Installation and Operation Manual
for
Talkaphone Voice over IP Interface

VOIP-1-2-4-8

Rev. 7/2/2014

CHAPTER 1

Introduction to Voice over IP Interfaces (VOIP-1, VOIP-2, VOIP-4, and VOIP-8)

The Voice over IP (VoIP) Interface allows all Talkaphone Emergency Phones to be used over an IP data network. The VOIPs integrate seamlessly with existing VoIP phone systems, and support standard VoIP protocols. For sites without existing VoIP systems, two VOIPs can be used in conjunction to send emergency calls over the IP network and then remotely “jump off” onto an existing PBX or PSTN phone network.

Figure 1-1: VOIP-1 Chassis

Figure 1-2: VOIP-2 Chassis

Figure 1-3: VOIP-4/VOIP-8 Chassis

Capacity. Talkaphone’s VOIP-8 model is an eight-channel unit, the model VOIP-4 is a four-channel unit, the model VOIP-2 is a two-channel unit, and the VOIP-1 is a single-channel unit. All of these VoIP units have a 10/100Mbps Ethernet interface and a command port for configuration.

Mounting. Mechanically, the VOIP-4 and VOIP-8 units are designed for a one-high industry-standard EIA 19-inch rack enclosure. By contrast, the VOIP-1 and the VOIP-2 are not rack mountable.

Phone System Transparency. These VOIP-1-2-4-8’s interoperate with a telephone switch or PBX, acting as a switching device that directs voice and fax calls over an IP network. The VOIP-1-2-4-8 units have “phonebooks,” directories that determine to whom calls may be made and the sequences that must be used to complete calls through the VOIP-1-2-4-8. The phonebooks allow the phone user to interact with the VOIP system just as they would with an ordinary PBX or telco switch. When the phonebooks are set, special dialing sequences are minimized or eliminated altogether. Once the call destination is determined, the phonebook settings determine whether the destination VOIP unit must strip off or add dialing digits to make the call appear at its destination to be a local call.

H. 323, SIP, & SPP. Being H.323 compatible, the VOIP-1-2-4-8 units can place calls to telephone equipment at remote IP network locations that also contain H.323 compatible voice-over-IP gateways. It will interface with H.323 software and H.323 gatekeeper units. H.323 specifications also bring to VoIP telephony many special features common to conventional telephony. H.323 features of this kind that have been implemented into the VOIP-1-2-4-8 units include Call Hold, Call Waiting, Call Identification, Call Forwarding (from the H.450 standard), and Call Transfer (H.450.2 from H.323 Version 2). The fourth version of the H.323 standard improves system resource usage (esp. logical port or socket usage) by handling call signaling more compactly and allowing use of the low-overhead UDP protocol instead of the error-correcting TCP protocol where possible.

The VOIP-1-2-4-8 is also SIP-compatible. (“SIP” means Session Initiation Protocol.) However, H.450 Supplementary Services features can be used under H.323 only and not under SIP. It can register with SIP proxy servers and call managers that are 100% SIP-compliant.

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

Data Compression & Quality of Service. The VOIP-1-2-4-8 unit comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

PSTN Failover Feature. The VOIP-2-4-8 can be programmed to divert calls to the PSTN temporarily in case the IP network fails. Enabling this feature will require a dedicated channel, therefore a VOIP-1 does not have the PSTN failover feature.

Management. Configuration and system management can be done locally with the VOIP-1-2-4-8 configuration software via a serial connection. After an IP address has been assigned locally, other configuration can be done remotely using the Web Interface GUI. All of these control software packages are included on the VOIP-1-2-4-8 CD.

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

The primary advantage of the Web GUI is remote access for control and configuration. The controller PC and the VOIP-1-2-4-8 unit itself must both be connected to the same IP network and their IP addresses must be known.

The Windows GUI gives access to commands via icons and pulldown menus, whereas the Web GUI does not. The Web GUI, however cannot perform logging in the same direct mode done in the Windows GUI. However, when the Web GUI is used, logging can be done by e-mail (SMTP).

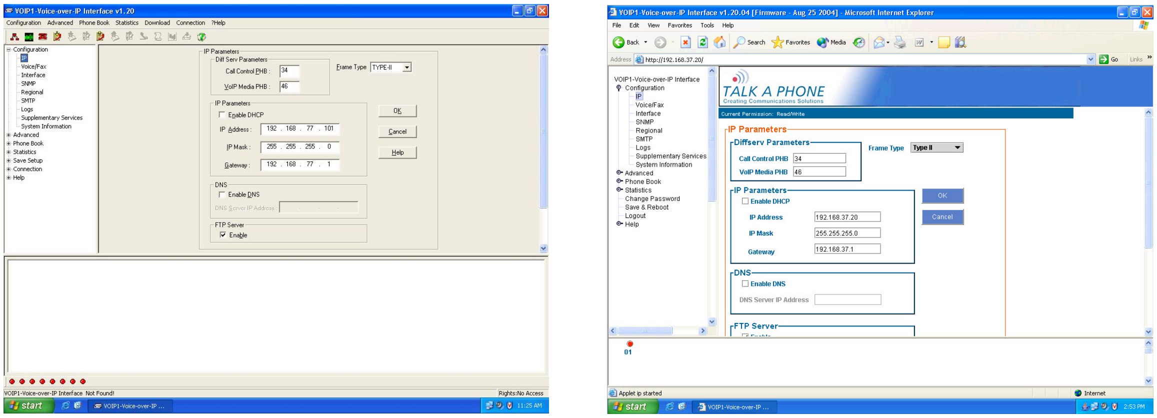


Figure 1-4: VOIP Interface Windows GUI (left) and Web Interface GUI (right)

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

Logging of System Events. The software for the VOIP-1-2-4-8 units has SysLog Server functionality. SysLog is a *de facto* standard for logging events in network communication systems.

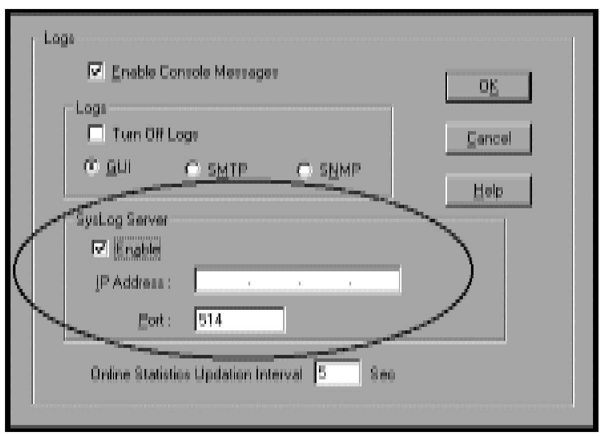


Figure 1-5: Syslog Functionality in VOIP-1-2-4-8 Interface Units

The SysLog Server resides in the VOIP-1-2-4-8 unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a "daemon").

Supplementary Telephony Services. The H.450 standard (an addition to H.323) brings to VoIP telephony more of the premium features found in PSTN and PBX telephony. VOIP-1-2-4-8 units offer five of these H.450 features: Call Transfer, Call Hold, Call Waiting, Call Name Identification (not the same as Caller ID), and Call Forwarding. (The first four features are found in the "Supplementary Services" window; the fifth, Call Forwarding, appears in the Add/Edit Inbound phonebook screen.) Note that the first three features are closely related. All of these H.450 features are supported for H.323 operation only; they are *not* supported for SIP or SPP.

VOIP-1-2-4-8 Front Panel LEDs

LED Types. The VOIP-1-2-4-8 units have two types of LEDs on their front panels:
(1) general operation LED indicators (for power, booting, and Ethernet functions), and
(2) channel operation LED indicators that describe the data traffic and performance in each VoIP data channel.

Active LEDs. On both the VOIP-4 and VOIP-8, there are eight sets of channel-operation LEDs. However, on the VOIP-4, only the lower four sets of channel-operation LEDs are functional. On the VOIP-8, all eight sets are functional.

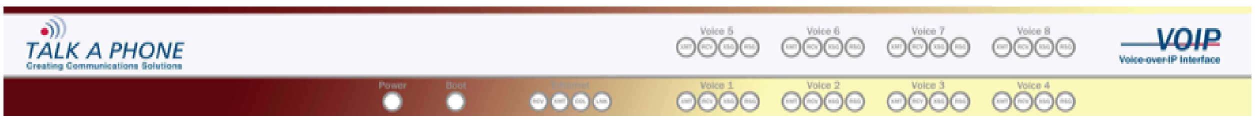


Figure 1-6. VOIP-4/VOIP-8 LEDs

Similarly, the VOIP-2 has the general-operation indicator LEDs and two sets of channel-operation LEDs, one for each channel.

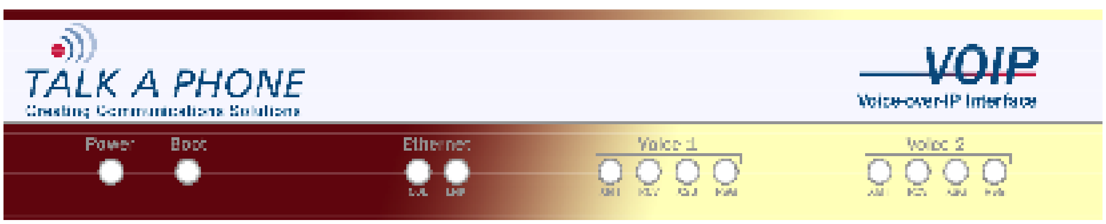


Figure 1-7. VOIP-2 LEDs

Finally, the VOIP-1 has the general-operation indicator LEDs and a set of channel-operation LEDs for its single VoIP channel.

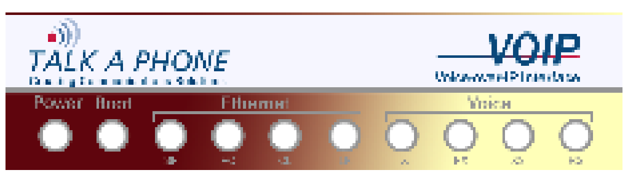


Figure 1-8. VOIP-1 LEDs

VOIP-1 LED Description

VOIP-1 Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
General Operation LEDs	
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on briefly while the VOIP-1 is booting. It lights whenever the VOIP-1 is booting or downloading a setup configuration data set.
Ethernet	**FDX.** LED indicates whether Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions. LNK. Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
Channel-Operation LEDs	
TX	**Transmit.** This indicator blinks when voice packets are being transmitted to the local area network.
RX	**Receive.** This indicator blinks when voice packets are being received from the local area network.
XS	**Transmit Signal.** This indicator lights when the FXS-configured channel is off-hook or the FXO-configured channel is receiving a ring from the Telco or PBX.
RS	**Receive Signal.** This indicator lights when the FXS-configured channel is ringing or the FXO-configured channel has taken the line off-hook.

VOIP-2-4-8 LED Descriptions

VOIP-2/VOIP-4/VOIP-8 Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
General Operation LEDs (one set on each VoIP Interface model)	
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on briefly while the VOIP-2-4-8 is booting. It lights whenever the VOIP-2-4-8 is booting or downloading a setup configuration data set.
Ethernet	**FDX.** LED indicates whether Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions. LNK. Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
Channel-Operation LEDs (one set for each channel)	
XMT	**Transmit.** This indicator blinks when voice packets are being transmitted to the local area network.
RCV	Receive. This indicator blinks when voice packets are being received from the local area network.
XSG	**Transmit Signal.** This indicator lights when the FXS-configured channel is off-hook, the FXO-configured channel is receiving a ring from the Telco, or the M lead is active on the E&M configured channel. That is, it lights when the VOIP-2-4-8 is receiving a ring from the PBX.
RSG	**Receive Signal.** This indicator lights when the FXS-configured channel is ringing, the FXO-configured channel has taken the line off-hook, or the E lead is active on the E&M-configured channel.

Computer Requirements

Minimum Requirements for Windows GUI:

The computer on which the VOIP-1-2-4-8 units' 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 VOIP-1-2-4-8 unit

However, this PC does not need to be connected to the VOIP-1-2-4-8 unit 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.

You will need an available COM port on the controller PC. You'll need to know which COM port is available for use with the VOIP-1-2-4-8 (COM1, COM2, etc.).

Minimum Requirements for Web GUI

- Local Windows GUI must have been used to assign IP address to VOIP-1-2-4-8.
- Internet Explorer 6.0 or higher; or Netscape 6.0 or higher
- Java Runtime Environment version 1.4.0_01 or higher

Placement

Mount your VOIP-1-2-4-8 in a safe and convenient location where cables for your network and phone system are accessible.

Specifications for VOIP-1-2-4-8 Units

Contents: The **VOIP-1-2-4-8** includes the following:

- VOIP unit
- 120VAC power supply
- 19” EIA rack-mount brackets (VOIP-4 and VOIP-8 only)

Model	VOIP-1	VOIP-2	VOIP-4	VOIP-8
Operating Voltage/Current	100-240VAC 1.0 A	External transformer: 3A @5V	100-240 VAC 1.2 - 0.6 A	100-240 VAC 1.2 - 0.6 A
Main Frequencies	50/60 Hz	50/60 Hz	50/60 Hz	50/60 Hz
Power Consumption	9.7 watts (with phone off hook)	19 watts	29 watts	46 watts
Mechanical Dimension	4.3" W x 5.6" D 1.0" H 10.8 cm W x 14.2 cm D x 2.95 cm H	6.2” W x 9” D x 1.4” H 15.8cm W x 22.9cm D x 3.6cm H	1.75” H x 17.4” W x 8.5” D 4.5cm H x 44.2 cm W x 21.6 cm D	1.75” H x 17.4” W x 8.5” D 4.5cm H x 44.2 cm W x 21.6 cm D
Weight	8 oz. (23 g)	1.8lbs (.82kg) 2.6lbs (1.17kg) with transformer	7.1 lbs. (3.2 kg)	7.7 lbs. (3.5 kg)

Identify Remote VOIP Site to Call

When you’re done installing the VOIP-1-2-4-8, you’ll want to confirm that it is configured and operating properly. To do so, it’s good to have another VoIP unit that you can call for testing purposes. You’ll want to confirm end-to-end connectivity. You’ll need IP and telephone information about that remote site. If this is the very first VoIP unit in the system, you’ll want to coordinate the installation of this VOIP-1-2-4-8 with an installation of another unit at a remote site.

Identify VOIP Protocol to be Used

Will you use H.323, SIP, or SPP? Each has advantages and disadvantages. Although it is possible to mix protocols in a single VOIP system, it is highly desirable to use the same VOIP protocol for all VOIP units in the system.

Hookup for VOIP-1

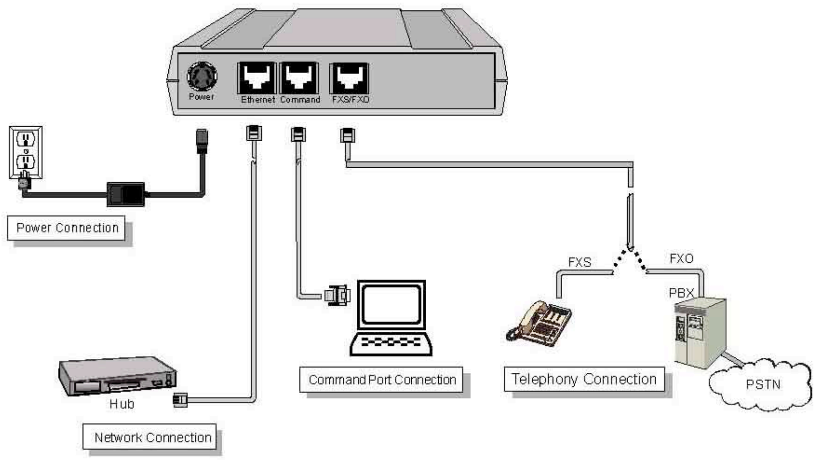


Figure 1-9: Sample hookup diagram for VOIP-1 Interface Unit

Hookup for VOIP-2

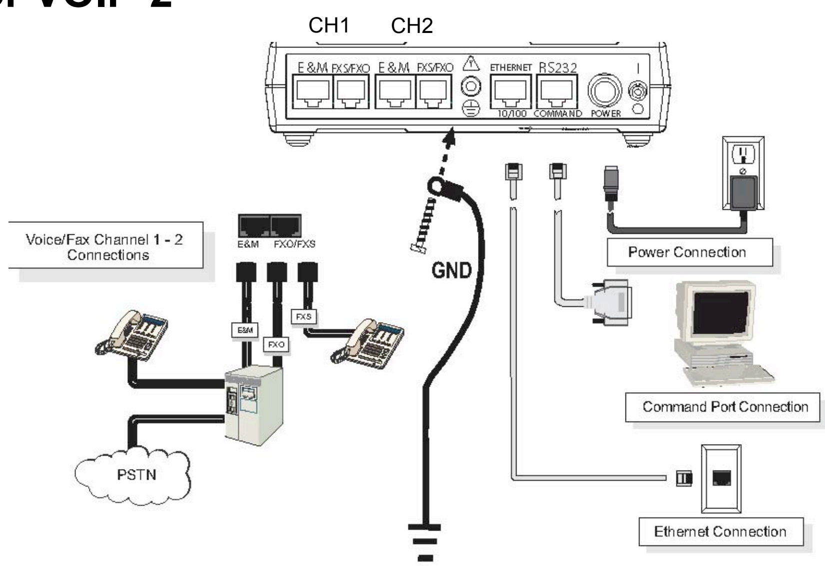


Figure 1-10: Sample hookup diagram for VOIP-2 Interface Unit

Hookup for VOIP-4 and VOIP-8

Connect the VOIP-4 or VOIP-8 as indicated in the following diagram. Connect the RJ-11 cables from the emergency phone(s), PSTN line(s), or analog extension(s) of the PBX to the ports labeled “FXS/FXO”.

Make sure to connect the chassis to Earth Ground at the grounding screw as indicated (VOIP-2, VOIP-4, and VOIP-8 only)

(VOIP-8 shown)

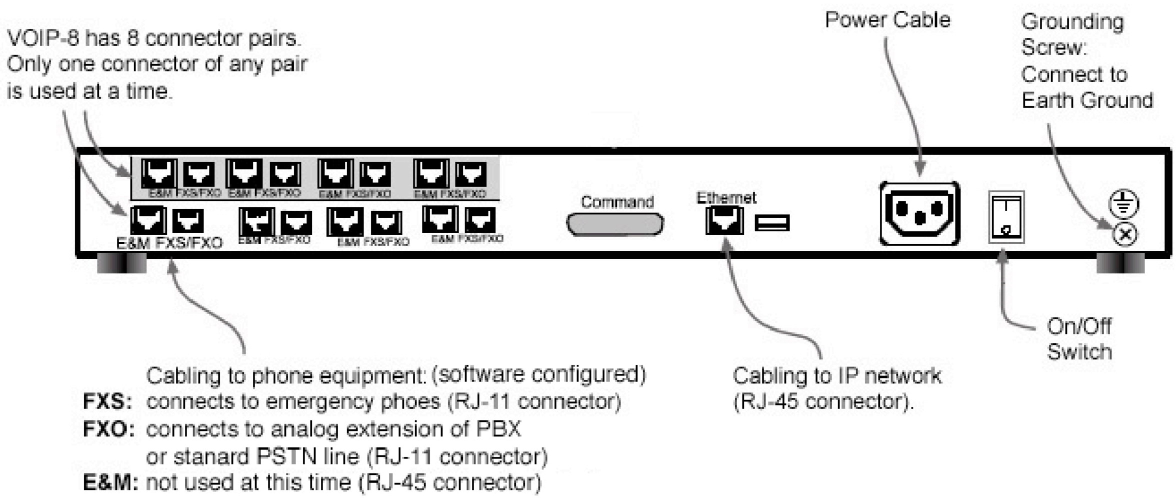


Figure 1-11: Sample hookup diagram for VOIP-4/VOIP-8 Interface Unit

Operation

When the VOIP unit is powered on, it will take approximately one minute to boot up. The red LED (second from the left) indicates that the unit is still booting. After the red LED clears, allow an extra twenty seconds to ensure the unit has fully booted before attempting to initiate a call.

The emergency phones will need to be programmed in accordance with the instructions in the **Quick Start** section in **Chapter 2**.

VoIP System Design

Before you begin programming the VOIP Interface units, it is recommended that you plan out your system layout. You should begin by choosing a setup type. There are two basic types of VoIP setups we can design for an emergency phone system:

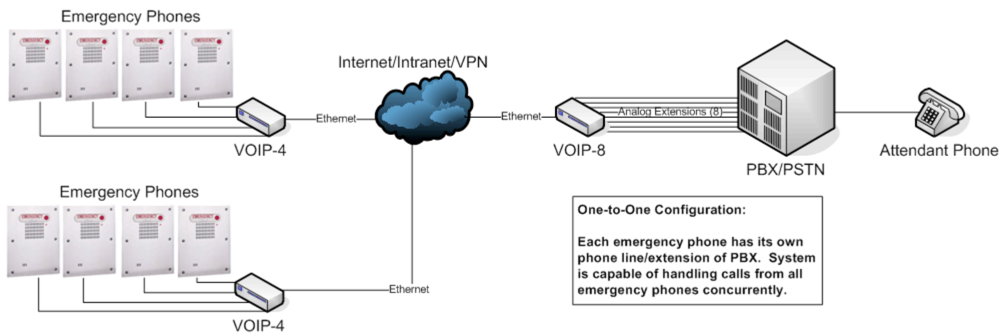


Figure 1-12: One-to-One Configuration

(1) The first setup type is a one-to-one configuration. In this scenario, each emergency phone has its own PBX extension or phone line. The number of calls that the head end is capable of receiving is equal to the number of emergency phones in the field.

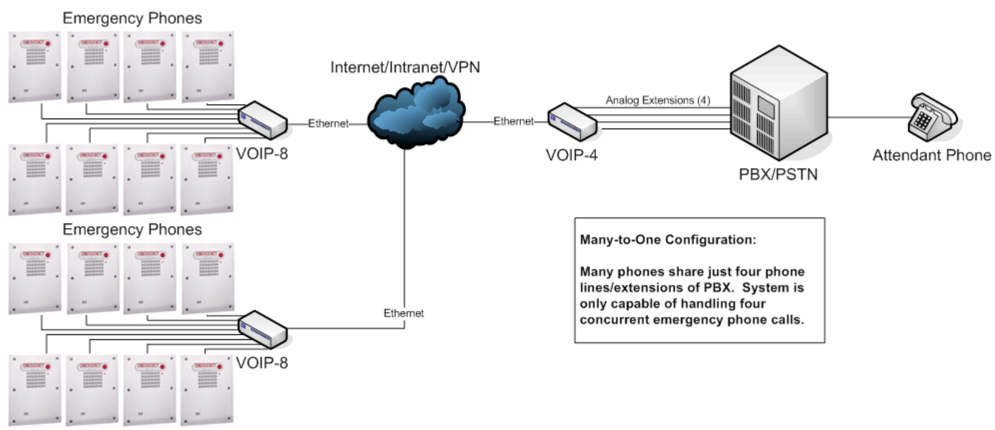


Figure 1-13: Many-to-One Configuration

(2) The second setup type is a many-to-one configuration. In this scenario, many Emergency Phones share PBX extensions or phone lines. The number of calls that the head end is capable of receiving is less than the number of emergency phones in the field.

Once you have chosen a setup type, it is recommended that you assign phone numbers/PBX extensions to the emergency phones and IP addresses to the VOIP units before programming any of the VOIP units. Keep in mind that the PBX extension assignments are separate from the VOIP phone book extensions. Please reference ***Phone Book Design*** (p. 18) for more information.

When designing your system layout, please keep in mind that All VOIP units must have fixed IP addresses. Also, ensure that the proper routing and switching hardware (routers, hubs, firewalls, VPNs, etc.) are in place for the VOIP units to communicate. It is critical that ensure network reliability, which includes sufficient bandwidth and minimizes packet loss and packet delays.

IMPORTANT NOTE: For the Emergency Phone System to work through a power outage, all components of the data path (i.e. the VOIP units, routers, hubs, switches, etc.) must be on back-up power.

IMPORTANT NOTE: For the Emergency Phone System to work through a power outage, all components of the data path (i.e. the VOIP units, routers, hubs, switches, etc.) must be on back-up power.

The following are examples of other types of VoIP setups you can design.

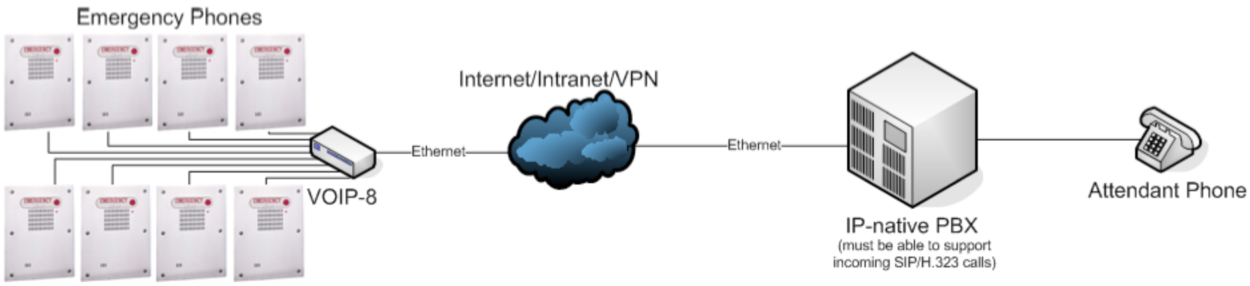


Figure 1-14: Many-to-One IP-Native Head End Configuration

(3) Figure 1-14 is an example of a configuration with a completely IP-native head end. With the IP-native PBX, there isn't a need for VOIP units at the head end.

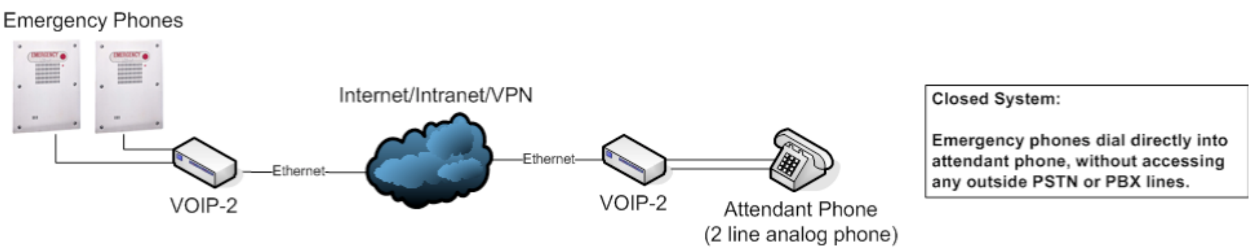


Figure 1-15: One-to-One Closed Configuration

(4) Figure 1-15 is an example of a closed configuration with no PBX or PSTN lines. This configuration type relies solely on the network infrastructure for call routing.

CHAPTER 2:
Quick Start Guide for VOIP-1-2-4-8 Units

Download VOIP-1-2-4-8 Configuration Software
You can download the configuration and firmware update utility by following one of the links below or by going to the product page at www.talkaphone.com/products/

VOIP-1: www.talkaphone.com/sites/default/files/software/VOIP-1-2-4-8/voip-1.zip

VOIP-2-4-8: www.talkaphone.com/sites/default/files/software/VOIP-1-2-4-8/voip-2-4-8.zip

Install VOIP-1-2-4-8 Configuration Software onto a PC
1. VOIP-1-2-4-8 must be properly cabled. Power must be turned on.
2. Extract the content of either **"voip-1.zip"** or **"voip-2-4-8.zip"** depending on the model used. Open the extracted folder and double-click on the **autorun.exe** icon.
3. At first dialog box, click on **Install Software**.
4. If you will be configuring the VOIP unit remotely from the network and the PC does not have the Java Runtime Environment installed, highlight **Java** and click **OK**. Otherwise, skip to **Step (7)**.
5. Follow the on-screen instructions to properly install the Java Runtime Environment.
6. Click on **Install Software** under the autorun.exe that was launched in **Step (2)**.
7. Highlight either **VOIP-1 Software** or **VOIP-2-4-8 Software** and click **OK**.
8. At the "Welcome" screen, click **Next.**
9. Follow on-screen instructions. Accept default program folder location and click **Next.**
10. Accept default icon folder location. Click **Next**. Files will be copied.
11. At completion screen, click **Finish**.
12. At the prompt "Do you want to run VOIP-1-2-4-8 Configuration?," click **No**. Software installation is complete.
13. Go to **Start** → **All Programs** → **VOIP-1** or **VOIP-2-4-8** → **Configuration Port Setup**. Select the proper COM port that will be used to configure the VOIP unit.

Notes on the Configuration of VOIP-1-2-4-8 Units
The initial configuration of the VOIP-1-2-4-8 units must be done locally using the Windows GUI. However, all additional configurations can be done via the Web GUI once you know the IP address of the VOIP unit being configured. The VOIP-1-2-4-8 unit can be reprogrammed remotely in almost every setup where a computer can access the web interface GUI.

Once you have finished programming the VOIP units, you may set each of the units to request a login and password each time the configuration software is launched. For more information on this topic, please reference ***Setting a Password*** (pp. 38-39).

IMPORTANT NOTE: After each configuration change, make sure to hit **OKAY** at each screen. Once you have completed configuring all options, make sure to also **"Save Settings and Reboot"**. If you do not click save and reboot, all changes will be lost.

Phone/IP Starter Configuration

The following are steps that Talk-A-Phone Co. recommends that you take in order to create a functional VoIP system.

IMPORTANT NOTE: After each configuration change, make sure to hit **OKAY** at each screen. Once you have completed configuring all options, make sure to also **"Save Settings and Reboot"**. If you do not click save and reboot, all changes will be lost.

1. Open the VOIP-1-2-4-8 configuration program: **Start | Programs | VOIP-*x-x-x* | Configuration**.

2. **Configuring the IP address.** Go to **Configuration | IP**. Enter the IP parameters for your VoIP site. Leave "Enable DHCP" unchecked unless your network setup requires DHCP.

   NOTE: If a phone's IP is established via DHCP, it can only make outgoing calls, not ingoing call, unless it is set up to register with an H.323 gatekeeper, SIP proxy, or SPP master VOIP. Also, your VoIP until **must** also specify a Gateway address in order for it to work properly even if a Gateway will never be used.

   If you will be using the VOIP-1-2-4-8 unit on an existing network, please consult your network administrator for IP addresses, subnet mask, and gateway information. If you will be creating a dedicated network, you may use proper private addressing (e.g. IP address 192.168.1.25, subnet mask 255.255.255.0, and a gateway 192.168.1.1).

3. If you would like to configure and operate the VOIP-1-2-4-8 unit using the web browser GUI, continue on to step (4). The Web GUI has the same functionality as the local Windows GUI, but offers remote access. If you would like to continue with the Windows GUI, skip to step (5).

4. **Enable Web browser GUI (Optional).** To do configuration and operation procedures using the web browser GUI, you must first enable it. Once you've begun using the web browser GUI, you can go back to the Windows GUI at any time. However, you must log out of the web browser GUI before using the Windows GUI. To do so, follow these steps:

   a. Close the Windows GUI.
   b. Make sure Java Runtime Environment 1.4.2_01 or greater is installed.
   c. Launch a compatible web browser (Internet Explorer 6.0 or above; or Netscape 6.0 or above).
   d. **IMPORTANT NOTE:** The PC being used must be connected to and have an IP address on the same
   e. IP network of the VOIP unit.
   f. Browse to IP address of the VOIP unit being configured.
   g. If a username and password have been established, enter them when prompted.
   h. Use web browser GUI to configure VOIP unit.

5. Go to **Configuration | Voice/Fax**. Select **Coder** | "Manual." Choose "G.711 u-law @ 64kbps" as the **Selected Coder** from the pulldown menu. Talkaphone recommends G.711 u-law for maximum line quality. It is especially recommended that this coder be selected when Talk-A-Lert will be used with VoIP.

   Under **DTMF**, select "Inband" from the **DTMF** pulldown menu.

Under **Advanced Features**, make sure that **Silence Compression** is “unchecked”.

Under **Automatic Disconnection**, set the **Call Duration** to “3600” secs.

For VoIP units connected to emergency phones in a many-to-one configuration or for one-to-one configurations (For more information on many/one-to-one configurations, see ***VoIP System Design*** p. 12), it is recommended that you enable the Auto Call feature.
The Auto Call feature automatically links the emergency phone to the dial tone of the PBX extension or phone line, so the emergency phone only needs to dial the extension or phone number to reach the attendant at the head end.

Under **Auto Call/OffHook Alert**, select “Auto Call” from the pulldown menu and specify the **Phone Number** that you would like the emergency phone to dial. Please reference ***Phone Book Design*** (p. 18) for clarification on the phone number you should enter in this field. For head end units of a many-to-one configuration, select “None” under **Auto Call/OffHook Alert**.

If you know of any other specific parameter values that will apply to your system, enter them. Most of the time, all the channels on the multi-channel VOIP units will share the same parameter values. To facilitate settings duplication, you can copy parameter settings from one channel to another. Click **Copy Channel**. Select **Copy to All**. Click **Copy**. At the main Voice/Fax Parameters screen, click **OK** to exit from the dialog box.

6. Go to **Configuration | Interface**. Select **Interface Type |** “FXS” if connecting to emergency phones or for head end of “closed system” (ringing phone directly off head end VOIP unit without any external phone lines or PBX), “FXO” for head end VOIP unit(s) (connecting to PBX extensions or analog phone lines).

   If the VoIP unit will be using an “FXS” interface, make sure that **FXS Options | Current Loss** is “checked”. This will allow the emergency phone to hang up automatically when a call is over.

   Go to **Disconnect on Call Progress Tone** and make sure that it is “checked”.

   Under **Flash Hook Options**, set the **Detection Range** to a minimum of “100” ms and a maximum of “150” ms.

   If you know of any other specific parameter values that will apply to your system, enter them. Most of the time, all the channels on the multi-channel VOIP units will share the same parameter values. To facilitate settings duplication, you can copy parameter settings from one channel to another. Click **Copy Channel**. Select **Copy to All**. Click **Copy**. At the main Voice/Fax Parameters screen, click **OK** to exit from the dialog box.

7. Go to **Configuration | Regional Parameters**. Select **Custom** from the **Country/Region** pulldown menu. Now change the entries for the following types:

   Unobtainable Tone:
   - Frequency 1 = 480
   - Frequency 2 = 620
   - Cadence 1, 2, 3, 4 = 500

   Survivability Tone:
   - Frequency 1 = 480
   - Frequency 2 = 620
   - Cadence 1, 2, 3, 4 = 500

   Reorder Tone:
   - Frequency 1, 2 = 999
   - Cadence 1, 2, 3, 4 = 0

These settings should be similar to that for USA except for the above changes.

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

8. **SMTP Configuration.** You can configure the VOIP units to send e-mail notifications. If you would like to receive phone-call logs from the VOIP-1-2-4-8 via e-mail (to your VoIP Administrator or someone else), continue with step (9). If not, skip to step (10).

9. Go to **Configuration | SMTP**. SMTP allows you to send phone-call log records to the VoIP Administrator via e-mail. Check **Enable SMTP**. You should have already obtained an e-mail address for the VOIP-1-2-4-8 unit itself (this serves as the origination e-mail account for e-mail logs that the VOIP-1-2-4-8 can e-mail out automatically).

   Enter this e-mail address in the “Login Name” field. Type the password for this e-mail account.

   Enter the IP \address of the e-mail server where the VOIP-1-2-4-8’s e-mail account is located in the “Mail Server IP Address” field.

   Typically the e-mail log reports are sent to the VoIP Administrator but they can be sent to any e-mail address.

   Decide where you want the e-mail logs sent and enter that e-mail address in the “Recipient Address” field. Whenever e-mail log messages are sent out, they must have a standard Subject line (e.g. “Phone Logs for VoIP N”). If you have more than one VOIP-1-2-4-8 unit in the building, you’ll need a unique identifier for each one (select a useful name or number for “N”). In this “Subject” field, enter a useful subject title for the log messages.

   In the “Reply-To Address” field, enter the e-mail address of your VoIP Administrator.

10. Go to **Configuration | Logs**. Select “Enable Console Messages.” To allow log reports by e-mail (if desired), click **SMTP**. Click **OK**. To do logging with a SysLog client program, check **Enable** under “SysLog Server” in the **Logs** screen and specify the SysLog Server’s IP Address. To implement this function, you must install a SysLog client program.

11. **Enable premium (H.450) telephony features.** Go to **Configuration | Supplementary Services**. Select any features to be used. For Call Hold, Call Transfer, and Call Waiting, specify the key sequence that the phone user will press to invoke the feature. For Call Name Identification, specify the allowed name types to be used and a caller-id descriptor.

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

12. **Naming the VoIP gateway.** Go to **Phone Book | Phone Book Configuration**. Enter the name you would like the VOIP unit to use. This name will be used when the Caller ID feature is enabled.

13. **Programming outbound phone book information.** Go to **Phone Book | Outbound Phone Book | List Entries**. Click on **Add** to create new entries. The Outbound phone book lists the phone numbers or extensions the VOIP unit can call. Please reference the ***Phone Book Design*** section (p. 18) for information and examples on how to program the phone book.

14. **Programming inbound phone book information.** Go to **Phone Book | Inbound Phone Book | List Entries**. Click on **Add** to create new entries. The Inbound phone book describes the dialing sequences that can be used to call the VOIP unit being programmed unit and how those

calls will be directed. Please reference the ***Phone Book Design*** section (p. 18) for information and examples on how to program the phone book.

15. **Changing user name and password (Web GUI only. See p. 38 for instructions on the Windows GUI).** Go to **Change Password**. Specify the **User Name** that you would like to use for this VOIP unit.

    If an **Old Password** exists, enter that password. Now proceed with assigning a **New Password** and then **Reconfirm Password**.

16. **Save configuration changes.** Go to **Save Setup | Save and Reboot**. Click **OK**. This will save the parameter values that you have just entered. The VOIP-1-2-4-8's "BOOT" LED will light up while the configuration file is being saved and loaded into the VOIP-1-2-4-8. Don't do anything to the VOIP-1-2-4-8 until the "BOOT" LED is off (a loss of power at this point could cause the VOIP-1-2-4-8 unit to lose the configuration settings you have made).

Phone Book Design

Phone book entries are critical when designing a VoIP setup for Emergency Phones. The phone book in the VOIP units serve in providing routing information for calls over a Voice-over IP setup. The Outbound phone book for a particular VOIP unit describes the dialing sequences required for a call to originate locally and reach any of its possible destinations at remote VoIP sites. The Inbound phone book for a particular VoIP unit describes the dialing sequences required for a call to originate remotely from any other VOIP sites in the system, and to terminate on that particular VOIP.

Concisely, the Outbound phone book lists the phone stations it can call and the Inbound phone book describes the dialing sequences that can be used to call that VOIP unit and how those calls will be directed. In general, the Inbound phone book entries of the local VOIP unit will match the Outbound phone book entries of the remote VOIP unit. Similarly, the Outbound phone book entries of the local VOIP unit will match the Inbound phone book entries of the remote VOIP unit. However, in most cases, the VOIP units will only have some matching entries.

Once you've programmed the VOIP units with a known IP address, you can remotely program the phone books of each unit through the Web GUI. The following steps will assist you in completing the task of programming the phone book(s).

1. Open a web browser to the VOIP-1-2-4-8 to be configured to access the Web GUI. Go to **Phone Book.**

2. Under **Outbound Phone Book**, add entries for the extensions and IP addresses for each emergency phone or extension/phone number that the VOIP unit will call. It is advised that every entry be configured for the H.323 protocol.

   To add an entry, click on **Add**. Enter the extension/phone number to be dialed in the **Destination Pattern** field. Specify the **Total Digits** and the **IP Address** to which that number is assigned.

   You can now continue configuring other parameters such as SIP proxy and H.323 gateway information. Click **OK** once you have completed configuring this entry. You should repeat step (2) for all outbound phone book entries.

3. Under **Inbound Phone Book**, add entries for the dialing sequences that can be used to call the VOIP unit being configured.

   To add an entry, click on **Add**. Enter the extension/phone number to be dialed in the **Remove Prefix** field. Under **Channel Number**, make sure that it is not set for "Hunting" mode. Assign a **Channel Number** to each extension/phone number to a port on the multi-channel VOIP units (VOIP-2, VOIP-4, VOIP-8).

   You can now continue configuring other parameters such as **Call Forward**. Click **OK** once you have completed configuring this entry. You should repeat step (3) for all inbound phone book entries.

Phone Book Design Example 1

In Figure 2-1, we have a one-to-one configuration. Each emergency phone will have an assigned PBX extension, so the head end VOIP-4 will only be using three of its four channels. Also, in this scenario, the PBX is set to ringdown mode, so every emergency phone is programmed with *13*5* and each VOIP unit connected to an emergency phone is configured to auto call/hotline to the attendant phone.

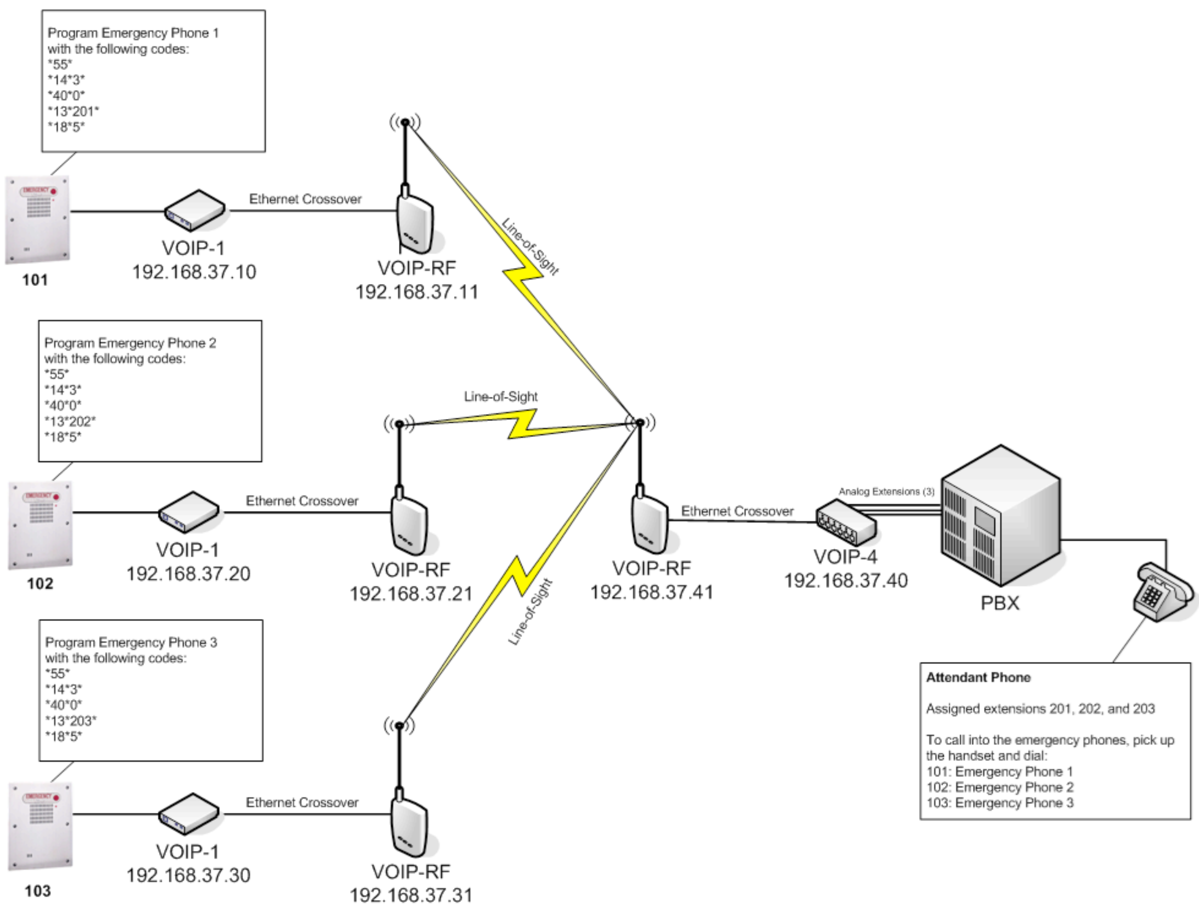


Figure 2-1: Example of a one-to-one VoIP System Configuration

For the VOIP units connected to emergency phones, we would program the phone books with the following information.

Emergency Phone 1:
Inbound Phone Book entry: 101
Outbound Phone Book entry: 201 assigned to 192.168.37.40

Emergency Phone 2:
Inbound Phone Book entry: 102
Outbound Phone Book entry: 202 assigned to 192.168.37.40

Emergency Phone 3:
Inbound Phone Book entry: 103
Outbound Phone Book entry: 203 assigned to 192.168.37.40

In this scenario, each emergency phone VOIP unit has been programmed to call one specific extension at the head end. There is no need to assign a channel to the inbound phone book entry because there exists only one channel for these VOIP units.

If we look at the phone book for the head end VOIP unit connected to the Attendant Phone, it should be programmed with:

Attendant Phone:

Inbound Phone Book entries:	201 assigned to Channel 1
	202 assigned to Channel 2
	203 assigned to Channel 3
Outbound Phone Book entries:	101 assigned to 192.168.37.10
	102 assigned to 192.168.37.20
	103 assigned to 192.168.37.30

In this case, each PBX extension has been assigned a number (201, 202, and 203). Thus, when Emergency Phone 2 places a call, it will connect to extension 202 by routing to Channel 2 on the head end VOIP unit. The attendant also has the ability to call into each of the emergency phones by dialing 101, 102, or 103 for the respective phones.

Phone Book Design Example 2

Let us look at another example:

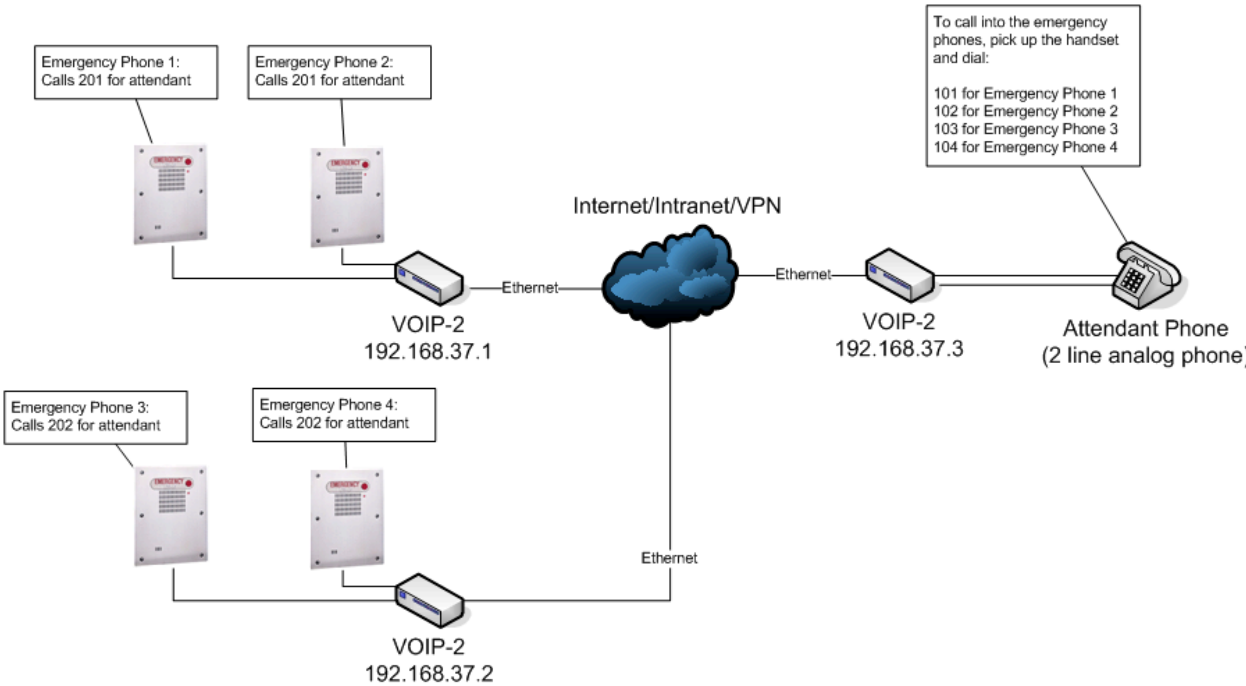


Figure 2-2: Example of a many-to-one VoIP System Configuration

In this scenario, we have a many-to-one closed system. This setup does not make use of a PBX or PSTN lines. The system has been designed so that Emergency Phones 1 and 2 will share line 201 and Emergency Phones 3 and 4 will share line 202. The emergency phones should have been properly programmed so that if a busy signal is received, the emergency phone will try to dial the same number or it will dial an alternate number.

If we look at the phone book configuration for the VOIP-2 unit connected to Emergency Phones 1 and 2, we will find the following:

Inbound Phone Book entry:	101 assigned to Channel 1
	102 assigned to Channel 2
Outbound Phone Book entry:	201 assigned to 192.168.37.3

Similar programming goes for the VOIP-2 unit connected to Emergency Phones 3 and 4.

If we look at the phone book configuration for the head end VOIP unit connected to the Attendant Phone, it should be programmed with:

Attendant Phone:

Inbound Phone Book entries:	201 assigned to Channel 1
	202 assigned to Channel 2
Outbound Phone Book entries:	101 assigned to 192.168.37.1

102 assigned to 192.168.37.1
103 assigned to 192.168.37.2
104 assigned to 192.168.37.2

In this case, the attendant phone has been assigned 201 and 202 and their extensions. So, when Emergency Phone 2 places a call, it will connect to extension 201 by routing to Channel 1 on the head end VOIP unit.

The attendant also has the ability to call into each of the emergency phones by dialing 101, 102, 103, or 104 for the respective phones. What happens when the attendant calls Emergency Phone 2 from line 1 (extension 201)? The call is routed from Channel 1 on the head end VOIP unit, through the LAN, to Channel 2 on the VOIP unit connected to Emergency Phone 2.

Operation and Maintenance

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

System Information screen

This screen presents vital system information at a glance. Its primary use is in troubleshooting. This screen is accessible via the **Configuration** pulldown menu, the **Configuration** sidebar menu, or by the keyboard shortcut **Ctrl + Alt + Y**.

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

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

Statistics Screens

Ongoing operation of the VOIP-1-2-4-8, whether it is in a VoIP/PBX setting or VoIP/telco-office setting, can be monitored for performance using the Statistics functions of the VOIP-1-2-4-8 software.

About Call Progress

Call Progress Details: Field Definitions		
Field Name	**Values**	**Description**
Channel	1-n	Number of data channel or time slot on which the call is carried. This is the channel for which call-progress details are being viewed.
Call Details		
Duration	Hours: Minutes: Seconds	The length of the call in hours, minutes, and seconds (hh:mm:ss).
Mode	Voice or FAX	Indicates whether the call being described was a voice call or a FAX call.
Voice Coder	G.723, G.729, G.711, etc.	The voice coder being used on this call.
Packets Sent	integer value	The number of data packets sent over the IP network in the course of this call.
Packets Rcvd	integer value	The number of data packets received over the IP network in the course of this call.
Bytes Sent	integer value	The number of bytes of data sent over the IP network in the course of this call.
Bytes Rcvd	integer value	The number of bytes of data received over the IP network in the course of this call.
Packets Lost	integer value	The number of voice packets from this call that were lost after being received from the IP network.
Outbound Digits	0-9, #, *	The digits transmitted by the VOIP-1-2-4-8 to the PBX/telco for this call.
Prefix Matched		Displays the dialed digits that were matched to a phonebook entry.

Call Progress Details: Field Definitions (cont'd)		
From – To Details		**Description**
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that handled this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address from which the call was received.
Options	SC, FEC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.
Silence Compression	SC	"SC" stands for Silence Compression. With **Silence Compression** enabled, the VOIP-1-2-4-8 will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel.
Forward Error Correction	FEC	"FEC" stands for Forward Error Correction. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off

Call Progress Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Supplementary Services Status		
Call on Hold	alphanumeric	Describes held call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers 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.

Call Progress Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Supplementary Services Status		
Caller ID	There are four values: "Calling Party + *identifier*"; "Alerting Party + *identifier";* "Busy Party + *identifier";* and "Connected Party + *identifier*"	This field shows the identifier and status of a remote VOIP (which has Call Name Identification enabled) with which this VOIP unit is currently engaged in some VOIP transmission. The status of the engagement (Connected, Alerting, Busy, or Calling) is followed by the identifier of a specific channel of a remote VOIP unit. This identifier comes from the "Caller Id" field in the **Supplementary Services** screen of the remote VOIP unit.
Status	hangup, active	Shows condition of current call.
Call Control Status	Tun, FS + Tun, AE, Mux	Displays the H.323 version 4 features in use for the selected call. These include tunneling (Tun), Fast Start with tunneling (FS + Tun), Annex E multiplexed UDP call signaling transport (AE), and Q.931 Multiplexing (Mux). See **Phonebook Configuration Parameters** (in T1 or E1 chapters) for more on H.323v4 features.

About Logs

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

Logs Screen Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Details (cont'd)		
Packets loss (lost)	integer value	The number of voice packets from this call that were lost after being received from the IP network.
Voice coder	G.723, G.729, G.711, etc.	The voice coder being used on this call.
Packets received	integer value	The number of data packets received over the IP network in the course of this call.
Bytes received	integer value	The number of bytes of data received over the IP network in the course of this call.
Outbound digits	0-9, #, *	The digits transmitted by the VOIP-1-2-4-8 to the PBX/telco for this call.
FROM Details		
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that originated this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address of the VOIP gateway from which the call was received.
Options	FEC, SC	Displays VOIP transmission options used by the VOIP gateway originating the call. These may include Forward Error Correction or Silence Compression.
TO Details		
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that completed (terminated) this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address of the VOIP gateway at which the call was completed (terminated).
Options		Displays VOIP transmission options used by the VOIP gateway terminating the call. These may include Forward Error Correction or Silence Compression.

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

About IP Statistics

IP Statistics: Field Definitions		
Field Name	**Values**	**Description**
		UDP versus TCP. (User Datagram Protocol versus Transmission Control Protocol). UDP provides unguaranteed, connectionless transmission of data across an IP network. By contrast, TCP provides reliable, connection-oriented transmission of data. Both TCP and UDP split data into packets called "datagrams." However, TCP includes extra headers in the datagram to enable retransmission of lost packets and reassembly of packets into their correct order if they arrive out of order. UDP does not provide this. Lost UDP packets are 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 appear as static).
"Clear" button		Clears packet tallies from memory.
Total Packets		Sum of data packets of all types.
Transmitted	integer value	Total number of packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the VOIP Interface software.
Received	integer value	Total number of packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the VOIP Interface software.

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

IP Statistics: Field Definitions (cont'd)		
RTP Packets		Voice signals are transmitted in Realtime Transport Protocol packets. RTP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the VOIP Interface software.
Received	integer value	Number of RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the VOIP Interface 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 VOIP Interface 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 VOIP Interface software.
Received	integer value	Number of RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the VOIP Interface 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 VOIP Interface software.

About Link Management

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

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

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

About Packetization Time

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

Packetization rates can be set separately for each channel. The table below presents the ranges and increments for packetization rates.

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

Once the packetization rate has been set for one channel, it can be copied into other channels.

About Registered Gateway Details

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

Registered Gateway Details: Field Definitions		
Field Name	**Values**	**Description**
Column Headings		
Description	alphanumeric	This is a descriptor for a particular VOIP gateway unit. This descriptor should generally identify the physical location of the unit (e.g., city, building, etc.) and perhaps even its location in an equipment rack.
IP Address	n.n.n.n, for n = 0-255	The RAS address for the gateway.
Port		Port by which the gateway exchanges H.225 RAS messages with the gatekeeper. .
Register Duration		The time remaining in seconds before the TimeToLive timer expires. If the gateway fails to reregister within this time, the endpoint is unregistered.
Status		The current status of the gateway, either registered or unregistered.
No. of Entries		The number of gateways currently registered to the Registrar. This includes all SPP clients registered and the Registrar itself.
Details		
Count of Registered Numbers		If a registered gateway is selected (by clicking on it in the screen), The "Count of Registered Numbers" will indicate the number of registered phone numbers for the selected gateway. When a client registers, all of its inbound phonebook's phone numbers become registered.
List of Registered Numbers		Lists all of the registered phone numbers for the selected gateway.

Connectivity Test

The procedures **Phone/IP Starter Configuration** and **Phone Book Design** must be completed before you can do this procedure.

1. These connections must be made:

   - VOIP-1-2-4-8 to local phone station
     or
   - VOIP-1-2-4-8 to extension of key phone system
   - VOIP-1-2-4-8 to command PC
   - VOIP-1-2-4-8 to Internet

2. Inbound Phonebook and Outbound Phonebook must both be set up with at least one entry in each. These entries must allow for connection between two VoIP units.
3. Console messages must be enabled. (If this has not been done already, go, in the GUI, to **Configuration | Logs** and select the **"Console Messages"** checkbox.
4. You now need to free up the COM port connection (currently being used by the VOIP-1-2-4-8 configuration program) so that the HyperTerminal program can use it. To do this, you can either (a) click on **Connection** in the sidebar and select **"Disconnect"** from the drop-down box, or (b) close down the configuration program altogether.
5. Open the **HyperTerminal** program.
6. Use HyperTerminal to receive and record console messages from the VOIP-1-2-4-8 unit. To do so, set up HyperTerminal as follows (setup shown is for Windows NT 4.0; details will differ slightly in other MS operating systems):

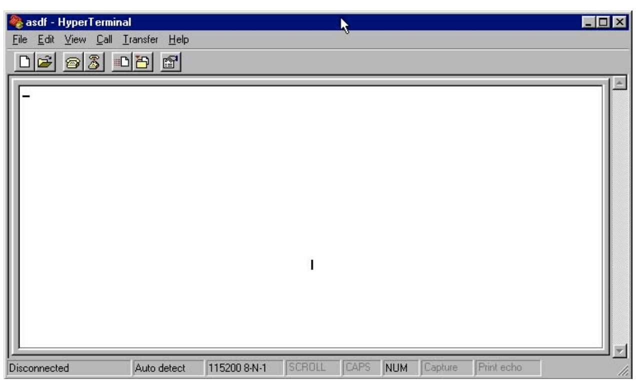


Figure 2-3: HyperTerminal window

In the upper toolbar of the HyperTerminal screen, click on the **Properties** button. In the **"Connect To"** tab of the **Connection Properties** dialog box, click on the **Configure** button. In the next dialog box, on the **"General"** tab, set **"Maximum Speed"** to 115200 bps.

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

Data bits: 8
Parity: none
Stop bits: 1

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

7. Make VOIP call on a local phone line accessing PSTN directly or through key system.

8. Read console messages recorded on HyperTerminal.

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

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

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

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

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

   “**PSTN: pstn call connected on X**”

   where X is the number of the VoIP channel carrying the call.

10. If the HyperTerminal messages do not confirm connectivity, refer to the ***Troubleshooting*** procedure in this manual.

Setting a Password (Windows GUI)

Talkaphone recommends that a user name and a **secure** password be assigned to each VOIP unit in your system configuration.

After a user name has been designated and a password has been set, that password is required to gain access to the configuration software under both the Windows and Web GUIs. Only one user name and password can be assigned to a VOIP unit.

IMPORTANT NOTE: Record your user name and password in a **safe** place. If the password is lost, forgotten, or irretrievable, the user must restore the VOIP-1-2-4-8 to factory defaults.

Follow these steps to set a user name and password for the VOIP-1-2-4-8 units:

1. Go to **Start | Programs | VOIP-*x-x-x* | Set Password.**
2. You will be asked to confirm that you want to set a password, which requires automatic rebooting of the VOIP unit. Click **OK** if you are sure you want to set a password.
3. The **Password** prompt will now appear. If you intend to use the built-in FTP Server function, enter a user name. A user name is not required to access the Windows GUI, the Web GUI, or the commands in the Program group menu. Enter your secure password in the **New Password** field of the Password screen. Type the same password again in the **Confirm Password** field to verify the password you have chosen.
4. Click **OK**.
5. A message box will appear indicating that a password has been set successfully. After the password has been set successfully, the VOIP unit will reboot itself (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 GUI and any of the software components listed in the Program group menu.
7. Both the User Name and Password are both needed for access to the FTP Server residing in the VOIP unit. When the Windows GUI asks for the password upon launch of the program, it will simply quit if **CANCEL** is selected. The Windows GUI will then produce an error message if an invalid password is entered.

Setting a Password (Web GUI)

Setting a password is optional when using the Web GUI. Only one password can be assigned and it works for all VOIP-1-2-4-8 software functions (Windows GUI, web browser GUI, FTP server, and all Program menu commands, e.g., Upgrade Software – only the FTP Server function requires a User Name in addition to the password). After a password has been set, that password is required to access the web browser GUI.

IMPORTANT NOTE: Record your user name and password in a **safe** place. If the password is lost, forgotten, or irretrievable, the user must restore the VOIP-1-2-4-8 to factory defaults.

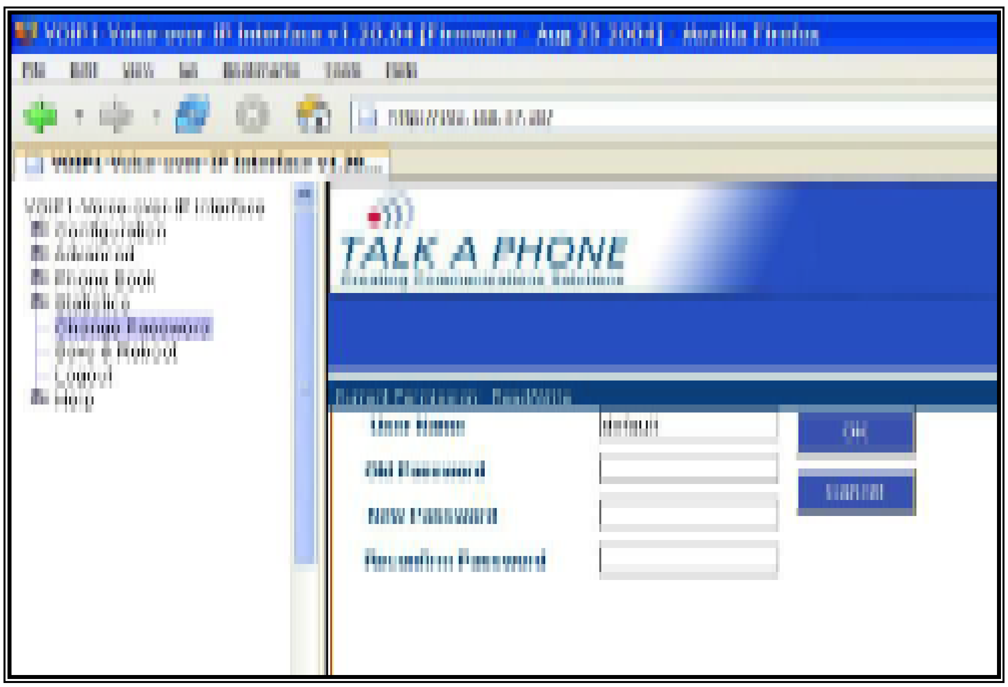


Figure 2-4: 'Change Password' preference pane under Web GUI

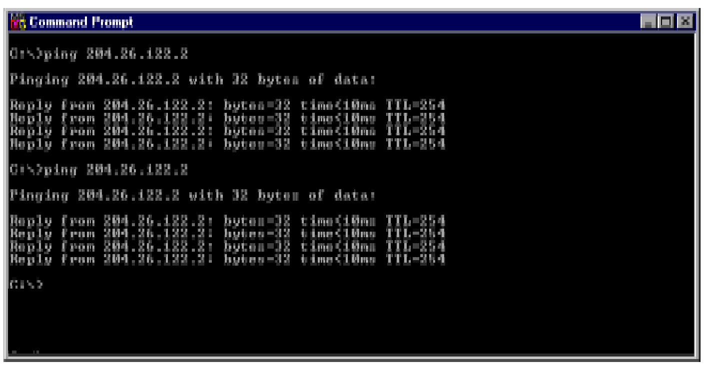
Appendix A:
Troubleshooting the Voice over IP Interfaces

I cannot connect to a VOIP-1-2-4-8 unit remotely via the Web GUI.

1. Check the power and cabling for the VoIP unit.
2. Be sure an IP address has been assigned to the VOIP-1-2-4-8 unit (this must be done in the Windows GUI).
3. Make sure that the configuration was saved in the Windows GUI. Each pane in the configuration software requires you to click on “Ok” before you “Save and reboot”.
4. Make sure that the latest version of Java is installed for your browser.
5. Make sure that the PC being used to access the VOIP unit remotely is on the same IP network.
6. Make sure that a firewall is not blocking traffic the addresses and ports that the VOIP-1-2-4-8 use. Also, check to make sure that any routers in between are configured properly.

I cannot establish connectivity between two VoIPs in the system. I have phones that will not connect in my setup.

1. Ping both VOIP-1-2-4-8 units to confirm connectivity to the network.



2. Verify the telephone connections. Check cabling. Are connections well seated? To correct receptacle?
Are telephone Interface Parameter settings correct?
3. Verify phonebook configuration.
4. Observe console messages while placing a call. Look for error messages indicating phonebook problems, network problems, voice-coder mismatches, etc.

The emergency phone consistently gets a busy signal or it is unable to dial out to the intended phone.

1. Make sure the voice codecs are the same on all VOIP-1-2-4-8 units.
2. Verify phonebook and ensure that the calls are being routed to the intended box.
3. Make sure that the VOIP-1-2-4-8 unit is configured with a proper gateway address.

The emergency phone(s) in my setup seem to be experiencing low microphone.

If your setup includes consolidators and an Iwatsu PBX, there might be volume issues where the microphone volume of the emergency phone is low. Test different GAIN settings to achieve optimal results, but note that if the gain is too high, the DTMF may become distorted and make the PBX unable to route the calls. Adjust the voice gain setting to achieve optimal volume. You may first want to try setting the VOIP-1-2-4-8 field units to:

VOICE GAIN: +5db IN, -5dB OUT

Also, increasing the trim pot on the emergency phones may help.

The attendant phone(s) cannot pick up microphone input from the emergency phone(s) in my setup.

Ensure that all Emergency Phones are programmed to operate on Mode III.

My configuration changes are not saved in the VOIP unit.

After each configuration change, make sure to hit OKAY at each screen, and do a "save settings and reboot" when you are done. If you do not click save and reboot, all changes will be lost.

Appendix B:
Frequently Asked Questions

How far can a phone be from the VOIP-1-2-4-8?

The VoIP Interface units work with 1000 ft. (24AWG) twisted pair copper wire.

Does the VOIP-1-2-4-8 provide power to the phone?

Yes, all of the VoIP Interface units provide power to the Emergency Phones.

My VOIP-1-2-4-8 is taking a long time to boot up. Is this normal?

Yes, the VOIP-1-2-4-8 units typically take about one minute to complete a cold boot.

How do I install a VOIP-1-2-4-8 behind a firewall or Proxy Server?

Your firewall or proxy server will need to support either H.323 or SIP. You may also open up the ports listed below.

H.323 protocol uses:

UDP:	5000-5075, 16000-20000
TCP:	1720, 16000-20000
Q.931:	900-916

SIP protocol uses:

UDP:	5000-5300

SPP protocol uses:

UDP:	10000 (or the port number specified in the outbound phone book entries)

NOTE: Opening up firewall ports can make your network vulnerable to an attack. Please consult with your network administrator before carrying out any firewall configuration changes.