
 - Silence Compression 142
 - Voice Gain 139
- Voice/FAX Parameters screen,
 - accessing
 - analog..... 213
 - T1/E1 136
- Voice/FAX parameters, setting
 - analog..... 213
 - T1/E1 136
- voip dialing digits
 - non-PSTN type..... 74
 - types used..... 73
- voip email account
 - analog..... 252
 - T1/E1 170
- voip management, remote
 - analog..... 241
 - T1/E1 159
- voip network, lab/simulated
 - use in startup 75
- voip software
 - host PC 41, 52
- voip software (analog)
 - host PC 197
- voip software (T1/E1)
 - host PC 121
- voip system example, conceptual (E1)
 - calls to remote PSTN 323
 - foreign calls, national rates 325
 - voip site to voip site 322
- voip system example, digital & analog, with phonebook details

- E1 356
- T1 307
- voip system example, digital only,
with phonebook details
 - E1 349
 - T1 301
- voip(E1)
 - basic functions of 20
- voip(T1)
 - basic functions of 13
- voltage, operating
 - analog models 44
 - E1 models 43
 - T1 models 42
- W**
- warnings, safety 88
- warranty 524
- web browser GUI and logs
 - analog 258
 - T1/E1 176
- web browser GUI, enabling
 - analog 59, 212
 - T1/E1 135
- web browser interface
 - browser version requirement 441,
444
 - general 441
 - Java requirement 441
 - prerequisite local assigning of IP
address 442
 - video useability 441
- web GUI
 - Java and 443
 - remote control/configuration and
..... 443
- web GUI vs. local Windows GUI
comparison 442
- web GUI vs. Windows GUI
 - BRI 37
- web GUI, logging and 442
- weight
 - analog models 44
 - E1 models 43
 - T1 models 42
- weight loading
 - in rack 95
- weight of unit
 - lifting precaution 88
 - personnel requirement 88
- Well Known Ports 542
- well-known port number, SMTP
 - analog 252
 - T1/E1 170
- well-known port, gatekeeper
registration
 - E1 333
 - T1 284
- well-known port, Q.931 params,
H.323
 - E1 333, 342
 - T1 284, 293
- well-known port, SIP
 - E1 343
 - T1 294
- well-known port, SNMP
 - analog 243
 - T1/E1 161
- Windows GUI vs. web GUI
 - BRI 38
- wink signaling (E&M) 234
- Wink Timer (E&M) field 234
- With H.245 Addr option (gatekeeper,
Network Parameters) 506
- worksheet
 - phonebook 79, 80
- X**
- XMT (channel) LED
 - analog models 33, 34
 - BRI models 40
- XMT (Ethernet) LED
 - analog models 33
 - BRI models 40
- XSG LED
 - analog models 33, 34
- Y**
- Yellow Alarm (E1 stats) field 399
- Yellow Alarm (T1 stats) field 396
- Yellow Alarm Format field (T1)... 152
- Z**
- zone management (gatekeeper) 482
- Zone Prefixes 1 & 2 gatekeeper
defined services 517

zone prefixes, example	518	definition of.....	483
zones, gatekeeper.....	482	establishing	482
definition.....	450		



S000249H

Free Manuals Download Website

<http://myh66.com>

<http://usermanuals.us>

<http://www.somanuals.com>

<http://www.4manuals.cc>

<http://www.manual-lib.com>

<http://www.404manual.com>

<http://www.luxmanual.com>

<http://aubethermostatmanual.com>

Golf course search by state

<http://golfingnear.com>

Email search by domain

<http://emailbydomain.com>

Auto manuals search

<http://auto.somanuals.com>

TV manuals search

<http://tv.somanuals.com>