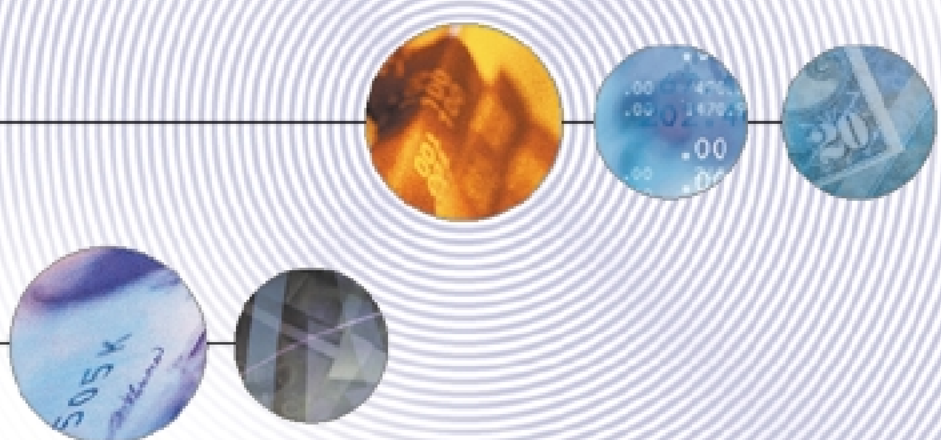
FX Digital Communication System VoIP Board Installation and Programming Instructions



This publication is applicable to software revision 17A and later.

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COMDIAL®
The Rising Star™

Comdial® strives to design the features in our communications systems to be fully interactive with one another. However, this is not always possible, as the combinations of accessories and features are too varied and extensive to insure total feature compatibility.

Accordingly, some features identified in this publication will not operate if some other feature is activated. Comdial disclaims all liability relating to feature non-compatibility or associated in any way with problems which may be encountered by incompatible features. Notwithstanding anything contained in this publication to the contrary, Comdial makes no representation herein as to the compatibility of features.

Issued: 2/12/97

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Warnings And Cautions Used In This Publication

*****WARNING***** Writers use warning notices in this publication to emphasize to the readers that hazardous voltages, currents, temperatures, or other conditions that could cause personal injury exist in this equipment or are associated with its use.

There are no warnings in this publication.

CAUTION Writers use caution notices to call attention to conditions where readers might cause equipment damage or improper operation if they do not exercise proper care.

This publication contains the following cautions:

CAUTION

VoIP circuit boards are susceptible to damage caused by electrostatic discharge, and you must keep this fact in mind as you handle the circuit boards. Refer to the Comdial publication IMI01-005, Handling Of Electrostatically Sensitive Components, for general information Specific handling precautions are also included in this installation instruction.

CAUTION

If you install the VoIP board in an FX expansion cabinet, you must insure that the SCSI interface cards (FXSOPT-SCX-1), which mount on the services board and provide the expansion cabinet interconnection, are at revision B or later.

CAUTION

When installing a board while power is on, use the pre-charge cable and be sure to connect the cable to the labeled pre-charge port. DO NOT mistakenly use the serial data port that is located along the board's right front edge.

NOTES Writers use notes to call attention to information that is important to the understanding and operation of the equipment.

NOTE: Notes are located through the publication wherever the writers deem them useful.

Notes

Installing and Programming the VoIP Board

Introducing the VoIP Board

In a modern, distributed business environment, there is often the need to pass both data and voice information back and forth between different locations. Traditionally, this voice and data traffic is handled separately. Devices that employ protocols such as frame relay handle the data passage, and leased lines (often T1 lines) or the public switched telephone network (PSTN) handle the voice element. This redundant connectivity and its associated tariffs and regulations result in substantial operating costs to the network owners. Eliminating the separate pathway for the voice element by sending it over the same path as the data would result in a sizable cost savings.

The Voice over Internet Protocol (VoIP) board (Comdial part number FXVOIP-xx) allows station-to-station calling between FX systems that are running the Internet Protocol (IP) networking software. Combining the features of the IP networking software, and the interconnection provided by the VoIP boards, telephone voice, facsimile (FAX) relay, and data modem communications are available between several widely-separated FX systems through the same site-supplied Wide Area Network (WAN) configuration that handles the data traffic for the site.

Call handling features available to telephone callers over the IP network are the same as those provided by the conventional system-to-system networking features. For a discussion of the conventional networking features, refer to Comdial publication IMI89–280, *Installing and Using System-To-System Networking*.

Defining VoIP

Voice over Internet Protocol (VoIP) is based upon the principal of transmitting digitized voice packets over IP networks. Basically, VoIP consists of converting voice signals into streams of digital packets and sending those packets of data through an IP environment. While it is beyond the scope of this publication to provide a tutorial on VoIP technology, there is certain information that you must have to help understand how the VoIP board functions, and you can find that information herein.

Locating Tutorial Information

If you want tutorial information, white papers are available from Comdial that provides a basis understanding of Comdial VoIP and IP-telephony technology. Navigate your Internet browser to ***http://www.comdial.com/ip/ip_fx.asp*** and download the following white papers:

- *Comdial Voice Over IP Quality*
- *The Future of Communications: The FX™ Evolves*

Describing Basic VoIP Operation

In a very general way, here is how a typical VoIP exchange can occur. Someone initiates a call on a system telephone, and that call travels over the house wiring to the FX digital communications system and through its installed VoIP board. Once the VoIP board receives the call, it communicates with a local router, forwarding IP packets through the IP network to another local router, and from there to a VoIP board installed in a remote system. From the remote VoIP board, the call passes through the remote FX digital communications system, through that site's house wiring, and rings at a telephone connected to that system.

Describing the VoIP Board

The VoIP board is a voice handling device that replaces the traditional line board in an FX telephony system. By connecting the VoIP board to a facilities' data network, you arrange for the voice element to pass over the same connection that handles the data transfer. Thus, the VoIP board enables a voice connection to exist through an established, site-provided circuit connection or wide area network (WAN) data arrangement. The board converts the telephone system's control signalling and voice traffic into IP packets for transport over an IP network. By doing this, the VoIP board and the IP network replaces the functions normally provided by a conventional T1 connection between the networked systems.

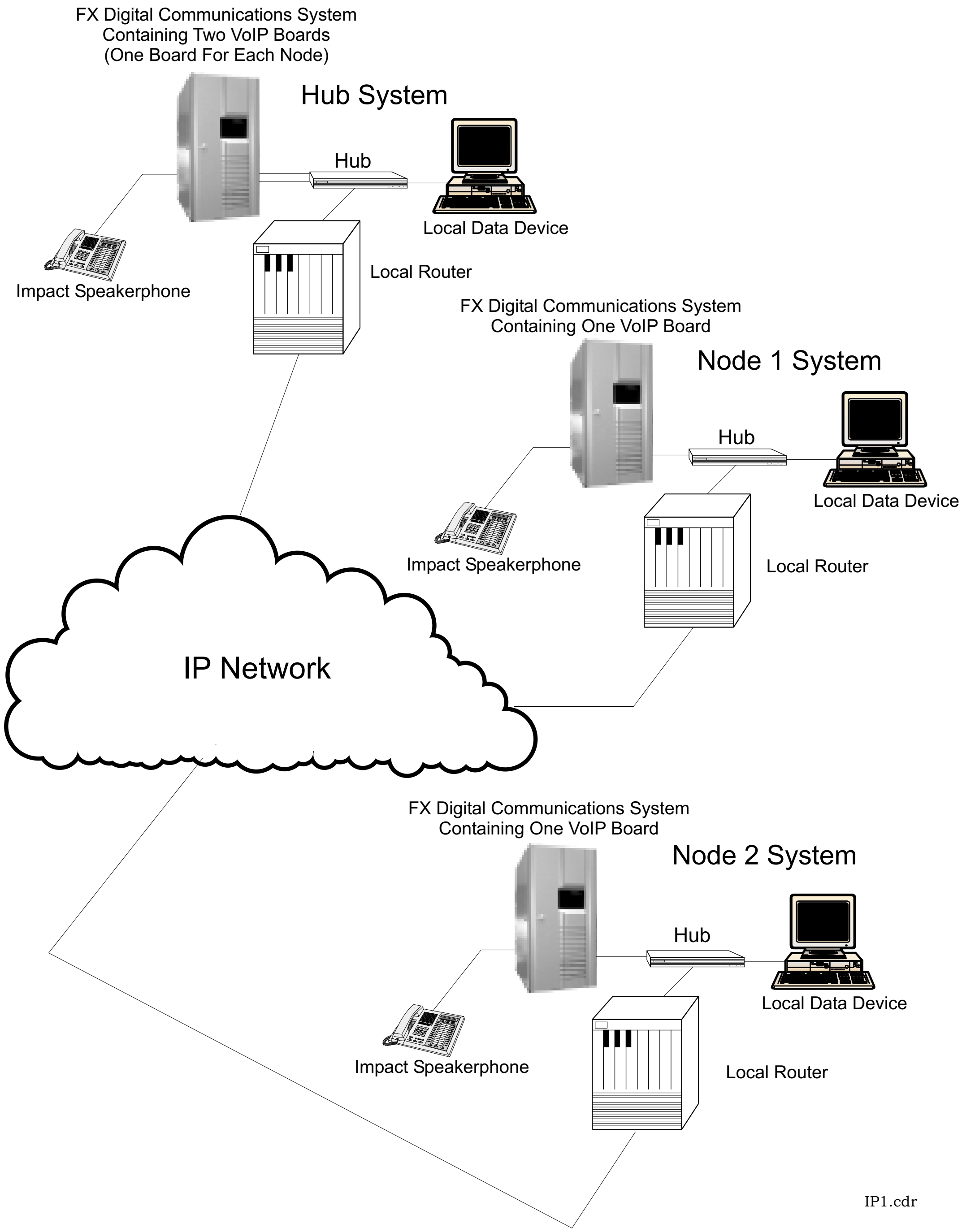
Currently there are two configurations of the VoIP board as defined in the following table:

Detaining the VoIP Configurations

Part Number	Description	Board ID	Voice Channels	Digital Signal Processor (DSP) Circuits
FXVOIP-S	12-Port VoIP Network Board	147	0-12	4
FXVOIP-L	24-Port VoIP Network Board	148	0-24	8

The current Comdial system-to-system networking scheme dictates that the hub system contain at least one VoIP board for each node participating in the IP telephony arrangement. Each node system must contain at least one VoIP board to communicate with the hub system.

Each VoIP board has its own address for incoming messages. While each VoIP board provides one end of a hub-to-node connection, you may need multiple boards to make the needed hub to node connections in installations involving high network traffic.



Visualizing a VoIP Network

Board Status

The system manages a heartbeat scheme between the hub and the IP network nodes. The purpose of the heartbeat is to allow the call processing from the hub to determine if node call processing is still operational. The system sends the heartbeat message through every VoIP board connected between the hub and a node. Sending the message over every board at the same time allows the hub system to detect an individual board disconnect at the node and take its counterpart out of service at the hub. The hub starts database synchronization as soon as it detects a no-response to a heartbeat signal. (The hub also starts database synchronization if the VoIP board reports a red alarm or if someone resets the board.)

Installing the VoIP Board

Complying with Underwriters Laboratories Regulations

Per The Underwriters Laboratories regulation 1950, be aware of the following precautions when installing telephone equipment that is to be directly connected to the telephone company network:

- never install telephone wiring during a lightning storm,
- never install telephone jacks in wet locations unless the jack is specifically designed for wet locations,
- never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface,
- use caution when installing or modifying telephone lines,
- avoid using a telephone (other than a cordless type) during an electrical storm—there may be a remote risk of electrical shock from lightning,
- do not use the telephone to report a gas leak in the vicinity of the leak.

Considering Electrostatic Discharge

CAUTION

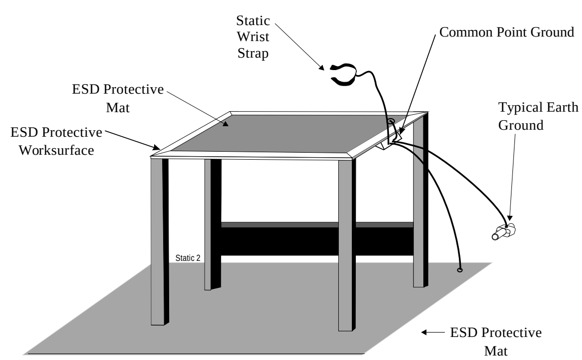
VoIP circuit boards are susceptible to damage caused by electrostatic discharge, and you must keep this fact in mind as you handle the circuit boards. Refer to the Comdial publication IMI01-005, Handling Of Electrostatically Sensitive Components, for general information Specific handling precautions are also included in this installation instruction.

When removing or installing circuit boards in the FX cabinet, you must install a static discharge wrist strap on your bare wrist, and adjust it for a snug fit. Be sure that the strap is touching bare skin and is not isolated by clothing. Connect the wrist strap cord between the wrist strap and a AC or earth ground.

Whenever you remove a circuit board from the cabinet, immediately place the board in a static protection bag while you still have your wrist strap in place and properly grounded.

Creating a Static-Safe Work Area

When removing circuit boards from an installation location for servicing, always transport them to a static-safe work area in static protection bags. If you do not already have a static-safe work area, you can create one by arranging a work area as detailed in the illustration.



Viewing the Static-Safe Work Area

Making the VoIP Board Installation

You can install the VoIP board in any available universal slot in any available FX system cabinet and connect it directly to the network router or to a WAN connection.

CAUTION

If you install the VoIP board in an FX expansion cabinet, you must insure that the SCSI interface cards (FXSOPT-SCX-1), which mount on the services board and provide the expansion cabinet interconnection, are at revision B or later.

Install the VoIP board per the following instructions:

1. Normally you should disconnect the AC power cord from the AC outlet and disconnect the optional battery back-up assembly from the power supply; however, when necessary, you can remove or install board in an operating system. If you must do this, connect one end of a standard telephone handset coil cord to the precharge port on the power supply chassis (on FXS systems— on FXT systems, the pre-charge cord is pre-wired to the power supply and hangs between it and the circuit board cage).
   During step 5, you will connect the other end of this coil cord to the precharge jack on the VoIP board.
2. Install your static discharge wrist strap on your bare wrist; adjust it for a snug fit. Be sure that the strap is touching bare skin and is not isolated by clothing. Connect the wrist strap cord between the wrist strap and an AC or earth ground

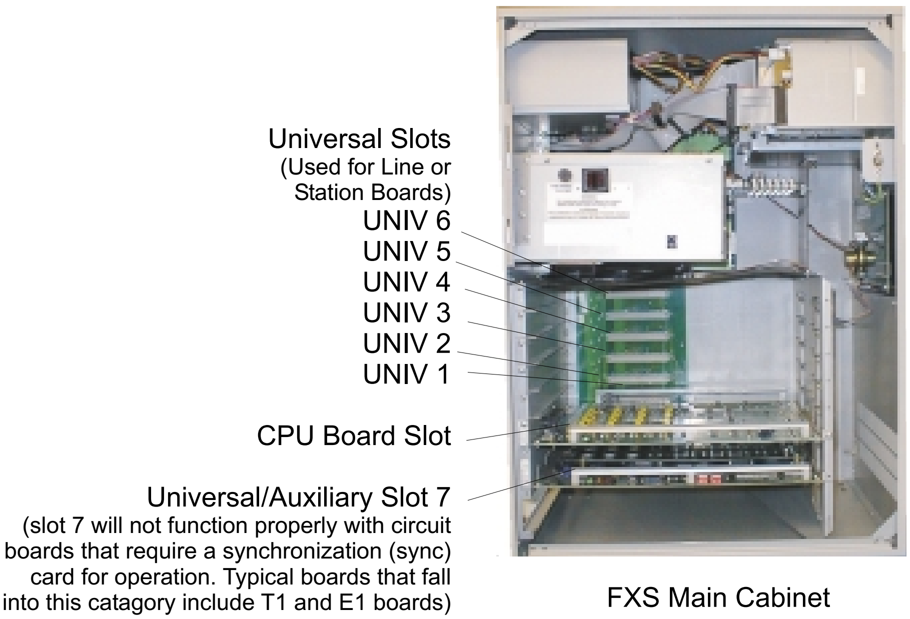
NOTE: With the common equipment in the installed position, the ground lug on the side of the cabinet is an appropriate grounding point since it should have a heavy ground wire connected between it and a good earth ground.

3. Each new board is supplied in a static protection bag for safe keeping. When you are ready to install the line board, remove it from its static protection bag. Conversely, when you remove a line board from the cabinet, immediately place it in a static protection bag.
4. If you are installing the board in an operating system, connect the free end of the precharge cord that you installed in step 1 to the precharge jack on the board. The precharge port is located along the left front edge of the VoIP board and labeled as such.

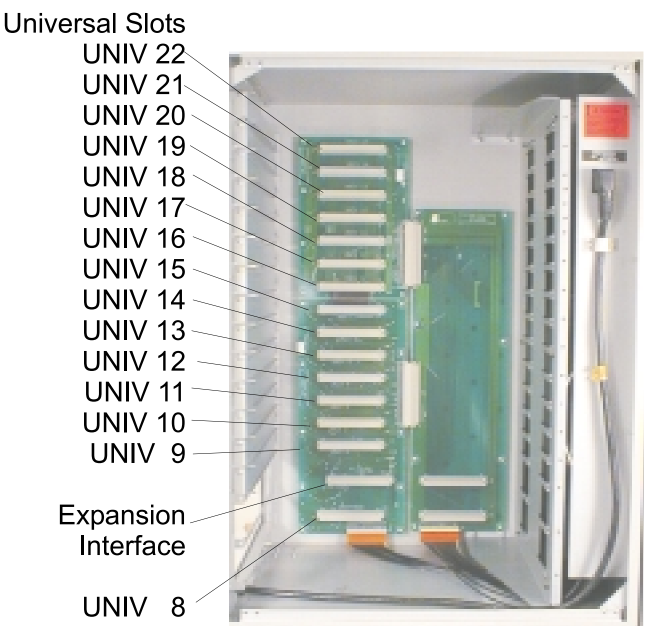
CAUTION

When installing a board while power is on, use the pre-charge cable and be sure to connect the cable to the labeled pre-charge port. DO NOT mistakenly use the serial data port that is located along the board's right front edge.

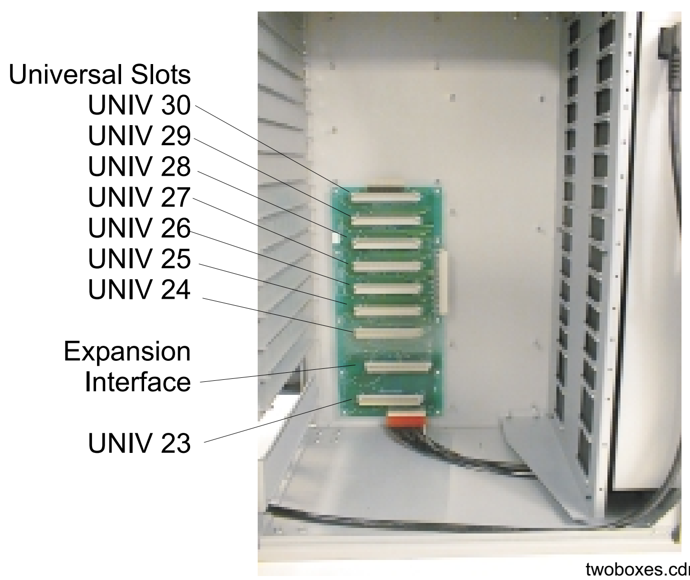
5. You can install the VoIP board in any universal slot either in the main cabinet or in any expansion cabinets that are part of the system. To install the boards, orient them with the left and right guides in cabinet board cage. and press the boards firmly until their board edge connectors properly mate with the connection on the cabinet's backplane.
   (If you are removing boards from the cabinet, remove the retaining hardware, pull the boards toward you until their board edge connectors separate from the cabinet's backplane connection, and slide the boards free of the cabinet.)
6. Make a final inspection to ensure that all boards are oriented correctly and mated properly then secure them to the board cage.



FXS Main Cabinet

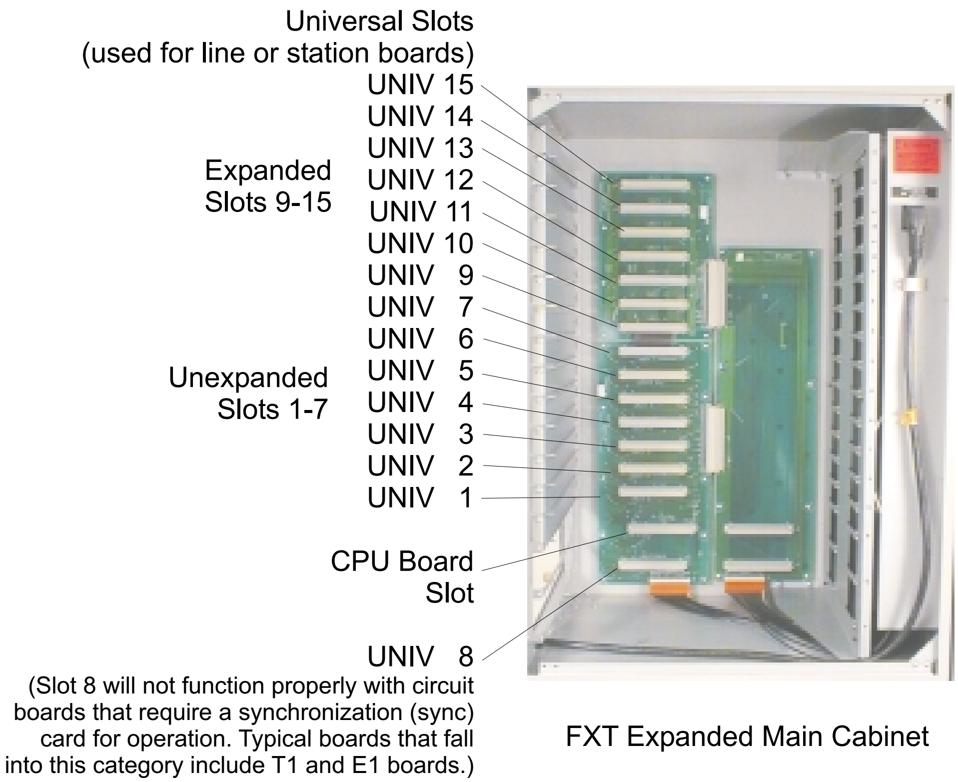


FXS First Expansion Cabinet

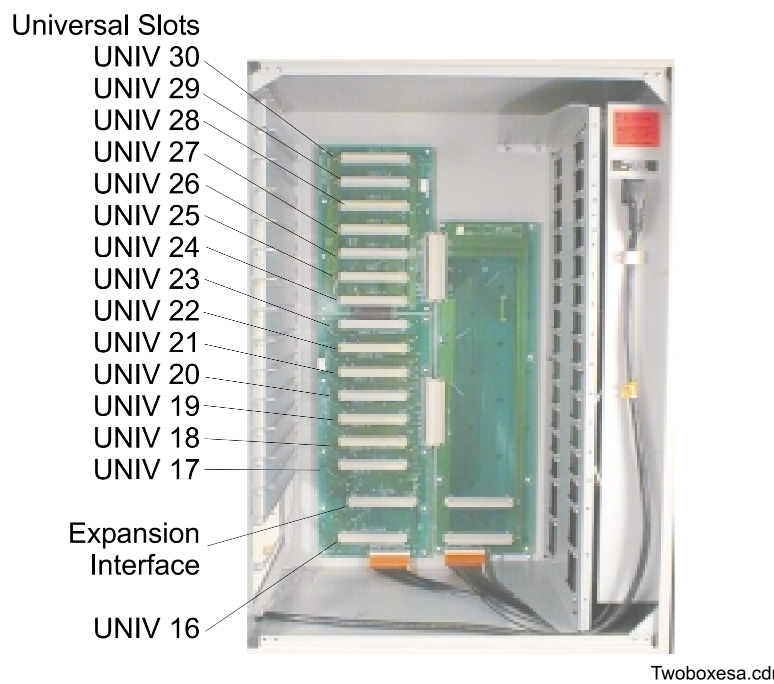


FXS Second Expansion Cabinet

Locating the FXS Circuit Boards

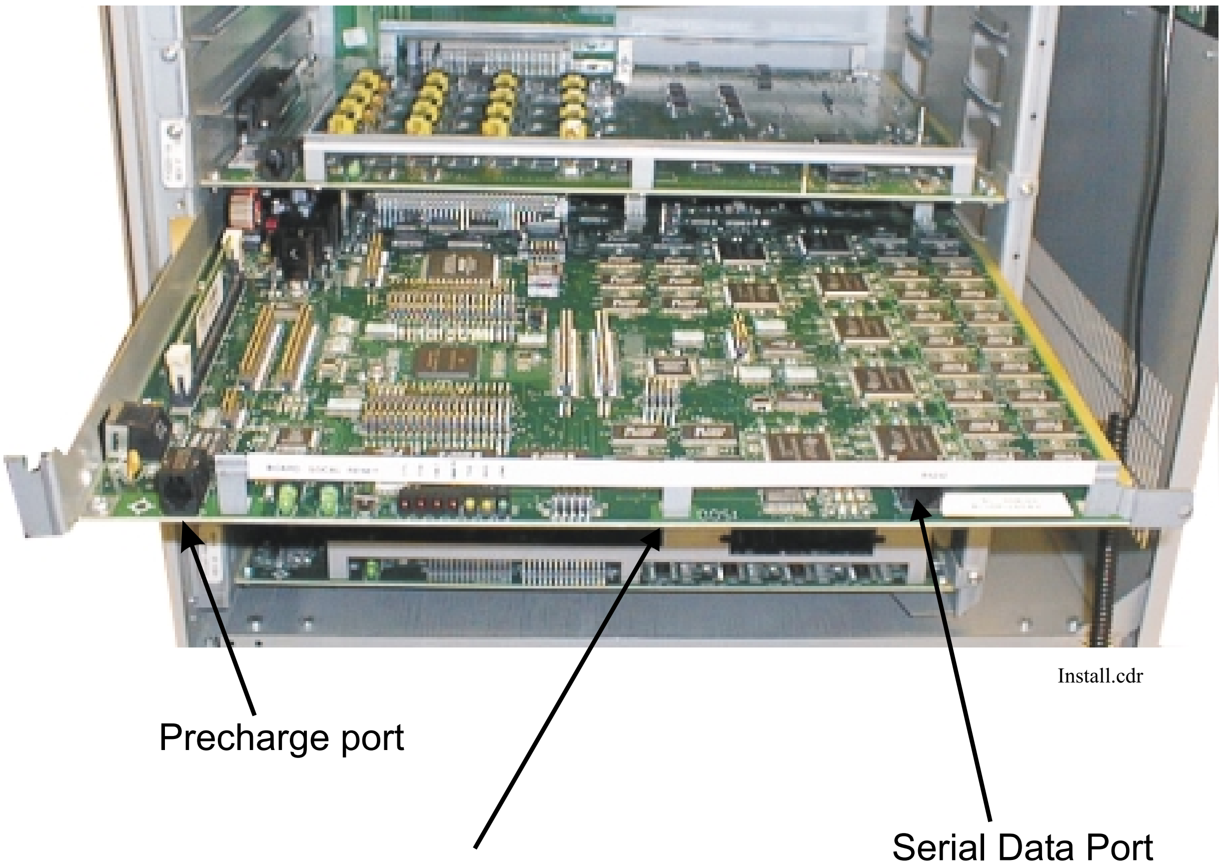


FXT Expanded Main Cabinet



FXT Expansion Cabinet

Locating the FXT Circuit Boards



1. Install the VoIP board in any universal board slot.
2. Seat the board firmly.
3. Attach the board to the cabinet in three places with the supplied hardware.

Installing the VoIP Board

Making the VoIP Interconnection

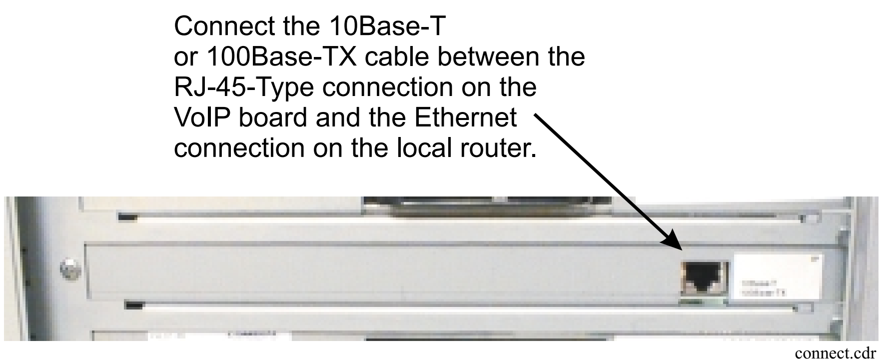
The connection between the VoIP board and the network is by either 10Base-T at 10 Mbps or 100Base-TX at 100 Mbps Ethernet connection using a category 5 cable. Comdial installation specialists suggest that you use 100Base-TX wiring to achieve the best operating results. The VoIP board connection is a RJ-45-type connector, which is an eight-pin connector used for data transmission over standard, twisted-pair telephone wire.

Interconnection consists of connecting a customer-supplied cable between the jack on the VoIP board and the Ethernet connection on the networking equipment (the router, hub, and so forth).

There are no other electrical connections such as straps or jumpers that you need to make.

NOTE: There is a modular connector on the side of the VoIP board that is labeled as an RS232 connection. It is a serial connection reserved exclusively for fail-safe programming. See the discussion titled Reviewing Fail-Safe Programming *shown on page 22 for connection details.*

Once the connections are complete and the equipment is fully operational, check the VoIP board's status lights for proper indication. Refer to the Troubleshooting information on page 31 for complete status light details.

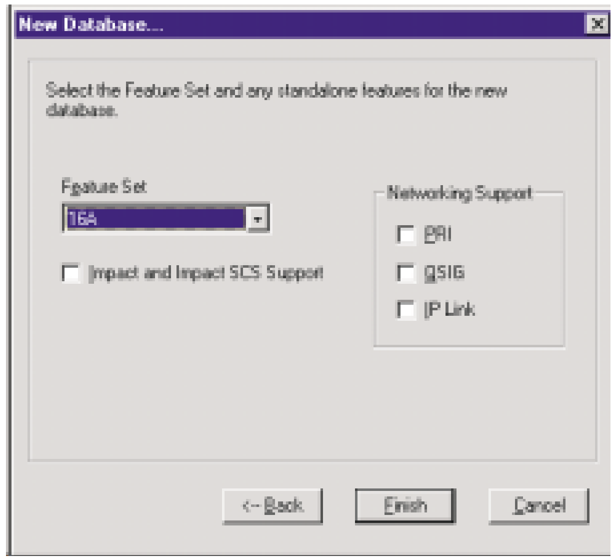


Making the VoIP Connection

Configuring the System for VoIP Board Operation

Setting the IP Link Networking Support

If you begin system configuration by creating a new database for the communications system, VMMI prompts you as to what features to enable. If you plan to operate a VoIP board in a system networking environment, you must select the *IP Link* box on the new database dialog.



Viewing the New Database Dialog

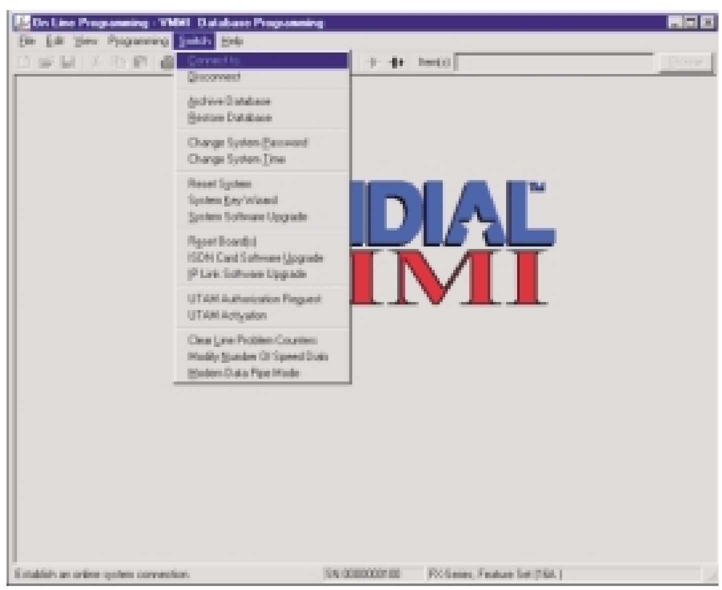
Setting the IP Link Channels

The software key that enables the system operating software and the CTI applications for the FX system also sets the number of IP channels that you can enable or modify on the VoIP board.

Once you have made the necessary arrangements with the Comdial representative and received your software key, you can use the System Key Wizard to enable or modify the IP channel.

You can also modify or enable IP channels at the same time that you use the System Software Upgrade feature to download a system software up-grade to the FX digital communications system.

Both the System Key Wizard and the System Software Upgrade are available from the VMMI Switch menu.



Locating the VMMI Switch Menu

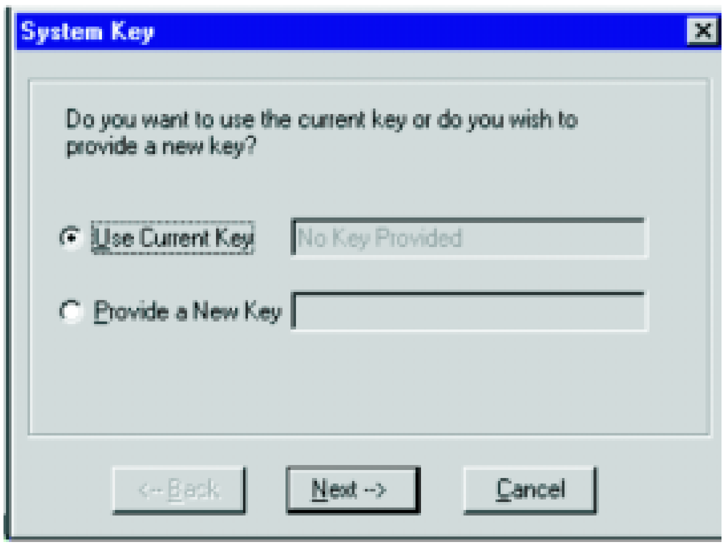
The System Key Wizard provides a summary of keyed channels including an indication if the configured channels exceeds the number allowed by the software key. Also refer to the *Number of Channels* discussion on page 13.

Entering or Modifying IP Channels

To enter or modify the software key for IP channel content without also up-grading the FX system software, perform the following procedure:

1. Navigate to the Switch/System Key Wizard and click open the dialog box. From there you can enter a new software key number or examine the features enabled by the current software key.
2. Click *Next* to examine the features that the software key enables. If you provided a new key and the features are correct, click *Finish* to set the key on the system.

NOTE: The system resets itself when you change the key.



NOTE: This screen shot shows an example of the Key Wizard after you have entered or modified IP channels.

Entering or Modifying IP Channels

Configuring Parameters for VoIP Board Operation

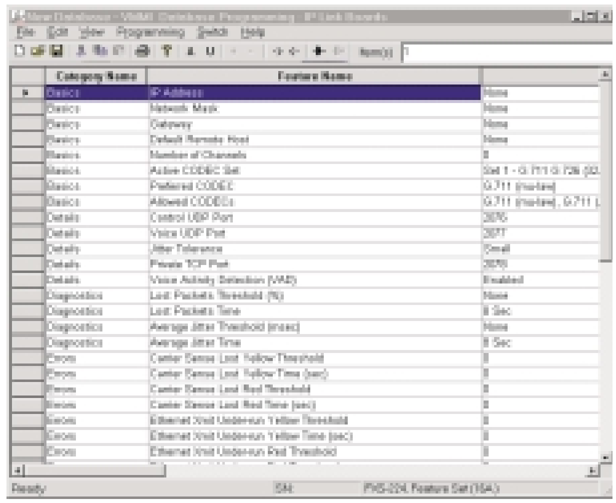
When you add VoIP boards to the FX system, you must use the VMMI programming utility to take several configuration steps to make the VoIP boards operational. It is very important to remember that reconfiguring a VoIP board results in the termination of all active calls on that board.

Setting the VoIP Configuration Parameters

You must set the operating parameters for each VoIP board. From the VMMI main menu, select Programming/Boards Menu/IP Link Boards. From this dialog, choose the desired parameters for the VoIP board.

It is very important that you set the values for the following board parameters to be the same at both ends of the IP link:

- Active CODEC set
- Preferred CODEC set
- Control UDP port
- Voice UDP port
- Private UDP port
- Packet Size



Reviewing the VoIP Configuration Parameters

Setting Basic Board Options

IP Address

The IP Address item defines the address of the VoIP board on the IP network. It is a four-byte field, presented in decimal *dotted-quad* format (that is; nnn.nnn.nnn.nnn where nnn represents one, two or three-digit numbers.) The default value is 0.0.0.0 (no address). The VMMI software prevents the IP address from being a class D, class E, loopback reserved, broadcast, or network address as defined by industry standards RFC 791 and RFC 1122. The network administrator assigns the IP address that you enter here. The system will not operate without an address. (For more details about IP addresses, see the discussion on page 37.)

Network Mask

The network mask defines the portion of the IP address that refers to the network. It is a four-byte field, presented in decimal *dotted-quad* format. Acceptable values range from 255.0.0.0 to 255.255.255.252. The default value is 0.0.0.0 (none). The network administrator assigns the network mask value that you enter here. (For details about the network mask part of the IP address, see the IP address discussion on page 37.)

NOTE: The VMMI software prevents the combination of IP address and network mask from forming a broadcast address.

Gateway

This value represents the LAN address of the gateway, or router, on the LAN where the VoIP board resides. It is a four-byte field, presented in decimal *dotted-quad* format. The default value is 0.0.0.0 (none). The network administrator assigns the gateway value that you enter here.

Default Remote Host

NOTE: The Default Remote Host parameter is not available when the Netorking or Phone *option is set for* IP Phone Only.

Configure the IP address of the far-end networking link board with this entry. It is a four-byte field, presented in decimal *dotted-quad* format. The default value is 0.0.0.0 (none). The network administrator assigns the remote host address that you enter here. If you do not have a default remote host, no communications can occur.

Number of Channels

This programming entry specifies the number of voice channels to be mapped for the board. For FXVOIP-S boards, when loaded with CODEC sets 3 and 4, the number of channels range from 0 to 12. For FXVOIP-L boards, when loaded with CODEC sets 3 and 4, the number of channels range from 0 to 24 (0 to 22 if G.711 is the active CODEC set and VAD is not enabled). The FXVOIP-S board provides four DSP circuits and the FXVOIP-L board provides eight DSP circuits. Notice from the discussion titled *Active CODEC Set* on page 14, CODEC sets 3 and 4 support three channels per DSP circuit. This specification determines that the FXVOIP-S provides 3 x 4 = 12 channels and the FXVOIP-L provides 3 x 8 = 24 channels. The default is 0 channels.

NOTE: The total number of IP networking voice channels is keyed. The number of keyed channels forms an available channel pool. Each time you configure a VoIP board with a number of channels, you decrease the software pool by the same amount. The software prevents you from bringing into service any more IP channels than the amount keyed. If you provision more IP channels than the software key allows, the System Key Wizard configuration screen informs you of this fact. See page 11 for an example of a typical configuration screen.

After you program the voice channels for the VoIP board, the software automatically maps the channels to corresponding logical line ports if the VoIP board is operating in the *Networking* channel mode or logical station ports if the VoIP board is operation in the *IP Phone Only* channel mode. (See the discussion titled *Networking or Phones* on page 14.)

You can use VMMI to verify this mapping by navigating to the Programming/Boards Menu/Line Port Location or Station Port Location menu. Once there, search for the VoIP board and examine the line or station port information presented for that board. The software assigns the FX system's logical ports based on the number of voice channels that you have configured on each VoIP board. Since the VoIP Boards default to having zero voice channels, the system maps no ports when you install the boards. **You must configure voice channels for system use to cause the port mapping to take place.**

Active CODEC Set

The active CODEC set is one among several pre-defined sets described in the following chart. The default value is Set 3. Refer to the charts on page 41 for CODEC packet size and bandwidth specifications.

Listing the Supported Coding Sets			
CODEC ID	CODEC Description	Available CODEC	Channel/DSP
3	Set 3 - includes G.729AB with long echo tail	G.711 mu-law (64Kbps), G.711 A-law (64 Kbps), G.726 (32Kbps), G.726 (16Kbps), and G.729AB (8Kbps)	3
4	Set 4 - includes G.723 with long echo tail	G.711 mu-law (64Kbps), G.711 A-law (64 Kbps), G.726 (32Kbps), G.726 (16Kbps), G.723 (6.3Kbps), and G.723 5.3Kbps	3

NOTE: The set selection that you make here limits the choices for the Preferred CODEC and Allowed CODECs features to the choices allowed by that selected CODEC set.
The particular DSP package that has been downloaded to the VoIP board limits the available CODEC sets. DSP package 2 contains CODEC sets 3 and 4.
See the Chapter titled Understanding CODECs *found on page 39 for more CODEC discussion.*

Preferred CODEC

This parameter defines the CODEC that the system uses for all outbound calls on the board. Choices are determined by which CODEC set is active. The default is the first available CODEC in the Active CODEC set. Refer to the charts on page 41 for CODEC packet size and bandwidth specifications.

Allowed CODECS

This list defines the allowable CODECs the system can use for outbound calls on the board. A homogeneously configured system will always use the Preferred CODEC. If you have programmed the Preferred CODEC differently on connected systems, you must choose a CODEC that both systems can use. A system may use either a Preferred or an Allowed CODEC. There is no default value for this feature.

NOTE: The software includes this feature for future use. For now, the value list will be empty because the preferred CODEC set is always used.

Networking or Phones

The VoIP board supports either Networking or IP speakerphone operation. In the *Networking* mode, several FX systems can network together into one communications system. In the *IP Phone Only* mode, the VoIP board provides interfacing for IP speakerphones that operate with the FX system through an IP network.

Setting Detailed Board Options

The following paragraphs discuss the detailed board options and explains their value choices. Normally, you need not change these options from their default values.

Control UDP Port

This parameter value defines the User Datagram Protocol (UDP) port that the VoIP board uses for passing call control information between systems. You must insure that this value is identical on connected boards. The parameter value may range from 1024 to 65535 with a default value of 2076. Network administrators may need to assign the programming value to this port to allow for data transfer across fire-walls.

Voice UDP Port

This value defines the UDP port that the VoIP board uses for passing voice packets between systems. You must insure that this value is identical on connected boards. The value may range from 1024 to 65535 with a default value of 2077. Network administrators may need to assign the programming value to this port to allow for data transfer across fire walls.

Private TCP Port

This item defines the TCP port the board uses for passing some non-voice call related networking messages between systems. You must program this item to have the same value for connected boards. The programming range is:1024 to 65535 with a default value of 2078. Network administrators may need to provision this port for transfer across intervening fire walls.

Voice Activity Detection

Voice Activity Detection (VAD) reduces the bandwidth required for calls. It does this by sending packets to the far-end only when speech has been detected at the near-end. This action can result in slight clipping at the beginning of speech, but usually provides better than 50 percent bandwidth reduction for G.711 and G.726. You can set VAD to be on or off with the default being on.

IP Precedence

The RFC 791, *Internet Protocol* defines values for use in the IP header's *Type of Service* field to optionally indicate varying levels of precedence at the packet level. The precedence setting may range from 0 to 5 with 5 indicating high-priority. The default value is 5.

FAX/Modem Switchover

NOTE: The FAX/Modem Switchover parameter is not available when the Netorking or Phone *option is set for* IP Phone Only.

When the VoIP board detects FAX or modem tones, it switches to a CODEC defined expressly for use with these types of data calls. Since this CODEC may consume more bandwidth than the Preferred CODEC, you may need to disable automatic switchover when operating on low-bandwidth links. FAX/Modem Switchover may be on or off with the default value being on.

Packet Size

The Packet Size setting controls the size of the voice information field in outgoing packets. Choices are small and large. Smaller packets reduce the delay in the voice path but increases the data space required for the protocol headers. Less delay leads to better voice quality at the expense of overall bandwidth. Large packets increase voice path delay slightly but lower the protocol header space. The default value is large. Refer to the charts on page 41 for CODEC packet size and bandwidth specifications.

NOTE: If the system loses packets when using a large packet size, the effect is more noticeable than when using a small packet size. The effect becomes more noticeable because large packet size means more voice data per packet.

Setting the Quality of Service Parameters *(Diagnostic Options)*

Quality of Service (QoS) parameters represent data that describes the performance of the VoIP board. QoS data gives real time information about the quality of the inbound voice, FAX, or modem data streams coming from the IP network for a given VoIP call. This information is available from the system software for any VoIP board participating in a call. Automatic delivery of QoS data is based on a set of programmable rules or conditions referred to as *traps*. When a trap condition is met, the system stores an error in the digital trunk error log.

You can employ the optional Network Management System (NMS) feature to monitor the VoIP board's QoS information. Complete NMS operational details are available in GCA40–237, *Comdial Network Management System User Instructions*.

You can set the diagnostic options for detecting QoS problems for each board from the *IP Links Board* dialog.

Lost Packets Time and Lost Packets Threshold

The software averages the number of lost packets over the configured time for each call. When the percentage of lost packets exceeds the configured threshold, the board generates a Lost Packets Trap. You choose the time from the provided choices and choose the percent in one percent increments. Recommended values for useful quality of service information are eight seconds for the time and 10 percent for the threshold.

Average Jitter Time and Average Jitter Threshold

Over the configured time for each call, the software averages out the jitter level that results from the different network-provided waiting times allowed for the different packets of data . When the average jitter time exceeds the configured threshold, the board generates an Average Jitter Trap. Recommended values for useful quality of service are eight seconds for the time and 500 milliseconds for the threshold.

Noting the Alarm Conditions *(Error Parameters)*

Low Level diagnostics (alarm conditions) represent data that describes the performance of the VoIP board. The system handles low level diagnostics in a slightly different manner from the QoS parameters described above. The lower level layers on a VoIP board are related to the physical layer of the IP network. The lower level diagnostic errors reported are either yellow or red alarms depending on the severity and duration of the condition. Yellow alarms indicate temporary problems, while red alarms indicate major problems that disrupt speech. The same error can create a yellow or red alarm. Yellow alarms call for a lower sense of concern than do red alarms; therefore, you should program yellow alarms to activate before red alarms. Just as for QoS, you need the optional NMS feature to monitor the VoIP board's low level diagnostics.

You must configure separately the time and threshold for each condition's yellow and red alarm. You choose the time from the provided choices and the threshold from a weighting value between 0 and 999. The threshold value represents a tolerance level for the conditional alarms. Some operating conditions may be stable and quiet and you can tolerate a high threshold limit while other situations may require that you maintain a much lower tolerance level.

Set the error parameters for the alarm conditions for each board from the *IP link Boards* dialog.

- **Carrier Sense Lost During Transmit**—If the count of carrier sense losses during frame transmission exceeds the configured threshold during the configured time, the board generates an alarm.
- **Ethernet Transmission Under-run**—If the count of transmission buffer under-runs exceeds the configured threshold during the configured time, the board generates an alarm.
- **Ethernet FIFO Over-run**—If the count of FIFO over-runs exceeds the configured threshold during the configured time, the board generates an alarm.
- **Retransmit Limit Expired**—If the count of retransmission attempt limit expirations exceeds the configured threshold during the configured time, the board generates an alarm.
- **Late Collision**—If the count of late collisions exceeds the configured threshold during the configured time, the board generates an alarm.
- **Ethernet Receive Buffer Busy/Full**—If the count of buffer busy errors exceeds the configured threshold during the configured time, the board generates an alarm.
- **Mis-aligned Frame Received**—If the count of non-byte-aligned frames received exceeds the configured threshold during the configured time, the board generates an alarm.
- **CRC Error**—If the count of CRC errors exceeds the configured threshold during the configured time, the board generates an alarm.

Upgrading the VoIP Board Software

The VoIP board is *plug-and-play* in the sense that, other than the VMMI configuration settings, there are no system-level programming requirements. As Comdial revises the VoIP board's operating software from time to time, you may find that you must up-grade the software at your installation. When you up-grade the software, you are storing new information in the VoIP board's flash memory.

At start up, the VoIP board copies its flash memory information into its RAM memory and then operates from the information it stored in the RAM.

Understanding the Software Up-grade Requirements

The software supports two downloading methods for making software up-grades to the VoIP board: Downloading over the system backplane wiring (Dual Port Random Access Memory—DPRAM) and downloading through the Ethernet connection (TFTP).

The DPRAM method is the typical serial port arrangement that you traditionally use to download system software up-grades to the FX digital communications system. It is slow and downloads software to one VoIP board at a time because the signalling uses the equipment cabinet's backplane wiring.

The TFTP method downloads the software up-grade from the PC into the VoIP board through the VoIP board's IP connection. This method provides a faster down-loading scheme. Because of the speed advantage, the IP down-loading scheme is the preferred method.

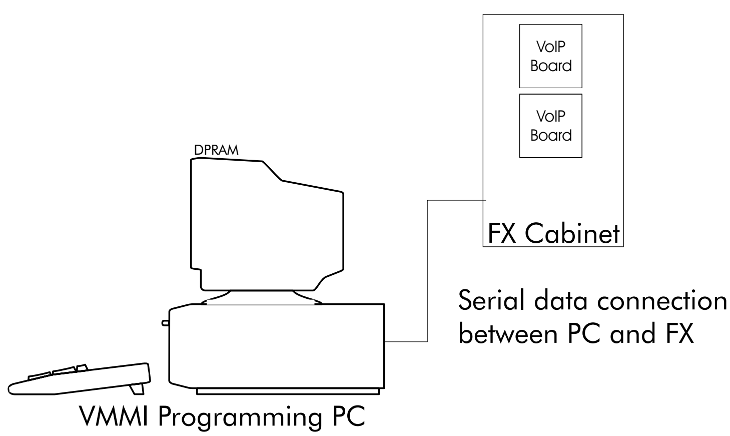
NOTE: While you can perform the VoIP software up-grade remotely through a modem and the IP network, usually you do this up-grade programming while on site and with your PC directly connected to the communications system and the VoIP board or the LAN.

There are two different techniques that you can use when up-grading the software with the TFTP method—the LAN connection and the direct VoIP board connection.

- If the site's network administrator grants you access for your programming PC on the site's LAN, connect the programming PC's network board to the LAN with a standard 10Base-T or 100Base-TX cable and execute simultaneous multiple board programming.
- If you do not have access to the site's LAN, make a direct connection between the programming PC's network board and the VoIP board's IP connection with a 10Base-T or 100Base-TX crossover cable. This arrangement permits programming of one board at a time.

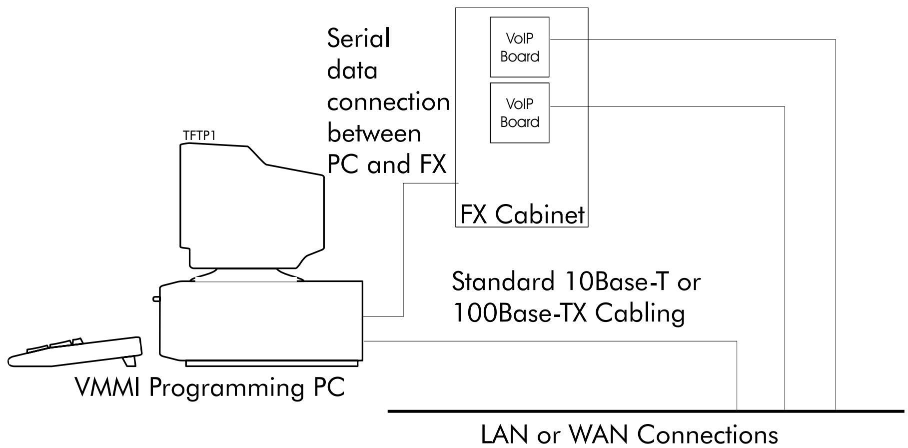
Preparing for the Up-grade

If you are using the DPRAM method, connect your PC's serial port to the FX digital communications system's serial data port and configure the communications settings appropriately.



Configuring the DPRAM Method

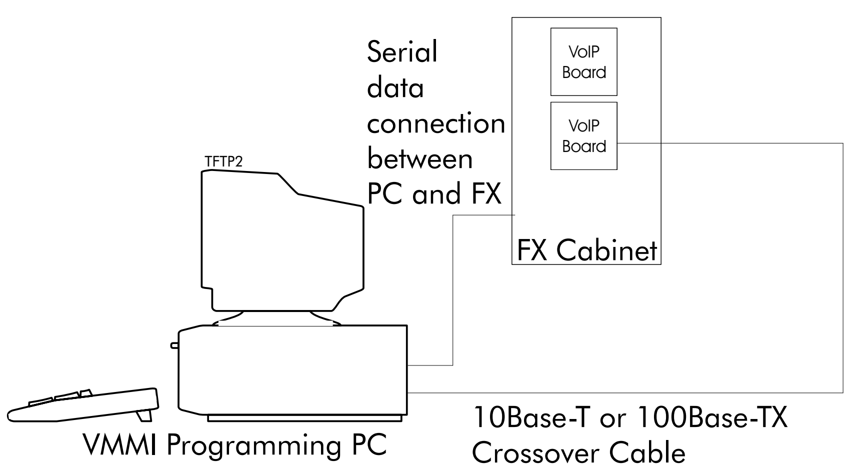
If you are using the TFTP method that includes LAN privileges for your programming PC, consult the site's LAN administrator for the appropriate network settings and permissions. Make the necessary parameter settings on the programming PC, and connect it's network board to the LAN using a standard 10Base-T or 100Base-TX cable. Also, you must connect your PC's serial port to the FX digital communications system's serial data port and configure the communications settings appropriately.



Configuring the TFTP Method with LAN Privileges

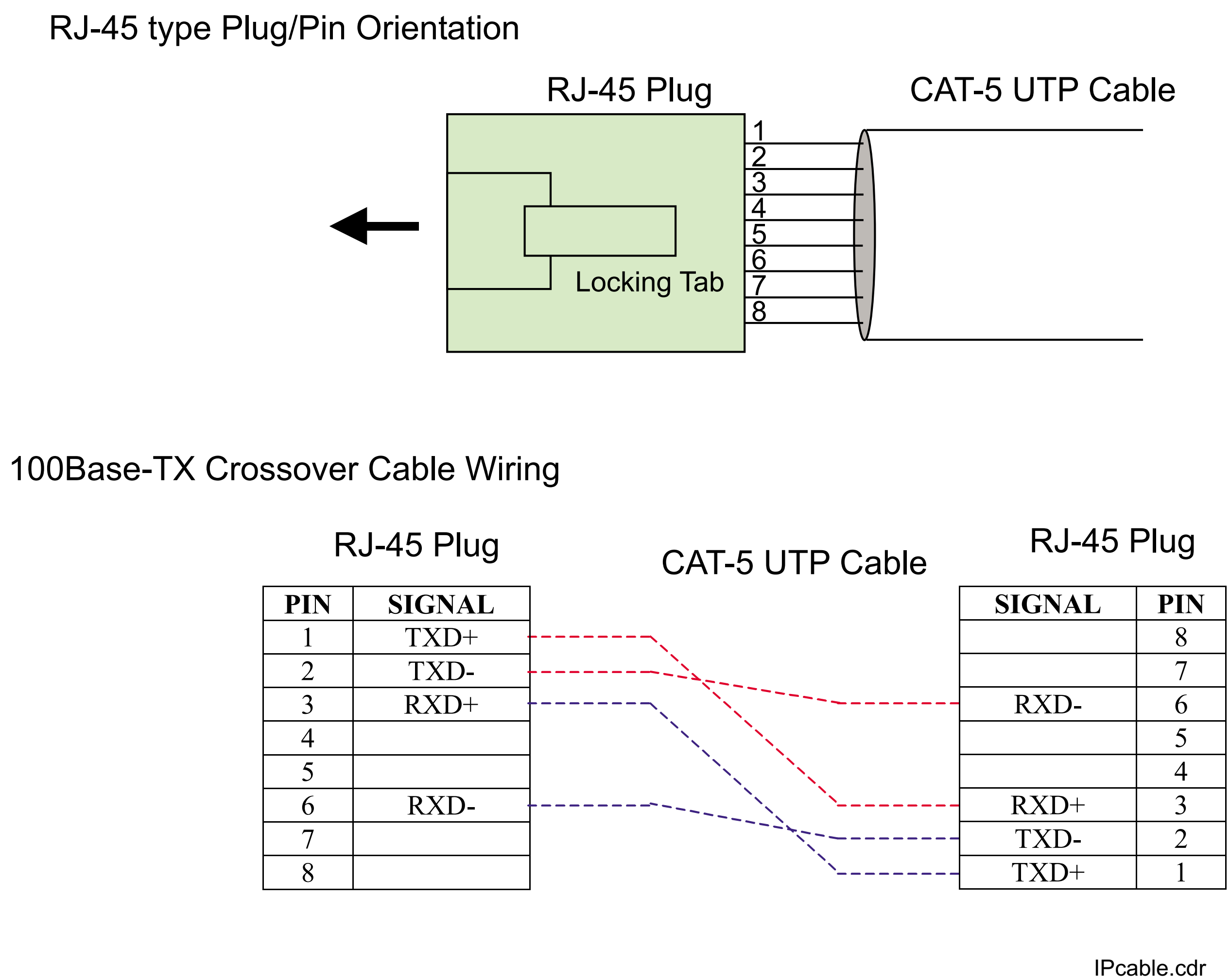
If you are using the TFTP method without LAN privileges for your programming PC, connect your PC's network board to the VoIP board's IP connection. Remember, using a direct crossover connection precludes downloading to multiple boards simultaneously. You need to supply (either by purchase or construction) a 10Base-T or 100Base-TX crossover cable for this connection.
— An acceptable off-the-shelf cable that you can purchase is the Belkin* part number: A3X126-10-YLW-M.
**Belkin Components, Compton, CA 90220.*
— If you wish to construct your own cable, wire it per the diagram shown on page 20. Alternately, you may use a 100Base-T4 crossover cable if you have one available.

Also, you must connect your PC's serial port to the FX digital communications system's serial data port and configure the communications settings appropriately.



Configuring the TFTP Method without LAN Privileges

Wiring the IP Crossover Cable



Wiring the IP Crossover Cable

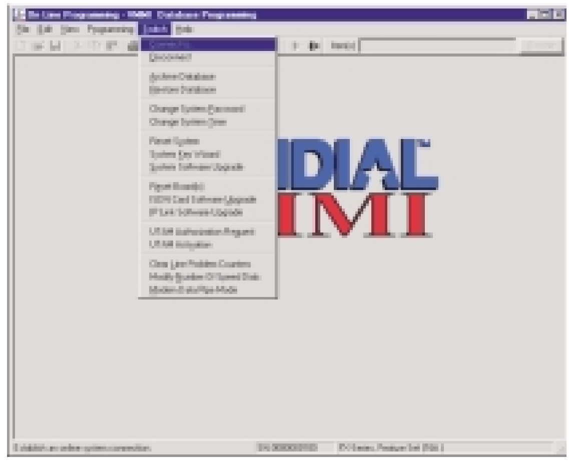
Downloading the Software Up-grade

The software up-grade consists of three files: the main application firmware; the different CODEC sets (the DSP images); and the bootloader file. Download the software up-grade in the following manner:

CAUTION

The software download sequence takes several minutes (maybe up to 10 minutes for TFTP and possibly up to 50 minutes for DPRAM) to complete. Once complete, VMMI displays a prompt to that effect.

1. Navigate to the VMMI Switch/VoIP Software Upgrade menu.



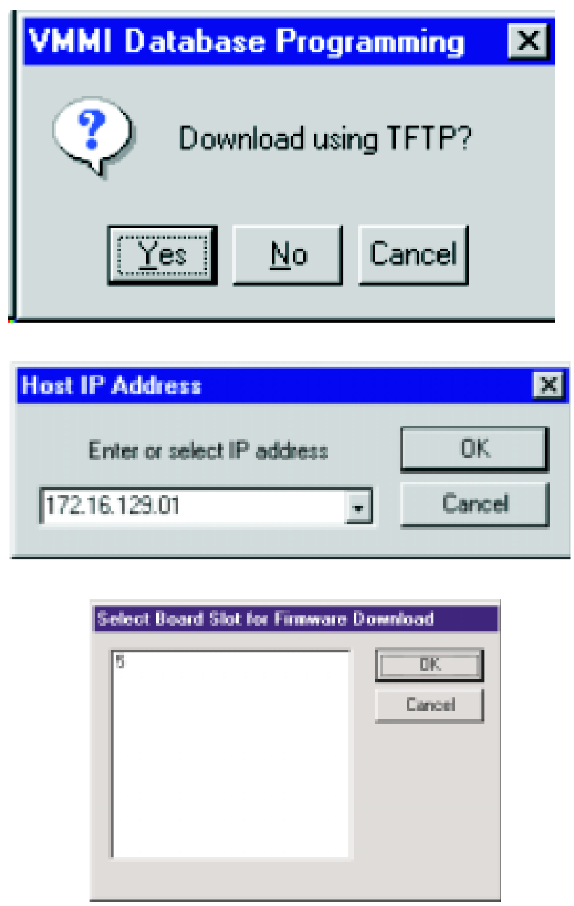
Locating the VMMI Switch Menu

2. Select the IP Link Software Upgrade line from the Switch menu. The resulting dialog offers you the option to use TFTP method of software download. Select *Yes* to employ the TFTP method or *No* to employ the DPRAM method.

3. If you select *Yes* to employ the TFTP method, a dialog prompts you for the IP address of your PC. Click **Ok** to use the PC's address.

   If you select *No* to employ the DPRAM method an therefore not send the up-grade through the IP interface, the software does not present the address dialog box.

4. The software prompts you to select the boards by slot location that you plan to up-grade. Remember, LAN connected PC and VoIP boards permit simultaneous multiple board programming while direct PC to VoIP board connection permits programming of one board at a time.



Downloading the Software Up-Grade

Downloading the Software Up-grade—continued

5. The software next prompts you to select a valid image file. This is the software up-grade file that you have previously obtained from your Comdial source and stored on your PC.
6. If you have chosen to use the TFTP download method, the screen first presents a busy cursor (the hour glass icon) until all board up-grades are complete. It then presents an up-grade summary message. With the DPRAM download method, the screen presents a progress bar followed by an up-grade complete message.
7. After the download is finished, wait for the board to reset then check the VoIP board's status lights for proper indication (see the status light details on page 31) and make a test call through the network. If the board seems inoperative, perform the software download sequence again. Because of the manner in which the software download operates, should a signal interruption occur at the short period of time that the software is loading in the VoIP board's flash memory, that signal interruption could leave the VoIP board in an inoperative state. **This is a highly unlikely occurrence**. However, in the event that it does occur, there is special Fail-Safe Programming action that you can take at the board location.

Reviewing Fail-Safe Programming

Fail-safe programming is a last resort method that you can use to make a VoIP board operational should the flash memory become corrupted during a software up-grade session. This is a rare occurrence and you will seldom, if ever, be called upon to perform the fail-safe programming procedure. Nevertheless, the fail-safe method will properly reload the application software and the CODEC images should they be corrupted. However, should the Boot ROM become corrupted, you must return the board to the factory for servicing.

NOTE: It is possible that certain future software upgrades may require that the use of fail-safe programming. Should this ever be necessary, documentation accompanying the software will direct you to that effect.

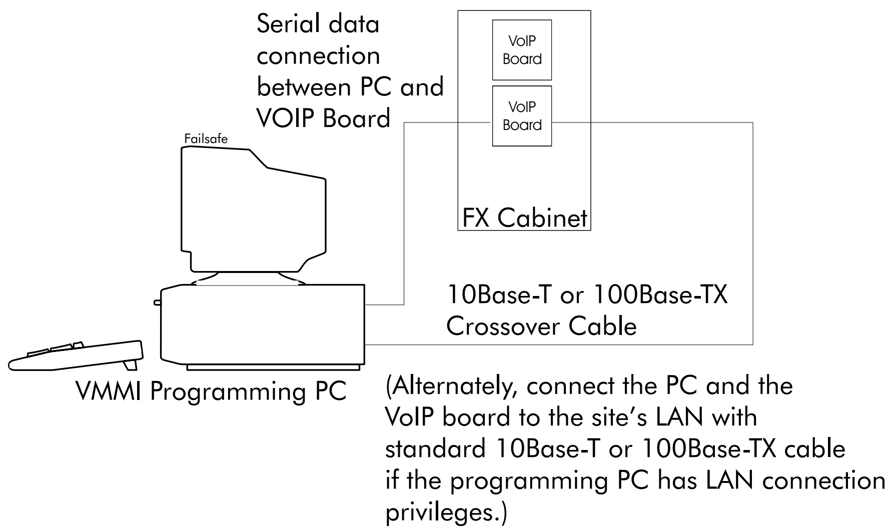
To properly perform fail-safe programming, you must meet the following requirements:

- you must perform this programming at the customer site,
- you must have a TFTP software utility program loaded on your PC (the VMMI can serve this function),
- you must have a terminal communications software utility program loaded on your PC (you can use the utility supplied with your PC or you can obtain terminal communications programs from almost all software supply channels),
- you must have the application up-grade software stored on your PC (obtain VoIP application up-grade software from Comdial sources),
- you must have a network board installed in your PC,
- you must connect the PC directly to the VoIP board's IP network connection with the proper crossover cable (or to the same IP network to which the VoIP board is connected if your PC has an IP address for that network),
- you must connect the PC directly to the serial connection labeled *RS232* on the edge of the VoIP board,
- you must be sure that the VMMI PC is disconnected from the FX system.

Performing Fail-Safe Programming

Setting Up the Equipment

1. Connect the programming PC's network board and the VoIP board or boards to the site's LAN if the PC has LAN privileges. If the PC does not have LAN privileges, disconnect the VoIP board from the IP network and connect the PC's network board connection to the VoIP board's IP network connection—use a 10Base-T or 100Base-TX crossover cable for this connection (see page 19 for cable details).
2. Connect the PC serial connection to the VoIP board's serial data connection (see page 8 for an illustration of its location). Configure the PC's serial communications link for 9600 baud, N81 (no parity bits, eight data bits and one stop bit).



Configuring for Failsafe Programming

3. If the PC is not already connected to the LAN with a valid network address, configure the programming PC's Network TCP/IP Properties to use a static address that is in the same subnet as the VoIP board's address.

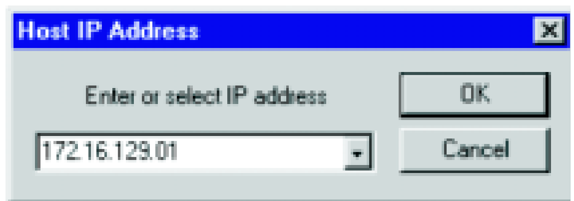
Starting the Programming Procedure

NOTE: If you do not use the VMMI as the TFTP server, start a TFTP server utility software of your choice on your PC. If you do this, you do not need to perform steps 1 and 2

1. Start the VMMI programming utility on the PC, and navigate to the VMMI Switch/VoIP Software Upgrade menu. Select the IP Link Software Upgrade line from the Switch menu.
   Since your PC is not connected to the FX system, the following dialog appears:

Recognizing the Non-Connect Dialog

2. Click *Yes* to use the VMMI as your TFTP server.
   This action causes a dialog to appear that prompts you for an IP address.
   At this time, **DO NOT Click OK—a later step instructs you to do so when needed.** For now, leave the dialog box on the desktop so you can return to it later, and proceed to step 3



3. Start and run a terminal communications software program on your PC (such as HyperTerminal® that you can find on your Microsoft® Windows® desktop at Start/Programs/Accessories).
4. Press the RESET button on the VoIP board to start a dialog between the VoIP board and your PC.

   The details in the following example session shows the dialog displays of a typical fail-safe software upgrade session along with comments concerning where you can alter the information if you choose to do so. This illustrative session demonstrates the changing of typical Boot Record parameters at board start up. It also demonstrates loading the Fail Safe Programmer application over the Ethernet connection. Do note that while the dialog seems intimidating, most of the displayed information is merely for diagnostic use.

   Normally, the Boot ROM loads the FLASH memory application and executes it. It does this using a file named *bootflash*. For Fail-Safe Programming, you instruct the VoIP board's Boot ROM application to load the flash programmer software file, *burner.ftp*, over the Ethernet connection and then execute that program. Using the flash programmer, the VoIP board will be able to load the application software.

NOTE: The Boot ROM does not commit the Boot Record parameters to FLASH. While the Boot ROM uses those records, you must continue the programming action to save the values to FLASH memory.

Running the Fail-Safe Program

```
-
  (c) 2000, Comdial VOIP BootRom Startup Dialog Version 1.4
  Load & Run FLASH via filename :  bootflash
  Load & Run TFTP  via filename :  path\filename
-
  VOIP IP address on LAN         :  199.75.30.37
  VOIP LAN subnet mask           :  255.255.255.0
  VOIP LAN hardware mac address :  00:e0:a6:00:02:87
  IP address of the TFTP host   :  199.75.30.38
  Filename to load and start    :  bootflash
  Dsp Type  1=549 2=5420 3=5402 :  1
  Boot Flag 1=MXP 2=APP          :  2

To change any of this, press any key within 1 seconds
```

1. Press any key to break the Boot ROM's boot-up process.

```
(M)odify any of this or (C)ontinue? [M]
```

2. Press the Enter to accept the default selection or press [M] to Modify the parameters.

```
  THESE CHANGES APPLY ONLY TO THE PRESENT WORKING VALUES.
Run application's bsp>bootrecord to permanently save the changes.
Press <Enter> to use the value in braces, or enter a new value.
```

3. Enter IP address of the VoIP board. In this display, you either enter the appropriate data for each parameter below or keep the present data shown in braces [] by pressing the **Enter** key. You must change the name of the load and start file that the software uses from *bootflash* to *burner.tfp*. You must set the TFTP host's IP address to be the same as the IP address of you TFTP server.

```
  VOIP IP address on LAN         ?  [199.75.30.37] 199.75.30.36
  Subnet mask for LAN            ?  [255.255.255.0] 255.255.255.128
  Set default routing gateway   ?  [N] Y
  IP address of default gateway ?  [0.0.0.0] 199.75.30.01
  Change hardware mac address   ?  [N] N
    IP address of the TFTP Host   ?  [199.75.30.38] 199.75.30.38
  Filename 79 characters maximum
  Filename to load and start    ?  [bootflash] burner.ftp
  Dsp Type  1=549 2=5420 3=5402 ?  [1]
  Boot Flag 1=MXP 2=APP          ?  [2]
```

NOTE: The program now shows the entire data again for confirmation.

```
----------------------------------------
  (c) 2000, Comdial VOIP BootRom Startup Dialog Version 1.4
  Load & Run FLASH via filename :  bootflash
  Load & Run TFTP  via filename :  path\filename
----------------------------------------
  VOIP IP address on LAN        :  199.75.30.36
  VOIP LAN subnet mask          :  255.255.255.128
  IP address of default gateway :  199.75.30.1
  VOIP LAN hardware mac address :  00:e0:a6:00:02:88
  IP address of the TFTP host   :  199.75.30.38
  Filename to load and start    :  flash.tfp
  Dsp Type  1=549 2=5420 3=5402 :  1
  Boot Flag 1=MXP 2=APP         :  2
----------------------------------------
(M)odify any of this or (C)ontinue? [M] c
```

4. Type C and press Enter to accept ALL of the changes.

```
Updating parameter storage. This may take a while...Done
```

NOTE: The changes are now in use; however, they are not yet saved to FLASH. You must to continue the exercise to complete that task.

```
Bootrom Ram Test value: 0x00018000

Testing Application Ram...
Last Pattern was 003e8000 ... Ram OK
--------------------------------
This program loads a TFTP file into ram and runs it.
The TFTP Server should be ready before you continue.
   Enter filename bootflash to exit and reboot
--------------------------------
```

```
File Name to load into ram and run      ? [burner.ftp]
Format: (S)rec (M)icro                  ? [M]
(M)odify any of the above or (C)ontinue ? [M] c
```

5. Return to the *Host IP Address* dialog box that you left open on your desktop, and click **OK** to accept the programming PC's address. This action starts VMMI as your TFTP server.

6. Keep the file name [*burner.ftp*] and the format [M]icro and press C to continue the program.

NOTE: The TFTP file loader now gets the file from your PC over the Ethernet connection.

```
Padding Region 0x00060000 Length 0x002a0000 with 0xFF

Start   TFTP download
Device  TFTP Initialized
.........
Header information:
_________________

Magic Number:        0xff544e49
Code length:         0x0004bd9e
Offset:              0x00000024
Destination Address: 0x00060000
Executable Address:  0x00060000
Header check Sum:    0x0010bdc2
Code check Sum:      0x01711735
Next Header Offset:  0x00000000
Options:             0x00000002
_________________-
Good    header checksum
Found   MICRO image.
Testing Image Checksum
Good    Image Checksum
```

NOTE: The Fail-Safe Programmer software download to the board is now complete and will automatically execute.

```
Transferring control to the loaded application
========================================================
=    Copyright (c) Integrated Systems, Inc., 1998.     =
=    Copyright (c) Telogy Networks Golden Gateway      =
=    Copyright (c) Comdial Corporation,      2000      =
= Comdial VOIP Ram Based Flash Programmer Version 1.3  =
========================================================
(The Fail Safe found discrepencies with the BootRecord in FLASH and SDRAM)
********************************************************
* BOOT ITEM        WORKING VALUE        SAVED VALUE    *
********************************************************
 voip mac      00:e0:a6:00:02:88  00:e0:a6:00:02:87
 voip ip         199.075.030.036    199.075.030.037
 voip mask       255.255.255.128    255.255.255.000
 voip gateway    199.075.030.001    000.000.000.000
 tftp host ip    199.075.030.038    199.075.030.038
 voip dsptype                  1                  1
 voip bootflag                 2                  2
********************************************************
```

7. For Fail-Safe Programming, **do not** save the changes as these parameters are only for this working session. You should type N in response to each question.

```
voip ip addresses are different.
Commit WORKING to SAVED ? (Y)es (N)o (R)emaining : [N]
voip masks are different.
Commit WORKING to SAVED ? (Y)es (N)o (R)emaining : [N]
voip gateway addresses are different.
Commit WORKING to SAVED ? (Y)es (N)o (R)emaining : [N]
```

```
__________________________

CAUTION: THIS PROGRAM REPLACES THE BOARD SOFTWARE.
The TFTP Server should be ready before you continue.
   Enter filename bootflash to exit and reboot
__________________________
```

8. You can now download the application software to the VoIP board. Enter the application file name *ggbasicai.ftp*, type M for the [M]icro format, and type C to[C]ontinue the operation.

```
File Name to program into FLASH       ?  [bootrom.ftp] ggbasicai.ftp
Format: (S)rec (D)sp (B)oot (M)icro   ?  [M]

(M)odify any of the above or (C)ontinue? [M] C
```

NOTE: At this point, the Fail-Safe Programmer downloads the application file over the Ethernet connection.

```
Padding Region 0x00650000 Length 0x002a0000 with 0xFF

Start   TFTP download
Device  TFTP Initialized
.....................................................
Header information:
_________________-
Magic Number:         0xff544e49
Code length:          0x001b4566
Offset:               0x00000024
Destination Address:  0x00060000
Executable Address:   0x00060000
Header check Sum:     0x0027458a
Code check Sum:       0x08ef675f
Next Header Offset:   0x00000000
Options:              0x00000002
_________________-
Good    header checksum
Found   MICRO image.
Testing Image Checksum
Good    Image Checksum
```

```
Image   OK, continue (y|n) ? [n] Y
```

9. Type Y to instruct the Fail-Safe Programmer to load the *ggbasicai.ftp* program into FLASH. When you make this response, the Fail-Safe Programmer programs the FLASH.

```
Erase   FLASH sector
23
Program FLASH Sector
23
Program FLASH complete
Loading Header
Header information:
-----------------
Magic Number:        0xff544e49
Code length:         0x001b4566
Offset:              0x00000024
Destination Address: 0x00060000
Executable Address:  0x00060000
Header check Sum:    0x0027458a
Code check Sum:      0x08ef675f
Next Header Offset:  0x00000000
Options:             0x00000002
-----------------
Padding Region 0x00650000 Length 0x002a0000 with 0xFF
Flash   Header Checksum 0x0027458a OK.
Loading Image
Reload  Image Checksum 0x08ef675f OK.
Flash   Image Processed.
Freeing Region
```

NOTE: The system has saved the ggbasicai.ftp file. Fail-Safe Programming is complete.

```
--------------------------
CAUTION: THIS PROGRAM REPLACES THE BOARD SOFTWARE.
The TFTP Server should be ready before you continue.
   Enter filename bootflash to exit and reboot
--------------------------
```

10. You are finished Fail-Safe Programming. Reboot the VoIP board by typing *bootflash*.

```
File Name to program into FLASH       ?  [ggbasicai.ftp] bootflash
```

NOTE: The VoIP board re-boots with the OLD network parameters still in FLASH as it should because you did not save the working values that you entered during the exercise—see step 7 on page 28.

```
pSOSystem V2.5.0
Copyright (c) 1991 - 1998, Integrated Systems, Inc.
--------------------------------------------------------------------
  (c) 2000, Comdial VOIP BootRom Startup Dialog Version 1.4
  Load & Run FLASH via filename :  bootflash
  Load & Run TFTP  via filename :  path\filename
--------------------------------------------------------------------
  VOIP IP address on LAN        :  199.75.30.37
  VOIP LAN subnet mask          :  255.255.255.0
  VOIP LAN hardware mac address :  00:e0:a6:00:02:87
  IP address of the TFTP host   :  199.75.30.38
  Filename to load and start    :  bootflash
  Dsp Type  1=549 2=5420 3=5402 :  1
  Boot Flag 1=MXP 2=APP         :  2
--------------------------------------------------------------------
To change any of this, press any key within 5 seconds
```

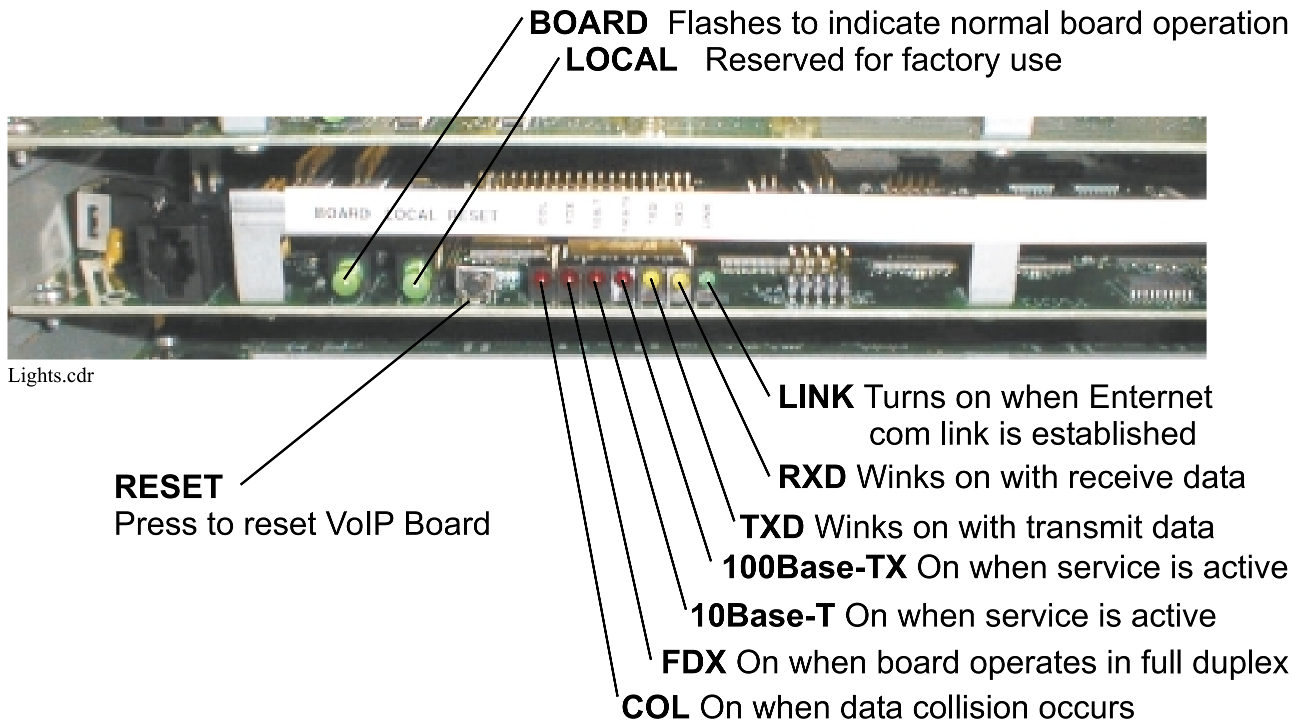
11. Disconnect from the serial port and, let the Boot ROM perform its normal boot-up sequence.

    If it is not successful, the VoIP board may require factory servicing. Contact your representative in Comdial's Technical Services Department for details. The Technical Services toll-free telephone number is: 1-800-366-8224.

Troubleshooting the VoIP Board Installation

Defining the Lights and Indicators

Along the front of the VoIP board is a row of indicating lights. The following illustration describes the indicating function of each light.



Defining the Lights and Indicators

The Ethernet status lights are useful to verify a functioning Ethernet connection. The **LINK** LED indicates a functioning network connection at the data speed indicated by either the **10Base-T** or **100Base-TX** LEDs. The other status lights indicate transmit and receive data, full or half duplex operation, and data collision alerts.

The **BOARD** LED synchronizes with the LEDs on other circuit boards installed in the FX system. A flash rate of approximately every five seconds indicate a normal operating board.

Defining Terms

Bandwidth The measure of the number of bits per second that can flow through a network link at a given time.

CODECS The term CODEC is an acronym that stands for **co**der/**dec**oder. A CODEC is a DSP-based procedure, or small computer program (also known as an algorithm) that reduces the number of bytes consumed by large or complicated files. File reduction is necessary in order to minimize the amount of storage space or transmission time required for large or complicated data files. Also see G.711 and refer to the in-depth discussion on CODECs found on page 39.

DSP Digital Signal Processing, refers to various techniques for improving the accuracy and reliability of digital communications. DSP works by standardizing the levels or states of a digital signal. A DSP circuit can differentiate between human-made signals, which are orderly, and noise, which is inherently chaotic.

Ethernet The physical medium of a local area network. Ethernet networking is typically wired with Category 5 cabling at 10Base-T and 100Base-TX data speeds.

Fire-Wall A combination of computer hardware and software that limits the exposure of a computer or a computer network to influence from data sources located outside the network. Fire walls are often used between local area networks (LANs) and the Internet.

Frame Relay This is a standardized data handling technique that passes data in a series of variable length packets, or frames. A frame relay network can handle data formatted in almost any accepted protocol without disturbing any specific control data associated with the data. Frame relay supports both data communications and compressed and packetized audio and video signals.

FRAD	Frame Relay Access Devices, devices used to connect data streams to a frame relay service provider's demarcation device. FRADs connect not only multiple data streams but also T1 and analog interfaces to customer-owned telephony equipment.
G.711	The ITU-T standard CODEC that is used in most modern telecommunications equipment. This CODEC is considered the reference for speech quality by which other CODEC's are judged. It is known as toll quality. Also see CODECS.
Gateways	VoIP gateways are translation devices that bridge calls between the circuit-switched PSTN and the IP networks.
Internet	A world-wide network of computers and computer networks joined together by a high-speed backbone of data links.
Intranet	A private network that uses Internet protocols and standards. An Intranet network is available to designated users (typically employees of a company, customers, or other people who have the use authority via password entry). Companies often provide Intranet service through a server that users access with browser software.
IP	Internet Protocol, the protocol used to transmit data over the Internet and other managed IP networks (such as Intranet).
IP Telephony	Internet Protocol Telephony, is a general term for the technologies that use the IP's packet-switched connections to exchange voice, FAX and other forms of information that traditionally have been carried over dedicated circuit-switched connections of the PSTN.
Jitter	The variable delay that individual packets can experience when traversing a network. A network introduces jitter if it causes data packets to arrive with variable delay at a destination. Jitter is disruptive to audio communications.
LAN	Local Area Network, a network of local service provider-provided lines or other privately-supplied signal transfer devices used to link data communications equipment together through a standardized protocol over a local geographical area.

PCM	Pulse Code Modulation. A standardized method of coding analog speech into binary (digital) code words.
PSTN	Public Switched Telephone Network, the traditional telephony network employing regular telephone lines.
QoS	Quality of Service, on IP networks is the idea that transmission rates, error rates, and other characteristics can be measured, improved, and to some extent, guaranteed in advance.
Red Alarm	A system generates a red alarm when it detects a local failure such as a loss of synchronization.
Router	A network device that directs IP packets to the appropriate destination based upon the IP address.
TFTP	Trivial File Transfer Protocol, a simplified file transfer protocol that transfers files but does not require password protection.
VAD	Voice Activity Detection, a processing algorithm that monitors the speech for the presence of silence and sends silence packets out-of-band (as a unique code) to the far-end, rather than sending empty data packets.
VoIP	Voice over Internet Protocol, a mnemonic that describes the real-time transmission of voice over IP networks.
WAN	Wide Area Network, a network that uses common carrier-provided lines and a standardized protocol to link together data communications equipment that is dispersed across a large geographical area.
UDP	User Datagram Protocol, a protocol that describes how messages reach application programs. UDP is a transport layer protocol that provides a specific mode of communication for delivery of packets to a remote or local user.
Yellow Alarm	A system sends a yellow alarm back toward the source of a failed transmit circuit to indicate that the input of a network element has failed.
µ-Law	One form of the ITU-T's G.711 CODEC that is typically used in the United States. A-law is another version that is used in most of the rest of the world.

10Base-T A 10Mbs Ethernet LAN that works on twisted-pair, home-run wiring with the look and feel of standard telephone cable. The 10Base-T Ethernet Network Interface Cards (NICs) that fit in the LAN-connected data devices operate with multiple categories of cabling but communications reliability increases as the cable categories increase.

100Base-TX A 100 Mbs Ethernet LAN implemented over Category 5 cabling. 100Base-TX uses two pairs of wires—one pair for data transmission and the other pair for data reception. 100Base-TX is the best connection for servers, hubs, switches, and routers because it supports full-duplex operation. The 100Base-TX installation requires compatible Network Interface Cards (NICs) and Category 5 cabling.

Notes

Detailing an IP Address

IP addressing is a broad subject that requires some research and study before you can completely understand it's every facet. The purpose of this discussion is to only provide a brief overview of the subject that compliments the installation and programming of the VoIP board.

For further information and in-depth understanding about the subject of IP Addressing, you may want to visit the following World Wide Web Internet locations:

- **www.IETF.org**
  Navigate to the site's **RFC pages** area, and examine the following documents that are stored there:
  — RFC 791, which discusses Internet Protocol
  — RFC 950, which discusses subnetting
  — RFC 1519, which discusses classless interdomain routing
- **www.3com.com/nsc/501302.html**
  Study the discussion by **Chuck Semeria** titled *Understanding IP Addressing: Everything You Ever Wanted to Know*.

The following paragraphs provide a general understanding of IP addressing for your information.

Overviewing IP Addressing

In the most widely installed version of the Internet Protocol (IP) today, an IP address is a 32-bit number that identifies each sender or receiver of information. This address is sent in packets over the IP network.

An IP address typically has two parts: the identifier of a particular network on the IP network and an identifier of the particular device (Host ID—can be a server or a workstation) within that network.

Expressing the IP Address

The 32-bit IP address is usually expressed as four decimal numbers, each representing eight bits, separated by periods. This is sometimes known as the *dot address* or, more technically, as *dotted quad notation*.
Here's an example: **130.5.5.25**

Each of the decimal numbers represents a string of eight binary digits. Thus, the above IP address really is this string of 0s and 1s:
10000010.00000101.00000101.00011001

The decimal version of the IP address is easier to read and is the form most commonly used.

Breaking down the Address

Some portion of the IP address represents the network number or address and some portion represents the local machine address (also known as the *host number* or address). IP addresses can be one of several classes, each determining how many bits represent the network number and how many represent the host number. Many large organizations have been assigned Class B addresses that allows 16 bits for the network number and 16 for the host number. (There are also Class A addresses that allow 8 bits and Class C addresses that allow 24 bit addresses.) Using the above example, here's how the IP address is divided:

<—Network address—><—Host address—>
130.5 . 5.25

Adding Subnetting to the Address

The purpose of subnetting is to set up multiple networks using the same IP address. If you wanted to add subnetting to an address, then some portion (in this example, eight bits) of the host address could be used for a subnet address. Thus:

<—Network address—><—Subnet address—><—Host address—>
130.5 . 5 . 25

This explanation divides the subnet into neat eight bit groupings; however, an organization could choose some other scheme using only part of the third quad or even part of the fourth quad.

Once a packet has arrived at an organization's gateway or connection point with its unique network number, the internal gateways can route it within the organization using the subnet number as well. The router knows which bits to look at by looking at a network mask. A mask is simply a screen of numbers that tells you which numbers to look at underneath. In a binary mask, a *1* over a number marks that number for consideration while a 0 causes the number to be ignored. Using a mask saves the router from having to handle every 32-bit address on the local network.

Using the previous example, the combined network number and subnet number occupy 24 bits or three of the quads. The appropriate internal network mask is as follows: **255.255.255.0** or a string of all 1's for the first three quads (telling the router to look at these) and 0's for the host number that the router ignores. Network masking allows routers to move the packets more quickly.

Understanding CODECs

Understanding CODECs

Comdial digital communications systems transport voice information from telephone to telephone in a digital format. The system captures the speech information as analog signals, then transforms it into coded bytes of data by a process called Pulse Code Modulation (PCM). The PCM data flows in a continuous stream of data bits.

Telephone conversations occur in both directions simultaneously so each end needs a coder and a decoder. These two components together are referred to as the CODEC. The CODEC implements standards-based mathematical algorithms to code the voice information as data. There are a variety of standardized CODEC's. Traditional telephony in North America uses the G.711 CODEC in its µ-law form. The alternate form, A-law, is common in other parts of the world. G.711 provides very good speech quality and is often referred to as *toll quality*.

The circuitry in the Comdial VoIP system converts the system's internal PCM-encoded speech into IP packets for transmission across an IP data network. The IP packets share the data network with any other data traffic already there.

Voice quality in VoIP depends on several interrelated things. The voice must be coded and decoded with a CODEC that has good intelligibility characteristics. The voice data must travel from the source to the destination without delay, loss or corruption. The amount of bandwidth available in the network connection, the delay that the packet experiences, and any packet loss or corruption that occurs all work to constrain the transport of the voice packets across the network.

Bandwidth

Bandwidth is a factor that affects VoIP Quality of Service (QoS) and is the measure of the number of bits per second that can flow through a network link at a given time. As various data devices (such as computers) use the data network, the bandwidth is consumed. Large file transfers can cause momentary loss of bandwidth. After the various data devices that use the network exercise their bandwidth requirements, any remaining portion is known as the available bandwidth. For a VoIP connection, lack of available bandwidth means that the connection will not support the added packet flow of the voice packets. Typically, the section with the smallest physical bandwidth is also the section with the smallest available bandwidth. This section will limit the VoIP data packet capacity. For a data network to support VoIP, it must have adequate bandwidth across the entire VoIP connection path. Cable interconnection specifications, network layout, and equipment type are all factors that combine to dictate the available bandwidth of the end-to-end connection. Adequate network bandwidth is necessary to support the maximum VoIP data traffic. Required bandwidth is the product of the number of voice channels supported (simultaneous calls) and the bandwidth usage of each voice channel. The bandwidth usage per channel is determined primarily by the CODEC used and its associated overhead.

To provide several options, Comdial offers several different CODEC’s as detailed in the tables shown on page 41. Different CODEC’s use different amounts of compression to reduce the amount of voice data to be transmitted thus reducing the amount of network bandwidth required to pass the voice packets. Note that the table lists the number of kilobits per second (kbps) required by the different CODEC’s for one direction of the full-duplex VoIP conversation. (As a reference, a typical 33.6 modem passes 33.6 kbps of data in each direction.)

As the tables indicate, CODEC’s such as G.723.1A and G.729AB significantly reduce the data bandwidth required. They accomplish this by using advanced algorithms that model voice in a highly compressed way. There is in general a tradeoff between using a high compression CODEC (with its low Bandwidth usage) and voice quality. The high compression CODEC’s typically have slightly reduced voice quality, and introduce additional delay due to the added computational effort. The highest bandwidth is required by the minimal compression G.711 CODEC, which is the standard toll quality CODEC. Conversely, some of the high-compression low-bandwidth CODEC’s are often considered to be comparable to cell phones in voice quality.

It is important to note that this data are for a VoIP call that is IP end-to-end. If a portion of the connection path is over Frame Relay or ATM, this technology may add additional overhead to change the IP packets into a different format. In this case, the bandwidth requirement may increase by as much as fifty percent.

The FX VoIP system implements a signal processing algorithm that can reduce the bandwidth requirements shown in the table by twenty to fifty percent. The feature is called Voice Activity Detection (VAD). With VAD, the speech processing circuit monitors the speech data for the presence of silence. In typical conversations, one of the parties is usually silent. When the circuitry detects silence, the system circuitry sends special *silence packets. The* G.723 and G.729 CODECs have VAD-like functionality built in to the compression algorithms and so gain little additional benefit from the VAD feature.

Detailing the CODEC Bandwidth Specifications

Detailing the CODEC Bandwidth Specifications					
CODEC	**Mono-directional Nominal Data Rate (Kbps)**	**Frame Relay Mono-directional Bandwidth (Kbps)**		**Ethernet Mono-directional Bandwidth (Kbps)**	
		Small Packet Size	**Large Packet Size**	**Small Packet Size**	**Large Packet Size**
G.711	64 Kbps	78	78	82	82
G.726	32 Kbps	46	46	50	50
G.726	16 Kbps	30	30	34	34
G.729AB	8 Kbps	19	15	21	17
G.723.1A	6.3 Kbps	21	14	24	15
G.723.1A	5.3 Kbps	21	14	24	15

Packet Delay

Another factor that significantly affects VoIP quality is the delay that the VoIP data packets experience as they traverse the network. Regular packet delays longer than 100 milliseconds begin to interfere with normal conversation. Longer delays can cause echoes, which can make normal conversation difficult. The total end-to-end packet delay is the result of the many incremental delays along the connection path. Some small delay occurs during the digital signal processing of the speech signals by the FX VoIP system. As the VoIP packets travel across the network, a number of sometimes large delays can occur.

The CODEC's specified for use with VoIP for the processing of the speech impose small processing delays. The high compression CODEC's (G.723.1A and G.729AB) require greater processing power and so introduce incrementally more delay than the low-compression CODEC's. The packet delay experienced during signal processing is typically on the order of tens of milliseconds.

VoIP packet delay in the IP network can be quite large and can vary significantly from one network to another. Each router that the packet traverses in its travel across a network adds an additional delay. The router's or switch's design and the configuration of its buffering size affects the delay each router adds to the voice packets. On the reception end, other buffering can add more delay. Careful network architecture, equipment selection and configuration all minimize delay as does the practice of minimizing the number of router hops along the path. Small router queue sizes and high bandwidth connections also reduce delay but, small router queue sizes may result in lost packets.

When cumulative delay from transmitter to receiver and back is too long, echo can result.

If the VoIP call passes through a network with very bad delay (such as the Internet) or if the call goes out over a long-distance PSTN line that has excessive delay (for example a line using a satellite connection), echo can occur. To suppress echo, the FX VoIP system implements a standards-based G.168 echo canceller.

In addition to the average delay that all the VoIP packets experience, individual packets can experience a small amount of extra delay (referred to as jitter) relative to the other packets in the data stream. This extra delay is due to instantaneous network usage and congestion or to the fact that this packet of data took different routes through the network. Jitter causes some packets to arrive at different times at the destination relative to one another and sometimes causing the packets to arrive out of sequence. This effect causes the CODEC to have difficulty recreating the speech on the receiving end, and voice quality is impaired. The FX VoIP system reduces the jitter effect by using a jitter buffer to buffer the incoming packets. The system then reassembles the packets in the correct order. This buffering makes it possible to reorganize late packets at the cost of an incremental delay.

Lost Packets

Yet another factor that affects VoIP quality is lost or damaged packets. In the process of traversing the network occasionally a packet can get lost. This can occur if a router queue along the network path is overloaded. In this case, the router will typically discard the packet. It is also possible for a data packet to become damaged or corrupted during its travel through the network. In both cases, these packets are unusable. The VoIP packet protocol assumes that the packets arrive correctly. The packets are not retransmitted if their delivery fails or they arrive in a corrupted state.

The voice CODEC on the receiving end extrapolates new packets to fill in the gaps caused by missing packets. If the packet damage or loss is severe, voice quality will degrade. The impact on voice quality depends upon the CODEC used. You can offset excessive packet loss by using a large packet size setting. This action increases the size of each individual voice packet. This increase in size results in less overhead in the data packets and a slightly improved network utilization.

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