
MultiVOIP[®] GSM

SIP-to-Cellular Gateways

Model: MVPGSM-2

User Guide



User Guide

S000450A

Wireless MultiVOIP GSM Unit (Model: MVPGSM-2)

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 © 2008, by Multi-Tech Systems, Inc.

Multi-Tech Systems, Inc. makes no representations or warranty 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. Check Multi-Tech's Web site for current versions of our product documentation.

Record of Revisions

Revision	Date	Description
A	10/27/08	Initial release.

Patents

This Product is covered by one or more of the following U.S. Patent Numbers: **6219708, 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

Registered trademarks of Multi-Tech Systems, Inc. are MultiVOIP GSM, Multi-Tech, and the Multi-Tech logo. Windows is a registered trademark of Microsoft.

World Headquarters

Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, Minnesota 55112
Phone: 763-785-3500 or 800-328-9717
Fax: 763-785-9874
<http://www.multitech.com>

Technical Support

Country	By Email	By Phone
Europe, Middle East, Africa:	support@multitech.co.uk	(44) 118 959 7774
U.S., Canada, all others:	support@multitech.com	(800) 972-2439 or (763) 717-5863

Warranty

Warranty information for this product can be found on the Multi-Tech Systems website at: www.multitech.com

CONTENTS

CHAPTER 1 – DESCRIPTION AND SPECIFICATIONS	4
Introduction	4
Specifications	5
CHAPTER 2 – INSTALLATION AND ACTIVATION	7
Introduction	7
Safety Warnings	7
Unpacking Your MultiVOIP GSM	7
Mounting Instructions.....	8
Cabling Procedure	9
GSM Instructions.....	10
CHAPTER 3 – SOFTWARE INSTALLATION	11
Introduction	11
Loading MultiVOIP GSM Software onto the PC	11
Setup Overview	14
CHAPTER 4 – CONFIGURING YOUR MULTIVOIP GSM	24
Introduction	24
Software Categories Covered in This Chapter	24
How to Navigate Through the Software	25
Web Browser Interface	25
CHAPTER 5 – PHONE BOOK CONFIGURATION	52
Introduction	52
Sample Configurations	53
Phone Book Descriptions	56
CHAPTER 6 – USING THE SOFTWARE	64
Introduction	64
Software Categories Covered in This Chapter	64
Statistics Section.....	66
MultiVOIP GSM Program Menu Items	76
FTP Server File Transfers (“Downloads”)	81
Web Browser Interface	85
SysLog Server Functions	88
APPENDIX A – CABLE PIN-OUTS & PORTS	89
APPENDIX B – REGULATORY INFORMATION	90
APPENDIX C – WASTE ELECTRICAL AND ELECTRONIC EQUIPMENT (WEEE) STATEMENT.....	92
APPENDIX D – C-ROHS HT/TS SUBSTANCE CONCENTRATION.....	93
INDEX	94

Chapter 1 – Description and Specifications

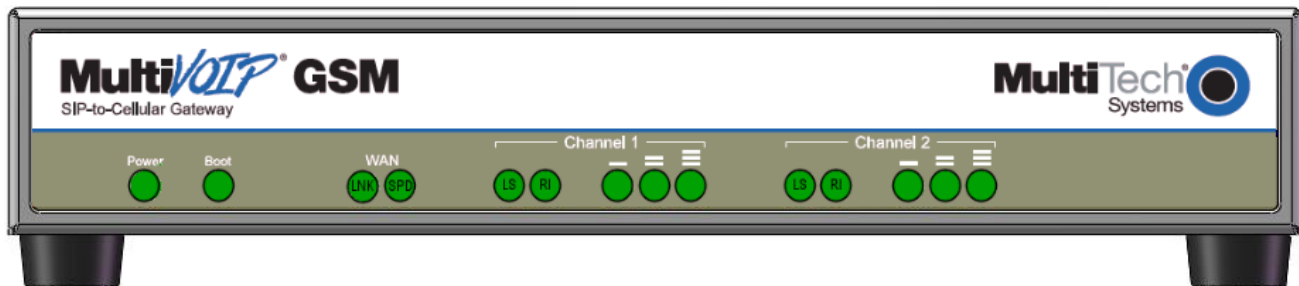
Introduction

The MultiVOIP GSM provides wireless voice communications over the Internet or an Intranet. By integrating wireless connectivity into your existing data network, you can realize substantial savings on inter-office long distance toll charges. The MVPGSM has “phone books,” which are directories set up to simulate dialing and connecting as though the call was in the local area.

Front Panel LEDs

LED Types. The MVPGSM has two types of LEDs on the front panel:

- (1) general operation LED indicators (for power, booting, and Ethernet functions), and
- (2) channel operation LED indicators that describe the voice traffic and performance for each channel.



MVPGSM LEDs

Front Panel LED Definitions	
LED	Description
General Operation LEDs	
Power	Indicates presence of power
Boot	After power up, the Boot LED will be on briefly while the MultiVOIP GSM is booting. It lights whenever the MultiVOIP GSM is booting, saving a configuration or receiving a firmware upgrade.
WAN	LNK. Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While the link is up, this LED will flash off to indicate activity. SPD. Speed indicator LED: this is lit when link speed is 100MB/s and off when it is 10MB/s.
Channel-Operation LEDs (one set for each channel)	
LS	Line Status. <ul style="list-style-type: none"> • Continuous “on” state indicates that the channel is connected. • Flashing states: <ul style="list-style-type: none"> • 600 ms on / 600 ms off – searching for network, no SIM card or authentication in progress • 75 ms on / 3000 ms off – IDLE mode; registered on network but no call in progress • 75 ms on / 75 ms off / 75 ms on / 3000 ms off – GPRS PDP contexts activated • Off state. Channel is off (not ready) or in sleep mode.
RI	Ring Indicator. This LED will light when an incoming Ring Signal is detected.
Signal Strength Bars (0-3)	There are three LEDs for each channel. If no LEDs are lit, there is not enough signal strength to maintain the wireless connection reliably. If at least one LED is lit, there is enough signal to maintain connection with Low signal strength. If the left two LEDs are lit, a Medium signal strength is present. If all three LEDs are lit, a High signal strength is present.

Computer Requirements

The computer on which the MVPGSM 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 GSM.

However, this PC does not need to be connected to the MultiVOIP GSM 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 using a web browser.

Specifications

	MVPGSM-2
Operating Voltage/Current	+9 to 12Vdc (9Vdc @ 750mA) (12Vdc @ 580mA)
Mains Frequencies	50 - 60 Hz
Power Consumption	14 watts
Mechanical Dimensions	1.71" H x (2.1" with feet attached) 9.792" W x 7.646" D ----- 4.343 cm H x (5.334 cm with feet attached) 24.871 cm W x 19.42 cm D
Weight	4.75 lbs (2.14 kg)
Operational Temperature	-5° to +55° C (UL listed @ 40° C limited by power supply) @ 20-90% non-condensing relative humidity.
Certifications	CE EMC: FCC Parts 2/15/22/24, RSS 132/133, EN 310 489-7, EN 55022, EN 55024 <u>Safety</u> : UL 60950, cUL, EN 60950, AS/NZS 60950:2000

Antenna Specifications

GSM/EGSM Antenna Requirements/Specifications

Frequency Range:	824 – 960 MHz / 1710 – 1990 MHz
Impedance:	50 Ohms
VSWR:	<2.0:1
Typical Radiated Gain:	3 dBi on azimuth plane
Radiation:	Omni
Polarization:	Vertical
Wave:	Half Wave Dipole

Antennas available from Multi-Tech Systems, Inc.

Description	Part Number
900/1800 MHz 1/2 Wave Antenna Mag Mount, 12.5", 1 Pack	ANF1-1MMHW
850/1900 MHz 1/2 Wave Antenna Mag Mount, 12.5", 1 Pack	ANF2-1MMHW

PTCRB Requirements Note

There cannot be any alteration to the authorized antenna system. The antenna system must be the same type with similar in-band and out-of-ban radiation patterns and maintain the same specifications.

FCC Requirements Note

The antenna gain, including cable loss, must not exceed 3.0 dBi at 1900 MHz / 1.6 dBi at 850 MHz for mobile operating configurations and 7.0 dBi at 1900 MHz / 2.3 dBi at 850 MHz for fixed mounted operations, as defined in 2.1091 and 1.1307 of the rules for satisfying RF exposure compliance.

Load Balancing

The MVPGSM uses round robin load balancing to ensure that one or more SIMs are not used excessively while others go largely ignored during times of light traffic. The channels are treated as a list and once a channel is used, it goes to the bottom of the list and the next channel comes to the top of the list and is ready for use. In the case of a MVPGSM-2, once channel 1 has been used, the next call will be routed through channel 2, then back to channel 1, and so on. When Hunting is enabled, load balancing is done automatically.

Chapter 2 – Installation and Activation

Introduction

The MultiVOIP GSM is equally usable as tabletop unit or mounted in a location with good reception. The initial setup is best performed before any mounting is done.

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 replacement.

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

General Safety

The MultiVOIP GSM is designed for, and intended to be used in fixed applications. “Fixed” means that the device is physically secured at one location and is not able to be easily moved to another location.

Caution: A separation distance of at least 20 cm (8 inches) is normally maintained between the transmitter’s antenna and the body of the user or nearby persons. The MVPGSM is not designed for or intended to be used in portable applications within 20 cm. (8 inches) of the body of the user.

RF Interference Issues

It is important to follow any special regulations regarding the use of radio equipment due in particular to the possibility of radio frequency (RF) interference. Please follow the safety advice given below carefully.

- Switch OFF your MVPGSM in hospitals and any other place where medical equipment may be in use.
- Respect restrictions on the use of radio equipment in fuel depots, chemical plants or where blasting operations are in progress.
- There may be a hazard associated with the operation of your MVPGSM close to inadequately protected personal medical devices such as hearing aids and pacemakers. Consult the manufacturers of the medical device to determine if it is adequately protected.

Operation of your MVPGSM close to other electronic equipment may also cause interference if the equipment is inadequately protected. Observe any warning signs and manufacturers’ recommendations.

Unpacking Your MultiVOIP GSM

When unpacking your MultiVOIP GSM, check to see that all of the items are included in the box. If any box contents are missing, contact Multi-Tech Tech Support at 1-800-972-2439.

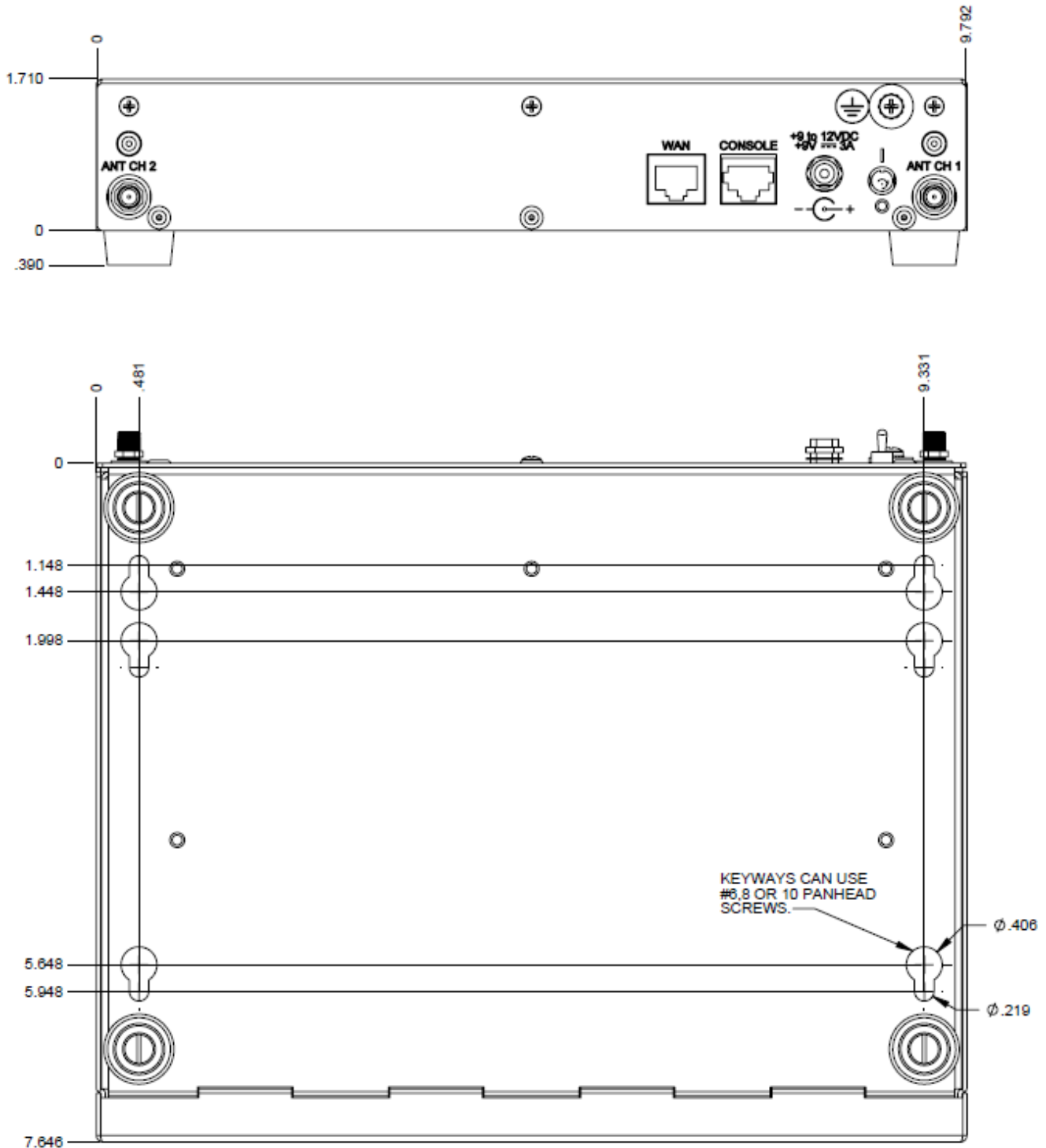
MVPGSM-2 model content list:

- MVPGSM-2
- DB9 to RJ45 cable
- Power transformer
- Power cord
- Printed Cabling Guide
- Product CD
- Antenna (2)

Mounting Instructions

When not used as a tabletop device, the bottom panel of the MVPGSM has six keyed slots for versatility of mounting. The dimensions (in inches) provided below allow for placement nearly anywhere.

Caution: Please make sure your signal strength is adequate for the planned site of mounting before actually finalizing placement. Verifying signal strength procedures can be found in the Setup Overview section.

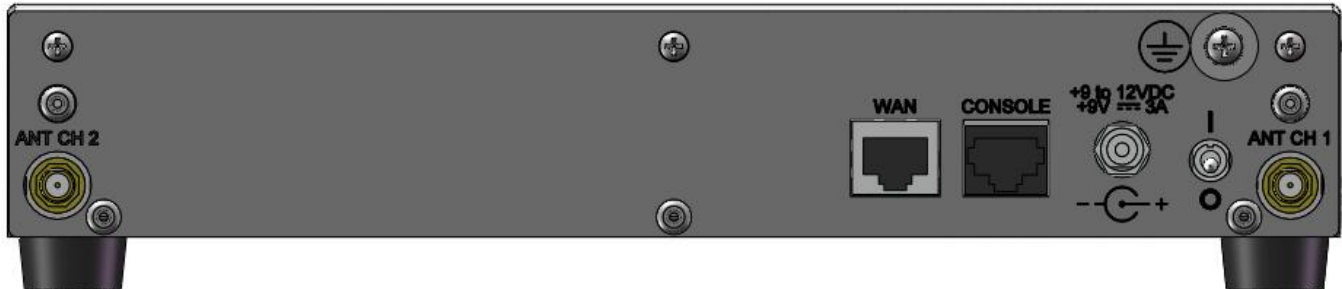


Dimensions for Mounting

Cabling Procedure

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

1. Connect a power cord to the transformer and to a live AC outlet, and then attach the barrel connector to the back of the MultiVOIP GSM as shown in the figure below.



Cabling for MVPGSM-2

2. Connect the MultiVOIP GSM to a PC by using a RJ-45 (male) to DB-9 (female) cable. Plug the RJ-45 end of the cable into the **CONSOLE** port of the MultiVOIP GSM and the other end into the PC serial port.
3. Connect a network cable to the **WAN** connector on the back of the MultiVOIP GSM. Connect the other end of the cable to your network.
4. Attach an antenna to both channel connectors on either side of the MultiVOIP GSM.
5. Turn on power to the MultiVOIP GSM by placing the ON/OFF switch on the back panel to the ON position. Wait for the **BOOT** LED on the MultiVOIP GSM to go off before proceeding. This may take a few moments.
6. Proceed to the *Software Installation* chapter to load the MultiVOIP GSM software.

GSM Instructions

Step 1 – Activate Your Wireless Account

Select a wireless network provider and follow their directions to activate your account and receive your SIM cards.

Phone Numbers

Every channel will have its own unique phone number. The phone number may simply be given to you by your wireless service provider or it may be on the SIM card or both. Wireless provider implementations may vary.

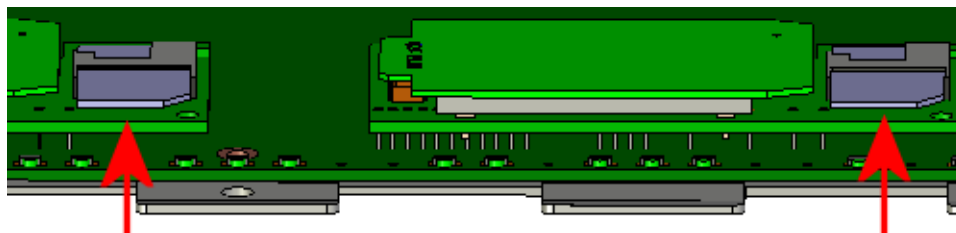
Step 2 – Installing the SIM Cards

The MVPGSM requires the power supply connection to begin operation. It also requires a SIM card (Subscriber Identity Module) to operate on a GSM network. To install the modem, do the following:

1. Turn the unit off and disconnect the power cord. Remove the three highlighted screws from the back as shown below.



2. Slide the cover forward and up to remove it.
3. Insert the SIM cards into the units at the front of the MVPGSM.



4. Verify that the SIM cards fit into their holders properly and then replace the cover. Attach the power cord.

Step 3 – Check Signal Strength

Turn the unit on and verify that the Power LED is lit and that the Boot sequence is finished (Boot LED is no longer active), then wait for the Link Status (LS) LED to show that the MVPGSM channel is registered on the wireless network (flashing 75 ms on and 3 seconds off). Once registered, the Signal Strength LEDs should be referenced for the strength of signal in its current location.

Caution: Before final placement or mounting, ensure that the wireless signal strength is strong enough for the chosen area. If no LEDs are lit, there is insufficient signal strength for proper operation. If one or more are lit, then there is good signal strength and operation will not be impaired. Finding a location with the strongest signal strength is desirable.



Chapter 3 – Software Installation

Introduction

Configuring software for your MultiVOIP GSM entails three tasks:

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

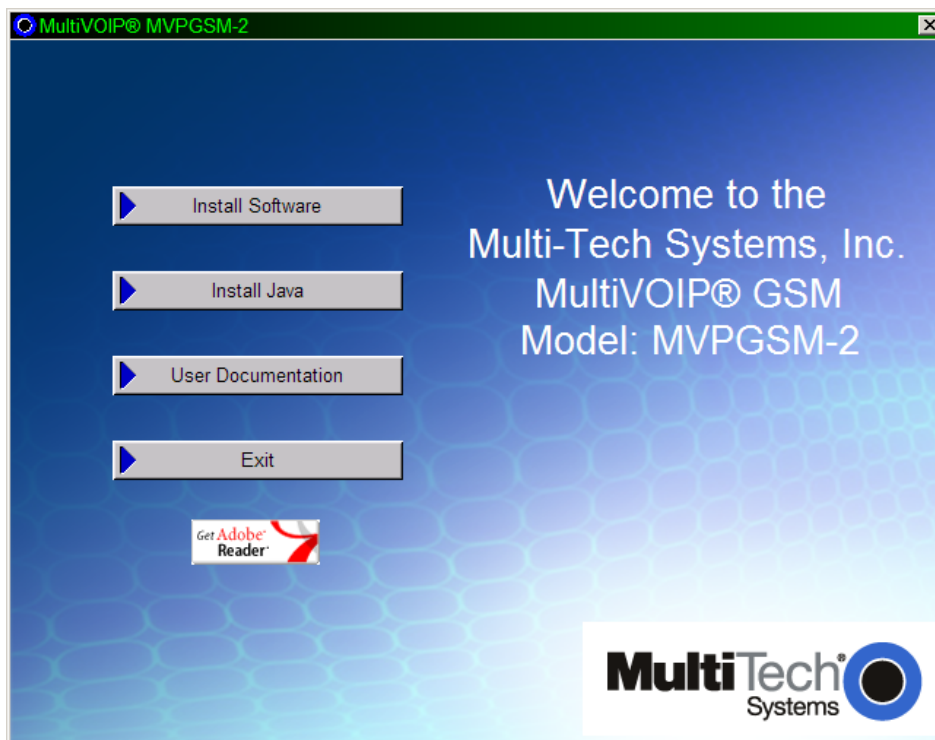
Setting values for telephony and IP parameters that will fit your system (details are in Chapter 4).

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

Loading MultiVOIP GSM Software onto the PC

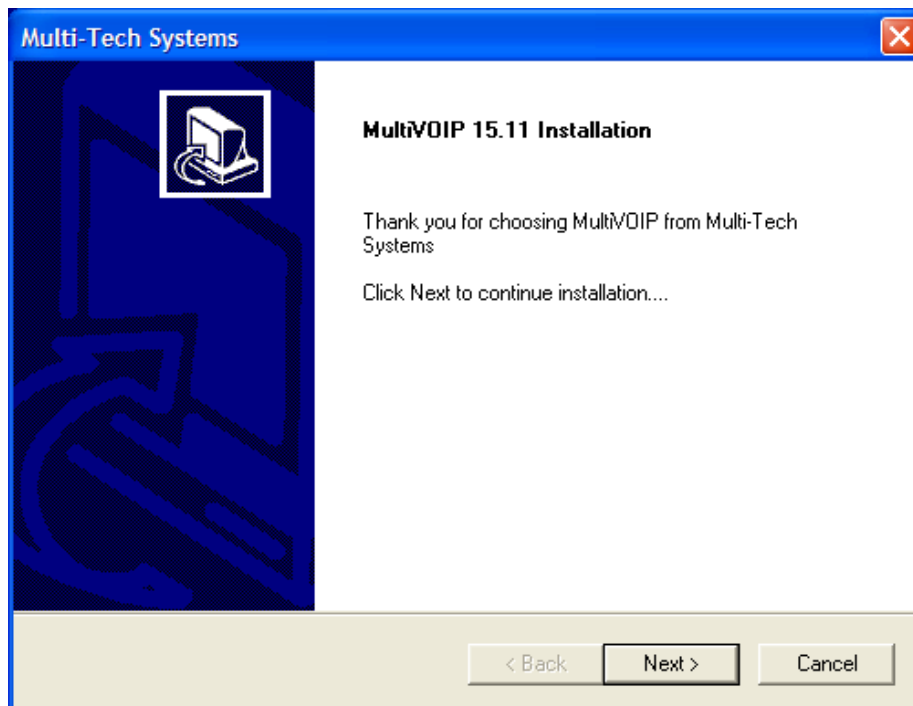
The software loading procedure does not present every screen or option in the loading process. It is assumed that someone with a thorough knowledge of Windows and the software loading process is performing the installation.

1. Be sure that your MultiVOIP GSM has been properly cabled and that the power is turned on.
2. Insert the MultiVOIP GSM CD into your CD-ROM drive. The CD starts automatically. It may take a few moments for the Multi-Tech CD installation window to display.



MVPGSM splash screen

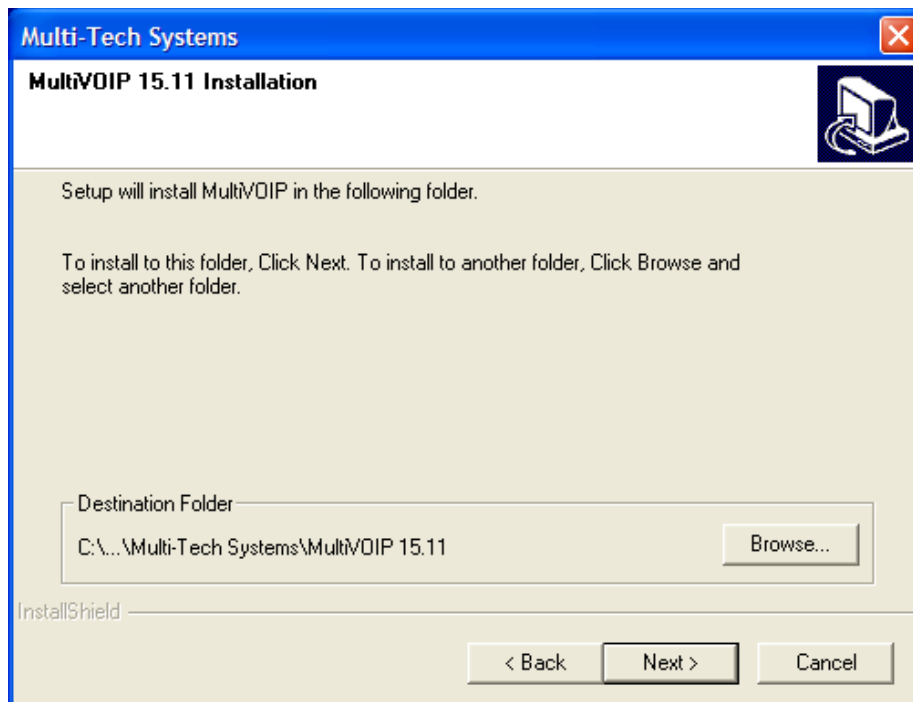
3. When the Multi-Tech Installation CD dialog box appears, click the **Install Software** icon.



Installation wizard screen

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

4. Follow the on-screen instructions to install your MultiVOIP GSM software. The first screen asks you to choose the destination for the MultiVOIP GSM software.



Destination screen

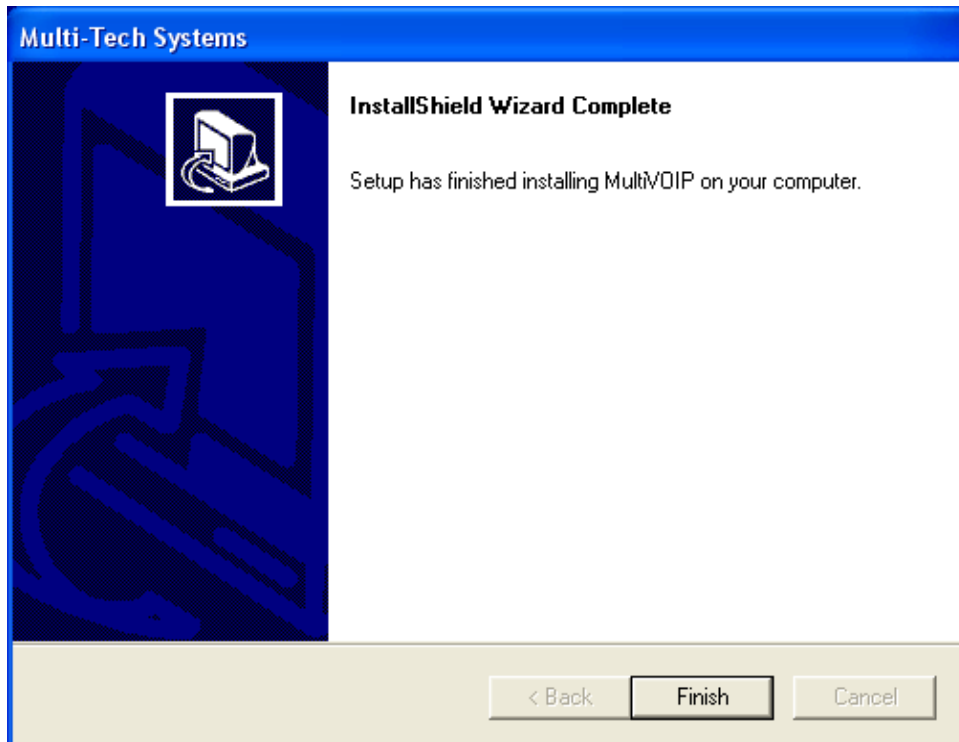
Choose a location and click **Next**.

5. At the next screen, you must select a program folder location for the MultiVOIP GSM software program icon. Click **Next**. Transient progress screens will appear while files are being copied.

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

Note: If the COM port setting made here conflicts with the actual COM port resources available in the command PC, the “Error in Opencomm handle” message will appear when the MultiVOIP GSM program is launched. If this occurs, you must reset the COM port.

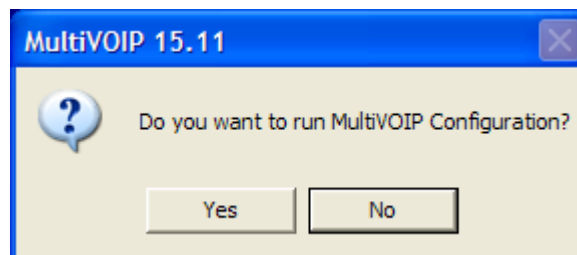
- A completion screen will appear.



Completion screen

Click **Finish**.

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



Configuration screen

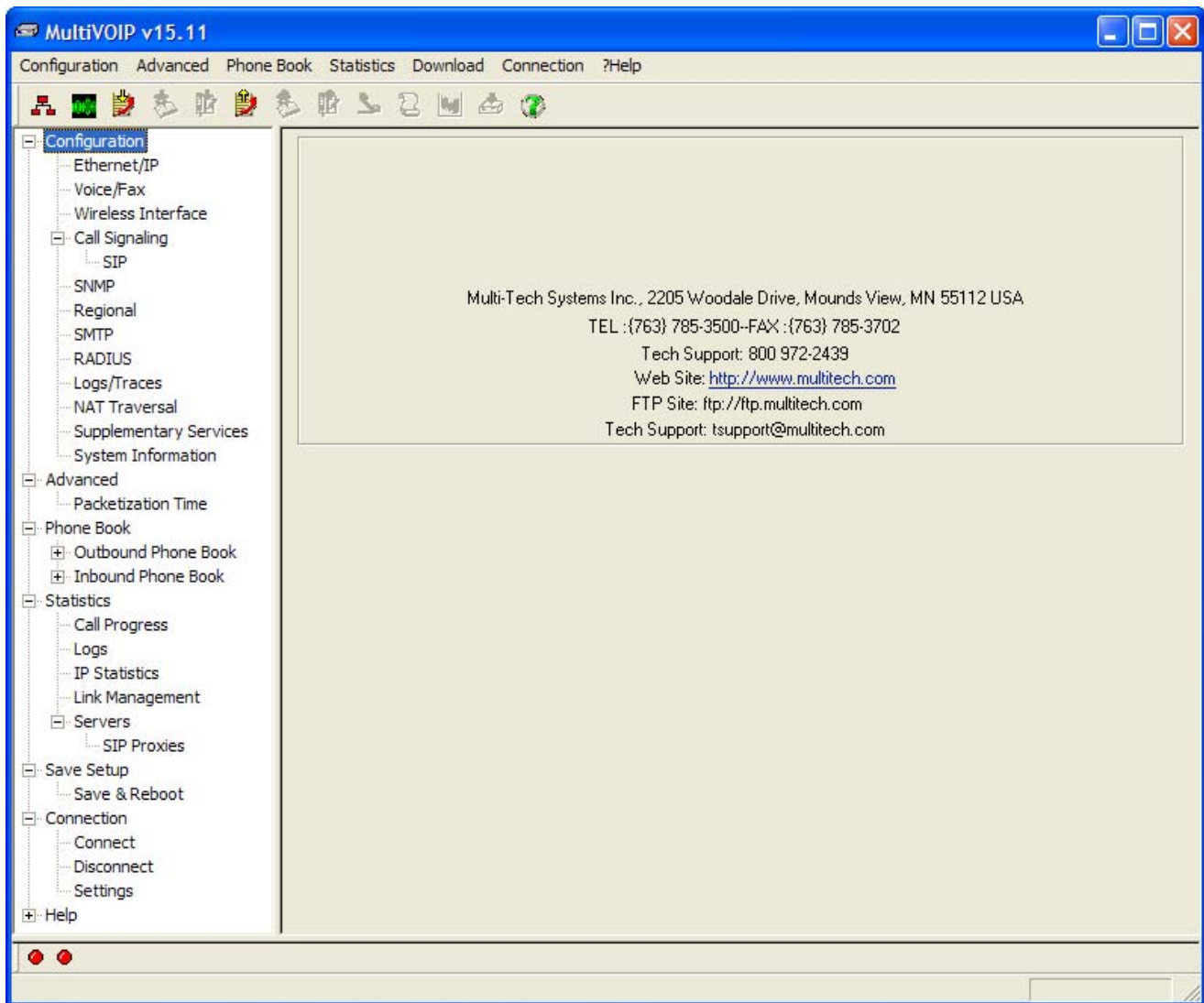
Software installation is now complete.

Setup Overview

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

- ⇒ **Ethernet/IP**
- ⇒ **Voice/Fax**
- ⇒ **Wireless Interface**
- ⇒ **Call Signaling**
- ⇒ **Regional**
- ⇒ **Phone Book**

This setup process must be followed by a **Save & Reboot** for the changes to take affect.



Main Screen

Ethernet/IP

A unique LAN IP address is required for the MultiVOIP GSM unit as well as a subnet mask and Gateway IP for minimal functionality. Other settings in this category pertain to specific features and protocols that can be used, but are not necessary for basic operation. Details for all settings are provided in chapter 4.

IP settings

Actions:

- Select Packet Prioritization if used
 - Set 802.1p Priority Parameters as needed
 - The Priority levels can be from 0 – 7, where 0 is lowest priority (details in Chapter 4)
 - VLAN ID identifies a virtual LAN by a number (1 to 4094)
- Set the Frame Type to match the network that the MultiVOIP GSM is attached to
 - TYPE II or SNAP
- Enter Gateway Name
 - Check to enable DHCP if used
- Enter IP Address for the MultiVOIP GSM unit (default is 192.168.3.143)
- Enter Subnet IP Mask for the MultiVOIP GSM unit
- Enter Gateway IP
- Enable DNS if desired
 - Enter DNS Server IP Address
- Enable SRV support if needed
- Diff Serv Parameters are for routers that are Diff Serv compatible
 - Setting both values to 0 effectively disables Diff Serv
- FTP Server Enable is only needed for firmware and software updates to the MultiVOIP GSM
- TDM Routing can be used if necessary

Voice

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

Voice/Fax Parameters

Select Channel Channel 1

Voice Gain
 Input 0 dB Output 0 dB

Dtmf
 Gain
 High -6 dB Low -8 dB
 Duration 100 ms
 DTMF : Out Of Band - Fixed Duration
 Out Of Band Mode: Rfc2833

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

Fax/Modem Parameters
 Fax Relay Enable
 Modem Relay 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

Auto Call
 Auto Call None Generate Local Dial Tone
 OffHook Alert Timer 10 secs
 Phone Number

Dynamic Jitter Buffer
 Minimum Jitter Value 60 ms
 Maximum Jitter Value 300 ms
 Optimization Factor 7

Automatic Disconnection
 Jitter Value 350 ms Consecutive Packets Lost 30
 Call Duration 180 secs Network Disconnection 300 secs

Configurable Payload Type

DTMF RFC 2833	96	RTP Redundancy	104
FRF11 Fax	101	Modem Relay	105
Fax Bypass	102	Modem Bypass	103

OK
 Cancel
 Copy Channel
 Default
 Help

Voice settings

Actions:

- Select Channel
 - Choose channel parameters:
 - Fax and modem parameters are not available at this time
 - Modem Relay Enable allows modem traffic through the VOIP system
 - Adjusting the Voice Gain and DTMF *should not be done* as it may adversely affect voice and DTMF quality
 - Select a Coder or allow Automatic negotiation
 - Advanced Features
 - Silence Compression, when enabled, will not send IP packets during times of silence
 - Echo Cancellation removes echo to improve voice quality
 - Forward Error Correction allows some bad packets to be recovered
 - Choose Auto Call / OffHook Alert settings
 - For automatically calling a remote VOIP without dialing (details in Chapter 4)
 - Change Dynamic Jitter values if necessary (details in Chapter 4)
 - Select any Automatic Disconnection options needed to ensure lines are not left “open”
 - Configurable Payload Types are best left at their defaults.
 - The Copy Channel button is available for easily transferring these settings to the other channels
- Repeat for all channels to be used

Wireless Interface

The Wireless Interface Parameters are the settings for the GSM connection. The Copy Channel button can save a lot of time during this step if channels are to be set with the same parameters.

The screenshot shows a dialog box titled "Wireless Interface Parameters". It contains several configuration options:

- Select Channel:** A dropdown menu currently set to "Channel 2".
- Disable Interface:** An unchecked checkbox.
- Use Module for DTMF and Tone Generation:** A checked checkbox.
- Enable CallerID:** An unchecked checkbox.
- No Response Timer:** A numeric input field set to "12" with a range of "(5-20) secs".
- Minimum Signal Quality Required:** A numeric input field set to "11" with a range of "(0-31)".
- SIM Pincodes:** Two input fields labeled "SIM PIN" and "SIM PUK".
- Modem Setup:** Five input fields labeled "Modem InitString1" through "Modem InitString5".

On the right side of the dialog, there are five buttons: "OK", "Cancel", "Copy Channel", "Default", and "Help".

Wireless Interface Parameters

Actions:

- Select Channel
 - Channel 1 or 2
- Disable Interface
 - Check box: Enable or Disable Interface
- Use Module for DTMF and Tone Generation
 - Check box: Enable or Disable DTMF and Tone generation (this is used in conjunction with out of band DTMF)
- Enable Caller ID
 - Check box: Enable or Disable Caller ID (this is for Caller ID information coming in from the wireless network)
- No Response Timer
 - Internal timer for command response time. Range 5-20 seconds
- Minimum Signal Quality Required
 - Range 0-31
- SIM Pincode
 - SIM PIN
 - Enter SIM PIN number (this is only necessary in cases where the PIN is required on power-up)
 - SIM PUK
 - Enter SIM PIN Unlock Key number (this would only be needed in conjunction with the above PIN number where 3 incorrect attempts lock an account)
- Modem Setup
 - Modem InitString1-5
 - Enter initialization strings (if necessary, internal modem commands can be added)
- Copy Channel (button)
 - Copy settings to another channel
- Default (button)
 - Reset unit to Factory Default Settings

Important: Verify Signal Strength

Before final placement or mounting, ensure that the wireless signal strength is strong enough for the chosen area. Once the unit has established connection to the wireless network, the Link Status LED will blink (75 ms on and 3 seconds off), then you can look to the signal indicators. To find your signal strength, look at the LED indicators on the front panel. If the first indicator is lit (“one bar”), there is sufficient signal strength for proper operation. If two or three are lit (two or three “bars”), then there is good signal strength and operation will not be impaired.

Call Signaling

The MultiVOIP GSM utilizes the SIP protocol for communication with other VOIP units. Additional details for all settings are found in Chapter 4.

SIP Parameters

Signaling Port :

Use SIP Proxy

Allow Incoming Calls Through SIP Proxy Only

SIP Proxy Parameters

	Proxy Domain Name / IPAddress	Port Number
Primary Proxy	<input type="text"/>	<input type="text" value="5060"/>
Alternate Proxy 1	<input type="text"/>	<input type="text" value="5060"/>
Alternate Proxy 2	<input type="text"/>	<input type="text" value="5060"/>

Append SIP Proxy Domain Name in User ID

Default Subscriber

Default Username

Default Password

Re-RegistrationTime secs

Proxy Polling Interval secs

TTL Value secs

SIP parameters

Actions:

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

Regional

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

Regional Parameters

Country/Region :

Standard Tones

Type	Frequency1	Frequency2	Cadence(secs)On/Off	Gain
DialTone	350	440	0.000/0.000/0.000/0.000	-16
RingTone	480	440	2.000/4.000/2.000/4.000	-16
BusyTone	480	620	0.500/0.500/0.500/0.500	-16
UnobtainableTone	480	620	0.000/0.000/0.000/0.000	-16
Survivability DialTone	650	650	0.000/0.000/0.000/0.000	-16
ReorderTone	480	620	0.250/0.250/0.000/0.000	-16
InterceptTone	440	620	0.000/0.000/0.000/0.000	-8

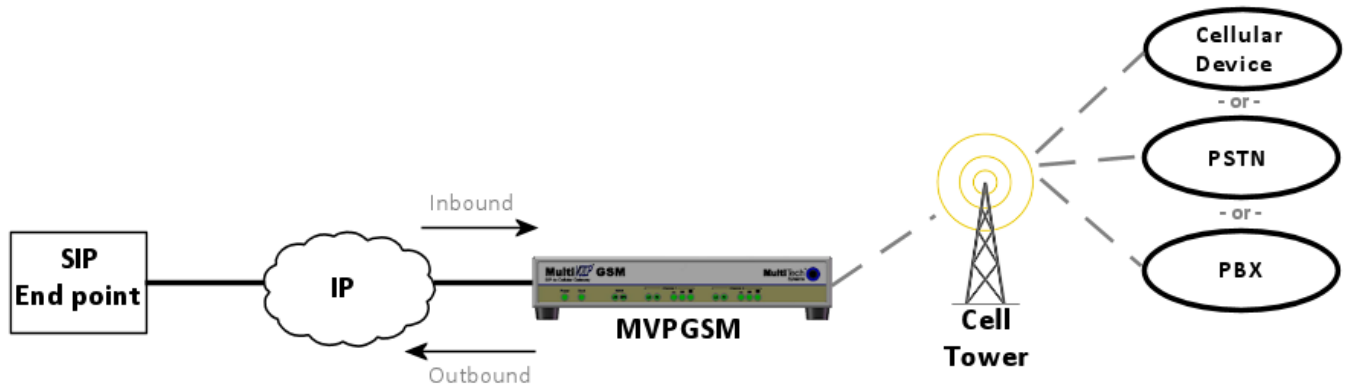
Regional Parameters

Actions:

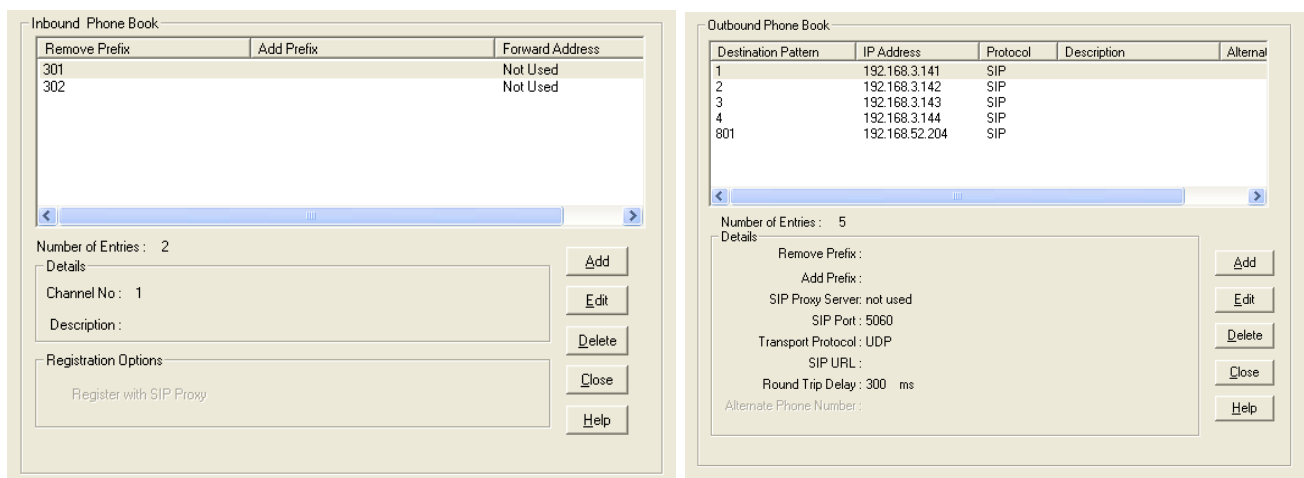
- Select the choice that matches the location of the MultiVOIP GSM from the Country/Region field
 - If there is not a selection to fit your needs, you may select Custom and set the tones manually

Phone Book

Without a populated phone book, the VOIP unit is unable to translate call traffic. You will need the information for both a local and any remote sites that are to be used. Detailed descriptions and examples are available in chapter 5. To better understand the meaning behind 'Inbound' and 'Outbound' please see the graphic below.



Inbound and Outbound directions



Phone Book screens

Actions:

- Select Outbound Phone Book
 - Select Add Entry
 - Accept Any Number: may be selected to allow any phone number from the wireless network to initiate a SIP call to the IP address of the SIP end point listed below
 - Destination Pattern: When a call from the wireless network is received, the subsequent digits received from the wireless side are used to match the destination pattern and route the call to the SIP end point with the IP address listed below
 - Remove Prefix: enter any access digits (e.g. a number to get out of a PBX system) or area codes to be taken off in this field – this manipulates the number dialed by the wireless-side caller before sending the digit string to the SIP end point of the IP address below
 - Add Prefix: any digits that need to be added to make a call appear as local should be put in this field – this manipulates the number dialed by the wireless-side caller before sending the digit string to the SIP end point of the IP address below
 - Enter the IP address of the SIP end point where the phone number is to be routed
 - Protocol type
 - SIP:
 - Select Transport Protocol, Proxy and URL if needed
 - The *Advanced Button* will allow routing to an Alternate IP Address to be entered for times when the primary does not respond within the allotted time
- Select Inbound Phone Book
 - Select Add Entry
 - Accept Any Number will allow any number received from a SIP end point to be routed to the selected channel below
 - Enter any access digits followed by the local calling code in the Remove Prefix field. Incoming SIP calls that match these digits will be routed to the selected channel below and have those digits removed from the incoming number
 - Enter any digits needed to access an outside line in the Add Prefix field. These will be added to the prefix of the SIP inbound call
 - Select Hunting in the Channel Number field to have the VOIP use the next available channel (this is necessary for load balancing), otherwise you can select a specific channel for the call to go out on
 - Add a description if you like
 - Call Forward may be set up (details available in Chapter 5)
 - Select Registration Option
- Repeat the Phone Book steps for any additional entries needed

Save & Reboot

Any time that you change settings on the VOIP unit, you must choose the **Save & Reboot** option; otherwise all changes made will not take affect and be lost completely when the MultiVOIP GSM is reset, shutdown or you exit from the management screen.

Chapter 4 – Configuring Your MultiVOIP GSM

Introduction

There are two methods of configuring your MultiVOIP GSM; one is through a web interface, and the other is through the Windows software interface. There are several necessary parameters that must be set for the MultiVOIP GSM unit to operate properly, with some additional settings that are optional. You must know the IP address that will be used, the IP mask, the Gateway IP, and the Domain Name Server information. The MultiVOIP GSM must be configured locally at first, but changes to this initial configuration can be done locally or remotely. Local configuration is done through a connection between the “Console” port of the MultiVOIP GSM and the COM port of the computer; the MultiVOIP GSM configuration software is used for this.

This chapter will explain the setup portion of the software pertaining to the list below, while Chapter 5 will cover the Phone Book setup and Chapter 6 will discuss the Statistics options and overall maintenance of the MultiVOIP GSM.

Software Categories Covered in This Chapter

- Ethernet/IP
- Voice/Fax
- Wireless Interface
- Call Signaling
 - SIP
- Regional
- SMTP
- RADIUS
- Logs/Traces
- NAT Traversal
- Supplementary services
- Save Setup
- Connection
 - Settings

How to Navigate Through the Software

The MultiVOIP GSM software is launched from the Start button and is found in the All Programs area under the title of MultiVOIP GSM x.xx (where x represents version number). The top option is “Configuration” – choose this.

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

Web Browser Interface

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

Set up the Web Browser interface (Optional). After an IP address for the MultiVOIP GSM unit has been established, you can choose to configure the unit by using the MultiVOIP GSM web browser interface. If you want to do configuration work using the web browser interface, you must first set it up:

- Set IP address of MultiVOIP GSM unit using the MultiVOIP GSM Configuration program (the Windows interface). The default IP address is 192.168.3.143.
- Save Setup in Windows interface.
- Close Windows interface.
- Install Java program from MultiVOIP GSM product CD (on first use only).
- Open web browser.
- Browse to IP address of MultiVOIP GSM unit.
- If a username and password have been set up, enter them when prompted.
- Set the browser to allow pop-ups. The MultiVOIP GSM Web interface makes use of pop-up windows.
- The configuration screens in the web browser will have the same content as their counterparts in the software; only the presentation differs.

Ethernet/IP

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

Ethernet / IP Parameters

Ethernet Parameters

Packet Prioritization (802.1p) Frame Type: TYPE-II

802.1p Parameters

Priority

Call Control: 6-Voice

VoIP Media: 3-Excellent Effort

Others: 0-Best Effort

VLAN ID: 1

IP Parameters

Gateway Name: MultiVoIP

Enable DHCP

IP Address: 192 . 168 . 3 . 143

IP Mask: 255 . 255 . 255 . 0

Gateway: . . .

Diff Serv Parameters

Call Control PHB: 34

VoIP Media PHB: 46

FTP Server

Enable

DNS

Enable DNS

Enable SRV

DNS Server IP Address: . . .

TDM Routing Option

Use TDM Routing For Intra-Gateway calls

OK

Cancel

Help

Network parameters

The **Ethernet/IP Parameters** fields are described in the tables and text passages below.

Ethernet/IP Parameter Definitions		
Field Name	Values	Description
Ethernet Parameters		
Packet Prioritization (802.1p)	Y/N	Select to activate prioritization under 802.1p protocol (described below).
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.
802.1p		<p>A draft standard of the IEEE about data traffic prioritization on Ethernet networks. The 802.1p draft is an extension of the 802.1D bridging standard. 802.1D determines how prioritization will operate within a MAC-layer bridge for any kind of media. The 802.1Q draft for virtual local-area-networks (VLANs) addresses the issue of prioritization for Ethernet networks in particular.</p> <p>802.1p enacts this Quality-of-Service feature using 3 bits. This 3-bit code allows data switches to reorder packets based on priority level. The descriptors for the 8 priority levels are given below.</p> <p><u>802.1p PRIORITY LEVELS:</u></p> <p><i>LOWEST PRIORITY</i></p> <p>1 – Background: Bulk transfers and other activities permitted on the network, but should not affect the use of network by other users and applications.</p> <p>2 – Spare: An unused (spare) value of the user priority.</p> <p>0 – Best Effort (default): Normal priority for ordinary LAN traffic.</p> <p>3 – Excellent Effort: The best effort type of service that an information services organization would deliver to its most important customers.</p> <p>4 – Controlled Load: Important business applications subject to some form of "Admission Control", such as preplanning of Network requirement, characterized by bandwidth reservation per flow.</p> <p>5 – Video: Traffic characterized by delay < 100 ms.</p> <p>6 – Voice: Traffic characterized by delay < 10 ms.</p> <p>7 – Network Control: Traffic urgently needed to maintain and support network infrastructure.</p> <p><i>HIGHEST PRIORITY</i></p>
Call Control Priority	0-7	Sets the priority for signaling packets. 0 is lowest priority.
VOIP Media Priority	0-7	Sets the priority for media packets. 0 is lowest priority.
Others (Priorities)	0-7, where 0 is lowest priority	Sets the priority for SMTP, DNS, DHCP, and other packet types. 0 is lowest priority.
VLAN ID	1 - 4094	The 802.1Q IEEE standard allows virtual LANs to be defined within a network. This field identifies each virtual LAN by number.
IP Parameter fields		
Gateway Name	alphanumeric	Descriptor of current VOIP unit to distinguish it from other units in system.
Enable DHCP	Y/N disabled by default	Dynamic Host Configuration Protocol is a method for assigning IP address and other IP parameters to computers on the IP network in a single message with great flexibility. IP addresses can be static or temporary depending on the needs of the computer.
IP Address	<i>n.n.n.n</i>	The unique IP address assigned to the MultiVOIP GSM. Default is 192.168.3.143
IP Mask	<i>n.n.n.n</i>	Subnetwork address that allows for sharing of IP addresses within a LAN.
Gateway	<i>n.n.n.n</i>	The IP address of the device that connects your MultiVOIP GSM to the Internet.

Table is continued on next page...

Ethernet/IP Parameter Definitions (continued)		
Field Name	Values	Description
Diff Serv Parameter fields		Diff Serv PHB (Per Hop Behavior) values pertain to a differential prioritizing system for IP packets as handled by Diff Serv-compatible routers. There are 64 values, each with an elaborate technical description. These descriptions are found in TCP/IP standards RFC2474, RFC2597, and, for present purposes, in RFC3246, which describes the value 34 (34 decimal; 22 hex) for Assured Forwarding behavior (default for Call Control PHB) and the value 46 (46 decimal; 2E hexadecimal) for Expedited Forwarding behavior (default for VOIP Media PHB). Before using values other than these default values of 34 and 46, consult these standards documents and/or a qualified IP telecommunications engineer. To disable Diff Serv, configure both fields to 0 decimal.
Call Control PHB	0 – 63 default = 34	Value is used to prioritize call setup IP packets. Setting this parameter to 0, in conjunction with VOIP Media PHB below will disable Diff Serv.
VOIP Media PHB	0 – 63 default = 46	Value is used to prioritize the RTP/RTCP audio IP packets. Setting this parameter to 0, in conjunction with Call Control PHB above will disable Diff Serv.
FTP Parameter fields		
FTP Server Enable	Y/N Default = disabled See “FTP Server File Transfers” in Chapter 6	MultiVOIP GSM unit has an FTP Server function so that firmware and other important operating software files can be transferred to the VOIP via the network.
DNS Parameter fields		
Enable DNS	Y/N Default = disabled	Enables Domain Name System function where computer names are resolved using a worldwide distributed database.
Enable SRV	Y/N	Enables ‘service record’ function. Service record is a category of data in the Internet Domain Name System specifying information on available servers for a specific protocol and domain, as defined in RFC 2782. Newer internet protocols like SIP, STUN, H.323, POP3, and XMPP may require SRV support from clients. Client implementations of older protocols, like LDAP and SMTP, may have been enhanced in some settings to support SRV.
DNS Server IP Address	<i>n.n.n.n</i>	IP address of specific DNS server to be used to resolve Internet computer names.

Voice/Fax

Setting the Voice Parameters. The Voice/Fax section needs to be set for each channel to be used. However, once you have established a set of Voice parameters for a particular channel, you can apply this entire set of Voice parameters to another channel by using the **Copy Channel** button and its dialog box. To copy a set of Voice parameters to all channels, select “Copy to All” and click **Copy**.

The majority of the settings should be left at their default settings as changes may introduce problems with audio quality. In each field, enter the values that fit your particular setup.

Voice/Fax Parameters

Select Channel: Channel 1

Voice Gain
 Input: 0 dB Output: 0 dB

Dtmf
 Gain: High -6 dB Low -8 dB
 Duration: 100 ms
 DTMF: Out Of Band - Fixed Duration
 Out Of Band Mode: Rfc2833

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

Fax/Modem Parameters
 Fax Relay Enable
 Modem Relay 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

Auto Call
 Auto Call: None Generate Local Dial Tone
 OffHook Alert Timer: 10 secs
 Phone Number:

Dynamic Jitter Buffer
 Minimum Jitter Value: 60 ms
 Maximum Jitter Value: 300 ms
 Optimization Factor: 7

Automatic Disconnection
 Jitter Value: 350 ms Consecutive Packets Lost: 30
 Call Duration: 180 secs Network Disconnection: 300 secs

Configurable Payload Type

DTMF RFC 2833	96	RTP Redundancy	104
FRF11 Fax	101	Modem Relay	105
Fax Bypass	102	Modem Bypass	103

OK
 Cancel
 Copy Channel
 Default
 Help

Voice/Fax parameters

The Voice/FAX Parameters settings 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	Channel to be configured is selected here.
Copy Channel	--	Copies the Voice/FAX attributes of one channel to another channel.
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 IP network to the remote SIP device. The default & recommended value is 0 dB .
Output Gain	+31dB to -31dB	Modifies audio level from IP to the voice channel. The default and recommended value is 0 dB .
DTMF Gain	--	The DTMF Gain (Dual Tone Multi-Frequency) controls the volume level of the DTMF tones sent out for Touch-Tone dialing.
DTMF Gain, High Tones	+3dB to -31dB & "mute"	Default value: -4 dB . Not to be changed except under supervision of Multi-Tech Technical Support.
DTMF Gain, Low Tones	+3dB to -31dB & "mute"	Default value: -7 dB . Not to be changed except under supervision of Multi-Tech Technical Support.
DTMF Parameters		
Duration (DTMF)	60 – 3000 ms	When DTMF: Out of Band is selected, this setting determines how long each DTMF digit 'sounds' or is held. Default = 100 ms.
DTMF In/Out of Band	Out of Band, or Inband	When DTMF Out of Band is selected, the MultiVOIP GSM detects and regenerates DTMF tones. When DTMF Inband is selected, the DTMF digits are passed through the MultiVOIP GSM unit as they are received.
Out of Band Mode	RFC 2833, SIP Info	RFC2833 method. Uses RTP mode defined in RFC 2833 to transmit the DTMF digits. SIP Info method. Generates DTMF tones on the telephony call leg. The SIP INFO message is sent along the signaling path of the call. You must set this parameter per the capabilities of the remote endpoint with which the VOIP will communicate. The RFC2833 method is the more common of the two methods.
FAX Parameters – This section is grayed-out as Fax is not supported by the wireless channels		
Fax Enable		
Modem Relay Enable		
Max Baud Rate (Fax)		
Fax Volume		
Jitter Value (Fax)		
Mode (Fax)		

Table is continued on next page...

Voice/Fax Parameter Definitions (continued)		
Coder Parameters		
Coder	Manual or Automatic	Determines whether selection of coder is manual or automatic. When Automatic is selected, the local and remote voice channels will negotiate the voice coder to be used by selecting the highest bandwidth coder supported by both sides without exceeding the Max Bandwidth setting. G.723, G.729, or G.711 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. Here 64K of digital voice is 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.
Selected Coder additional choices	G.711, G.729 -or- G.729, G.711	Coder Priority has two options (G.711,G.729 or G.729, G711) on the Selected Coder listing of the Coder group on the Voice/Fax screen. If G.711 is the higher priority, i.e., G.711 is preferred to G729 on the sending side, then G.711, G.729 option is selected. Similarly, if G.729 has the higher priority, then G.729, G.711 option is selected. It is used whenever a user wants to advertise both G.711 and G.729 coders with higher preference to a particular coder. It is useful when the calls are made from a particular channel on the VOIP to two different destinations where one supports G.711 and the other supports G.729.
Max bandwidth (coder)	11 – 128 kbps	This drop-down list enables you to select the maximum bandwidth allowed for this channel. The Max Bandwidth drop-down list is enabled only if the Coder is set to Automatic. If coder is to be selected automatically ("Auto" setting), then enter a value for maximum bandwidth.
Advanced Features		
Silence Compression	Y/N	Determines whether silence compression is enabled (checked) for this voice channel. With Silence Compression enabled, the MultiVOIP GSM will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel (<i>default = on</i>).
Echo Cancellation	Y/N	Determines whether echo cancellation is enabled (checked) for this voice channel. Echo Cancellation removes echo and improves sound quality (<i>default = on</i>).
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 doubles the overhead to the total network bandwidth consumed by the voice channel (<i>default = Off</i>).

Table is continued on next page...

Voice/Fax Parameter Definitions (continued)		
Field Name	Values	Description
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall,	The AutoCall option enables the local MultiVOIP GSM to call a remote SIP endpoint without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP GSM voice/fax channel, the MultiVOIP GSM immediately connects to the SIP end point identified in the Phone Number box of this option. This function requires an entry in the Outgoing phonebook of the local MultiVOIP GSM.
Generate Local Dial Tone	Y/N	If selected, dial tone will be generated locally while the call is connecting between gateways. This would be useful if there is a long network delay.
Offhook Alert Timer		This section is grayed-out as Offhook Alert is not used.
Phone Number	--	Phone number used for Auto Call function. This phone number must correspond to an entry in the Outbound Phonebook of the local MVPGSM.
Dynamic Jitter		
Dynamic Jitter Buffer		Dynamic Jitter defines a minimum and a maximum jitter value for voice communications. When receiving voice packets from an SIP end point, varying delays between packets may occur due to network traffic problems. This is called Jitter. To compensate, the MVPGSM uses a Dynamic Jitter Buffer. The Jitter Buffer enables the MVPGSM to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An Optimization Factor adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly affects the voice delay between MultiVOIP GSM gateways.
Minimum Jitter Value	60 - 400ms	The minimum dynamic jitter buffer of 60 milliseconds is the minimum delay that would be acceptable over a low jitter network. Default = 150 ms
Maximum Jitter Value	60 - 400ms	The maximum dynamic jitter buffer of 400 milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 ms
Optimization Factor	0 - 12	The Optimization Factor determines how quickly the length of the Dynamic Jitter Buffer is changed based on actual jitter encountered on the network. Selecting the minimum value of 0 means low voice delay is desired, but increases the possibility of jitter-induced voice quality problems. Selecting the maximum value of 12 means highest voice quality under jitter conditions is desired at the cost of increased voice delay. Default = 7.
Auto Disconnect		
Automatic Disconnection	--	The Automatic Disconnection group provides four options which can be used singly or in any combination.
Jitter Value	1-65535	The Jitter Value defines the average inter-arrival packet deviation before the call is automatically disconnected. The default is 300 milliseconds. A higher value means voice transmission will be more accepting of jitter. A lower value is less tolerant of jitter. Inactive by default. When active, default = 300 ms. However, value must equal or exceed Dynamic Minimum Jitter Value.
Call Duration	1-65535	Call Duration defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. Inactive by default. When active, default = 180 sec.
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	Specifies how long to wait before disconnecting the call when IP network connectivity with the remote site has been lost. Default = 30 sec

Configurable Payload Type

The Configurable Payload Type is located on the bottom of the Voice/Fax screen. The Configurable Payload Type is used when the remote side uses a different payload type for the associated features.

Wireless Interface

The Wireless Interface parameters are set individually for each channel. In each field, enter the values that fit your particular setup. Once you have established a set of Interface 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.

Wireless Interface Parameters

Select Channel: Channel 2

Disable Interface

Use Module for DTMF and Tone Generation

Enable CallerID

No Response Timer: 12 (5-20) secs

Minimum Signal Quality Required: 11 (0-31)

SIM Pincodes

SIM PIN: []

SIM PUK: []

Modem Setup

Modem InitString1: []

Modem InitString2: []

Modem InitString3: []

Modem InitString4: []

Modem InitString5: []

Buttons: OK, Cancel, Copy Channel, Default, Help

Wireless parameters

Wireless Interface parameter definitions		
Field Name	Values	Description
Select Channel	1 – 2	Select which channel you want to set parameters for.
Disable Interface	Check box	Enable or Disable the interface.
Use Module for DTMF Generation	Check box	Enable or Disable DTMF generation. When checked, the wireless module will generate DTMF tones to the cell network. If this is enabled, out of band DTMF must be set in Voice/Fax and out of band DTMF must be set to the remote SIP end point.
Enable Caller ID	Check box	Enable or Disable Caller ID reporting. This allows Caller ID from the cell network to be passed over IP.
No Response Timer	Range 5 - 20	Set the timeout in seconds, for the wireless module to wait for a valid response from AT commands.
Minimum Signal Quality Required	Range 0 - 31	Set the lowest allowed signal quality for the wireless channel to initiate a call.
SIM Pincode	-	
SIM PIN	<i>Numeric</i>	Should your SIM card get locked out, your provider can give you the PIN code for the SIM on this channel.
SIM PUK	<i>Numeric</i>	Should your SIM card get locked out, your provider can give you the PUK code for the SIM on this channel.
Modem Setup	-	Five areas are available for initialization strings to be sent to the internal wireless modules.
Copy Channel (button)	-	Will copy all settings from the current channel to the other available channels.
Default (button)	-	Resets parameters back to their factory default settings.

Call Signaling

The MultiVOIP GSM uses SIP call signaling.

SIP

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

SIP Parameters

Signaling Port :

Use SIP Proxy

Allow Incoming Calls Through SIP Proxy Only

SIP Proxy Parameters

	Proxy Domain Name / IPAddress	Port Number
Primary Proxy	<input type="text"/>	<input type="text" value="5060"/>
Alternate Proxy 1	<input type="text"/>	<input type="text" value="5060"/>
Alternate Proxy 2	<input type="text"/>	<input type="text" value="5060"/>

Append SIP Proxy Domain Name in User ID

Default Subscriber

Default Username

Default Password

Re-Registration Time secs

Proxy Polling Interval secs

TTL Value secs

SIP call signaling

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

Regional

The Regional Parameters are used to set the phone signaling tones and cadences. For the country selected, the standard set of frequency pairs will be listed for dial tone, busy tone, 'unobtainable' tone (fast busy or trunk busy), ring tone, and other, more specialized tones. If you need settings that are not available, the Custom selection will let you set the tones to what is necessary. The Regional Parameters fields are described in the table below.

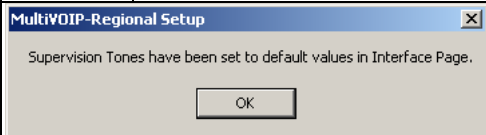
Regional Parameters

Country/Region :

Standard Tones

Type	Frequency1	Frequency2	Cadence(secs)On/Off	Gain
DialTone	350	440	0.000/0.000/0.000/0.000	-16
RingTone	480	440	2.000/4.000/2.000/4.000	-16
BusyTone	480	620	0.500/0.500/0.500/0.500	-16
UnobtainableTone	480	620	0.000/0.000/0.000/0.000	-16
Survivability DialTone	650	650	0.000/0.000/0.000/0.000	-16
ReorderTone	480	620	0.250/0.250/0.000/0.000	-16
InterceptTone	440	620	0.000/0.000/0.000/0.000	-8

Regional parameters

"Regional Parameter" Definitions		
Field Name	Values	Description
Country/Region	USA, Japan, UK, Custom	<p>Name of a country or region that uses a certain set of tone pairs for dial tone, ring tone, busy tone, unobtainable tone (fast busy tone), survivability tone (tone heard briefly, 2 seconds, after going off hook denoting survivable mode of VOIP unit), re-order tone (a tone pattern indicating the need for the user to hang up the phone), and intercept tone (a tone that warns a party that has gone off hook but has not begun dialing, within a prescribed time, that an automatic emergency or attendant number will be called; the automatic call can be used to direct an attendant's attention to a disabled or distressed caller, allowing an appropriate response to be made).</p> <p>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.</p> <p>Note 1: Intercept tone is applicable only when the FXS telephony interface has been chosen in the Interface screen and when the AutoCall / OffHook Alert field is set to OffHook Alert in the Voice/Fax Parameters screen. The time allowed for dialing before the automatic calling process begins is set in the OffHook Alert Timer field of the Voice/Fax Parameters screen.</p>
Advisory screen		This message screen appears whenever the Country field is changed. It informs the operator that, upon change of the Country field value, all User Defined Tones will be deleted.
Standard Tones fields		
Type column	dial tone, ring tone, busy tone, unobtainable tone, 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 GSM outputs as audio to the wireless 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 GSM outputs as audio to the wireless port. Default: -16dB
Cadence (ms) On/Off	n/n/n/n four integer time values in milliseconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable (fast busy), dial tone ("0" indicates continuous tone), survivability, and re-order. Default values differ for different countries/regions. Although most cadences have only two parts (an "on" duration and an "off" duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.
Custom (button)	Only available when Country/Region is set to "Custom"	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.

Setting Custom Tones and Cadences (optional). The Regional Parameters dialog box has a secondary dialog box that allows you to customize DTMF tone pairs to create unique ring-tones, dial-tones, busy-tones or “unobtainable” tones or “re-order” tones or “survivability” tones for your system. This screen allows the user to specify tone-pair attributes that are not found in any of the standard national/regional telephony toning schemes. To access this customization feature, click on the **Custom** button on the **Regional Parameters** screen. The “Custom” button is active only when “Custom” is selected in the **Country/Region** field.

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.
Frequency 1	Frequency in Hertz	Lower frequency of pair.
Frequency 2	Frequency in Hertz	Higher frequency of pair.
Gain 1	+3dB to -31dB and “mute” setting	Amplification factor of lower frequency of pair. This figure describes amplification that the MultiVOIP GSM applies to outbound tones entering the MultiVOIP GSM at the input port. Default: -16dB
Gain 2	+3dB to -31dB and “mute” setting	Amplification factor of higher frequency of pair. This figure describes amplification that the MultiVOIP GSM applies to outbound tones entering the MultiVOIP GSM at the input port. Default: -16dB
Cadence 1	integer time value in milliseconds; 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.
Cadence 2	duration in milliseconds	Cadence 2 is duration of first “off” period in signaling cadence.
Cadence 3	duration in milliseconds	Cadence 3 is duration of second “on” period in signaling cadence.
Cadence 4	duration in milliseconds	Cadence 4 is duration of second “off” period in the signaling cadence.

SMTP

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

Email Address for VOIP (for email call log reporting)

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

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

MultiVOIP GSM as Email Sender. When SMTP is used, the MultiVOIP GSM 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 GSM 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 GSM 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 GSM 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

“SMTP Parameters” Definitions		
Field Name	Values	Description
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select “SMTP” in the Logs screen.
Requires Authentication	Y/N	If this checkbox is checked, the MultiVOIP GSM will send Authentication information to the SMTP server. The authentication information indicates whether or not the email sender has permission to use the SMTP server.
Login Name	<i>alpha-numeric</i>	This is the User Name for the MultiVOIP GSM unit’s email account.
Password	<i>alpha-numeric</i>	Login password for MultiVOIP GSM unit’s email account.
Mail Server IP Address	<i>n.n.n.n</i>	This is the mail server’s IP address. This mail server must be accessible on the IP network to which the MultiVOIP GSM is connected.
Port Number	25	25 is a standard port number for SMTP.
Mail Type	text or html	Mail type in which log reports will be sent.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP GSM unit.
Reply-To Address	<i>email address</i>	User specified. This email address functions as a source email identifier for the MultiVOIP GSM, 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 GSM (esp. to indicate when log report email was undeliverable or when an error has occurred).
Recipient Address	<i>email address</i>	Email address where log reports are sent.
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>whichever 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, accessed by the *Select Fields* button, that allows you to customize email logging. The MultiVOIP GSM software logs data about many aspects of the call traffic going through the MultiVOIP GSM. The Custom Fields screen lets you pick which aspects will be included in the email log reports.

“Custom Fields” Definitions

Field	Description
Select All	Log report to include all fields shown.
Channel Number	Channel carrying call.
Duration	Length of call.
Packets Sent	Total packets sent in call.
Bytes Sent	Total bytes sent in call.
Packets Lost	Packets lost in call.
Outbound Digits Received	The DTMF dialing digits received by this gateway from the remote gateway presuming that DTMF is set to "Out of Band."
Call Status	Successful or unsuccessful.
Call Direction	Indicates originating party.
Server Details	The IP address of the traffic control server (if any) being used will be displayed here if the call is handled through that server.
Disconnect Reason	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause occasioned disconnection (e.g., a technical error or failure). Values are "Normal" and "Local" disconnection.
From Details	
Gateway Name	Originating gateway
IP Address	IP address where call originated.
Description	Identifier of site where call originated.
Options	When selected, log will record Silence Compression and Forward Error Correction by call originator.

Field	Description
Start Date, Time	Date and time the phone call began.
Call Mode	Voice or fax.
Packets Received	Total packets received in call.
Bytes Received	Total bytes received in call.
Coder	Voice Coder /Compression Rate used for call will be listed in log.
Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Type	Indicates the Call Signaling protocol used for the call.
DTMF Capability	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." It can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition.
Outbound Digits Sent	The dialing digits sent by this gateway to the remote gateway presuming that DTMF is set to "Out of Band."
To Details	
Gateway Name	Completing or answering gateway
IP Address	IP address where call was completed or answered.
Description	Identifier of site where call was completed or answered.
Options	When selected, log will record Silence Compression and Forward Error Correction by party answering call.

RADIUS

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

RADIUS

Enable Accounting

Server Address : 0 . 0 . 0 . 0

Accounting Port : 1813

Retransmission Interval : 2000 ms

Number of Retransmissions : 3

Shared Secret :

OK

Cancel

Select Attributes

Help

RADIUS Attributes

Select All

Attributes

Channel Number

Duration

Packets Sent

Bytes Sent

Packets Lost

Outbound Digits

Call Status

Start Date, Time

Call Mode

Packets Received

Bytes Received

Codec

Prefix Matched

From Details

Gateway Name

IP Address

Description

Options

To Details

Gateway Name

IP Address

Description

Options

OK

Cancel

Help

RADIUS settings

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

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

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

"RADIUS Attributes" Definitions			
Field	Description	Field	Description
Select All	Log report to include all fields shown.		
Channel Number	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.
Outbound Digits Sent	DTMF digits received by this gateway from remote gateway (if that DTMF set to "Out of Band").	Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Status	Successful or unsuccessful.		
<i>Server Details</i>	The IP address of the traffic control server being used will be displayed here if the call is handled through that server. The Options field refers to server features that might be activated.		
From Details		To Details	
Gateway Name	Originating gateway	Gateway Name	Completing or answering gateway
IP Address	IP address where call originated.	IP Address	IP address where call was completed/answered.
Description	Identifier of where call originated.	Description	Identifier of where call was completed/answered.
Options	When selected, log will record Silence Compression and Forward Error Correction by call originator.	Options	When selected, log will record Silence Compression and Forward Error Correction by party answering call.

Logs/Traces

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

- in the MultiVOIP GSM program (interface), or
- via email (SMTP)

The image shows two screenshots of configuration windows from a software interface.

The top screenshot is titled "Logs". It contains several sections:

- Console message Settings:** A checkbox labeled "Enable Console Messages" is checked. Below it is a "Filters" button.
- Logs:** A checkbox labeled "Turn Off Logs" is unchecked. Below it are two radio buttons: "GUI" (selected) and "SMTP".
- SysLog Server:** A checkbox labeled "Enable" is unchecked. Below it are two text input fields: "Server IP address" (containing "0 . 0 . 0 . 0") and "Port Number" (containing "514").
- Online Statistics Update Interval:** A text input field containing "10" followed by "sec".

 On the right side of the "Logs" window are three buttons: "OK", "Cancel", and "Help".

The bottom screenshot is titled "Console Messages Filter Settings". It contains four list boxes and two sets of navigation buttons:

- Trace Off for Functions:** A list box containing "T.38".
- Trace On for Functions:** A list box containing "GateKeeper", "H.323", "H450", "HUNTING", "LOGS", "PDD", "PRI", and "PSTN".
- Trace Off for Channels:** A list box containing "Channel 2".
- Trace On for Channels:** A list box containing "Channel 1".

 Between the "Trace Off for Functions" and "Trace On for Functions" boxes are two buttons: ">>" and "<<". Between the "Trace Off for Channels" and "Trace On for Channels" boxes are two buttons: ">>" and "<<". On the right side of the "Console Messages Filter Settings" window are three buttons: "OK", "Cancel", and "Help".

Logs and Filters screens

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

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

NAT Traversal

Setting the NAT (Network Address Translation) Traversal parameters. STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.

NAT Traversal

Descriptions for **NAT Traversal** screen fields are presented in the table below.

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

Supplementary Services

Supplementary Services features derive from the H.450 standard, - even though the H.450 standard refers only to H.323, Supplementary Services are still applicable to the SIP VOIP protocols.

Of the features implemented under Supplementary Services, three are very closely related: Call Transfer, Call Hold, and Call Waiting.

Call Transfer. Call Transfer allows one party to re-connect the party with whom they have been speaking to a third party. The first party is disconnected when the third party becomes connected. Feature is used by a programmable phone keypad sequence (for example, #*1).

Call Hold. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function. Feature is used by a programmable phone keypad sequence (for example, #*2).

Call Waiting. Call Waiting notifies an engaged caller of an incoming call and allows them to receive a call from a third party while the party with whom they have been speaking is put on hold. Feature is used by a programmable phone keypad sequence (for example, #*3).

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*.

Supplementary Services Parameters

Select Channel: Channel 1

Call Transfer

Enable

Transfer Sequence: #*1

Call Hold

Enable

Hold Sequence: #*2

Call Waiting

Enable

Retrieve Sequence: #*3

OK

Cancel

Copy Channel

Default

Help

Supplementary Services

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

Supplementary Services Parameter Definitions		
Field Name	Values	Description
Select Channel	1-2	The channel to be configured is selected here.
Call Transfer Enable	Y/N	Select to enable the Call Transfer function in the VOIP unit. This is a "blind" transfer and the sequence of events is as follows: Callers A and B are having a conversation. Caller A wants to put B into contact with C. Caller A dials call transfer sequence. Caller A hears dial tone and dials number for caller C. Caller A gets disconnected while Caller B gets connected to caller C. A brief musical jingle is played for the caller on hold.
Transfer Sequence	Any phone keypad character	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call transfer. The call-transfer sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #).
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.
Default	When this button is clicked, all Supplementary Service parameters are set to their default values.	
Copy Channel	Copies the Supplementary Service attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.	

Save Settings

Save & Reboot

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

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

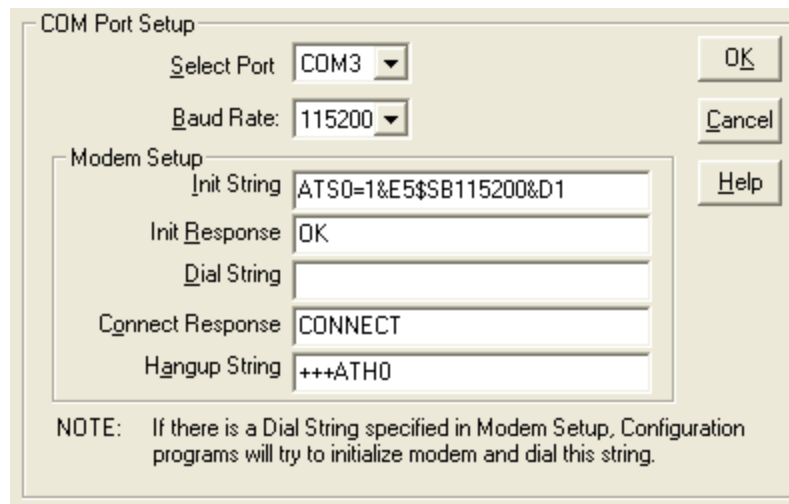
Connection

Settings

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

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

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



COM Port Setup

Select Port: COM3

Baud Rate: 115200

Modem Setup

Init String: ATS0=1&E5\$B115200&D1

Init Response: OK

Dial String:

Connect Response: CONNECT

Hangup String: +++ATH0

NOTE: If there is a Dial String specified in Modem Setup, Configuration programs will try to initialize modem and dial this string.

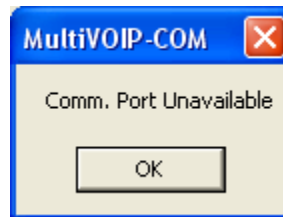
COM port setup

Troubleshooting Software Issues

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

Fixing a COM Port Problem

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

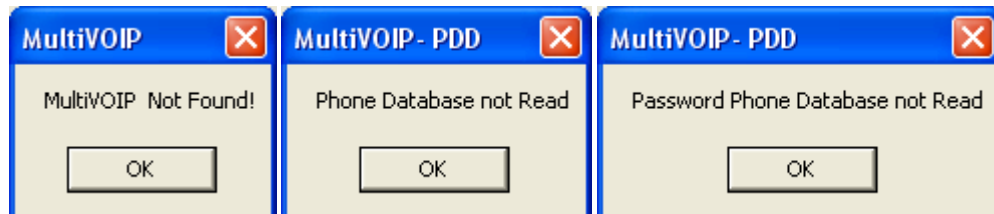


Error pop-up

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

Fixing a Cabling Problem

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



Cabling errors

In this case, the MultiVOIP GSM is disconnected from the PC running the MultiVOIP software – the cause of which may be a bad or unplugged cable, the VOIP is turned off or the COM port is wrong or bad. For instructions on MultiVOIP GSM cable connections, see the Cabling section of Chapter 3.

Chapter 5 – Phone Book Configuration

Introduction

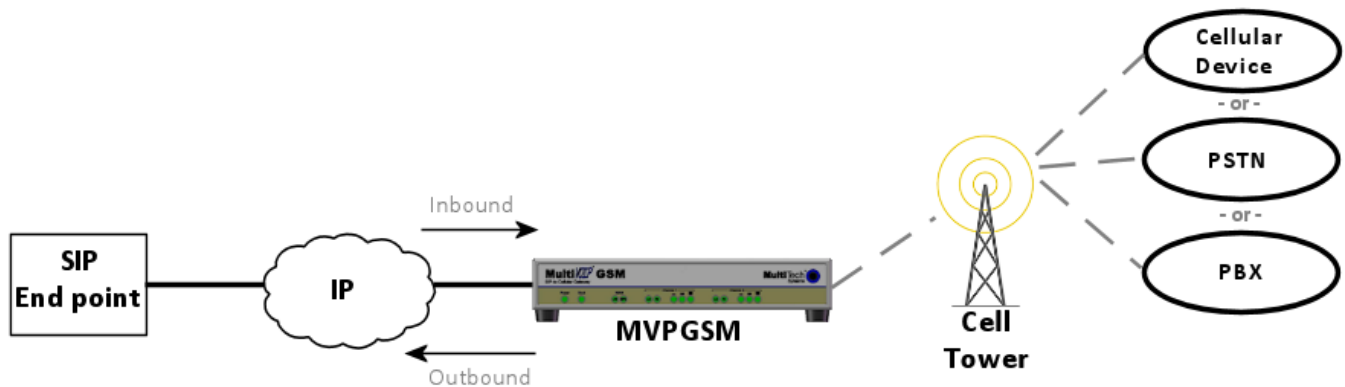
The wireless connectivity of the MVPGSM provides the cost-savings of wireless-to-wireless connections and inexpensive long distance to an existing location. By flagging calls that would connect to a remote site with another MultiVOIP unit or calls that are to be long distance, the MVP GSM can save money by routing those calls through the wireless connection instead of the standard PSTN. By configuring the phone books and setting up these routes, you can ensure that the calls placed are using the least cost possible to connect. The phone books also provide some security by limiting what numbers may be routed through them.

The “Inbound” phonebook is created with the dialing rules required for a call to be received on that VOIP. The “Outbound” phonebook describes dialing rules used to send calls from this VOIP, over IP, to a SIP end point.

Briefly stated, *the MVPGSM’s Outbound phone book lists the phone numbers it will call; its Inbound phone book describes the dialing sequences that can be used to call in and how those calls will be directed.* The phone numbers are not literally “listed” individually, but are, instead, described by rule, such as an area code or beginning digit.

Below you will find some basic configurations and the accompanying phone book setups to aid in the understanding of how the phone books work. These are not necessarily the situations where your MVPGSM would be used, but do serve to illustrate the functionality of the phone books.

Here is a basic setup example of how the MVPGSM can be put to use:



Sample Configurations

- **MVPGSM-to-MVPGSM**
- **MVPGSM as a Stand Alone Unit**
- **Analog VOIP and MVPGSM**
- **Load Balancing**

General Notes

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” 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). For PBX units that have a trunk access code, it’s important to enter the trunk access code (often a “9”) in the “Remove Prefix” field in the Outbound Phonebook.

MVPGSM-to-MVPGSM

The basic setup would consist simply of two VOIPs able to communicate via IP address, with preset phone numbers entered into the Phone Book of each unit, with the Inbound Phone Book of one unit containing the exact same information as the Outbound Phone Book of the other unit and vice-versa. This is the underlying concept behind the Phone Books – they are the reverse of each other so that they match information to accomplish the connections. This is more applicable to analog VOIP units than the wireless, but it helps with understanding the purpose of the Phone Book sections.

Example 1: The *Local VOIP* has channel 1 populated with a SIM using the phone number 612-555-1234 and the *Remote VOIP* has 763-555-4321 as the phone number for the SIM on channel 1. By populating the Phone Book of each unit with the information from the other, calls can be changed as to appear local to the receiving unit. This is the most basic setup and illustrates the concept behind the Phone Book entries and standard VOIP operation. This direct connection then takes advantage of the wireless connection and the calls are not charged as long-distance.

Example 2: Multi-Tech VOIP units can also be set to Auto Call, in which case, either or both sides can be set to automatically dial the other VOIP (‘612-555-1234’ or ‘763-555-4321’) when a phone number matches the entry set to Auto Call, the MVPGSM will automatically dial the corresponding phone number of the other VOIP unit.

MVPGSM as a Standalone Unit

Operating as a standalone unit (that is, with no corresponding VOIP unit on the receiving end), the MVPGSM can be used as a SIP gateway to handle all long distance calls and any other call that would be less expensive if the connection is wireless. In this case, the phone system call control module (e.g. an IP PBX) would determine the nature of the call being placed and route it to the proper gateway. Incoming calls would be automatically routed to the voice mail system that would then provide options for the caller to select from, e.g. a voice menu system that informs the caller to press “0” for the operator or simply enter the digits of the extension they wish to reach.

Example 3: Your company has an IP PBX system with an analog VOIP unit for local calls and direct calls to the office in another city and an MVPGSM that will handle all long distance calls. With this setup, a call control module handles the call routing – when an extension requests an outside line (by dialing a “9” first, for example) the call control module then watches what the first digit after the 9 is. If that first digit is anything other than one of the local area codes, the call is routed through the MVPGSM to take advantage of the cost savings provided by the wireless network. If the first digit is a local call, it is sent through the standard PSTN. The analog VOIP could also be setup to connect with a remote office VOIP as well. In this case then, there would be another access digit (an “8” for example) to have an extension use the analog VOIP and connect to the remote location as if it were a local call.

Analog VOIP and MVPGSM

The Any Number setup entails determining a series of numbers that the VOIP unit will recognize as needing to be routed to the remote VOIP.

Example 4: An analog VOIP and an MVPGSM are connected to the same network. Calls that are local or on the PBX are routed by the analog VOIP, while calls that are dialed out of the local area are sent to the MVPGSM to use the wireless network and save money. In this case, the MVPGSM is the SIP gateway to the wireless network. Incoming calls to the MVPGSM from an external source will receive the auto attendant and from this, external calls can access the extensions attached to the analog VOIP.

Load Balancing

The MVPGSM uses round robin load balancing to ensure that one or more SIMs are not used excessively while others go largely ignored during times of light traffic. When Hunting is enabled, load balancing is done automatically.

Example 5: Your MVPGSM has both channels populated with Pay-as-you-go SIM cards. You do not want the Channel 1 SIM to carry the majority of the traffic while channel 2 sits idle waiting for channel 1 to be busy so that it is used.

Example Inbound Phone Books

The basic purpose of the Inbound Phone Book is to create rules for routing incoming IP calls. Below you will find the Inbound Phone Book settings for the examples that were detailed in the sample Phone Book Configurations above.

Example 1: Local VOIP has 763-555-4321 in the Inbound Phone Book while the Remote VOIP has 612-555-1234 in the Inbound Phone Book. A user at the Local VOIP can dial 612-555-1234 and it will ring the Remote VOIP, a user at the Remote VOIP can dial 763-555-4321 and it will ring the Local VOIP. Both calls are treated as local area code calls.

Example 2: Local VOIP has 763-555-4321 in the Inbound Phone Book while the Remote VOIP has 612-555-1234 in the Inbound Phone Book. A user at the Local analog VOIP can pick up an attached phone and it will ring the Remote GSM VOIP. The call is treated as a local area code call. *The Auto-Call feature is set in the Voice Parameters section.*

Example 3: The Inbound Phone Book for this setup will be set to Any Number for all channels, allowing all digits and relying on the call control module to handle the routing.

Example 4: The outbound number on the analog side matches the inbound number on the MVPGSM. The Inbound Phone Book of the analog VOIP will list the phone number of the MVPGSM.

Example 5: Your Inbound phone book will have one entry that has “Hunting” selected for the ‘Channel Number’ selection box. The other fields will be populated per your setup, but the ‘Any Number’ selection is used in this example to allow any call, ***all other entries here must have ‘Hunting’ enabled as well to use the load balancing feature.***

Example Outbound Phone Books

The Outbound Phone Book is the director of where calls will be routed – whether that is to a specific phone number (Any Number) or to the IP address of a remote VOIP unit for resending as a local call or out to the wireless network to take advantage of wireless plan savings. Below you will find the Outbound Phone Book settings for the examples that were detailed in the sample Phone Book Configurations above.

Example 1: Local VOIP has 612-555-1234 in the Outbound Phone Book while the Remote VOIP has 763-555-4321 in the Outbound Phone Book a. A user at the Local VOIP can pick up an attached phone and dial 763-555-4321 and it will ring the Remote VOIP, a user at the Remote VOIP can pick up an attached line and dial 612-555-1234 and it will ring the Local VOIP. Both calls are treated as local area code calls.

Example 2: Local VOIP has 612-555-1234 in the Outbound Phone Book while the Remote VOIP has 763-555-4321 in the Outbound Phone Book a. A user at the Local VOIP can pick up an attached phone and it will ring the Remote VOIP, a user at the Remote VOIP can pick up an attached line and it will ring the Local VOIP. Both calls are treated as local area code calls.

The Auto-Call feature is set in the Voice Parameters section.

Example 3: The Outbound Phone Book is set to the destination pattern number of the auto attendant. In this example the MVPGSM would also be setup to use a proxy (the call control module).

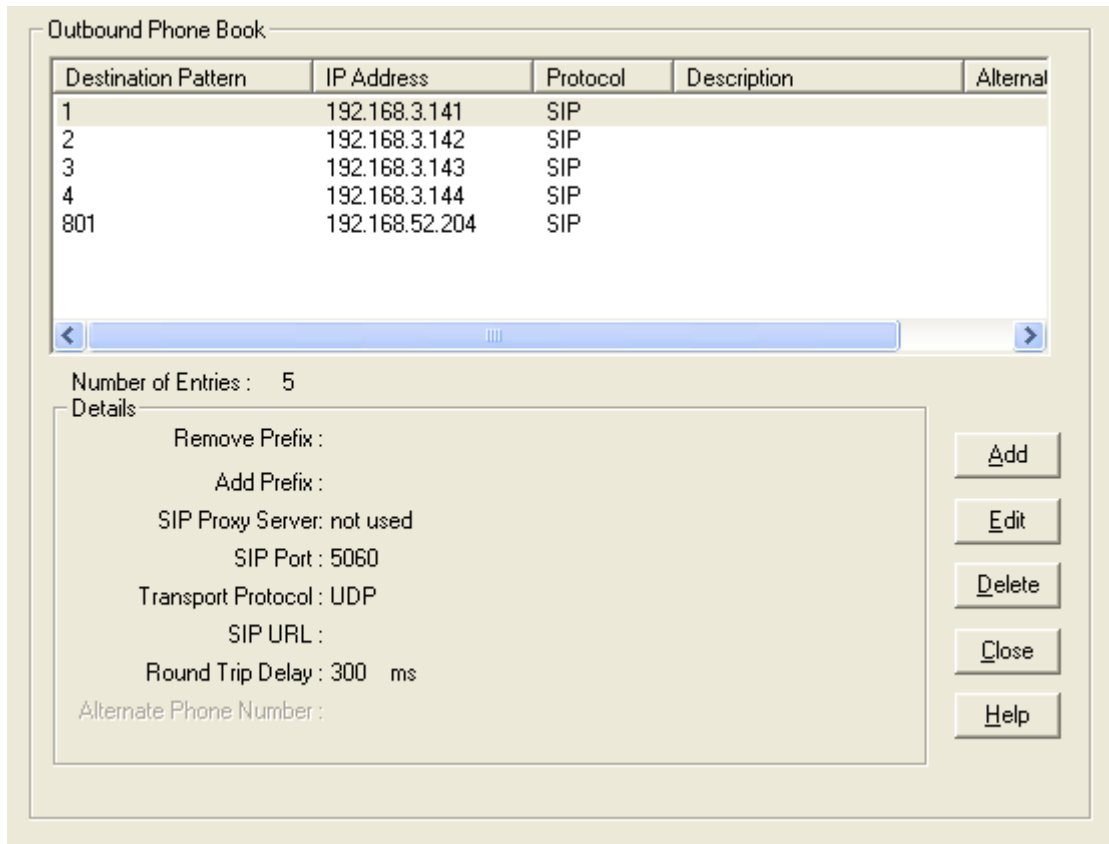
Example 4: The Outbound Phone Book of the MVPGSM will have the phone number of the analog VOIP routed to the IP address of the analog VOIP. The Outbound Phone Book of the analog VOIP will have the phone number of the MVPGSM.

Example 5: The Outbound phone book in this example is simply going to be set at 'Any Number' as well – the purpose is to have load balancing occur between the channels and allow all traffic to pass through the MVPGSM in a round robin manner to avoid the channel one SIM using more minutes than the other channel SIMs.

Phone Book Descriptions

Outbound Phone Book/List Entries

Fields in the “Details” section that are grayed out are not used by the SIP protocol.



Outbound Phone Book

Add/Edit Outbound Phone Book

Add/Edit Outbound Phone Book

Phone Number Details

Accept Any Number

Destination Pattern : Any Number

Total Digits : 0

Remove Prefix : |

Add Prefix :

IP Address : 192 . 168 . 3 . 141

Description :

Protocol Type

SIP

SIP

Use Proxy

Transport Protocol

ICP UDP

SIP Port Number: 5060

SIP URL:

OK

Cancel

Help

Advanced

Add/Edit screen

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

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

Add/Edit Outbound Phone Book: Field Definitions		
Field Name	Values	Description
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Destination Pattern field. The Any Number feature works differently depending on whether or not an external routing device is used (Proxy for SIP protocol). When no external routing device is used. If Any Number is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the IP Address in the Add/Edit Outbound Phone Book screen. "Any Number" can be used in addition to one or more Destination Patterns. When external routing device is used. If Any Number is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the external routing device used (Proxy for SIP protocol). The IP Address of the external routing device must be set in the Phone Book Configuration screen.
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be routed to a SIP end point on the IP network. This is the phone number or beginning portion of the phone number for calls that are to be routed to the IP address listed below.
Total Digits	as needed	Number of digits the phone user must dial to reach specified destination. <i>This field not used in North America</i>
Remove Prefix	dialed digits	Portion of dialed number to be removed before remaining digits are sent to their destination.
Add Prefix	dialed digits	Digits to be added before routing the call to the address below.
IP Address	<i>n.n.n.n</i>	The IP address to which the call above will be routed with the destination pattern given.
Description	alpha-numeric	This description will be sent as Caller ID information.
Protocol Type	SIP	Indicates protocol to be used in outbound transmission.

SIP Fields		
Use Proxy	Y/N	Select if proxy server is used.
Transport Protocol	TCP or UDP	VOIP administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other *See RFC 3087 ("Control of Service Context using SIP Request-URI," by the Network Working Group).	The SIP Port Number is a UDP logical port number. The VOIP will "listen" for SIP messages at this logical port. If SIP is used, 5060 is the default, standard or "well known" port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	<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.

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 or does not respond within the timeframe of the Round Trip Delay. 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 SIP end point fails. The user must specify the IP address of the alternate route for each destination pattern entry in the Outbound Phonebook.

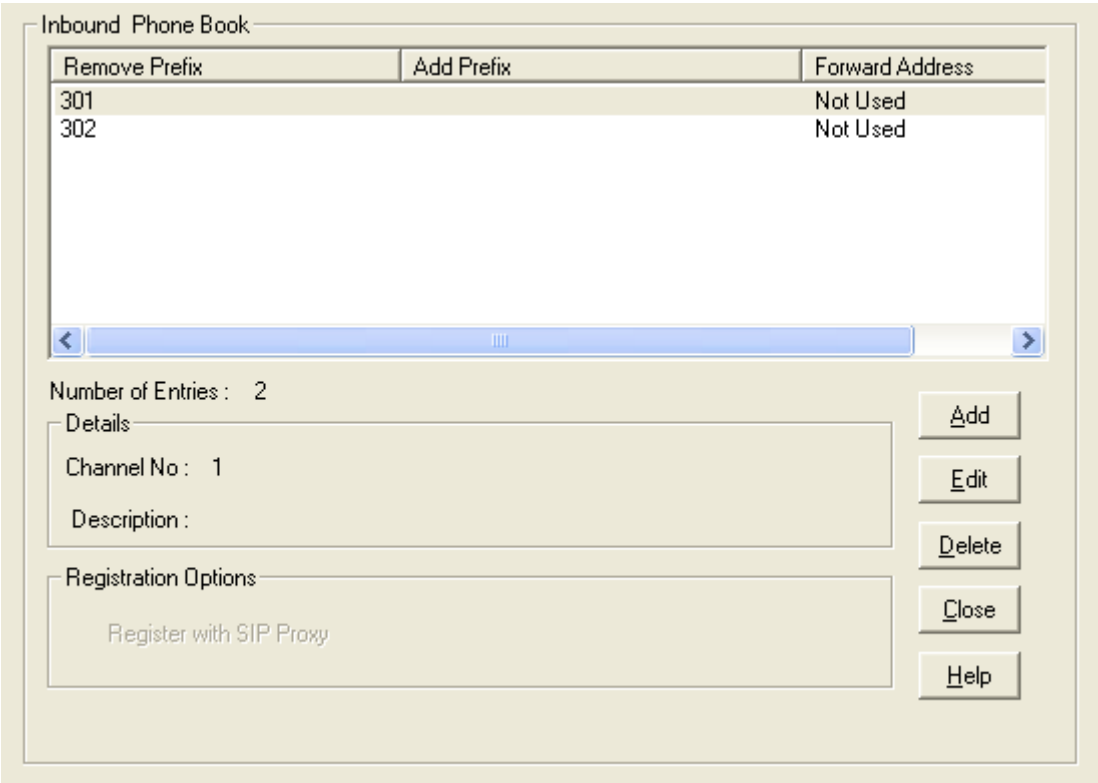
Advanced button

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

The Alternate Routing function facilitates PSTN Failover protection, that is, it allows you to re-route VOIP calls automatically if the VOIP system fails. The MultiVOIP GSM can be programmed to respond to excessive delays in the transmission of voice packets, which the MultiVOIP GSM 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 GSM diverts the call to another of its channels, allowing the call to go back out to the wireless network to reach the remote end using a land line.

Inbound Phone Book/List Entries

The “Details” and “Registration Options” sections will display information based on the setup and protocols chosen.



Inbound phonebook entries

Add/Edit Inbound Phone Book

Add/Edit Inbound Phone Book

Accept Any Number

Remove Prefix :

Add Prefix :

Channel Number :

Description :

Call Forward

Enable

Forward Condition

Unconditional Busy No Response

Forward Destination :

SIP call: Phone # or IP address or IP address:port or
Phone #:IP address:port or SIP URL or Ph#:IP address

Ring Count :

Registration Options

SIP

Register with SIP Proxy

Username

Password

Add/Edit Inbound Phone Book

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

Add/Edit Inbound Phone Book: Field Definitions		
Field Name	Values	Description
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Remove Prefix field. The Any Number feature of the Inbound Phone Book does not work when an external routing device is used (Proxy for SIP protocol). When no external routing device is used. If Any Number is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the VOIP on the channel listed in the Channel Number field. "Any Number" can be used in addition to one or more Prefixes.
Remove Prefix	dialed digits	Portion of digits to be removed from the incoming IP call before initiating a call on the wireless network using the remaining digits.
Add Prefix	dialed digits	Digits to be added to the incoming IP call after digits are removed (if any) but before initiating a call on the wireless network.
Channel Number	channel, or "Hunting"	Wireless VOIP channel which the incoming IP call will be routed to. "Hunting" directs the call to any available channel and enables load balancing to keep one channel from being used more often than another.
Description	--	Optional description field.
Call Forward Parameters		
Enable	Y/N	Click the check-box to enable the call-forwarding feature.
Forward Condition	Unconditional, Busy, No Response	Unconditional. When selected, all calls received will be forwarded. Busy. When selected, calls will be forwarded when station is busy. No Response. When selected, calls will be forwarded if called party does not answer after a specified number of rings, as specified in Ring Count field. Forwarding can be conditioned on both "Busy" and "No Response"
Forward Destination	IP address, phone number, port number, etc	Phone number or IP address to which calls will be directed. For SIP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address, (c) IP address: port number, (d) phone number: IP address: port number, (e) SIP URL, or (f) phone #: IP address.
Ring Count	integer	When "No Response" is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.
Registration Option Parameters	In a SIP VOIP system, gateways can register with the SIP Proxy.	

Authorized User Name and Password for SIP

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

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

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

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

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

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

Phone Book Save and Reboot

When 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 MultiVOIP GSM setup must be done locally using the MultiVOIP GSM program. After the initial configuration is complete, all of the MultiVOIP GSM units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVOIP GSM web interface software program or the MultiVOIP GSM program.

Chapter 6 – Using the Software

Introduction

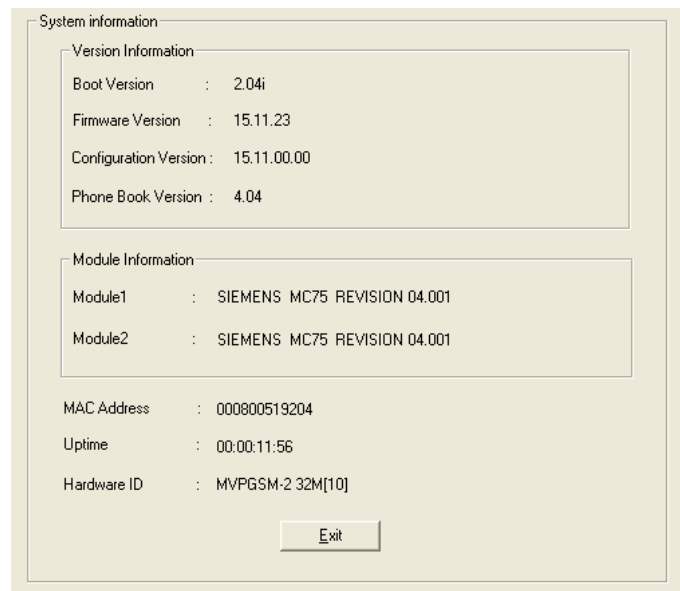
This chapter will primarily cover the day to day operation and maintenance sections of the MultiVOIP GSM software. How to update the firmware and software are also covered here should either be needed. This section will mainly focus on the Statistics section of the configuration software, but there are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

Software Categories Covered in This Chapter

- **System Information**
- **Call Progress**
- **Logs**
- **IP Statistics**
- **Link Management**
- **Registered Gateway Details**
- **Servers**
 - **SIP Proxies**
- **Advanced**
 - **Packetization Time**

System Information screen

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



System information screen

System Information Parameter Definitions		
Field Name	Values	Description
Boot Version	<i>nn.nn</i> alpha-numeric	Indicates the version of the boot code that is running on the VOIP. The boot code version is independent of the software version.
Firmware Version	<i>nn.nn.nn</i> alpha-numeric	Indicates the version of the MultiVOIP GSM firmware.
Configuration Version	<i>nn.nn. nn.nn</i> alpha-numeric	Indicates the version of the MultiVOIP GSM configuration software.
Phone Book Version	<i>nn.nn</i> alpha-numeric	Indicates the version of the MultiVOIP GSM phone book being used.
Module <i>n</i>	Module Info	This will display the model information of the internal wireless modems.
Mac Address	numeric	Denotes the number assigned as the VOIP unit's unique MAC address.
Up Time	days: hours: mm:ss	Indicates how long the VOIP has been running since its last booting.
Hardware ID	alpha-numeric	Indicates version of the MultiVOIP GSM circuit board assembly being used.

The frequency with which the System Information screen is updated is determined by a setting in the Logs/Traces screen ("Online Statistics Updation Interval").

Statistics Section

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

Call Progress

Call Progress Details

Call Progress Details for Channel 1

Call Details	Packet Details
Duration : 00:01:52	Packets Sent : 5,628
Mode : Voice	Packets Received : 5,560
Voice Coder : G.711 U-law @ 64 kbps	Bytes Sent : 968,016
IP Call Type : SIP	Bytes Received : 956,320
IP Call Direction : Incoming	Packets Lost : 59

From -- To Details	
From ----> To : Guilderland	----> 5119
Gateway Name : Guilderland	MVPGSM
IP Address : <input style="width: 100px;" type="text" value="217 . 161 . 64 . 145"/>	----> <input style="width: 100px;" type="text" value="216 . 133 . 69 . 140"/>
Options : None	None

DTMF/Other Details	Supplementary Services Status
Prefix Matched :	Call On Hold : -
Outbound Digits Sent :	Call Waiting : -
Outbound Digits Rcvd : 7637175119	Caller Id :
Server Details :	
DTMF Capability : Out of Band RFC2833	

Call Status : Active

Wireless Module Status : Ready

SIM Registration Status : Registered to home network

Signal Strength : 12 (Medium)

Note

Options : SC - Silence Compression FEC - Forward Error Correction

Signal Strength Range : 20 - 31 (High) , 10 - 19 (Medium) , 4 - 9 (Low) , 0 - 3 (Insufficient) , 99 (No Signal)

Call progress screen

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

Table is continued on next page...

Call Progress Details: Field Definitions (continued)		
Field Name	Values	Description
Supplementary Services Status		
Call on Hold	alphanumeric	Describes held call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers come from Gateway Name field in Phone Book Configuration screen of remote VOIP.
Call Waiting	alphanumeric	Describes waiting call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers come from Gateway Name field in Phone Book Configuration screen of remote VOIP.
Caller ID	“Calling Party + <i>identifier</i> ”; “Alerting Party + <i>identifier</i> ”; “Busy Party + <i>identifier</i> ”; “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.
Call Status fields		
Call Status	hangup, active	Shows condition of current call.
Wireless Module Status	ready, registering	This displays the current status of the wireless module on this channel.
SIM Registration Status	“registered to...”, not registered	This shows the registration status of the module on this channel.
Signal Strength	0-99	0-3 is an insufficient signal for use. 4-9 is a low strength signal that is usable. 10-19 is a medium strength and is a good connection. 20-31 is a high strength and is an excellent connection. 99 means there is no signal at all.
Note section (Options & Signal Strength Ranges)		“SC” stands for Silence Compression. With Silence Compression enabled, the MultiVOIP GSM will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. “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.
Reset (button)	-	Clicking this button will reset the channel including the wireless module.
Disconnect (button)	-	Clicking this button will disconnect the current connection on this channel.

Logs

Logs
Total Number of Logs: 0

Log#	StartDate, Time	Duration	Type	Status	IP Dir...	Mode	From

Call details

Voice coder :	Packets sent :
Disconnect Reason :	Packets recvd :
DTMF Capability :	Packets lost :
Outbound Digits Recvd :	Bytes sent :
Outbound Digits Sent :	Bytes recvd :
Server Details :	

From details

Gateway Name :

IP Address :

Options :

To details

Gateway Name :

IP Address :

Options :

SC - Silence Compression FEC - Forward Error Correction

Supplementary Services Info

Call Transferred To :

Call Forwarded To :

Previous
Next
First
Last
Exit
Help
Delete File

Log statistics screen

The table below describes the fields of the Logs screen.

Logs Screen Details: Field Definitions		
Field Name	Values	Description
Log # column	1 or higher	All calls are assigned an event number in chronological order, with the most recent call having the highest event number.
Start Date, Time column	dd:mm:yyyy hh:mm:ss	The starting time of the call. The date is presented as a day and a month of one or two digits, and a four-digit year, followed by a time-of-day.
Duration column	hh:mm:ss	This describes how long the call lasted in hours, minutes, and seconds.
Type	SIP	Indicates the Call Signaling protocol used for the call (SIP).
Status column	success or failure	Displays the status of the call (whether the call was completed or not).
IP Direction	Incoming, outgoing	Shows call "incoming" or "outgoing" with respect to the gateway.
Mode column	voice or FAX	Shows if the event being described was a voice call or a FAX call.
From column	gateway name	Displays the name of the voice gateway that originates the call.
To column	gateway name	Displays the name of the voice gateway that completes the call.
Special Buttons		
Previous	--	Displays log entry before currently selected one.
Next	--	Displays log entry after currently selected one.
First	--	Displays first log entry
Last	--	Displays last log entry.
Delete File	--	Deletes selected log file.
Call Details		
Voice coder	Coder protocol	The voice coder being used on this call.
Disconnect Reason	"Normal" or "Local"	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause.
DTMF Capability	inband, out of band	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different VOIP protocols. For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition.
Outbound Digits Received	0-9, #, *	These are the digits received from the remote SIP end point that are used to initiate a call out on the wireless network.
Outbound Digits Sent	0-9, #, *	These are the digits received from the wireless network and sent to the remote SIP end point to place a call.
Server Details	<i>n.n.n.n</i>	When the MultiVOIP GSM is operating in the non-direct mode (with proxy in SIP mode), this field shows the IP address of the server that is directing IP phone traffic.
Packets sent	integer value	Number of voice packets sent over the IP network during this call.
Packets received	integer value	Number of voice packets received over the IP network in the course of this call.
Packets lost	integer value	Number of voice packets from this call that were lost after being received from the IP network.
Bytes sent	integer value	Number of bytes of voice sent over the IP network during this call.
Bytes received	integer value	Number of bytes of voice received over the IP network during this call.
FROM Details		
Gateway Name	alphanumeric	Identifier for the VOIP gateway that originated this call.
IP Address	<i>n.n.n.n</i>	IP address for the VOIP gateway that originated this call.
Options	FEC, SC	Displays transmission options used by the gateway originating the call.
TO Details		
Gateway Name	alphanumeric	Identifier for the VOIP gateway that received this call.
IP Address	<i>n.n.n.n</i>	IP address for the VOIP gateway that received this call.
Options		Displays transmission options used by VOIP gateway receiving the call.
Supplementary Services Info		
Call Transferred To	phone number	Number of party called in transfer.
Call Forwarded To	phone number	Number of party called in forwarding.

IP Statistics

The screenshot shows a window titled "IP Statistics". At the top, there is an "IP Address:" label followed by a text input field containing ". . .". Below this are five grouped sections, each with a title and several input fields:

- Total Packets:** Transmitted (0), Received (0)
- UDP Packets:** Transmitted (0), Received (0), Received with Errors (0)
- TCP Packets:** Transmitted (0), Received (0), Retransmitted (0), Received with Errors (0)
- RTP Packets:** Transmitted (0), Received (0), Received with Errors (0)
- RTCP Packets:** Transmitted (0), Received (0), Received with Errors (0)

On the right side of the window, there are three buttons: "Clear", "Exit", and "Help".

IP statistics screen

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

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

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

IP Statistics: Field Definitions		
Field Name	Values	Description
IP Address	<i>n.n.n.n</i>	IP address of the MultiVOIP GSM. For an IP address to be displayed here, the MultiVOIP GSM must have DHCP enabled. Its IP address, in such a case, is assigned by the DHCP server.
"Clear" button	--	Clears packet tallies from memory.
Total Packets		Sum of 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 GSM 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 GSM software.
Received with Errors	integer value	Total number of error-laden packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM 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 GSM 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 GSM 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 GSM 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 GSM 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 GSM 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 GSM software.
RTP Packets		Voice signals are transmitted in Realtime Transport Protocol packets. RTP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP GSM 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 GSM 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 GSM 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 GSM 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 GSM 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 GSM software.

Link Management

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

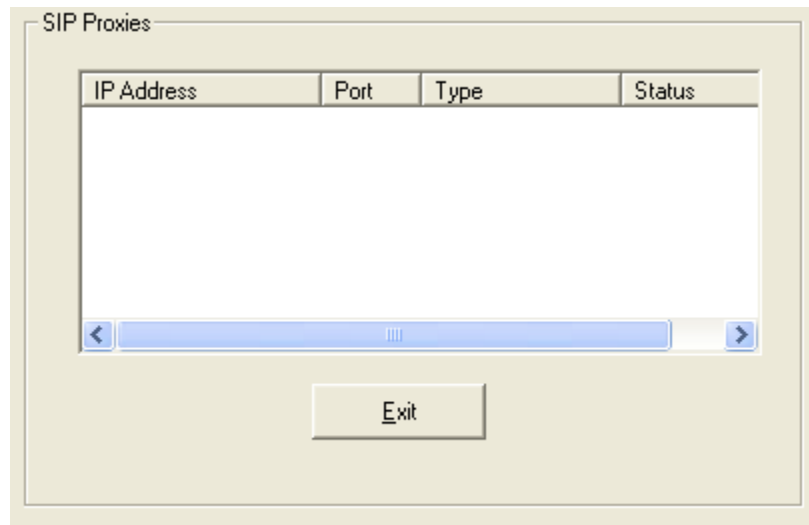
Link management

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

Servers

SIP Proxies

This window lists all the SIP proxy servers configured and the active server with which the system is registered.



SIP proxies

SIP Proxies (Statistics, Servers): Field Definitions		
Field Name	Values	Description
Column Headings		
IP Address	<i>n.n.n.n</i>	The IP address of the SIP proxy by which the MultiVOIP GSM is governed.
Port	<i>port</i>	SIP signaling port used for communication between MultiVOIP GSM unit and the SIP Proxy that governs it.
Type	Primary, Alternate	This field describes the type of gateway the MultiVOIP GSM is defined as with respect to the SIP proxy server.
Status	registered, not registered	The current status of the MultiVOIP GSM gateway with respect to the SIP proxy either registered or unregistered.

Advanced

Packetization Time

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

Packetization time

Packetization rates can be set separately for each channel.

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

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

Once the packetization rate has been set for one channel, it can be copied into other channels by using the Copy Channel button on the Packetization Time screen. Simply click the boxes next to the channels you wish to copy the settings for.

MultiVOIP GSM Program Menu Items

After the MultiVOIP GSM program is installed on the PC, it can be launched from the **Programs** group of the Windows **Start** menu (**Start** | **Programs** | **MultiVOIP GSM x.xx** | ...). In this section, we describe the software functions available on this menu.

Several basic software functions are accessible from the MultiVOIP GSM software menu, as shown below.

MultiVOIP GSM Program Menu	
Menu Selection	Description
Configuration	Select this to enter the Configuration program where values for IP, telephony, and other parameters are set.
Configuration Port Setup	Select this to access the COM Port Setup screen of the MultiVOIP GSM Configuration program.
Date and Time Setup	Select this for access to set calendar/clock used for call logging.
Download ATTable	Select this to download the current set of SIP protocols. This is generally only needed in conjunction with a firmware update.
Download Factory Defaults	Select this to return the configuration parameters to the original factory values.
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 GSM.
Set Password	Select this to create a password for access to the MultiVOIP GSM software programs (Program group commands, Windows interface, web browser interface, & FTP server). Only the FTP Server function <i>requires</i> a password for access. The FTP Server function also requires that a username be set along with the password.
Uninstall	Select this to uninstall the MultiVOIP GSM software (most, but not all components are removed from computer when this command is used).
Upgrade Software	Loads firmware and settings from the controller PC to the MultiVOIP GSM unit. User can choose whether to load Factory Default Settings or Current Configuration settings.

“Downloading” here refers to transferring program files from the PC to the nonvolatile “flash” memory of the MultiVOIP GSM. Such transfers are made via the PC’s serial port. This can be understood as a “download” from the perspective of the MultiVOIP GSM unit.

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

Setting and Downloading User Defaults

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

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



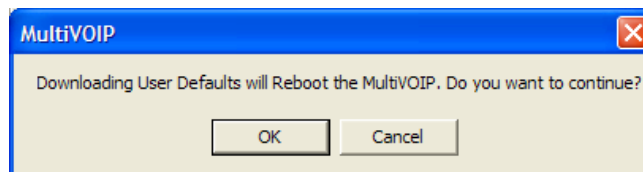
Save & Reboot

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

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

3. To download the user defaults, go to **Start | Programs | MultiVOIP GSM x.xx | Download User Defaults**.

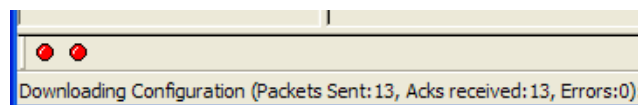
4. A confirmation screen will appear indicating that this action will entail rebooting the MultiVOIP GSM.



Confirmation screen

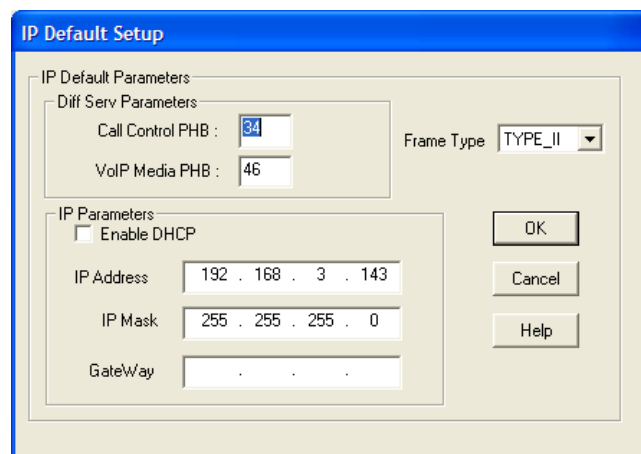
Click **OK**.

5. Progress bars will appear during the file transfer process.



Progress bars

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



Dialog screen

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

Setting a Password

Windows Interface

After a user name has been designated and a password has been set, that password is required to gain access to any functionality of the MultiVOIP GSM 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 GSM via the web browser interface.

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

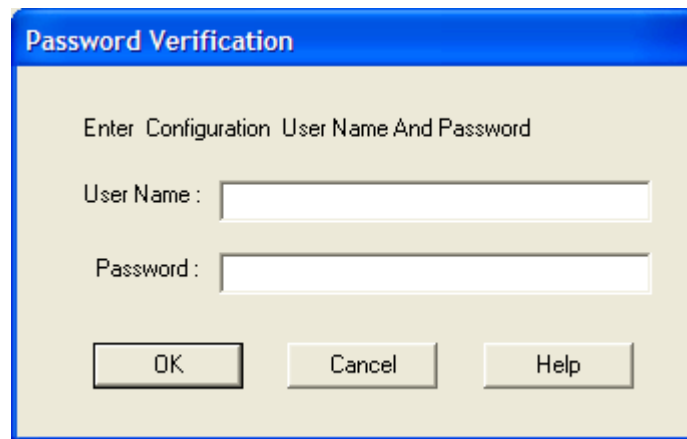
1. The MultiVOIP GSM configuration program must be off when invoking the **Set Password** command. If it is on, the command will not work.
2. To use the **Set Password** command, go to **Start | Programs | MultiVOIP GSM x.xx | Set Password**.
3. You will be prompted to confirm that you want to establish a password, which will entail rebooting the MultiVOIP GSM (which is done automatically).
Click **OK** to proceed with establishing a password.
4. The **Password** screen will appear. If you intend to use the FTP Server function that is built into the MultiVOIP GSM, enter a user name. (A User Name is not needed to access the local Windows interface, the web browser interface, or the commands in the **Program** group.) Type your password in the **Password** field of the **Password** screen. Type this same password again in the **Confirm Password** field to verify the password you have chosen.

NOTE: Be sure to write down your password in a convenient but secure place. If the password is forgotten, contact Multi-Tech Technical Support for advice.

Password screen

Click **OK**.

5. A message will appear indicating that a password has been set successfully.
After the password has been set successfully, the MultiVOIP GSM 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 interface and any part of the MultiVOIP GSM software listed in the **Program** group menu. User Name and Password are both needed for access to the FTP Server residing in the MultiVOIP GSM.



Password verification

When the MVPGSM program asks for the password at launch of program, if **CANCEL** is selected, the program will simply shut down.

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

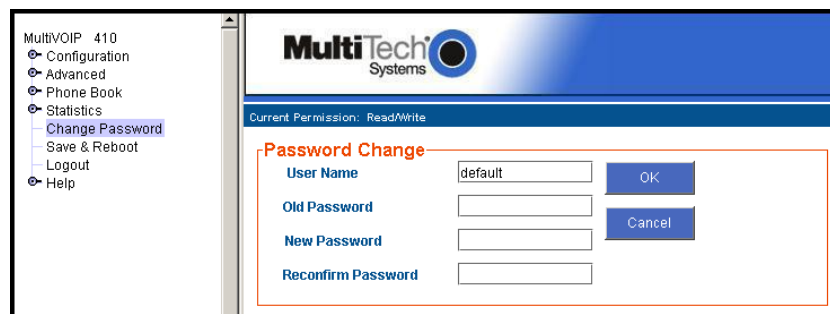


Invalid password

Web Browser Interface

Setting a password is optional when using the MultiVOIP GSM web browser interface. Only one password can be assigned and it works for all MultiVOIP GSM software functions (Windows interface, web browser interface, 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 GSM web browser interface.

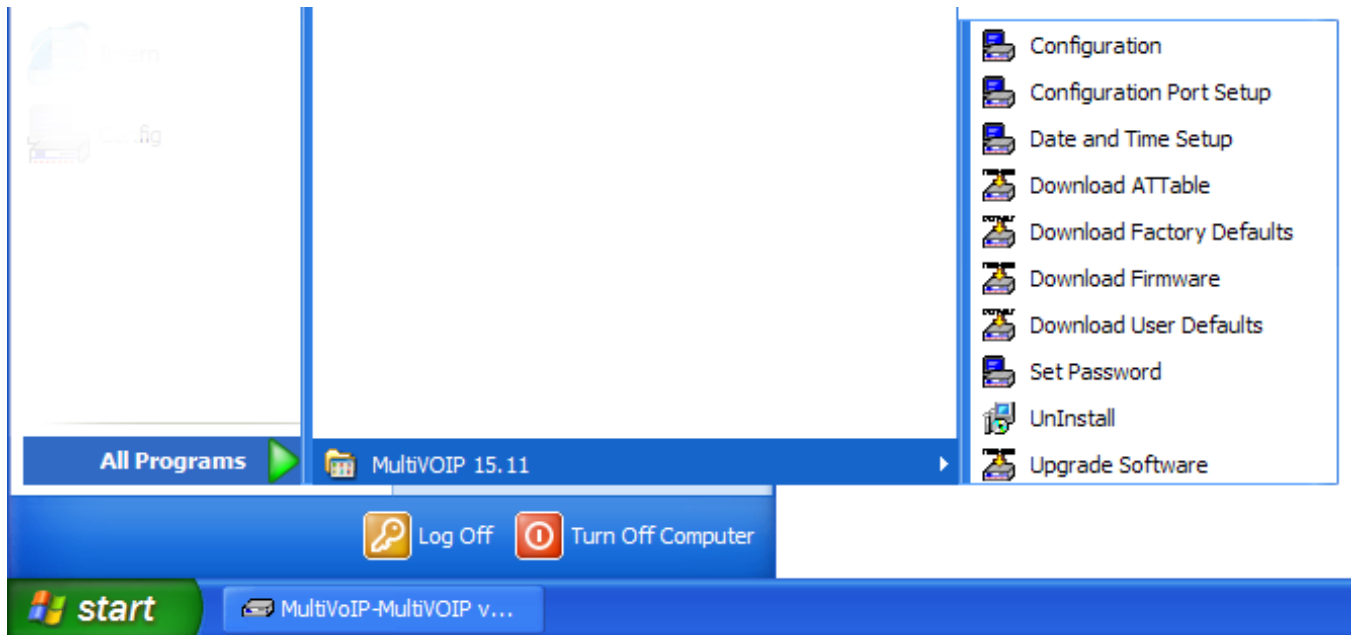
NOTE: Record your user name and password in a safe place. If the password is lost, forgotten, or irretrievable, the user must contact Multi-Tech Tech Support in order to resume use of the MultiVOIP GSM web browser interface.



Web interface password change

Upgrading Software

As noted earlier, the Upgrade Software command transfers from the controller PC to the MultiVOIP GSM unit. The settings can be either Factory Default Settings or Current Configuration Settings.



Upgrade software path

FTP Server File Transfers (“Downloads”)

Multi-Tech has built an FTP server into the MultiVOIP GSM 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, Netscape, or Firefox, used in conjunction with Windows Explorer).

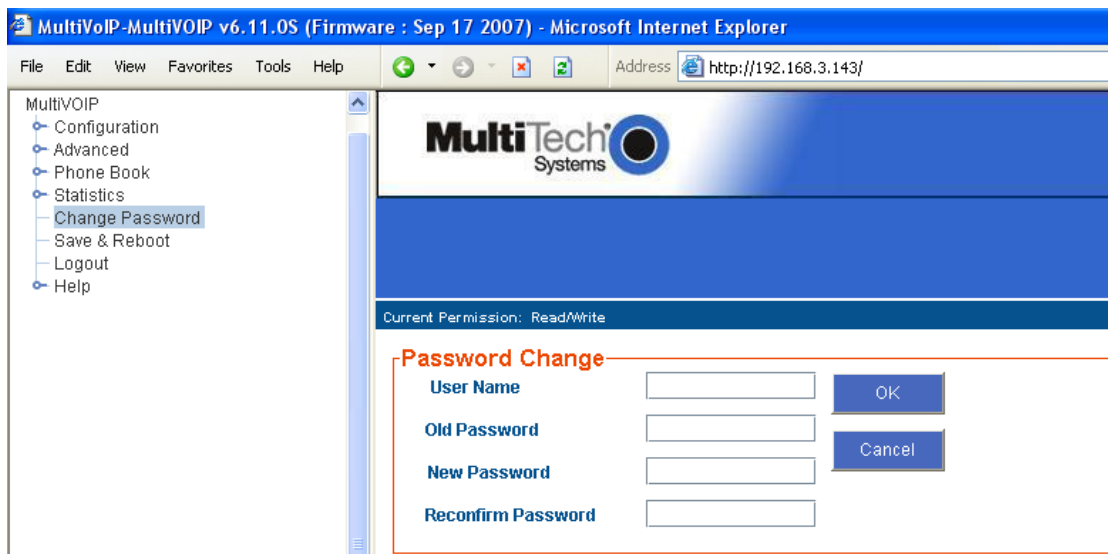
The terminology of “downloads” and “uploads” gets a bit confusing in this context. File transfers from a client to a server are typically considered “uploads.” File transfers from a large repository of data to machines with less data capacity are considered “downloads.” In this case, these metaphors are contradictory: the FTP server is actually housed in the MultiVOIP GSM 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, default configuration parameters, and phonebook data for the MultiVOIP GSM 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 GSM 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 GSM, follow these directions.

1. **Establish Network Connection and IP Addresses.** Both the controller PC and the MultiVOIP GSM unit(s) must be connected to the same IP network. An IP address must be assigned for each.
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.)



Change password

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

3. **Install FTP Client Program or Use Substitute.** You *should* install an FTP client program on the controller PC. FTP file transfers can be done using a web browser (e.g., Firefox or Internet Explorer) in conjunction with a local Windows browser (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 Multi-Tech does not provide an FTP client program with the MultiVOIP GSM software or endorse any particular FTP client program, we remind our readers that adequate FTP programs are readily available under retail, shareware and freeware licenses. (Read and observe any End-User License Agreement carefully.) Two examples of this are the “WSFTP” client and the “SmartFTP” client, with the former having an essentially text-based interface and the latter having a more graphically oriented interface, as of this writing. User preferences will vary.

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

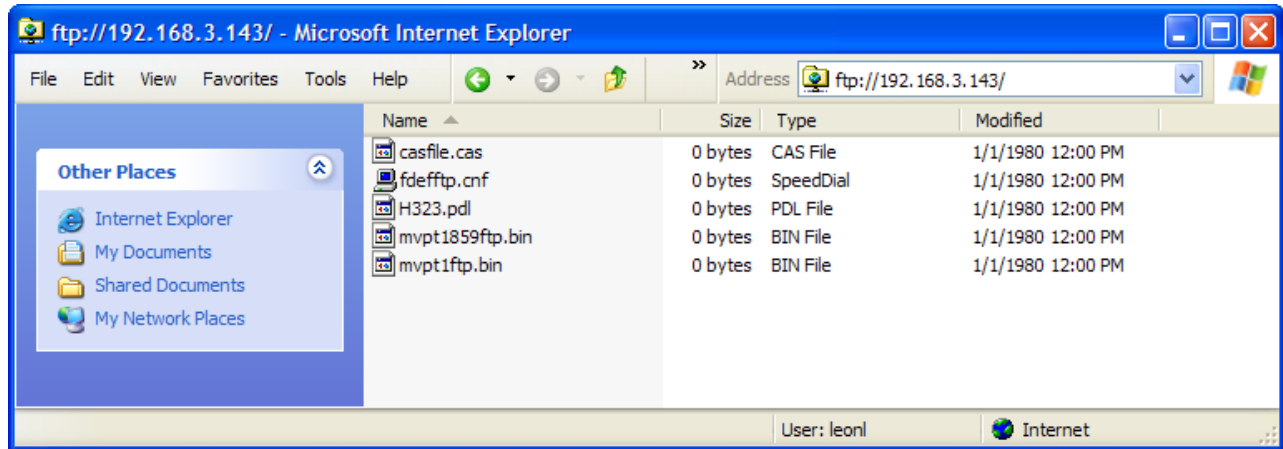
The screenshot shows the 'Ethernet / IP Parameters' configuration window. It is divided into two main sections: 'Ethernet Parameters' and 'IP Parameters'.
 In the 'Ethernet Parameters' section, 'Packet Prioritization (802.1p)' is checked, and 'Frame Type' is set to 'TYPE-II'. Below this, the '802.1p Parameters' section contains three dropdown menus: 'Call Control' (6-Voice), 'VoIP Media' (3-Excellent Effort), and 'Others' (0-Best Effort). A 'VLAN ID' field is set to '1'.
 In the 'IP Parameters' section, 'Gateway Name' is 'MultiVoIP', 'Enable DHCP' is unchecked, 'IP Address' is '192 . 168 . 3 . 143', 'IP Mask' is '255 . 255 . 255 . 0', and 'Gateway' is empty. The 'Diff Serv Parameters' section shows 'Call Control PHB' as '34' and 'VoIP Media PHB' as '46'.
 At the bottom, the 'FTP Server' section has a checkbox labeled 'Enable' which is checked and circled in red. To the right of the window are 'OK', 'Cancel', and 'Help' buttons.

Enable FTP server

5. **Identify Files to be Updated.** Determine which files you want to update. See table below.

File Type	File Names	Description
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.
inbound phonebook	InPhBk.tmr	This file updates the inbound phonebook in the MultiVOIP GSM unit.
outbound phonebook	OutPhBk.tmr	This file updates the outbound phonebook in the MultiVOIP GSM unit.

6. **Contact MultiVOIP GSM 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 GSM's FTP Server. If you are using a browser, the address must be preceded by "ftp://" (otherwise you'll reach the web interface within the MultiVOIP GSM unit).



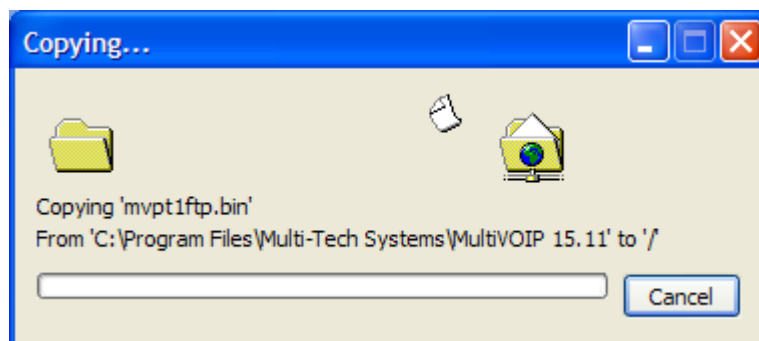
FTP address

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 or with an FTP client program (varies).

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

Download with Web Browser:

- In the local Windows browser, locate the directory holding the MultiVOIP GSM program files. The default location will be C:\Program Files\Multi-Tech Systems\MultiVOIP GSM xxxx yyyy (where x and y represent MultiVOIP model and software version numbers).
- Right-click on the file and select Copy files from Windows Explorer.
- Change focus to the Web browser window (left-clicking on the title bar will accomplish this), then right-click in the browser window showing the file and select Paste.
- You may be asked to confirm the overwriting of files on the MultiVOIP GSM. Do so.
- File transfer between PC and VOIP will look like a transfer within VOIP directories.



Download with FTP Client Program:

- In the local directory browser of the FTP client program, locate the directory holding the MultiVOIP GSM program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP GSM xxxx yyyy (where x and y represent MultiVOIP GSM model numbers and software version numbers).
 - In the FTP client program window, drag-and-drop files from the local browser pane to the pane for the MultiVOIP GSM FTP server. FTP client interface operations vary. In some cases, you can choose between immediate and queued transfer. In some cases, there may be automated capabilities to transfer to multiple destinations with a single command.
9. **Verify Transfer.** The files transferred will appear in the directory of the MultiVOIP GSM. Check to insure that the new file is in place by verifying the file size and date (this is easy if this is the first time you have done this as it will change from a file size of '0' and a date of 01/01/1980).
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 GSM Windows interface.

Web Browser Interface



Web interface main page

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

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

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

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

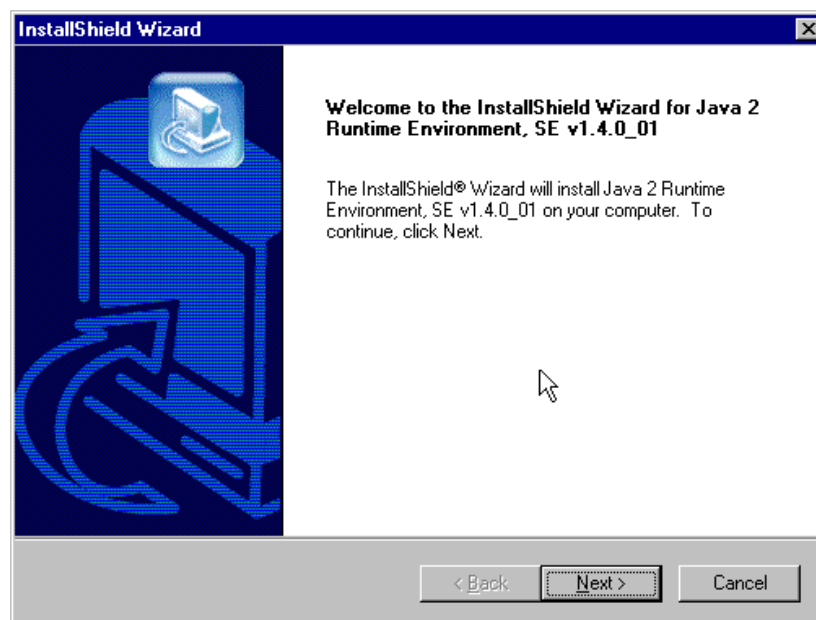
The Windows interface gives access to commands via icons and pull-down menus whereas the web interface does not. The web interface, however, cannot perform logging in the same direct mode done in the Windows interface. However, when the web interface is used, logging can be done by email (SMTP).

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

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

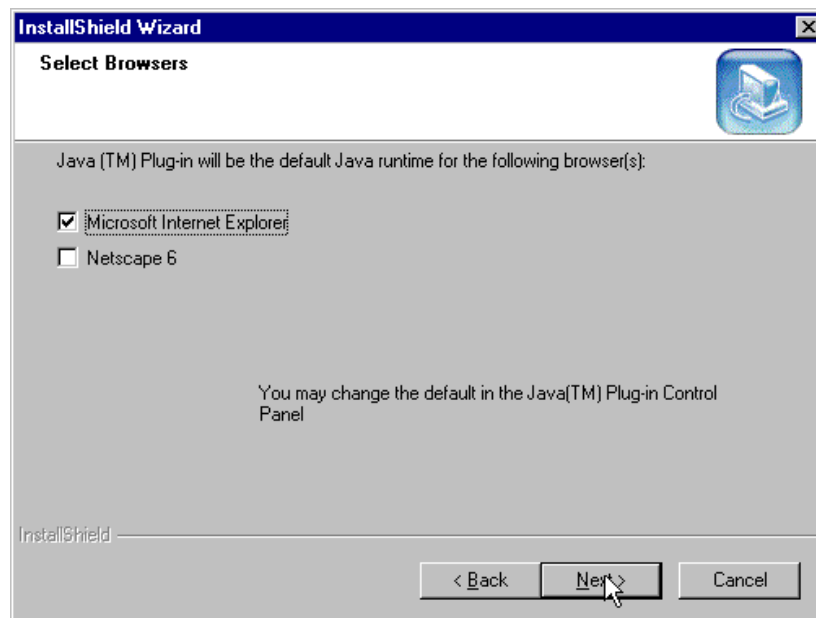
In order to use the web interface, you must also install a Java application program on the controller PC. This Java program is included on the MultiVOIP GSM product CD. Java is needed to support drop-down menus and multiple windows in the web interface.

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



Java install screen

During the installation, you may be asked to specify which browser(s) you'll use in the **Select Browsers** screen.



Browser choice

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

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

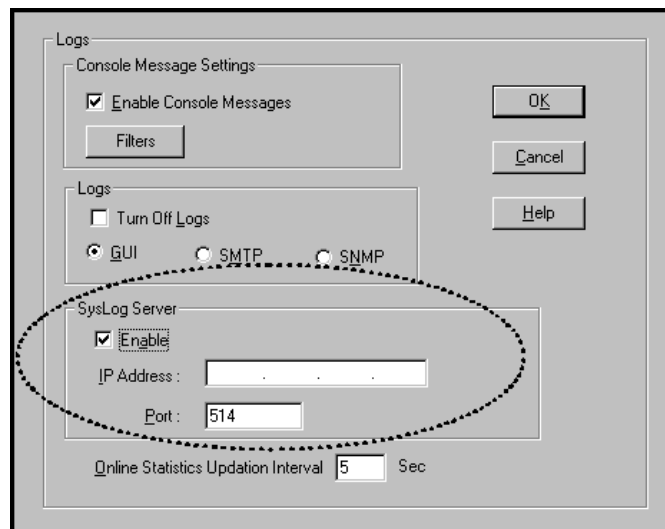
SysLog Server Functions

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

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

Multi-Tech Systems does not endorse any particular SysLog client program. SysLog client programs by qualified providers should suffice for use with MultiVOIP GSM units.

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



Enable SysLog

The IP Address used will be that of the MultiVOIP GSM itself.

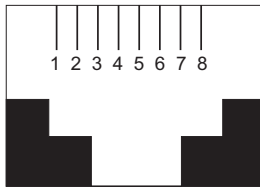
In the **Port** field, entered by default, is the standard (‘well-known’) logical port, 514.

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, 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.).

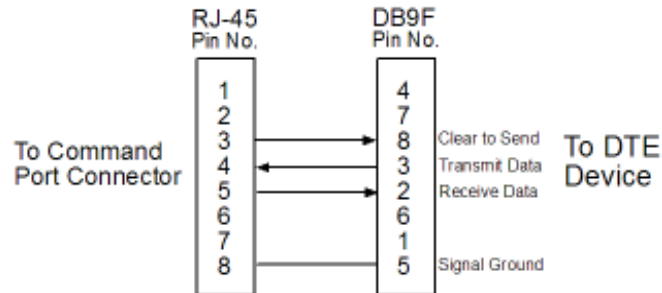
Appendix A – Cable Pin-outs & Ports

Command Cable

RJ-45 Connector



End-to-End Pin Info



RJ-45 connector plugs into Command Port of MultiVOIP GSM.

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

Ethernet Connector

The functions of the individual conductors of the MultiVOIP GSM'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

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 GSM operation are listed below.

Port Number Assignment List

Function	Port Number
telnet	23
tftp	69
snmp	161
snmp tray	162
http	80
ftp (simple)	115
SIP	5060
SysLog	514

Appendix B – Regulatory Information

EMC, Safety, and R&TTE Directive Compliance

The CE mark is affixed to this product to confirm compliance with the following European Community Directives: Council Directive 2004/108/EC of 31 December, 2004 on the approximation of the laws of Member States relating to electromagnetic compatibility,

and

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

and

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

FCC Part 15 Declaration

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 their 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.

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 C – Waste Electrical and Electronic Equipment (WEEE) Statement

July, 2005

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

Instructions for Disposal of WEEE by Users in the European Union

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



Appendix D – C-ROHS HT/TS Substance Concentration

依照中国标准的有毒有害物质信息

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

成分名称	有害/有毒物质/元素					
	铅 (PB)	汞 (Hg)	镉 (CD)	六价铬 (CR6+)	多溴联苯 (PBB)	多溴二苯醚 (PBDE)
印刷电路板	0	0	0	0	0	0
电阻器	X	0	0	0	0	0
电容器	X	0	0	0	0	0
铁氧体磁环	0	0	0	0	0	0
继电器/光学部件	0	0	0	0	0	0
IC	0	0	0	0	0	0
二极管/晶体管	0	0	0	0	0	0
振荡器和晶振	X	0	0	0	0	0
调节器	0	0	0	0	0	0
电压传感器	0	0	0	0	0	0
变压器	0	0	0	0	0	0
扬声器	0	0	0	0	0	0
连接器	0	0	0	0	0	0
LED	0	0	0	0	0	0
螺丝、螺母以及其它五金件	X	0	0	0	0	0
交流-直流电源	0	0	0	0	0	0
软件/文档 CD	0	0	0	0	0	0
手册和纸页	0	0	0	0	0	0
底盘	0	0	0	0	0	0

- X 表示所有使用类似材料的设备中有害/有毒物质的含量水平高于 SJ/Txxx-2006 限量要求。
- 0 表示不含该物质或者该物质的含量水平在上述限量要求之内。

INDEX

Auto Disconnect	32	Logs (Statistics) field definitions.....	70
AutoCall/Offhook.....	32	MultiVOIP GSM Not Found.....	51
Cabling	9	NAT Traversal screen fields.....	47
Call Hold	48	Packet Prioritization 802.1p	27
Call Progress fields.....	67	Packetization rates.....	75
Call Transfer.....	48	Phone Database not Read.....	51
Call Waiting.....	48	RADIUS Screen field definitions.....	44
Coder Parameters fields.....	31	Regional parameter definitions.....	37
Comm. Port Unavailable	51	Saving the MultiVOIP GSM Configuration	50
Creating a User Default Configuration	50	Set Baud Rate.....	50
Custom Tones and Cadences	39	Set Log Reporting Method.....	45
Diff Serv PHB value.....	28	Setting Ethernet/IP parameters	26
DTMF inband	30	Setting password	78
DTMF out of band.....	30	Setting user defaults	77
Dynamic Jitter	32	SIP Call Signaling parameter definitions	36
Email log reports	40	SMTP parameters definitions.....	41
FTP Server function.....	81	Specifications	5
FTP Server, logging out.....	84	STUN clients and servers.....	47
IP Statistics fields	72	Supplementary Services parameter definitions.....	48
LED descriptions	4	Voice/FAX parameter definitions	29
Link Management fields.....	73	Wireless Interface parameters	33

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>