V/IP Phone/Fax IP Gateway

Analog Models

NetWare and DOS

User's Manual

Part Number 800-1886-11, Rev. B

October 1997

Notice of Filing

Declaration of CE Conformance (for International sales)

A Declaration of CE Conformance is on file at the MICOM addresses shown below. The declaration lists the models described in this manual. If the unit carries the CE mark, this declaration certifies that it meets the specific EMC standards and safety (LVD) standards required for CE marking. If the product is a card, the card is CE-compliant only if it is placed in a CE-marked host unit.

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Any units not carrying the CE approval are not CE-compliant. Cards placed in these units may not meet emission standards for CE compliance.

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Specifications, tolerances, and design characteristics described in this manual are subject to change without notice.

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NOTE: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Changes or modifications to this product, that could increase the amount of Radio Frequency Emissions from this product, without expressed written approval of MICOM Communications Corp., could cause the product and the user to violate the FCC's Rules and Regulations, thus requiring the product to be turned off or disconnected.

Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Notification of Canadian Requirements

This digital apparatus does not exceed the Class A limits for radio noise emissions from digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques de classe A prescrites dans le règlement sur le brouillage radioélectrique édicté par le Ministère des Communications du Canada.

Host Computer Requirement

V/IP Phone/Fax IP Gateway Analog Voice Interface Cards must be installed only in PC chassis that have power supplies that meet IEC 950 reinforced insulation requirements.

Note: This card must be installed in a CSA/UL Certified Fire Enclosure.

United Kingdom Requirements

Host Computer Compliance Warning

The host computer must comply with the General Approval NS/G/1234/J/100003.

Safety Cover

The supplied safety cover must be installed on the V/IP Phone/Fax IP Gateway Analog Voice Interface Card.

Interconnection of Ports Warning

Interconnection directly, or by way of other apparatus, of ports marked “SAFETY WARNING. See Instructions for use”, with ports marked or not so marked may produce hazardous conditions on the network. The advice of a competent engineer must be obtained before such a connection is made. None of the ports provide isolation sufficient to satisfy the relevant parts of BS 6301. Apparatus connected to the ports, must either have been approved to the relevant parts of BS 6301 or to have been previously evaluated against BS 6301 British Telecom Technical Guides 2 or 26, and given permission to attach. Other usage will invalidate any approval given to this apparatus.

The 1 or 2 voice ports on the card may be configured as non-network ports.

WARRANTY

MICOM warrants that to the extent that the equipment delivered is hardware, such equipment shall be free from defective material and workmanship for a period of 3 years from the date of shipment of equipment from MICOM when given normal, proper and intended usage. MICOM further agrees to provide, without cost, emergency replacement equipment, shipped freight prepaid, for a period of ninety (90) days from date of shipment of the equipment and factory repair for the remainder of the warranty period provided that:

(a) MICOM is promptly notified upon discovery that the equipment is defective;

(b) The equipment is returned freight prepaid to MICOM;

(c) MICOM's examination of the equipment shall disclose that any defect was not caused by failure of electrical power or air conditioning, damage from lightning or weather-related causes, accident, misuse, neglect, alteration, improper installation, unauthorized repair or improper testing.

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INSTALLING NON MICOM SOFTWARE IN MICOM EQUIPMENT SHALL VOID THIS WARRANTY.

MICOM may, in its sole discretion, except for the first ninety (90) days of warranty, elect to repair or replace the equipment, in which event MICOM shall have a reasonable time to make repairs or to replace the equipment. MICOM will return the equipment freight prepaid.

THE PROVISIONS OF THIS WARRANTY ARE IN LIEU OF ANY OTHER WARRANTY, WHETHER EXPRESS OR IMPLIED, WRITTEN OR ORAL (INCLUDING ANY WARRANTY OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE), AND MICOM'S LIABILITY ARISING OUT OF THE MANUFACTURE, SALE, OR SUPPLYING OF THE EQUIPMENT OR ITS USE, WHETHER BASED UPON WARRANTY, CONTRACT, NEGLIGENCE, PRODUCTS LIABILITY OR OTHERWISE, SHALL NOT EXCEED THE ORIGINAL AMOUNT PAID BY THE BUYER FOR THE EQUIPMENT. IN NO EVENT SHALL MICOM BE LIABLE TO THE BUYER OR ANY OTHER PERSON OR ENTITY FOR UNINTENDED OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, LOSS OF PROFITS OR USE DAMAGES) ARISING OUT OF THE MANUFACTURE, SALE OR SUPPLYING OF THE EQUIPMENT.

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If you experience difficulty with this product, contact your MICOM Certified Distributor for prompt assistance.

MICOM offers complete factory repair for both in-warranty and out-of-warranty equipment. Before returning any equipment, you *must* obtain a Return Authorization number. Contact your distributor for assistance.

MICOM warrants all out-of-warranty repairs or upgrades performed at its factory location or performed by MICOM Customer Service for a period of 90 days after completion.

Shipping charges must be prepaid.

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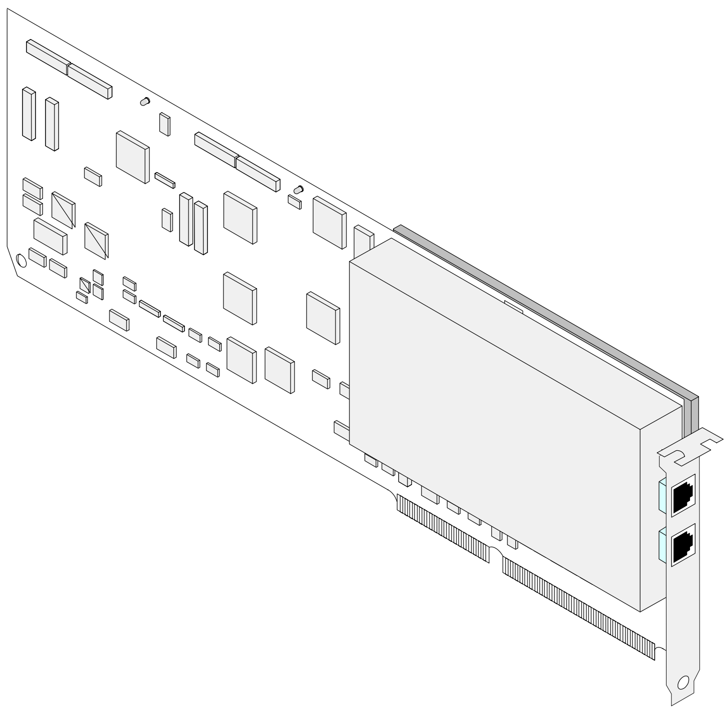
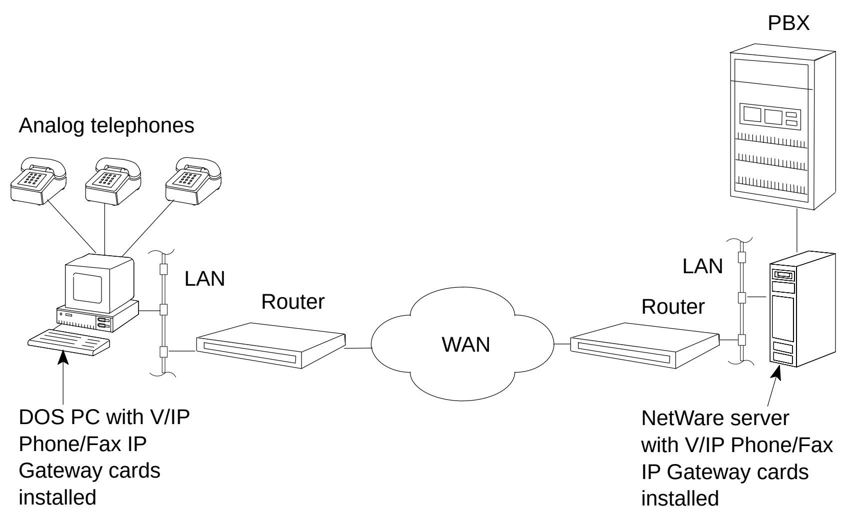


V/IP Phone/Fax IP Gateway Analog Voice Interface Card

Description Of Operation 1

V/IP (Voice Over IP) Phone/Fax IP Gateway makes it possible to transmit voice and fax traffic over any network. The gateway digitizes voice and fax signals, then encapsulates the information within IP packets that can be routed across a wide area network. The gateway consists of cards and software that is installed in DOS PCs and Novell NetWare servers.



This manual describes the V/IP Phone/Fax IP Gateway Analog Voice Interface Cards operating under NetWare and DOS. For brevity, the cards will be referred to as Voice Interface Cards hereafter in this manual.

The following are described in this manual:

Part number . . .	consists of . . .
VIP-2001-ISA-FXE	• Single channel, ISA bus, FXS and E&M analog interface card, model number VIP-1-ISA-FXE. • V/IP software for NetWare (3.12 and 4.x), DOS (6.0 and above), and Windows 95.
VIP-2002-ISA-FXE	• Dual channel, ISA bus, FXS and E&M analog interface card, model number VIP-2-ISA-FXE. • V/IP software for NetWare (3.12 and 4.x), DOS (6.0 and above), and Windows 95.
VIP-2001-ISA-FXO	• Single channel, ISA bus, FXO analog interface card, model number VIP-1-ISA-FXO. • V/IP software for NetWare (3.12 and 4.x), DOS (6.0 and above), and Windows 95.
VIP-2002-ISA-FXO	• Dual channel, ISA bus, FXO analog interface card, model number VIP-2-ISA-FXO. • V/IP software for NetWare (3.12 and 4.x), DOS (6.0 and above), and Windows 95.
VIP-2001-ISA-FXO–UK	• Single channel, ISA bus, FXO analog interface card, model number VIP-1-ISA-FXO, tested for BABT compliance. • V/IP software for NetWare (3.12 and 4.x), DOS (6.0 and above), and Windows 95.
VIP-2002-ISA-FXO–UK	• Dual channel, ISA bus, FXO analog interface card, model number VIP-2-ISA-FXO, tested for BABT compliance. • V/IP software for NetWare (3.12 and 4.x), DOS (6.0 and above), and Windows 95.

The analog interfaces of these cards will work with:

- standard office phones and fax machines
- legacy telephone switches (PBXs, key telephone systems) found in today's corporate offices

Note: The Voice Interface Cards do not support modems on the analog interfaces.

The LAN environment in which the V/IP will operate is determined by the Network Interface Cards (NICs) installed in the PC chassis. This V/IP release has been tested to operate correctly within Ethernet and Token Ring LANs.

The WAN environment the cards can operate across is determined by the routers used in the network. The WAN can be any public or private WAN environment (frame relay, ISDN, leased line, etc.). The cards use packet loss error correction to handle any IP packets that are dropped in transit or received in error across the WAN during a phone or fax call.

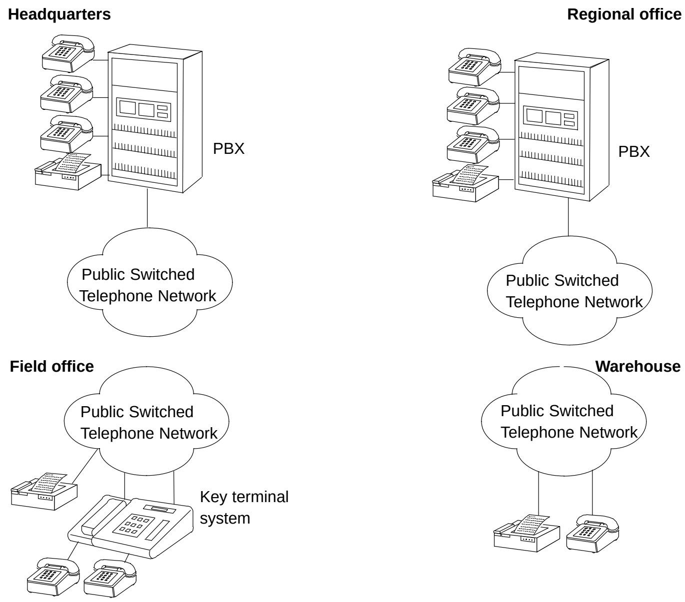
The cards use MICOM's ClearVoice technology to deliver the best quality voice using very little bandwidth. ClearVoice is based on the ITU-standard G.729 compression algorithm.

The cards support Group 3 fax up to 9600 bits per second. Fax calls are automatically detected.

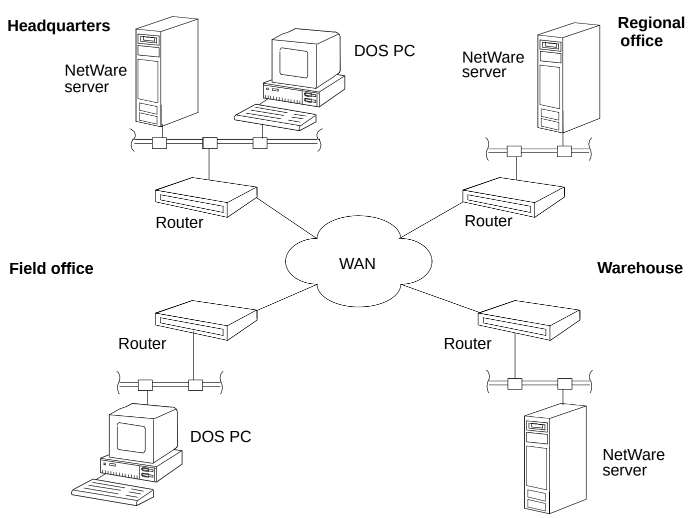
SNMP network management is supported, including GETs, GETNEXTs, and SETs. Traps are not supported.

Application and Benefit

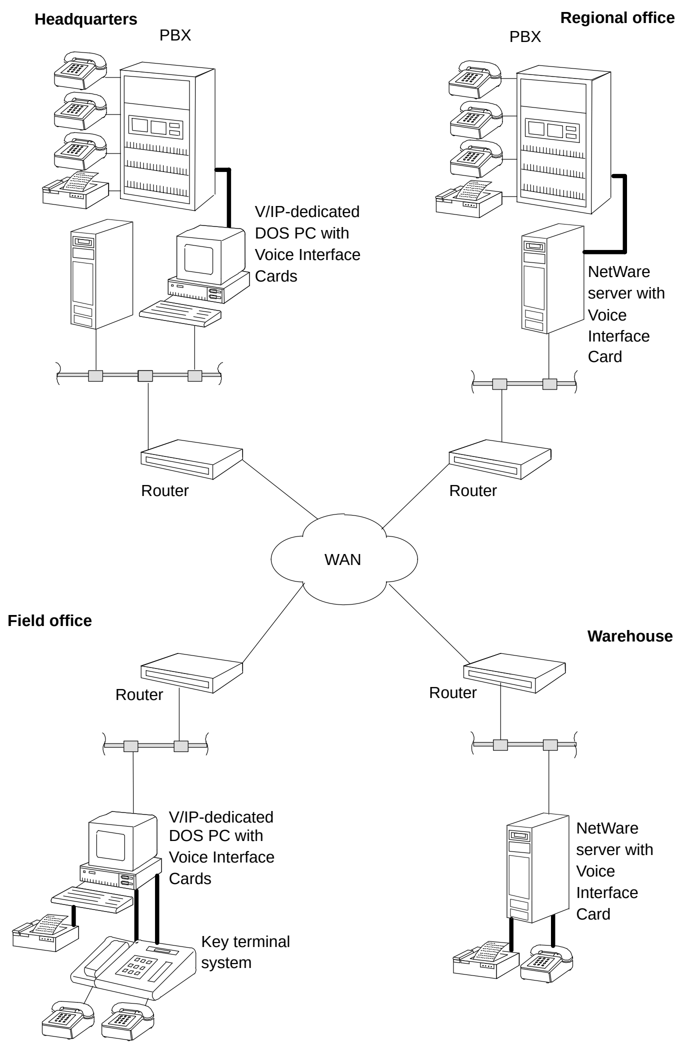
If your company's phone system is similar to this:



And your computer networks are interconnected, like this:



You can install Voice Interface Cards in your DOS PCs or NetWare servers and put phone and fax calls to your company's offices over the network that services your LANs, like this:



Your phone and fax calls will ride for free over your existing WAN.

How It Operates

1. The Voice Interface Cards digitize analog voice signals at 8 Kbps.

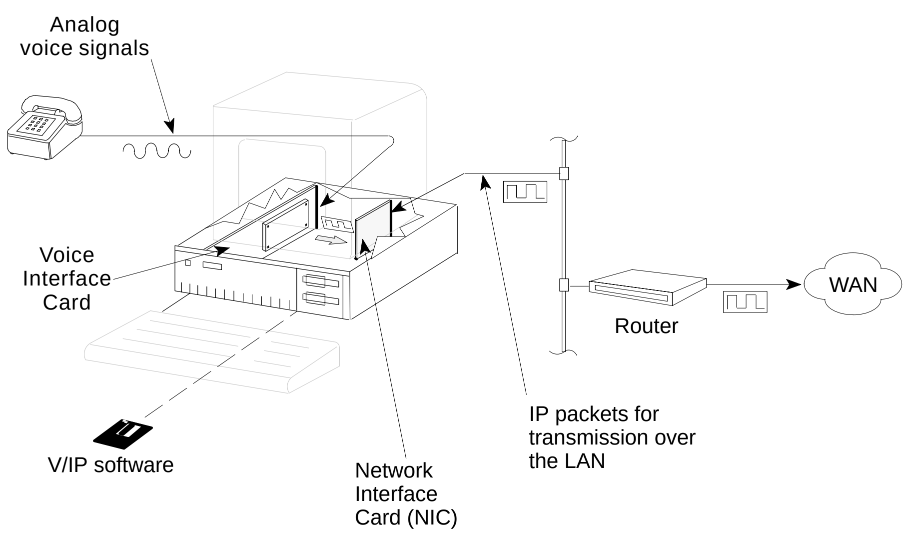
2. The V/IP system software handles the:
   - management of calls
   - encapsulation of digital voice into IP packets
   - phone directory database

3. The V/IP system software passes the IP packets via the PC's ISA bus to a Network Interface Card (NIC).

4. The NIC transmits the IP packets onto the LAN.

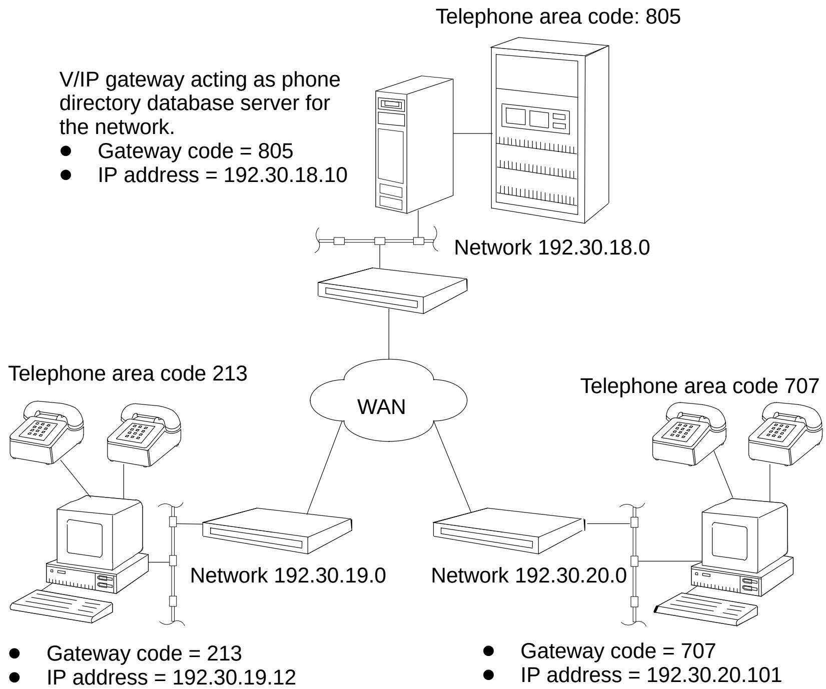
5. A router attached to the LAN forwards the IP packets across the WAN, where they will be received by another V/IP gateway PC at the remote end.

6. The process is reversed at the remote end.



Phone Directory Database

To allow you to easily dial a telephone or fax on the network, the V/IP gateway maps a series of dialed digits (called the gateway code) to the IP address of the remote V/IP gateway whose phone or fax you are calling. This mapping information is contained in a database called the phone directory database. You must designate one V/IP gateway to be the phone directory database server for the network. The other V/IP gateways update their databases from this server at regular intervals (the default is every 24 hours).



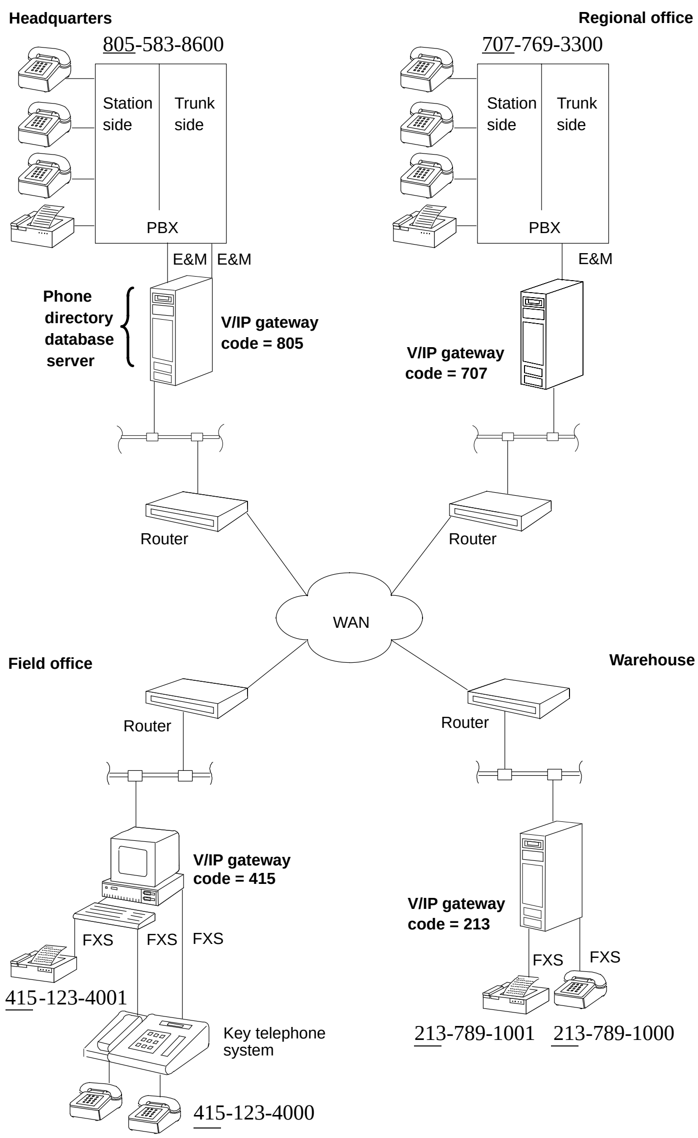
The phone directory database for the above network would look like this:

Gateway code	IP address
805	192.30.18.10
213	192.30.19.12
707	192.30.20.101

When you dial 805 on one of the phones at the site on the lower left (telephone area code 213), the V/IP gateway will map the 805 to IP address 192.30.18.10 and make a connection to one of the channels on the top V/IP gateway (telephone area code 805).

Making Phone Calls Over the Network

Here is an example application to illustrate phone/fax calls across the network (the V/IP gateway dialing plan):



For this example application, the company's standard telephone system is as follows:

- The headquarters on the upper left has a main number of 805-583-8600 with extension numbers going to individual employee offices (i.e., 2423, 2424, etc.).
- The regional office on the upper right has a main number of 707-769-3300 with extension numbers going to individual employee offices (i.e., 3301, 3302, 3303).
- The field office on the lower left has multiple lines coming into a key telephone system, whose phone number is 415-123-4000. Also there is a separate line for a fax machine, whose phone number is 415-123-4001.
- The warehouse on the lower right has two numbers: 213-789-1000 (the phone) and 213-789-1001 (the fax).

The company wants to match the V/IP dialing plan to the Public Switched Telephone Network area codes. This is to minimize retraining of the users of the company's phone system. With this in mind, the V/IP gateways are configured as follows:

- The V/IP gateway at the company headquarters is configured for gateway code 805. Also, this gateway is designated as the network's phone directory database server.
- The V/IP gateway at the regional office is configured for gateway code 707.
- The V/IP gateway at the field office is configured for gateway code 415.
- The V/IP gateway at the warehouse is configured for gateway code 213.

Note: The V/IP gateway provides great flexibility in choosing the dialing plan. This example application is just one way of setting up the dialing plan. Many alternative dialing plans are possible.

For this application, the configurations would be as follows:

Voice Interface Card	Gateway code	Number of Digits Assigned for Channel	Maximum Number of Dial Digits	Channel number
Headquarters, all cards	805	0	4	0, all channels All channels are one hunt group. Only the gateway code must be dialed
Regional office, all cards	707	0	4	0, all channels All channels are one hunt group. Only the gateway code must be dialed.
Field office, card 1	415	4	0	4000, all channels to the key terminal system These two channels are one hunt group that must be dialed.
Field office, card 2				4001, channel to the fax This channel must be individually dialed.
Warehouse, card 1	213	4	0	First channel: 1000 Second channel: 1001 These channels must be individually dialed.

Here are comparisons of the dialing required between calls placed over the public telephone network and calls placed using the V/IP network:

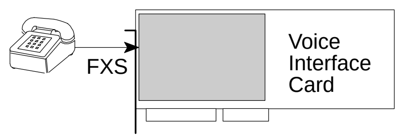
To call from . . .	**using the public telephone network, dial . . .**	**using the V/IP network, dial . . .**
Headquarters to the regional office	9 1 (707) 769-3300, extension 3302 You dial 9 for an outside line, then 1 707 769 3300 to get the PBX operator at the regional office and then ask for extension 3302.	8 707 3302 You dial 8[1] to get the PBX V/IP line, then dial 707 followed by 3302 to call directly to the extension.
Regional office to the headquarters	9 1 (805) 583-8600, extension 2423 You dial 9 for an outside line, then 1 805 583 8600 to get the PBX operator at the headquarters and then ask for extension 2423.	8 805 2423 You dial 8[1] to get the PBX V/IP line, then dial 805 followed by 2423 to call directly to the extension.
Field office to the warehouse	1 (213) 789-1000 You dial 1 213 789 1000 to call directly to that phone.	213 1000 You dial 213 followed by 1000 to call directly to that phone.
Warehouse to the field office	1 (415) 123-4000 You dial 1 415 123 4000. Your call is routed to the first idle line. The call can be answered by anyone sharing the line (who hears the phone ringing).	415 4000 You dial 415 followed by 4000. Your call is routed to an idle line. The call can be answered by anyone sharing the line (who hears the phone ringing).
Warehouse to the field office fax machine	1 (415) 123-4001 You dial 1 415 123 4001 to call directly to the fax machine.	415 4001 You dial 415 followed by 4001. Your call is routed directly to the fax machine.

[1] The 8 is an example PBX configuration parameter. You can configure any number as the PBX V/IP line access code.

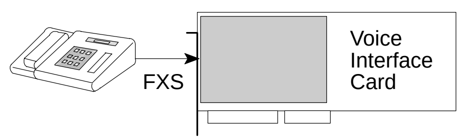
Telephone Interface Types

The cards support one or more of the following telephone interfaces, depending on the model you have ordered:

- FXS (Foreign Exchange Station) - a station loop start operation that provides a connection to:
  - a standard, single-line analog telephone instrument



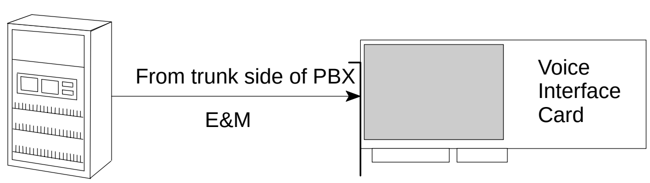
  - the line circuit of a key telephone system



  - a loop start trunk circuit of a Private Branch Exchange (PBX) that normally connects to incoming Central Office circuits

  This interface type provides power and ringing signals to its interfacing equipment. It is ***not*** intended for connection to the Public Switched Telephone Network.

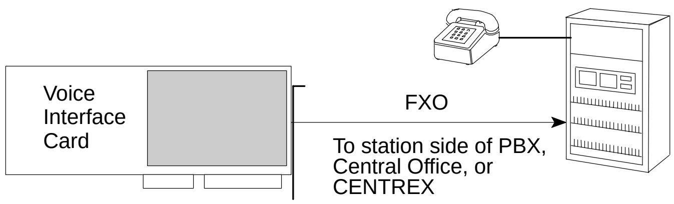
- E&M (Ear and Mouth) - a tie line trunk circuit used to connect between PBXs or other voice switching system.



Note: E&M AC15 or Pulsed DC is not supported.

- FXO (Foreign Exchange Office) - a trunk loop start operation that emulates a single-line telephone to:
  - Central Office lines,
  - CENTREX®,
  - or PBX stations.

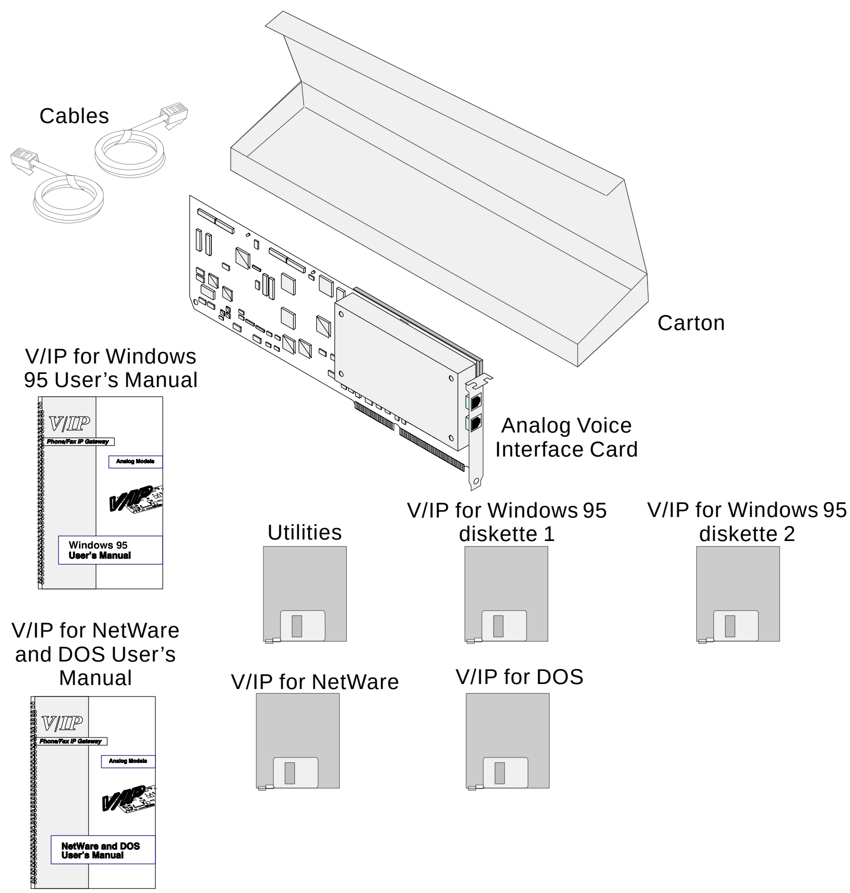
It recognizes ringing signals and draws current to indicate an active state.



Installation 2

Package Contents

You should have received the following in the V/IP Phone/Fax IP Gateway package:



Readme.txt File

The Utilities diskette includes a file called `readme.txt`. You should read this file (using any PC text editor program) to find out the latest information about the V/IP product.

Network Requirements

The Voice Interface Card is designed to operate within Ethernet and Token Ring networks that are interconnected using the following:

- Routers

  The Voice Interface Card will operate with any routers that support IP protocols across the remote link (WAN), such as Bay Networks, Cisco, 3Com, etc.

- WANs

  56 Kbps minimum link speed or faster

System Requirements

Processor

Voice Interface Cards must be installed only in IBM-compatible, UL/CSA/CE listed *grounded* computers.

The complete processor requirements are determined by the operating system that the PC is running and the size and activity level of the network. However, the specific processor requirements for running V/IP are as follows:

- 80486sx 25 MHz minimum, any 80486dx or Pentium
- $\leq$ 8 MHz ISA bus
- 4 MB minimum system memory, 16 MB recommended (or, per Novell specifications for NetWare)
- one full length, 16-bit ISA bus card slot per Voice Interface Card
- 5 MB available hard disk space
- 3½-inch diskette drive. For NetWare, this is required only for the workstation PC that will be used to copy files to the NetWare server.
- Monochrome, CGA, EGA, or VGA display

When running in a DOS environment, up to eight Voice Interface Cards may be installed in a PC chassis.

When running in the Novell NetWare environment, up to four Voice Interface Cards may be installed in a PC chassis.

Note: The Voice Interface Cards might not fit into some "pizza box" style chassis. Also, the cards might not fit into slots where the fan or heat sink for the CPU is taking up excessive vertical space.

Note: The Voice Interface Cards must be installed in a CSA/UL Certified Fire Enclosure.

Power Supply

The following are special notices about the reinforced insulation requirement for the power supply of the PC chassis within which Voice Interface Cards can be installed.

Important: Voice Interface Cards require that the PC's power supply have reinforced insulation.

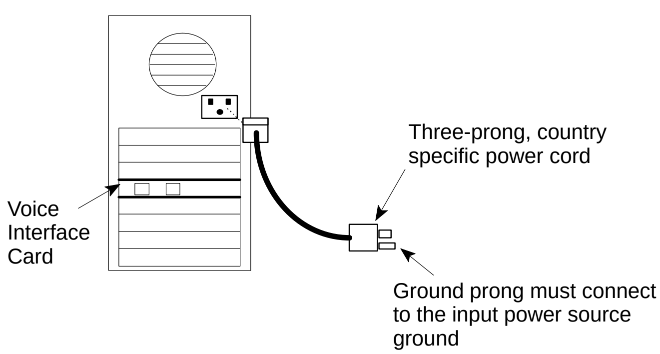
Caution

Voice Interface Cards can be installed only in a PC chassis that has a power supply that meets IEC 950 reinforced insulation requirements.

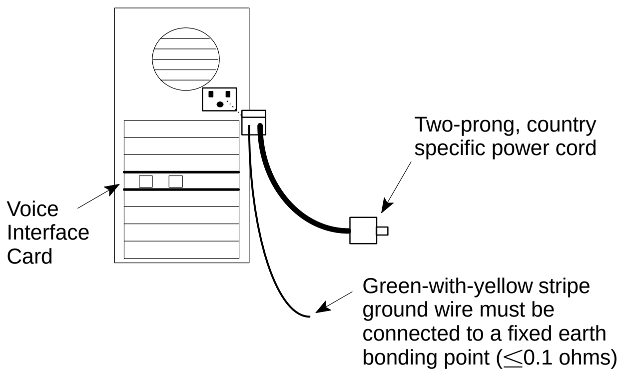
The following is special information about earth grounding that is required before connecting the card to the network.

WARNING

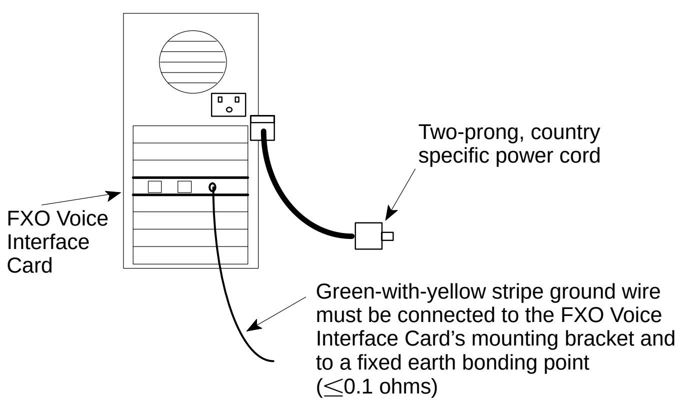
There must be a separate ground wire connection between the PC chassis and the input power source ground. A proper ground connection can be provided in one of three ways, as shown in the following three illustrations:



Connection Using A Three-Prong Power Cord



Connection Using A Two-Prong Power Cord With Separate Ground Wire



Connection Using A Ground Wire Between the Voice Interface Card and A Fixed Earth Bonding Point (FXO Voice Interface Card *only*)

For planning purposes, each card typically draws the following current from the PC's power supply:

FXS/E&M card:	+5V, 1 ampere -5V, 0.03 amperes +12V, 1.5 amperes during startup, 0.4 amperes sustained -12V, 0.03 amperes
FXO card:	+5V, 1 ampere -5V, 0.03 amperes +12V, 0.03 amperes -12V, 0.03 amperes

A typical PC's 200 watt power supply can supply the following current:

+5V, 22 amperes
-5V, 0.5 amperes
+12V, 8.0 amperes
-12V, 0.5 amperes

This PC would be limited to 5 FXS/E&M cards, due to the +12V 1.5 ampere startup current draw.

Operating System

- NetWare 3.12 or 4.x
- MS DOS 6.x

Network Interface Card (NIC)

See the `readme.txt` file in the Utilities diskette for a list of Network Interface Cards that have been tested with V/IP.

Configure the Card's Jumpers

You must perform the following steps before installing the card into the PC chassis.

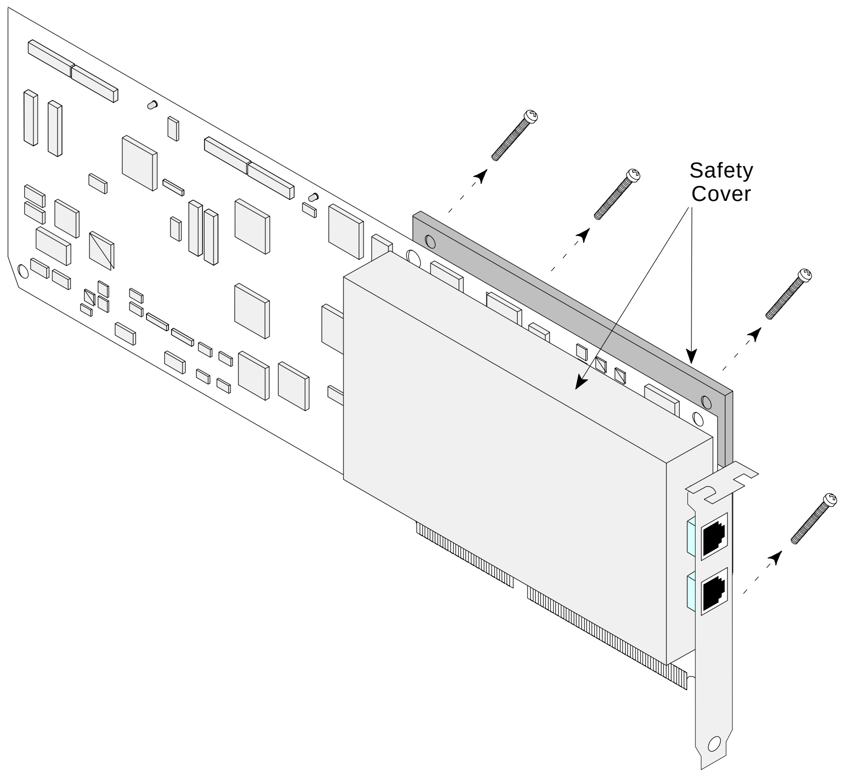
! WARNING

The following procedure must be performed only by a qualified technician.
The Voice Interface Card must not be installed in the computer chassis while performing the following procedure.

! Caution

The Voice Interface Card contains static-sensitive components. Be sure to use the standard methods for handling static-sensitive components. Wear a grounding wrist strap or touch the computer chassis before handling the card.

1. **Remove the safety cover:**



2. **Select the required analog interface (FXS/E&M Voice Interface Card *only*).**

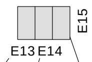
   Each channel can be configured individually for the type of analog interface you require: FXS, E&M type I, E&M type II, or E&M type V. The card is shipped from the factory preconfigured for FXS interface on all channels. However, even if you do plan on using the FXS interface, you should still check the jumper settings before installing the card in the PC chassis.

   The illustrations that follow show you how to configure each channel for each type of interface.

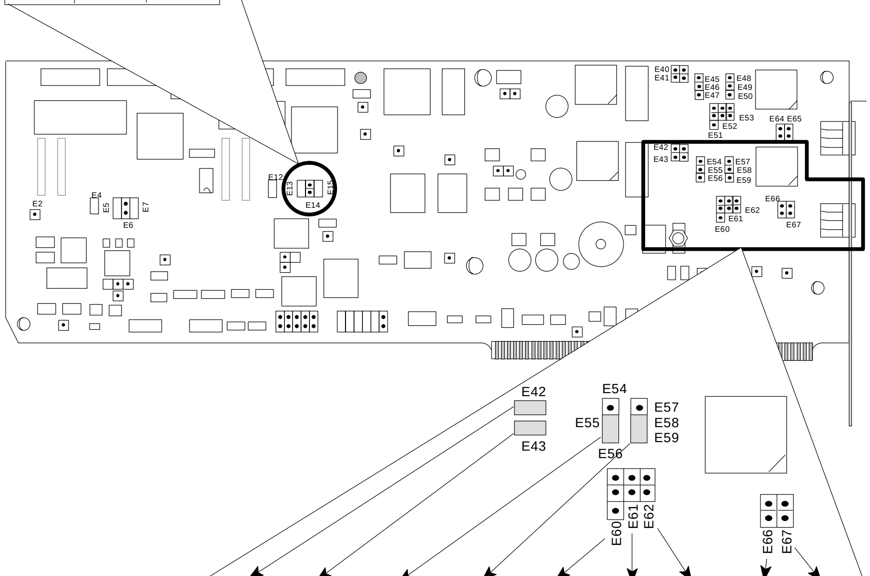
Note: The following diagrams show how to set the voice signaling for each channel of a Voice Interface Card. Note that Port 2 of a 2-channel Voice Interface Card is in the same physical location as Port 1 of a 1-channel card. Pages 2-14 to 2-17 show the settings for Port 1 of a 2-channel card.

FXS Interface

Port 2 of a 2-channel card
Port 1 of a 1-channel card



E13	E14	E15
In	In	In



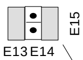
E42	E43	E54 E55 E56	E57 E58 E59	E60	E61	E62	E66	E67
In	In	E55 to E56	E58 to E59	Out	Out	Out	Out	Out

= jumper Out =

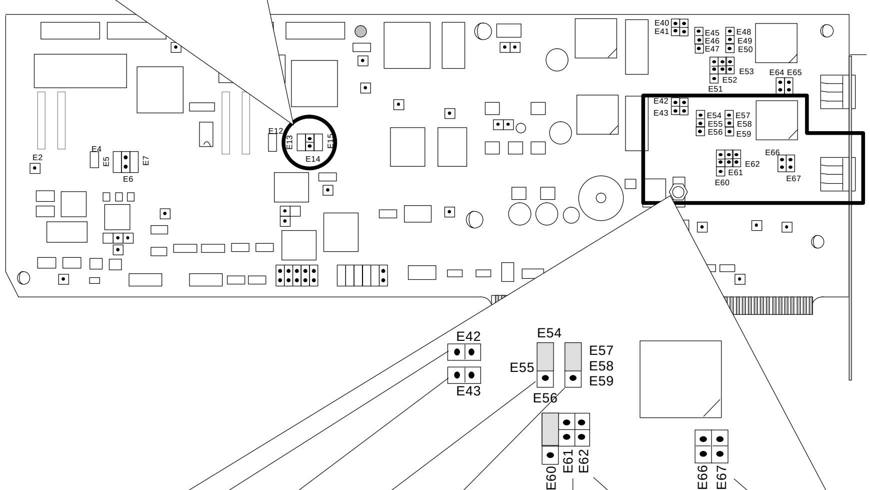
= jumper In =

E&M Type I Interface

Port 2 of a 2-channel card
Port 1 of a 1-channel card



E13	E14	E15
In	Out	In



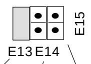
E42	E43	E54 E55 E56	E57 E58 E59	E60	E61	E62	E66	E67
Out	Out	E54 to E55	E57 to E58	1 to 2	Out	Out	Out	Out

= jumper Out =

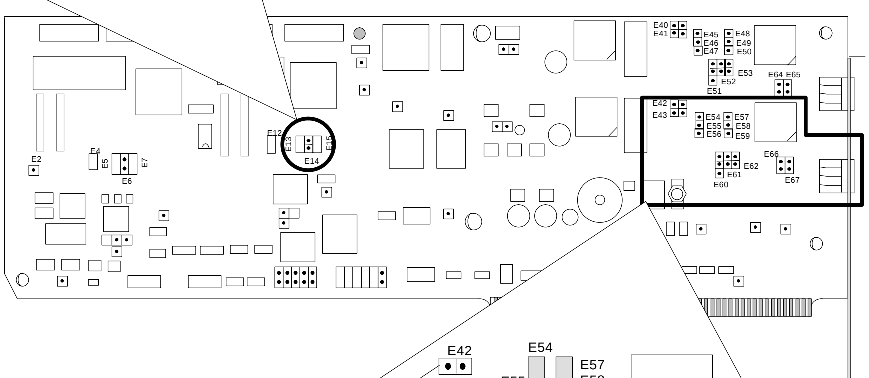
= jumper In =

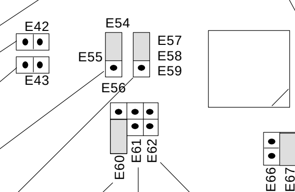
E&M Type II Interface

Port 2 of a 2-channel card
Port 1 of a 1-channel card

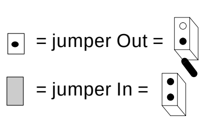


E13	E14	E15
In	Out	Out



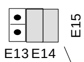


E42	E43	E54 E55 E56	E57 E58 E59	E60	E61	E62	E66	E67
Out	Out	E54 to E55	E57 to E58	2 to 3	Out	Out	Out	In

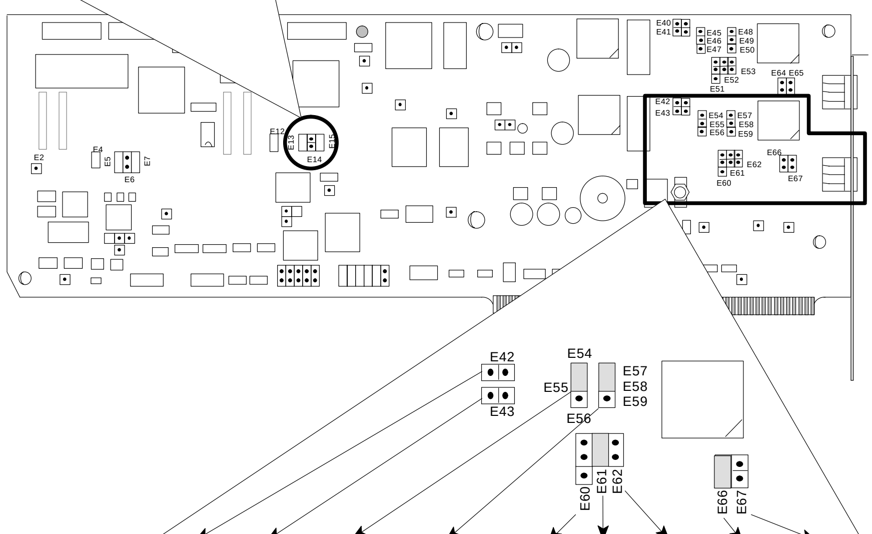


E&M Type V Interface

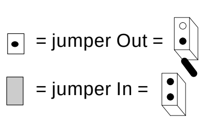
Port 2 of a 2-channel card
Port 1 of a 1-channel card



E13	E14	E15
Out	In	In

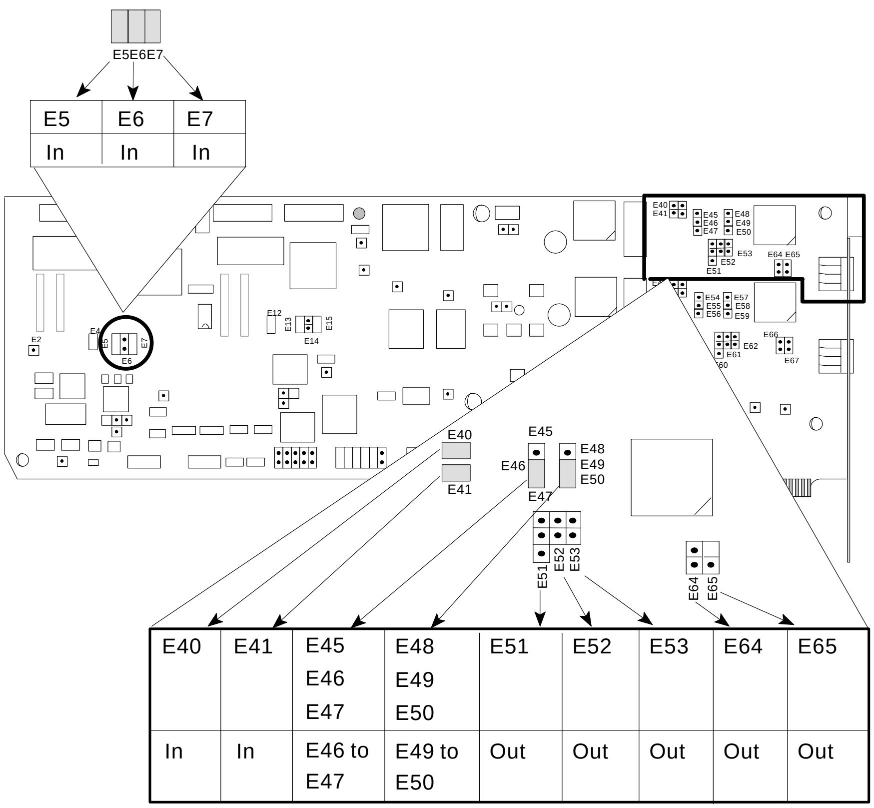


E42	E43	E54 E55 E56	E57 E58 E59	E60	E61	E62	E66	E67
Out	Out	E54 to E55	E57 to E58	Out	In	Out	In	Out



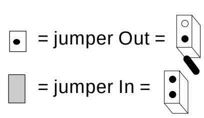
FXS Interface

Port 1 of a 2-channel card only



E5	E6	E7
In	In	In

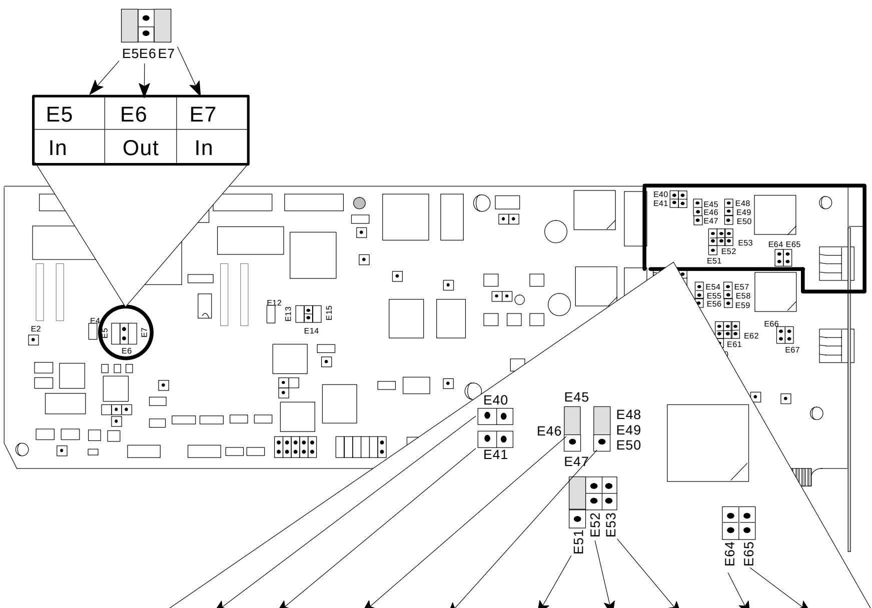
E40	E41	E45 E46 E47	E48 E49 E50	E51	E52	E53	E64	E65
In	In	E46 to E47	E49 to E50	Out	Out	Out	Out	Out



E&M Type I Interface

Port 1 of a 2-channel card only

E5	E6	E7
In	Out	In



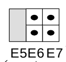
E40	E41	E45 E46 E47	E48 E49 E50	E51	E52	E53	E64	E65
Out	Out	E45 to E46	E48 to E49	1 to 2	Out	Out	Out	Out

= jumper Out =

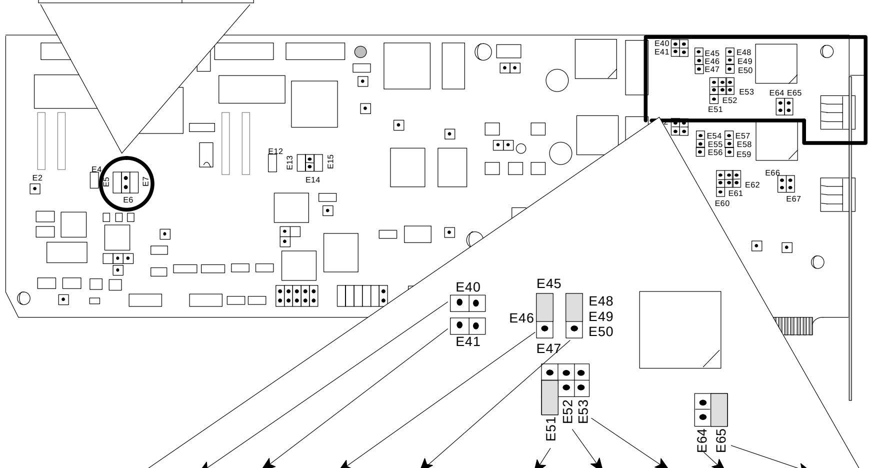
= jumper In =

E&M Type II Interf9ace

Port 1 of a 2-channel card only



E5	E6	E7
In	Out	Out



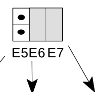
E40	E41	E45 E46 E47	E48 E49 E50	E51	E52	E53	E64	E65
Out	Out	E45 to E46	E48 to E49	2 to 3	Out	Out	Out	In

= jumper Out =

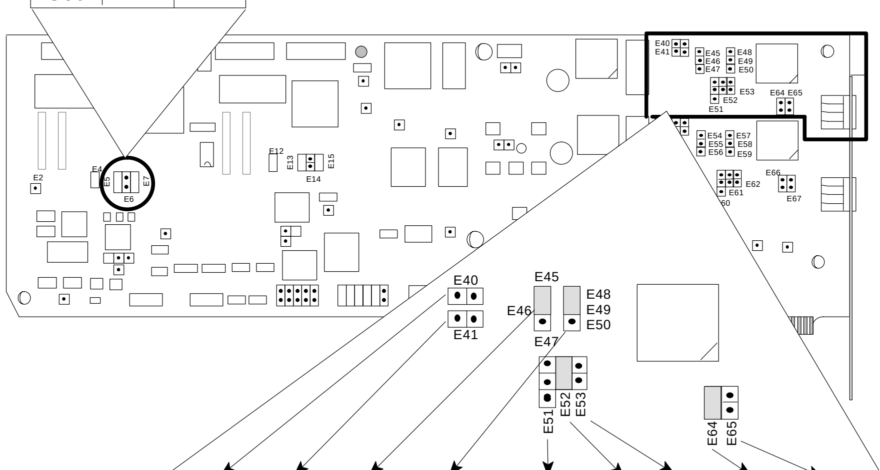
= jumper In =

E&M Type V Interface

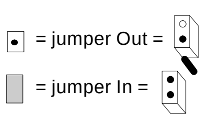
Port 1 of a 2-channel card only



E5	E6	E7
Out	In	In



E40	E41	E45 E46 E47	E48 E49 E50	E51	E52	E53	E64	E65
Out	Out	E45 to E46	E48 to E49	Out	In	Out	In	Out



3. **Configure the Input/Output Port Base Address.**

You must assign the base address for the I/O port addresses that the card requires. The I/O port addresses you can select from are:

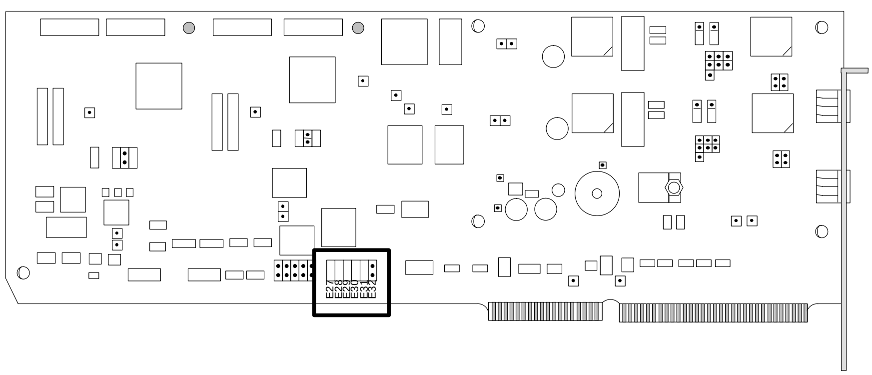
200, 210, 220, 230, 240, 250, 260, 270, 280, 290, 2A0, 2B0, 2C0, 2D0, 2E0, 2F0

300, 310, 320, 330, 340, 350, 360, 370, 380, 390, 3A0, 3B0, 3C0, 3D0, 3E0, 3F0

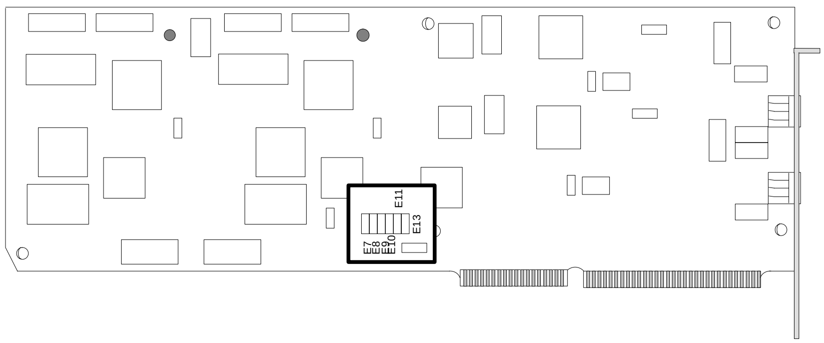
When choosing an I/O port base address, you must make sure there are no devices that are already using this address. For a list of potential conflicting devices, see Potential I/O Port Address Conflicts on page 2-21. The jumper settings are shown in the table on page 2-20.

Note: Be sure to write down the I/O port base addresses you assign to the Voice Interface Cards. You must enter these addresses when installing the software.

FXS/E&M card: set the I/O port base address on jumpers E27 to E32:

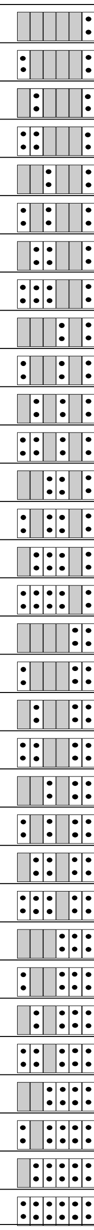


FXO card: set the I/O port base address on jumpers E7 to E13:



I/O Port Base Address

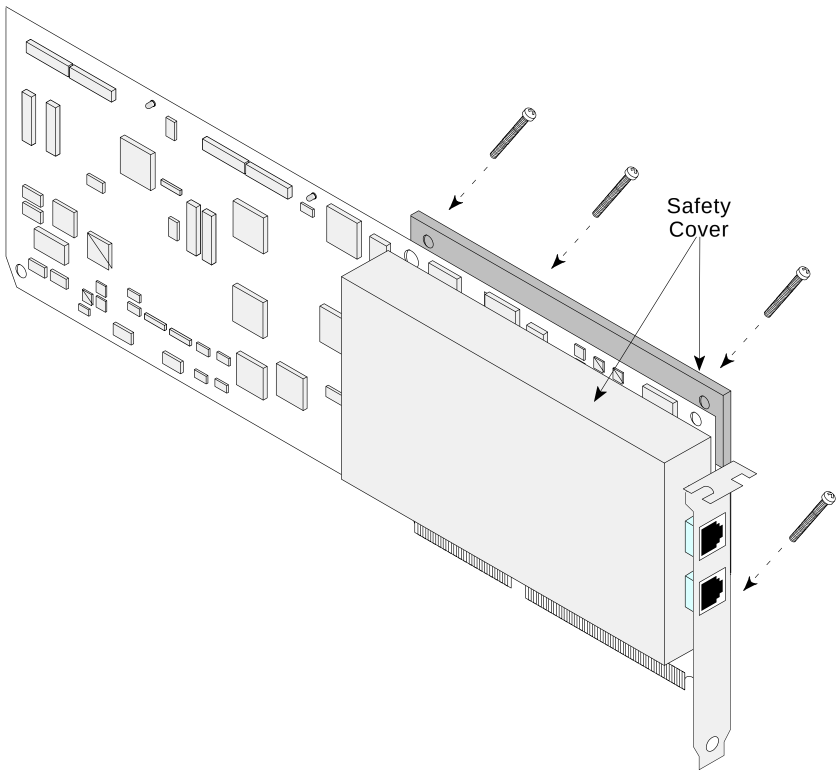
FXS/E&M Card → FXO Card →	E27 E7	E28 E8	E29 E9	E30 E10	E31 E11	E32 E13	Jumper Appearance
200 – 20F	In	In	In	In	In	Out	
210 – 21F	Out	In	In	In	In	Out	
220 – 22F	In	Out	In	In	In	Out	
230 – 23F	Out	Out	In	In	In	Out	
240 – 24F	In	In	Out	In	In	Out	
250 – 25F	Out	In	Out	In	In	Out	
260 – 26F	In	Out	Out	In	In	Out	
270 – 27F	Out	Out	Out	In	In	Out	
280 – 28F	In	In	In	Out	In	Out	
290 – 29F	Out	In	In	Out	In	Out	
2A0 – 2AF	In	Out	In	Out	In	Out	
2B0 – 2BF	Out	Out	In	Out	In	Out	
2C0 – 2CF	In	In	Out	Out	In	Out	
2D0 – 2DF	Out	In	Out	Out	In	Out	
2E0 – 2EF	In	Out	Out	Out	In	Out	
2F0 – 2FF	Out	Out	Out	Out	In	Out	
300 – 30F	In	In	In	In	Out	Out	
310 – 31F	Out	In	In	In	Out	Out	
320 – 32F	In	Out	In	In	Out	Out	
330 – 33F	Out	Out	In	In	Out	Out	
340 – 34F	In	In	Out	In	Out	Out	
350 – 35F	Out	In	Out	In	Out	Out	
360 – 36F	In	Out	Out	In	Out	Out	
370 – 37F	Out	Out	Out	In	Out	Out	
380 – 38F	In	In	In	Out	Out	Out	
390 – 39F	Out	In	In	Out	Out	Out	
3A0 – 3AF	In	Out	In	Out	Out	Out	
3B0 – 3BF	Out	Out	In	Out	Out	Out	
3C0 – 3CF	In	In	Out	Out	Out	Out	
3D0 – 3DF	Out	In	Out	Out	Out	Out	
3E0 – 3EF	In	Out	Out	Out	Out	Out	
3F0 – 3FF	Out	Out	Out	Out	Out	Out	



Potential I/O Port Address Conflicts

I/O Port Address (hexadecimal)	**Potential Conflicting Device**
200 - 20F	Game card, joystick (200-20F)
210 - 21F	
220 - 22F	Sound card, Token Ring
230 - 23F	Sound card
240 - 24F	Sound card
250 - 25F	
260 - 26F	
270 - 27F	LPT2 (278-27F)
280 - 28F	Network interface card (280-29F)
290 - 29F	Network interface card (280-29F)
2A0 - 2AF	
2B0 - 2BF	
2C0 - 2CF	
2D0 - 2DF	
2E0 - 2EF	COM4 (2E8-2EF)
2F0 - 2FF	COM2 (2F8-2FF)
300 - 30F	Network interface card, modem, etc. (300-31F)
310 - 31F	Network interface card, modem, etc. (300-31F)
320 - 32F	Network interface card (320-33F)
330 - 33F	Network interface card (320-33F)
340 - 34F	
350 - 35F	
360 - 36F	
370 - 37F	LPT1 (378-37F)
380 - 38F	
390 - 39F	
3A0 - 3AF	
3B0 - 3BF	
3C0 - 3CF	
3D0 - 3DF	
3E0 - 3EF	COM3 (3E8-3EF)
3F0 - 3FF	Floppy diskette controller (3F0-3F7), COM1 (3F8-3FF)

4. **Reinstall the safety cover after completing the jumper configurations.**



Install the Card Into the PC Chassis

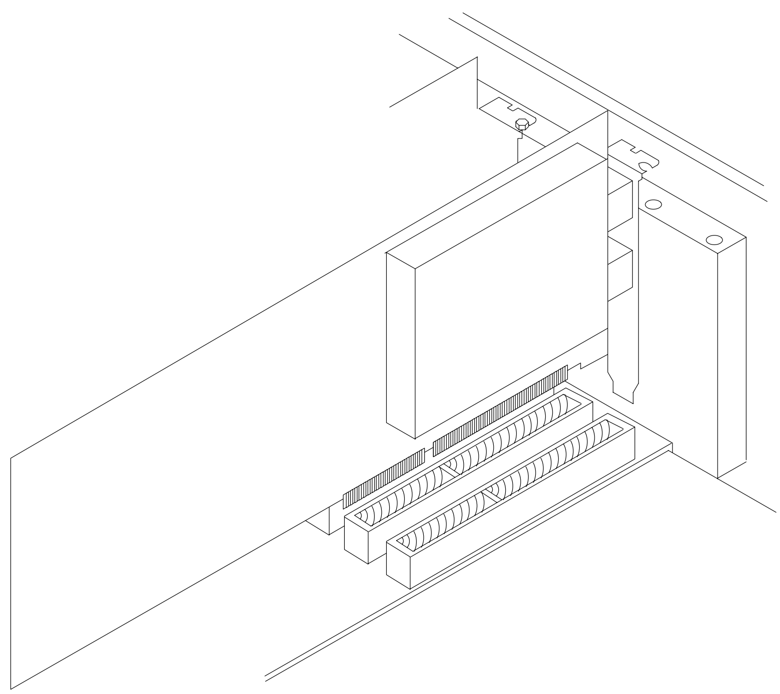
⚠ *WARNING*

Remove power from the computer chassis before installing the Voice Interface Card.

⚠ *Caution*

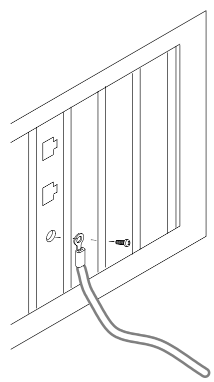
The Voice Interface Card contains static-sensitive components. Be sure to use the standard methods for handling static-sensitive components. Wear a grounding wrist strap or touch the computer chassis before handling the card.

You should refer to the hardware installation documentation that came with the computer chassis for the procedure on installing cards. Here is an example installation:



Connect to Earth Ground (FXO Voice Interface Card *only*)

If your installation site requires the connection of a separate ground wire between the FXO Voice Interface Cards and a fixed earth bonding point (see page 2-4 and 2-5 for earth ground information), you must make that connection now. The mounting bracket of the card must be connected directly to earth ground, as follows:



To earth ground

Connect to PBX, Telephone, or Fax Line

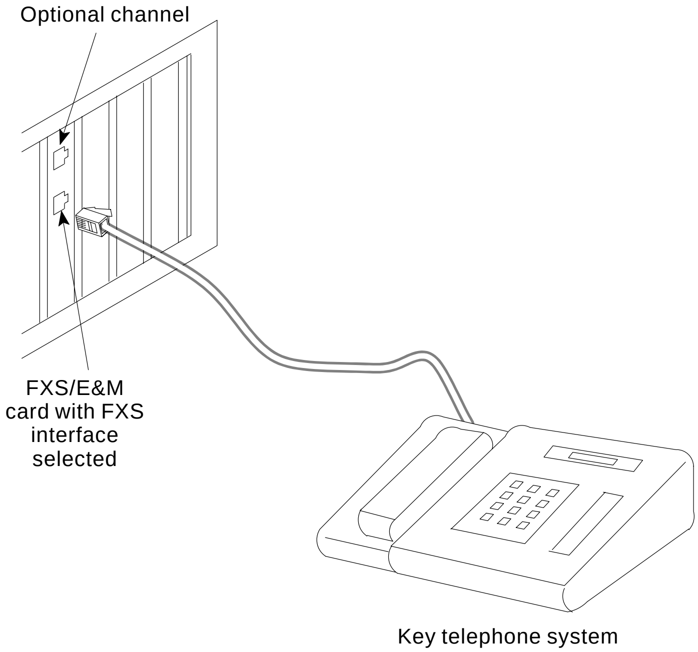
> ! **WARNING**
>
> Remove power from the computer chassis before making any connections to the analog interfaces on the Voice Interface Card.

Using the cable(s) supplied with the card, connect the PBX, telephone, or fax equipment to the channel(s) on the card. For pin assignments and technical details on the analog interfaces, refer to Appendix B.

! ***Caution***

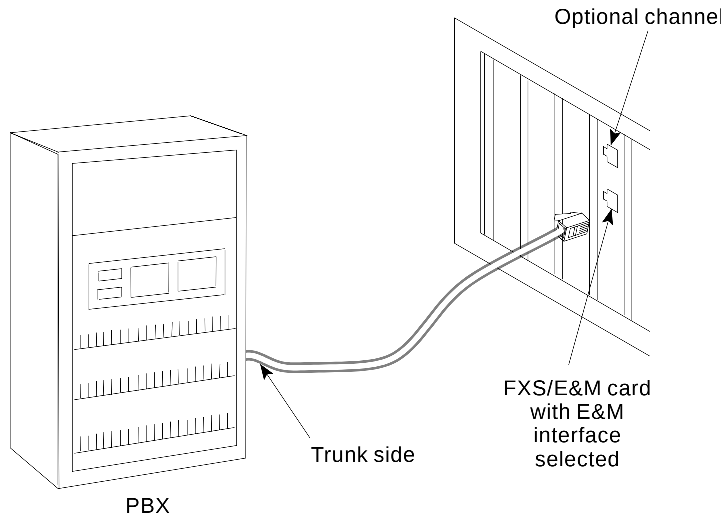
The RJ1CX connectors on the FXS/E&M card are 8-pin modular jacks. If you attempt to plug in 6-pin RJ11-type connectors, the outer pins of the RJ1CX connectors may be bent beyond the point where they can be used. The connectors will work for FXS interface, but may no longer function for E&M interface.

Here are some example analog interface connections:

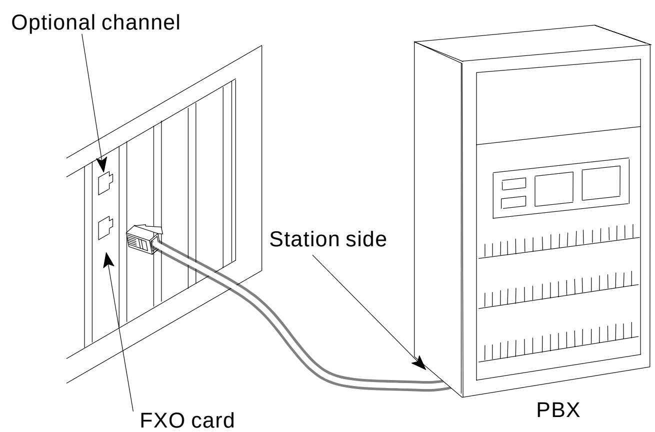


Note: The FXS interface is ***not*** intended for connection to the Public Switched Telephone Network.

FXS Interface to Key Telephone System



E&M Interface to PBX



⚠ *Caution*

When connecting FXO Voice Interface Cards to either the Public Switched Telephone Network, or to a PBX, you *must* use the factory-supplied cables for these connections.

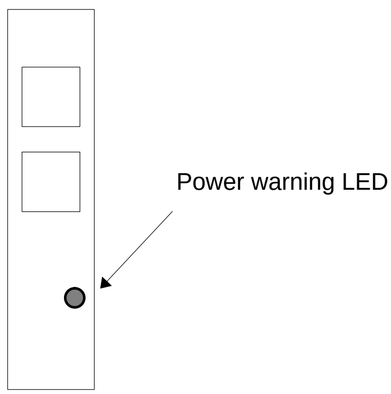
FXO Interface to PBX

Power up the PC

Power up the PC and verify that the machine and Voice Interface Card(s) are operating correctly. Then go on to Section 3 for software installation and configuration.

FXS/E&M Revision D (or later) Voice Interface Cards Only

The FXS/E&M Voice Interface Card has a power warning LED installed in the mounting bracket. You can view this LED as follows:



FXS/E&M Voice
Interface Card,
front view

The LED should turn on momentarily as the Voice Interface Card is powering up. If the LED stays on continuously, this means the card is drawing too much current from the PC's +12V supply. If this happens:

1. Shut down the PC.
2. Check the cable(s) from the Voice Interface Card channels to your telephone equipment. The LED can come on if certain pins are shorted.
3. Powerup the PC once again.
4. If the LED stays on, shut down the PC and remove the Voice Interface Card from the PC chassis.
5. Verify the setting of the jumpers. An incorrect configuration of the jumpers could cause the LED to come on.
6. Reinstall the Voice Interface Card and power up the PC once again. If the LED comes on again, contact your distributor for service.

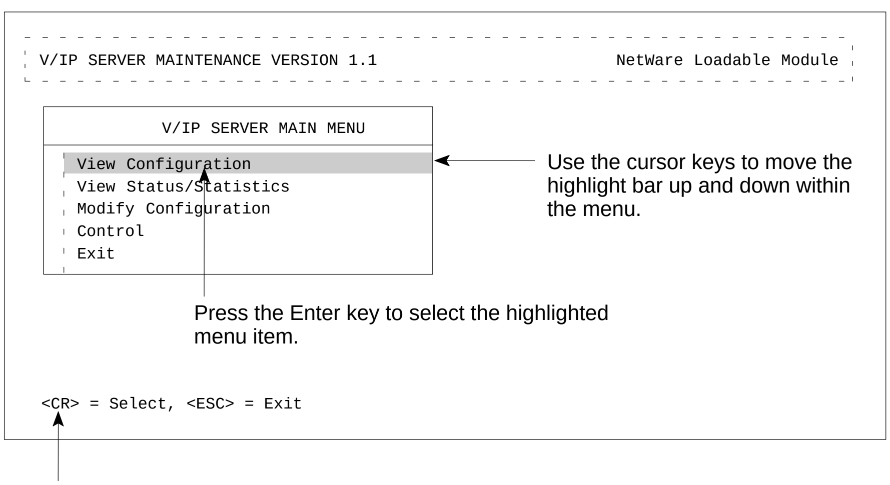
Software Installation and Configuration 3

This section will take you through the steps you must perform to install and configure the V/IP gateway software and get your Voice Interface Cards operating on your network. The contents of this section are:

- User Interface Description
- Planning Your V/IP Network
- Installing the Software
- Configuring the System Parameters
- Configuring the Voice/Fax Channels
- Configuring the Routing Protocol and Priority
- Hardware Parameter Conflicts

User Interface Description

The V/IP gateway software is controlled from a simple-to-use, menu-driven interface. Here is how you operate the menu-driven interface:



This line that tells you what keys are valid. They can be any of the following:

<CR>	= Enter key: selects the highlighted menu item.
<ESC>	= Esc key: exits (or, backs out) from the current menu.
<INSERT>	= Ins key: used in a very few places to add a new Voice Interface Card to the local V/IP gateway.
<DEL>	= Del key: used in a very few places to delete a Voice Interface Card from the local V/IP gateway

Planning Your V/IP Network

For initial software installation, there is a first time V/IP gateway configuration that is required. To quickly perform this configuration, you should first plan your V/IP network. The following charts will assist you in planning the configuration of the V/IP gateways.

When you have finished planning your V/IP network, continue on to *Installing the Software.*

V/IP Gateway Configuration Planning Chart

V/IP gateway PC:		Enter an identifying name for your reference purposes.
Gateway code:	____________________	1 to 9 digits. Must be unique for each V/IP gateway.
IP address:	____ . ____ . ____ . ____	Enter the IP address in decimal digits. This is the IP address defined for the Network Interface Card in the PC that is to be used for the V/IP gateway.
Phone directory database server?	Yes/No – if no, enter the IP address of the phone directory database server: ____ . ____ . ____ . ____	One PC must be designated the phone directory database server for the V/IP network.
Number of digits assigned for channel:	_____	0 to 7 digits. 0 defines all V/IP channels in this PC as one hunt group.
Maximum number of dial digits:	_____	0 to 16 digits. The maximum number of digits a Voice Interface Card in this PC can dial to the devices attached to its channels (phone lines).

Channel numbers

0 to 7 digits. The number to be dialed to call directly to this channel.

If *Number of digits assigned for channel* is 0, all channel numbers in the V/IP gateway will be set to 0 as well.

Specific channels can be assigned the same channel number. This makes them part of one hunt group.

Card 01 I/O address:	Port 1:	Port 2:
Card 02 I/O address:	Port 1:	Port 2:
Card 03 I/O address:	Port 1:	Port 2:
Card 04 I/O address:	Port 1:	Port 2:
Card 05 I/O address:	Port 1:	Port 2:
Card 06 I/O address:	Port 1:	Port 2:
Card 07 I/O address:	Port 1:	Port 2:
Card 08 I/O address:	Port 1:	Port 2:

Here is an example of a completed planning chart reflecting the configuration of the headquarters PC in the network diagram shown on page 1-8.

V/IP gateway PC:	***Headquarters***	Enter an identifying name for your reference purposes.
Gateway code:	***805***	1 to 9 digits. Must be unique for each V/IP gateway.
IP address:	***192*** . ***30*** . ***18*** . ***11***	Enter the IP address in decimal digits. This is the IP address defined for the Network Interface Card in the PC that is to be used for the V/IP gateway.
Phone directory database server?	Yes/No – if no, enter the IP address of the phone directory database server: ____ . ____ . ____ . ____	One PC must be designated the phone directory database server for the V/IP network.
Number of digits assigned for channel:	***0***	0 to 7 digits. 0 defines all V/IP channels in this PC as one hunt group.
Maximum number of dial digits:	***4***	0 to 16 digits. The maximum number of digits a Voice Interface Card in this PC can dial to the devices attached to its channels (phone lines).

Channel numbers	0 to 7 digits. The number to be dialed to call directly to this channel. If *Number of digits assigned for channel* is 0, all channel numbers in the V/IP gateway will be set to 0 as well. Specific channels can be assigned the same channel number. This makes them part of one hunt group.

Card 01 I/O address: ***200***	Port 1: ***0***	Port 2: ***0***
Card 02 I/O address: ***220***	Port 1: ***0***	Port 2: ***0***
Card 03 I/O address: ***300***	Port 1: ***0***	Port 2: ***0***
Card 04 I/O address: ***320***	Port 1: ***0***	Port 2: ***0***

Here is an example of a completed planning chart reflecting the configuration of the field office PC in the network diagram shown on page 1-8.

V/IP gateway PC:	***Field office***	Enter an identifying name for your reference purposes.
Gateway code:	***415***	1 to 9 digits. Must be unique for each V/IP gateway.
IP address:	***192*** . ***30*** . ***19*** . ***123***	Enter the IP address in decimal digits. This is the IP address defined for the Network Interface Card in the PC that is to be used for the V/IP gateway.
Phone directory database server?	Yes/No – if no, enter the IP address of the phone directory database server: ***192*** . ***30*** . ***18*** . ***11***	One PC must be designated the phone directory database server for the V/IP network.
Number of digits assigned for channel:	***4***	0 to 7 digits. 0 defines all V/IP channels in this PC as one hunt group.
Maximum number of dial digits:	***0***	0 to 16 digits. The maximum number of digits a Voice Interface Card in this PC can dial to the devices attached to its channels (phone lines).

Channel numbers

0 to 7 digits. The number to be dialed to call directly to this channel.

If *Number of digits assigned for channel* is 0, all channel numbers in the V/IP gateway will be set to 0 as well.

Specific channels can be assigned the same channel number. This makes them part of one hunt group.

Card 01 I/O address: ***200***	Port 1: ***4000***	Port 2: ***4000***
Card 02 I/O address: ***220***	Port 1: ***4001***	Port 2:
Card 03 I/O address:	Port 1:	Port 2:
Card 04 I/O address:	Port 1:	Port 2:

Installing the Software

Perform the software load procedure beginning with the PC that is going to be the V/IP phone directory database server for the network.

Important: If the PCs used as V/IP gateways are to be taken down for any reason, you must bring up the PC that is the V/IP phone directory database server first. The rest of the PCs will download the phone directory database from that PC when they load the V/IP software.

You can perform either *For NetWare 3.12 and 4.x Servers* (below) or *For DOS PCs* on page 3-17, whichever applies to your PCs.

For NetWare 3.12 and 4.x Servers

The software installs as NetWare Loadable Modules (NLMs).

1. Prepare the NetWare Servers.

   You must set the default gateway/router. This is the router that provides the WAN access. To set this in a NetWare server, proceed as follows:

   a. Copy the file `gateways` from the `/etc/samples` directory to the `/etc` directory.

   b. Add the following line to the new `/etc/gateways` file:

      ```
      net 0.0.0.0 gateway n.n.n.n metric 2 active
      ```

      where: *n.n.n.n* is the IP address of the gateway/router

   If a *net 0.0.0.0* line already exists in the `/etc/gateways` file, then modify the IP address as required.

Note: If there is no gateway/router on the LAN segment to which the V/IP PC is connected, you should enter the PC's own IP address as the gateway/router.

2.. Run the V/IP for NetWare software setup program.

a. Insert the *V/IP for NetWare* diskette into the diskette drive of the NetWare server. Enter the following command at the server prompt:

`TECH_PUBS:` **`load a:setup.nlm`**

You will see this display:

```
NIC Parameters

IP Address                 []
IP Mask                    []
Port                       []
Network Driver Filename    []
Done
```

b. Select `IP Address` from the display and enter the IP address of the Network Interface Card:

`IP Address :` **`192.30.18.10`**

c. Select `IP Mask` from the display and enter the IP subnet address mask required for the network within which the server is located. You may not have to fill in this information, as the V/IP software will automatically enter a subnet address that is appropriate to the IP address you entered previously.

`Mask :` **`255.255.255.0`**

d. Select `Port` from the display and enter the I/O port address of the PC's Network Interface Card:

`Port :` **`300`**

e. Select `Network Driver Filename` and enter the path and file name of the driver for the PC's Network Interface Card. This file must have the extension .lan.

`Network Driver Filename :` **`c:\exp16.lan`**

f. Select **Done**.

3. The setup program will copy the following files to the server:

micmdrv.nlm to \SYSTEM\micmdrv.nlm

micmvoic.nlm to \SYSTEM\micmvoic.nlm

micmcfg.nlm to \SYSTEM\micmcfg.nlm

You will see the following prompt:

```
               Are you using an Ethernet NIC?

No
Yes
```

Select Yes if you are using an Ethernet Network Interface Card. Select No if you are using a Token Ring Network Interface Card. You will then see this display:

Note: From here on, the screens shown in this text represent those that would be displayed had you selected Ethernet Network Interface Card to the previous prompt. The displays for Token Ring do vary for those shown for Ethernet.

```
                       Message

File Micmdrv.nlm is copied to \SYSTEM\Micmdrv.nlm
File Micmvoic.nlm is copied to \SYSTEM\Micmvoic.nlm
File Micmcfg.nlm is copied to \SYSTEM\Micmcfg.nlm
```

Press the Enter key.

4. You will see this message:

```
                       Message

The following lines will be added to AUTOEXEC.NCF

load snmp ControlCommunity=public
load TCPIP
load c:\exp16.lan port=300 frame=ETHERNET_II name=VIP
bind ip to VIP addr=192.30.18.11 mask=255.255.255.0
load ipconfig
load micmvoic.nlm
```

This sequence of entries in `autoexec.ncf` is required to allow the V/IP for NetWare to startup automatically when the server is booted. Press the Enter key to this message. You will see this message:

```
Message

The following lines will be added to AUTOEXEC.NCF
If any of the above lines already appear in the
autoexec.ncf file they will be added a second time
which will result in errors on boot up. If this is
the case we suggest that you note the lines above
and manually edit your AUTOEXEC.NCF file rather
then run this setup utility.
```

Press the Enter key. You will see this message:

```
Message

AUTOEXEC.NCF is updated!

The original AUTOEXEC.NCF is backed up as autoexec.vip.
```

Press the Enter key.

5. You will see this message.

```
Message

To load V/IP:

Restart the server and V/IP will be loaded
```

Press the Enter key. You will see a message about steps to take if the V/IP software does not start after rebooting the server:

```
                         Message
If V/IP does not load properly retry loading the MICOM
Voice Interface Card using different resources. If
V/IP still does not load, shutdown the server and try
another MICOM Voice Interface Card. If you continue to
experience difficulties loading V/IP, please contact
your MICOM reseller. If you wish to setup V/IP using
different parameters, remember to first restore
AUTOEXEC.NCF from AUTOEXEC.VIP
```

6. Press the Enter key. You will see this message:

```
Do you want to undo the setup?
No
Yes
```

Select **No**, to save the installation. (The only reason to select Yes, is if there was a mistake make during the installation prompts and you wish to start the software setup from the beginning.) Press any key to return to the server prompt.

7. Down the server and reboot the server. This will start the V/IP gateway function. Several displays will flash as the software comes up. Use the Alt-Esc key sequence until the following display is shown on the screen.

```
          INSTALLATION
   IP Address         []
   Gateway Code       []
   Directory Server   [  ]
   Password           []
   Hardware Parameters
   Done
```

8. You must enter the following information:

Menu Item	What To Enter
IP Address	Enter the Internet Protocol address assigned to the Network Interface Card that the V/IP gateway is to use. Only one Network Interface Card should be used by all Voice Interface Cards in the PC. The IP address must be expressed decimally in the form: n.n.n.n The subnet portion of the IP address must match the subnet address of the local area network to which the PC is connected. If you need to change the IP address after you have entered it, see page 5-40 for instructions.
Gateway Code	Enter the numeric code assigned to this PC for identification purposes within the V/IP network. This code is part of the dialing sequence to call a Voice Interface Card channel on this PC. • The gateway code can be from one to nine digits long. • The gateway code must be the same length for all PCs in the V/IP network. • The gateway code must not begin with 0 or 9.

Menu Item	What To Enter
Directory Server	Select: • Yes, to designate this PC to be the V/IP gateway phone directory database server. One PC must be designated the phone directory database server for the V/IP network. • No, for all other PCs. A data entry form will pop up for you to enter the IP address of the phone directory database serve.
Password	Enter the password to be used to authenticate requests to the V/IP phone directory database server. As there is only one phone directory database server, the password must be entered the same at all PCs. • The password can be from 1 to 31 alphanumeric characters. • The password is not case sensitive, thus PASSWORD, password, and PassWord are treated as the same password. • The password is never displayed on the screen. The * character is used to represent password character entries.

Menu Item	**What To Enter**

HardwareParameters	When you select Hardware Parameters, the Card menu will appear:
	`CARD`
	The menu will be blank. Press the Insert key to add a card. CARD 01 will appear in the menu. Select **CARD 01**. The Hardware Parameters menu will appear.
	`HARDWARE PARAMETERS`
	`IRQ`
	`I/O Address`
	`Shared Memory Address`
	`Done`
	The parameters you set here will define card number 01. Card numbers determine the priority of the channels when they are used as a hunt group. *The lower the card number and the lower the channel number, the higher the priority. The highest priority available channel will be selected when a call is placed to a hunt group*.

Menu Item	**What To Enter**

Hardware Parameters (continued)	The parameters you must configure are:
	• IRQ. Select the Interrupt Request number for this card. The Interrupt Request numbers you can select from are:
	3, 4, 5, 7, 9, 10, 11, 15
	• I/O Address. Select the I/O port base address that you configured on the jumpers of this card.
	• Shared Memory Address. Select the starting shared memory address for this card. The address range is A0000 to EF000 hexadecimal. Each card requires a 4K block of shared memory. Valid starting shared memory addresses are:
	a0000, a1000, a2000, a3000, a4000,
	a5000, a6000, a7000, a8000, a9000,
	aa000, ab000, ac000, ad000, ae000,
	af000, b0000, b1000, b2000, b3000,
	b4000, b5000, b6000, b7000, b8000,
	b9000, ba000, bb000, bc000, bd000,
	be000, bf000, c0000, c1000, c2000,
	c3000, c4000, c5000, c6000, c7000,
	c8000, c9000, ca000, cb000, cc000,
	cd000, ce000, cf000, d0000, d1000,
	d2000, d3000, d4000, d5000, d6000,
	d7000, d8000, d9000, da000, db000,
	dc000, dd000, de000, df000, e0000,
	e1000, e2000, e3000, e4000, e5000,
	e6000, e7000, e8000, e9000, ea000,
	eb000, ec000, ed000, ee000, ef000
	Select **Done.** You will see the following popup menu:
	`Are you sure?`
	`No`
	`Yes`
	Select **Yes** to confirm your entries.
	If you have additional cards installed in the PC, press the Insert key to add CARD 02 and select the hardware parameters for card number 02. Continue in this way for each of the cards in the PC.

9. Finally, select **Done** from the menu. You will see a series of messages about the Voice Interface Card DSP ports being initialized. Once this is complete, the V/IP gateway is operational on your network.

```
Initializing DSP port No. 1 in Card No. 1. Please wait.
DSP port No. 1 in Card No. 1 is initialized.

Initializing DSP port No. 2 in Card No. 1. Please wait.
DSP port No. 2 in Card No. 1 is initialized.

Initializing DSP port No. 1 in Card No. 2. Please wait.
DSP port No. 1 in Card No. 2 is initialized.

Initializing DSP port No. 1 in Card No. 2. Please wait.
DSP port No. 2 in Card No. 2 is initialized.
```

10. Press Alt-Esc to return to the system console.

You will see the following information on the system console:

This line tells you what version of MICMCFG.NLM and revision of MICMDRV.NLM that MICMVOIC.NLM will work with.

```
VoiceOverIP(V/IP)  Vers. 1.1  Aug.29,1997
User-Interface Level Version 1.04  Driver Revision 0
(c) Copyright 1996 MICOM Communications Corp.
All Rights Reserved
Patent Pending – MICOM Communications Corp.
14:46:21 RSVP Raw Socket Not Supported
14:46:21 Using UDP encapsulation
```

Note: If you encounter problems with conflicts between the IRQ numbers and shared RAM addresses of various boards, refer to Hardware Parameter Conflicts on page 3-43 to aid in resolving the conflicts.

12. Start the V/IP console by entering the following:

TECH_PUBS: **load micmcfg**

You will see the V/IP console main menu:

```
      V/IP SERVER MAINTENANCE VERSION 1.1                  NetWare Loadable Module

            V/IP SERVER MAIN MENU
     View Configuration
     View Status/Statistics
     Modify Configuration
     Control
     Exit

<CR> = Select, <ESC> = Exit, <F1> = Help
```

Now, go to page 3-25, *Configuring the System Parameters*.

For DOS PCs

Important: V/IP gateway software for DOS is designed to operate within a PC dedicated entirely to V/IP. The software requires that the PC's conventional and extended memory be made available completely to the V/IP application. Thus, the PC must not be running any application other than V/IP. Specifically,

- The PC cannot be running Windows, Windows 95, Windows NT, OS/2, or any environment other than DOS.
- No terminate and stay resident programs may be loaded.
- No programs may be loaded into high memory. That is, you cannot load HIMEM.SYS to use the computer's upper memory.
- Simply speaking, we recommend a PC clean booted in DOS, with no CONFIG.SYS or AUTOEXEC.BAT files.

Here are some other items you must address when setting up a DOS PC for V/IP:

- Motherboards and Network Interface Cards must have any plug and play features disabled at the BIOS level.
- We recommend that you disable the COM ports, LPT ports, and CD ROM drive, if applicable.
- You should disable any power saving features in the BIOS.
- IRQs should be assigned to ISA/EISA instead of PCI in the BIOS.

1. Clean boot the PC. (This means no CONFIG.SYS or AUTOEXEC.BAT files.)

2. Make sure you have configured your PC's Network Interface Card (NIC) using the utilities supplied with the card.

   As part of this step, you must have the Network Interface Card's LAN driver file located in the root directory (`C:\`) of the PC. This file will have the extension ".lan". Here are a couple of examples:

   ```
   3c90x.lan
   epro.lan
   ```

3. Run the V/IP for DOS software setup program.

   a. Insert the *V/IP for DOS* diskette into the PC's diskette drive. Log on to the diskette drive and enter `setup`, as follows:

      C:\> **a:**

      A:\> **setup**

      You will see this display:

```
            V/IP Parameters
   Working Directory          []
   Network Driver Filename    []
   Done
```

   b. Select `Working Directory` from the display and enter the directory path where you want the V/IP software installed. The suggested path is `c:\micmvoic`:

      Working Directory : **c:\micmvoic**

      Select `Network Driver Filename` from the display and enter the path and file name of the driver for the PC's Network Interface Card. This file must have the extension `.lan`.

      Network Driver Filename : **c:\3c90x.lan**

   c. Select **Done**.

4. You will see these messages:

   copying a:\micmvoic.exe to c:\micmvoic\micmvoice.exe

   copying a:\dos4gw.exe to c:\micmvoic\dos4gw.exe

   You will see this prompt:

```
   Please insert the Utilities disk now.
   <press Enter to continue>
```

   Remove the V/IP for DOS diskette and insert the Utilities diskette. You will see this message:

   copying a:\d4grun.exe to c:\micmvoic\d4grun.exe

5. You will see this message:

```
                        Message
 File Micmvoice.exe is copied to c:\micmvoic
 File Dos4gw.exe is copied to c:\micmvoic
 File D4grun.exe is copied to c:\micmvoic
 File c:\3c90x.lan is copied to c:\micmvoic
```

Press the Enter key to the above message. You will see this message:

```
Do you want to update autoexec.bat to run V/IP automatically?
   Yes
   No
```

- You should normally select **Yes**, so that the V/IP gateway comes up automatically in the event of a power failure. You will then receive a message that the original `autoexec.bat` file has been backed up as `autoexec.vip`.
- If you choose not to have the V/IP gateway come up automatically, you will see a message that details how to manually start the V/IP software. Make sure you record this information.

6. Press the Enter key. You will see this message:

```
 Do you want to undo the setup?
 Yes
 No
```

Select **No**, to save the installation. (The only reason to select Yes is if there was a mistake made during the installation prompts and you wish to start the software setup from the beginning.)

7. Press any key to return to the DOS prompt.

8. Take the Utilities diskette out of the disk drive and reboot the PC. When the PC comes up, you should see a display similar to the following:

```
    NIC Configuration Utility
  Primary I/O Port = [0x270]
  Finished Config
```

Configure the Network Interface Card's I/O port address (and other parameters listed, if any), and then select `Finished Config`.

9. Once the Network Interface Card is initialized, you will see the Installation Menu:

```
                INSTALLATION
 IP Address                [] 
 Default Router Address    [] 
 IP Network Mask           [0.0.0.0]
 Gateway Code              [] 
 Directory Server          [  ]
 Password                  [] 
 Hardware Parameters
 Done
```

You must enter the following information:

Menu Item	What To Enter
`IP Address`	Enter the Internet Protocol address assigned to the Network Interface Card that the V/IP gateway is to use. Only one Network Interface Card should be used by all Voice Interface Cards in the PC. The IP address must be expressed decimally in the form: `n.n.n.n` The subnet portion of the IP address must match the subnet address of the local area network to which the PC is connected. If you need to change the IP address after you have entered it, see page 5-40 for instructions.
`Default Router Address`	Enter the IP address of the router that provides the WAN access. If there is no router on the LAN segment to which this PC is connected, enter the PC's own IP address in this field.
`IP Network Mask`	Enter the IP address subnet mask for the network.
`Gateway Code`	Enter the numeric code assigned to this PC for identification purposes within the V/IP network. This code is part of the dialing sequence to call a Voice Interface Card channel on this PC. • The gateway code can be from one to nine digits long. • The gateway code must be the same length for all PCs in the V/IP network. • The gateway code must not begin with 0 or 9.

Menu Item	What To Enter
`Directory Server`	Select: • Yes, to designate this PC to be the V/IP gateway phone directory database server. One PC must be designated the phone directory database server for the V/IP network. • No, for all other PCs. A data entry form will pop up for you to enter the IP address of the phone directory database server.
`Password`	Enter the password to be used to authenticate requests to the V/IP phone directory database server. As there is only one phone directory database server, the password must be entered the same at all PCs. • The password can be from 1 to 31 alphanumeric characters. • The password is not case sensitive, thus PASSWORD, password, and PassWord are treated as the same password. • The password is never displayed on the screen. The * character is used to represent password character entries.

<table>
<tr><th>Menu Item</th><th>What To Enter</th></tr>
<tr><td>Hardware
Parameters</td><td>When you select Hardware Parameters, the Card menu will appear:

CARD

Initially, the menu will be blank. Press the Insert key to add a card. CARD 01 will appear in the menu. Select CARD 01.

You will see this prompt:

Is it a T1/E1 Expansion Card?
Yes
No

Select No.

The Hardware Parameters menu will appear:

HARDWARE
IRQ
I/O Address
Shared Memory Address
Done

The parameters you set here will define card number 01. Card numbers determine the priority of the channels when they are used as a hunt group. The lower the card number, the higher the priority. The highest priority available channel will be selected when a call is placed to a hunt group.
The parameters you must configure are:
• IRQ. Select the Interrupt Request number for this card. The Interrupt Request numbers you can select from are:
3, 4, 5, 7, 9, 10, 11, 15
• I/O Address. Select the I/O port base address that you configured on the jumpers of this card.</td></tr>
</table>

Menu Item	What To Enter

`Hardware Parameters`, continued	• Shared Memory Address. Select the starting shared memory address for this card. The address range is A0000 to EF000 hexadecimal. Each card requires a 4K block of shared memory. Valid starting shared memory addresses are:
	
	a0000, a1000, a2000, a3000, a4000, a5000, a6000, a7000, a8000, a9000, aa000, ab000, ac000, ad000, ae000, af000, b0000, b1000, b2000, b3000, b4000, b5000, b6000, b7000, b8000, b9000, ba000, bb000, bc000, bd000, be000, bf000, c0000, c1000, c2000, c3000, c4000, c5000, c6000, c7000, c8000, c9000, ca000, cb000, cc000, cd000, ce000, cf000, d0000, d1000, d2000, d3000, d4000, d5000, d6000, d7000, d8000, d9000, da000, db000, dc000, dd000, de000, df000, e0000, e1000, e2000, e3000, e4000, e5000, e6000, e7000, e8000, e9000, ea000, eb000, ec000, ed000, ee000, ef000
	
	Select **`Done.`** You will see the following popup menu:
	
	`Are you sure ?`
	`No`
	`Yes`
	
	Select **`Yes`** to confirm your entries.
	
	If you have additional cards installed in the PC, press the Insert key to add CARD 02 and select the hardware parameters for card number 02. Continue in this way for each of the cards in the PC.

After you have completed the form, select **Done** from the Installation menu. The MICMVOIC module will initialize the Voice Interface Cards.

Note: At this point, the V/IP gateway is functional over your network.

Once the Voice Interface Card(s) has been initialized, you will see the V/IP console main menu:

```
V/IP SERVER MAINTENANCE VERSION 1.1

        V/IP SERVER MAIN MENU
  View Configuration
  View Status/Statistics
  Modify Configuration
  Control
  Exit

<CR> = Select, <ESC> = Exit
```

Now, go to page 3-25, *Configuring the System Parameters*.

Configuring the System Parameters

1. Select `Modify Configuration` from the Main Menu and `V/IP System` from the Module Selection menu:

```
   V/IP SERVER MAIN MENU
   View Configuration
   View Status/Statistics
1  Modify Configuration
   Control
   Exit
```

```
   MODULE SELECTION
   Analog Voice/Fax
   Digital Voice (DVC)
   T1/E1
   RSVP
   SNMP
2  V/IP System
   Hardware
   Done
```

When you select `V/IP System`, a screen similar to the following will display:

```
                         V/IP SYSTEM CONFIGURATION
Gateway Code                                      [12]
IP Address                                        [199.30.19.11]
Default Router Address                            [199.30.19.1]
IP Netowrk Mask                                   [255.255.255.0]
Phone Directory IP Address                        [199.30.25.223]
Password for Phone Database                       [**]
Synchronizing Interval                            [24]
UDP Port                                          [65535]
Number of Digits Assigned for Channel             [1]
Maximum Number of Dial Digits                     [0]
Default Number of Digits                          [5]
Default Inter-Digit Time                          [2]
Inter-Digit Time                                  [10]
Call Progress Tone                                [NORTH AMERICA]
Load Directory Database at Startup                [ENABLED]
Call Diconnect Timeout (min)                      [2]
Done
```

2. Configure the system parameters as required by your network.

You will have configured some of the parameters already while loading the software. Refer to the following table for information about each parameter.

System Configuration

Option	Default	Description
Gateway Code		The numeric code assigned to this PC for identification purposes within the V/IP network. This code is part of the dialing sequence to call a Voice Interface Card channel on this PC. • The gateway code can be from one to nine digits long. • The gateway code must be the same length for all PCs in the V/IP network. • The gateway code must not begin with 0 or 9.
IP Address		The IP address assigned to this PC for V/IP purposes. It must match the IP address assigned to the Network Interface Card being used for the V/IP gateway. The address must be entered as decimal digits in this form: `n.n.n.n` **Note:** If you need to change the IP address, see page 5-40 for instructions.
Default Router Address		DOS version only. The IP address of the router that provides access to the WAN. The address must be entered as decimal digits in this form: `n.n.n.n`
IP Network Mask		DOS version only. The IP address mask used on the network.
Phone Directory IP Address		The IP address of the PC that is the V/IP network’s phone directory database server. The address must be entered as decimal digits in this form: `n.n.n.n`

System Configuration (cont'd)

Option	Default	Description
Password for Phone Database		This password accompanies all requests by Voice Interface Cards to the phone directory database server. The password provides a level of security to prevent unauthorized access to the database. The password is also used to authenticate call setup and disconnect. • The password can be from 1 to 31 alphanumeric characters. • The password is not case sensitive. • The password is never displayed on the screen. The * character is used to represent each password character entry. **Note:** If you change the password, all calls or connections in progress will be disconnected. After that, the password will be changed, and new calls can be made.
Synchronizing Interval	24 hours	This is the interval, in hours, after which the PC will synchronize its phone directory database with the network's phone directory database server. The range is 1 to 255 hours.
UDP Port	65535	Controlled UDP Port number. Used by call set-up code to reject check channel request packets and for code downline load.
Number of Digits Assigned for Channel	1 or 2 digits, depending on the number of Voice Interface Cards in the PC	The number of digits in the dialing string that will specify a channel on the card. This parameter must be set the same for all cards in one server. The range is zero to seven digits. To quickly make all channels in the V/IP gateway as members of one hunt group, simply set this parameter to 0.

System Configuration (cont'd)

Option	Default	Description
Maximum Number of Dial Digits	0 digits	For an incoming call, this is the maximum number of digits a Voice Interface Card channel in this V/IP gateway will dial to the attached device (a PBX). The range is 0 to 16 digits.
Default Number of Digits	5 digits	An outgoing call attempt will be made if the Default Number of Digits has been dialed and the Default Inter-Digit Time has elapsed since the last digit was dialed. The range is 1 to 31 digits.
Default Inter-Digit Time	2 seconds	This is the maximum time expected between dialed digits. When the Default Number of Digits has been dialed, the V/IP gateway will detect the end of dialing for an outgoing call after the Default Inter-Digit Time has elapsed. The range is 1 to 15 seconds. **Note:** You should set the Default Inter-Digit Time to a value less than the Inter-Digit Time.
Inter-Digit Time	10 seconds	An outgoing call attempt will be made if this time has expired after the last digit has been dialed (regardless of the number of digits dialed). The range is 1 to 15 seconds. **Note:** You should set the Inter-Digit Time to a value greater than the Default Inter-Digit Time.

System Configuration (cont'd)

Option	Default	Description
Call Progress Tone	North America	Selects one of eight countries or areas where the V/IP network is installed. This controls the dial, busy (engaged), congestion (reorder), and ringback (audible ringing) tones. • North America • Japan • U.K. • Europe • France • Central America • Chile • Australia
Load Directory Database at Startup	Enabled	Normally, when the V/IP software starts up, the phone directory database is downloaded from the phone directory database server. You can disable this action to save bandwidth during the startup of the V/IP gateway. • Enabled • Disabled
Call Disconnect Timeout	2 minutes	Disconnects calls on FXO channels if the call setup cannot be completed by the specified time. The range is 1 to 240 minutes. 0 disables this function.

Select **Done** from the System Configuration menu when you are finished setting the system parameters.

Now, continue on to *Configuring the Voice/Fax Channels*.

Configuring the Voice/Fax Channels

Quick Start

For initial operation, you may need to configure only a couple of options, as listed below. In many cases, you can use the default values for the other options.

- **Channel Number**. If the channel is to be part of a hunt group, you must configure the channel number as follows:
  - Set the channel number to 0 for all channels in this V/IP gateway if all ports are to be members of one hunt group.

    Alternatively, you can set the parameter *Number Of Digits Assigned For Channel* to 0. This will set the channel numbers of all voice/fax channels in the V/IP gateway to 0.
  - Set more than one channel to the same channel number to make those individual channels members of one hunt group.

  If you want to allow calls to be placed directly to this channel, you must assign a channel number in accordance with the *Number Of Digits Assigned For Channel* parameter setting in the System Configuration menu.
- **Analog Operation** (E&M analog interface only). Set to match the requirements of the associated PBX.
- **Line Impedance**. Set to match the interfacing telephone equipment.

These options are presented when you modify the configuration of a voice/fax channel, as follows:

1

```
   V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
 Control
 Exit
```

2

```
  MODULE SELECTION
 Analog Voice/Fax
 Digital Voice (DVC)
 T1/E1
 RSVP
 SNMP
 V/IP System
 Hardware
 Done
```

3

```
        CARD – PORT
 CARD # 01 – PORT # 01
 CARD # 01 – PORT # 02
 CARD # 02 – PORT # 01
 CARD # 02 – PORT # 02
```

Select the channel you want to configure.

```
                  VOICE/FAX CONFIGURATION – FXS
 Channel Number                         [1]
 UDP Port Number                        [65534]
 Mode                                   [Voice/Fax]
 Input Level Gain                       [ 0 dB]
 Output Level Attenuation               [ 0 dB]
 Busyout Mode                           [SYSTEM CONTROLLED]
 Background                             [REGENERATED]
```

Summary Of Voice/Fax Channel Options

The following table summarizes the configuration options associated with each voice/fax channel. The options will vary, depending on the type of analog interface selected for that voice/fax channel.

Voice/Fax Channel Configuration Options

Option	Telephone Interface	Default	Description
Channel Number	All	The number of the channel being configured.	The number to be dialed as part of the dialing string in order to call directly to this channel. The range is zero to seven digits. When the *Number of Digits Assigned for Channel* parameter in the System Configuration menu is set to 0, all channels in that server become a hunt group. In this case, the V/IP gateway will automatically set all channel numbers to 0. When channels are members of a hunt group, an incoming call will be placed to an available channel in the hunt group. The lowest card number and second-lowest physical channel number has the highest priority. See page B-13 if you need more information about channel numbers. See page 4-5 for more information about hunt groups.

Voice/Fax Channel Configuration Options (cont'd)

Option	Telephone Interface	Default	Description
UDP Port Number	All	65534 (for card 01, channel 01), 65533 (for card 01, channel 02), 65532 (for card 02, channel 01), etc.	Specifies the UDP port number this voice/fax channel is to bind with when connecting to a remote Voice Interface Card. The port number must be unique for each channel. The valid UDP port number range is 1024 to 65535. You might need to change the value of the UDP port numbers if you have a firewall installed. However, it may be more convenient to make changes to the firewall itself to accommodate V/IP. **Note:** If you change the UDP port number for a channel that has a call or connection in progress, that channel's call or connection will be disconnected. See page B-13 if you need more information about UDP port numbers.

Voice/Fax Channel Configuration Options (cont'd)

Option	Telephone Interface	Default	Description
Mode	All	Voice/Fax	Selects the operating mode of the channel. • Voice/Fax, sets the channel to automatically detect a fax call. • Voice Only, the channel will process all calls as voice calls. See page B-13 for more information about the mode parameter.
Input Level Gain	All	0 dB	Selects one of 13 input signal gain values in 1 dB increments. -6 = maximum attenuation 0 = no gain 6 = maximum gain Must be set to 0 dB for U.S.A. and Canada. See page B-14 if you want more information about input level gain.
Output Level Attenuation	All	0 dB	Selects one of 20 output level attenuation values in 1 dB increments. 0 = no attenuation 19 = maximum attenuation Must be set to 0 dB for U.S.A. and Canada. See page B-14 if you need more information about output level attenuation.

Voice/Fax Channel Configuration Options (cont'd)

Option	Telephone Interface	Default	Description
Busyout Mode	All	System Controlled	Selects one of two methods of busyout control: • Forced On • System Controlled See page B-14 if you want more information about busyout mode
Background	All	Regenerated	Selects either regenerating or suppressing background noise during idle periods. • Regenerated • Silence See page B-14 if you want more information about the background parameter.
Regeneration Type	All	Dial Pulse	Specifies one of two types of regeneration format: • Dial Pulse • DTMF See page B-14 if you need more information about the regeneration type parameter.
Fax Digitizing Rate	All	9600	Selects one of four digitizing rates for fax operation: • 9600 • 7200 • 4800 • 2400 See page B-15 if you want more information about fax digitizing rates.

Voice/Fax Channel Configuration Options (cont'd)

Option	Telephone Interface	Default	Description
Regeneration Delay	All	1 second	Delays from 1 to 15 seconds the forwarding of dial digits to the destination PBX. See page B-15 for more information about regeneration delay.
Automatic Level Enhancement	All	Disabled	Enables or disables the automatic level enhancement function for the input level of this channel. • Enabled • Disabled See page B-15 for information about automatic level enhancement.
Call Inhibit	All	Disabled	When set to Enabled, the channel will not be able to call another channel. The channel can only answer calls (if Receive Inhibit = Disabled). • Enabled • Disabled See page B-15 if you want more information about call inhibit.

Voice/Fax Channel Configuration Options (cont'd)

Option	Telephone Interface	Default	Description
Receive Inhibit	All	Disabled	When set to Enabled, the channel will not answer calls. The channel can only originate calls (if Call Inhibit = Disabled). • Enabled • Disabled **Note:** Setting this parameter to Disabled will not stop the PBX from attempting to connect to this channel. See page B-16 if you want more information about receive inhibit.
Autocall Extension Number	All		When the equipment connected to this channel goes off-hook, the card will automatically place a call to the configured extension number. The autocall extension number range is from 1 to 32 digits. **Note:** The autocall number must *not* include the PBX access code for E&M interface. The caller must enter the PBX access code before the autocall is placed. See page B-16 if you need more information about autocall extension numbers.

Voice/Fax Channel Configuration Options (cont'd)

Option	Telephone Interface	Default	Description
Jitter Buffer	All	Static	Reserved for future use. For this release, only the static jitter buffer is supported. See page B-17 if you need more information about the jitter buffer.
Jitter Buffer Size	All	50 milliseconds	Jitter buffer adjustments are needed when excessive delays are occurring between the two ends of a calls. The jitter buffer takes effect on the receiving side of a call. Jitter Buffer Size sets the size of the static jitter buffer. You enter the desired jitter buffer size directly in increments of milliseconds. The Jitter Buffer Size range is from 50 to 5000 milliseconds. (5000 milliseconds = 5 seconds.) See page B-17 for more information about jitter buffer size.
Forward Error Correction	All	Enabled	Enables or disables the error correction feature. If the ability of packets to transit the network is fairly certain, you can disable the error correction and save on bandwidth. However, if there is a risk to packets being dropped, or being received in error, you should keep the error correction enabled. • Enabled • Disabled See page B-18 for more information about error correction.

Voice/Fax Channel Configuration Options (cont'd)

Option	Telephone Interface	Default	Description
Echo Canceller	All	Disable	Allows for disabling the echo canceller. This feature is used in selected loopback test applications. • Enable • Disable See page B-18 if you need more information about the echo canceller parameter.
DTMF Detector	All	Disable	When enabled, this feature ensures that DTMF signals at the remote end are regenerated with a uniform on/off time of 100 milliseconds. • Enable • Disable See page B-22 if you need more information about the DTMF detector.
Analog Operation	E&M	2-wire	Selects the type of operation for the E&M channel: • 2-wire • 4-wire
Line Impedance	FXS, E&M 2-wire, and FXO	600 ohms	Selects one of two line impedance matching types: • 600 ohms • Complex See page B-19 for more information about line impedance.

Voice/Fax Channel Configuration Options (cont'd)

Option	Telephone Interface	Default	Description
Maximum Output Level	E&M 4-wire	0 dB	Selects one of two maximum output levels: • 0 dB • +7 dB See page B-19 for more information about maximum output level.
Port Emulation	E&M	DC	Selects one of two port emulations: • DC • Wink Start See page B-19 for information about the port emulations.
Ringing Frequency	FXS	25 Hz	Selects the signal used to ring the bell on the called telephone. • 25 Hz • 50 Hz See page B-21 if you need more information about ringing frequency.
Number of Ring Cycles	FXO	1	Selects the number of ring backs before receipt of the dial tone. The range is 1 to 9. See page B-21 for more information about the number of ring cycles.
Disconnect Supervision	FXO	Tone	Selects the type of disconnect supervision. • Tone • Power Interrupt. See page B-22 for more information about disconnect supervision.

Select **Done** from the Voice/Fax Configuration menu when you are finished configuring the parameters for this channel. You will see this popup menu:

```
SAVE CONFIGURATIONS ?
No
Yes
```

Select **Yes** to save the configuration.

After all channels have been configured, continue on to *Configuring the Routing Protocol and Priority*.

Configuring the Routing Protocol and Priority

Routing Protocol

For the NetWare Environment

Novell TCP/IP runs the Routing Information Protocol (RIP).

- If your routers use RIP as the routing protocol, you must do the following configuration at each Novell server:
  1. Assign an IP address to the Network Interface Card.
  2. Bind `micmvoic.nlm` to the Network Interface Card.
- If your routers use a routing protocol other than RIP, you must:
  1. Assign an IP address to the Network Interface Card.
  2. Bind `micmvoic.nlm` to the Network Interface Card.
  3. Enter the static/default route in the `/etc/gateways` file in the Novell servers.

For the DOS Environment

The V/IP gateway only requires a static/default route be defined. You have already made this entry during the software installation. This was done when you entered the IP address of the router that provides WAN access (the Default Router entry shown on page 3-20).

Priority Configuration

You may need to configure the routers in your network to use a form of priority handling for the voice and fax packets transmitted by the V/IP gateway. These packets cannot be retransmitted if dropped due to network congestion. The priority schemes can be any of the following, in order of preference:

- Weighted Fair Queuing.
- Prioritize IP packets over IPX packets.
- Prioritize by UDP port numbers. The voice/fax packets sent out by the V/IP gateway are UDP packets with the source UDP port = 1EA hexadecimal (490 decimal). You should prioritize UDP packets with the source UDP port = 1EA hexadecimal.
- Prioritize by IP addresses.

Hardware Parameter Conflicts

Here are some potential conflicts regarding the hardware parameters:

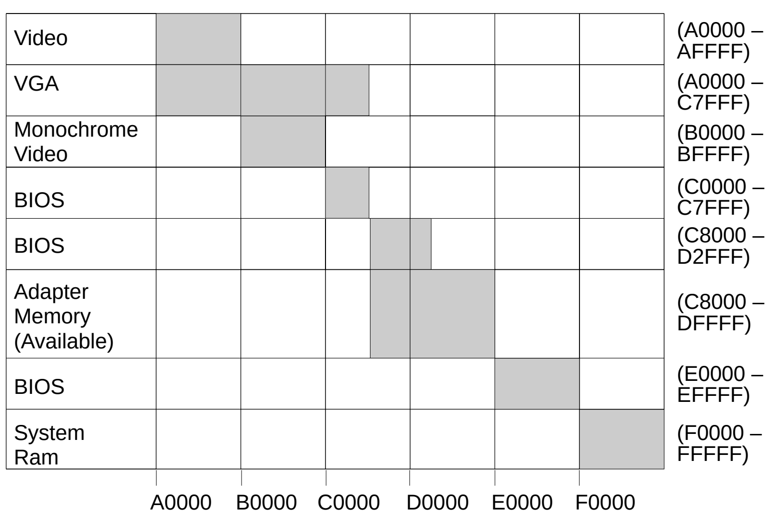
Potential Interrupt Request Conflicts

IRQ	Device
3	COM2
4	COM1
5	LPT2, Network Interface Card
7	LPT1
9#	VGA, video, PCI*
10	Network Interface Card, PCI video*
11	Network Interface Card, PCI*
15	CD ROM drive, SCSI Interface, PCI*

* PCI bus will automatically take up IRQ 9 through 15, unless the BIOS is reconfigured to make the PC work as an ISA bus.

\# IRQ9 is the same interrupt as IRQ2.

Potential Shared RAM Address Conflicts



Operation 4

Placing A Call

The sequence for placing a call over the V/IP gateway is as follows:

1.	Lift the receiver and wait for the dial tone.	
2.	Dial the PBX access code or extension.	*This step is optional.* If the Voice Interface Cards are connected to a PBX, you might need to dial an access code to connect to one of the local V/IP gateway channels. Then, wait for a second dial tone.
3.	Dial the gateway code.	This is the gateway code assigned to the remote V/IP gateway of the telephone you want to call.
4.	Dial the channel number.	*This step is optional.* You might need to dial the channel number if you are calling a telephone connected directly to a V/IP gateway channel. If all channels in a V/IP gateway are members of one hunt group, and the hunt group number is 0, you do not need to dial the channel number.
5.	Dial any digits to be forwarded to the PBX.	*This step is optional.* This could be an extension number. The number of these digits is controlled by a parameter called Maximum Number Of Dial Digits.

Here is an example string of dialed digits:

8 805 1 3423

Digits forwarded to the PBX (extension number)

Channel number

Gateway code

PBX access code for V/IP gateway

A call attempt is made immediately when the V/IP gateway has received the following dialed digits:

gateway code

+ channel number

+ digits forwarded to the PBX

To end the call, simply hang up the telephone.

A full dialing string (gateway code + channel number + digits forwarded to the PBX) can be a lot of digits to dial to get an immediate call attempt. The following parameters allow you to use less then a full dialing string and still place a call.

- Default Number of Digits and Default Inter-Digit Time. These two parameters in the V/IP System Configuration menu work together. A call attempt will be made after a brief pause if the Default Number of Digits has been dialed and the Default Inter-Digit Time has expired. Here is an example:
  - Default Number of Digits = 4 digits
  - Default Inter-Digit Time = 2 seconds
  - User dials **8** to access the V/IP gateway.
  - User dials **805 1**, which meets the criteria for the Default Number of Digits (4 digits), and pauses for more than 2 seconds (the Default Inter-Digit Time).
  - A call attempt will be made using the dialed digits. The gateway code of the called V/IP gateway is 805, and the channel number is 1. This dialed string would connect the user to whatever was connected to the channel at that time.
- Inter-Digit Time. This parameter in the V/IP System Configuration menu sets the amount of time allowed between dialed digits. A call attempt is made if the Inter-Digit Time has expired after the last digit is dialed, regardless of how many digits are dialed.

Fast Dialing

When dialing less than a full string of digits, you normally will have to wait for a timer to expire to have the V/IP gateway attempt to make the call. To avoid having to wait for either the Default Inter-Digit Time or the Inter-Digit Time, you can dial the **#** symbol at the end of the dialing string. This will cause the V/IP gateway to immediately make the call. Here are four examples to illustrate how this works:

V/IP System Configuration for the examples:

- Default Number Of Digits = 5 digits
- Default Inter-Digit Time = 2 seconds
- Inter-Digit Time = 11 seconds

- First example dialed string: **8 805 01** (PBX access code + V/IP gateway code + channel number)

  Since the Default Number Of Digits was dialed, you will have to wait 2 seconds (the Default Inter-Digit Time) before the V/IP gateway will make the call.

- Second example dialed string: **8 805 01#**

  The V/IP gateway will make the call immediately.

- Third example dialed string: **8 805**

  Since less than the Default Number Of Digits was dialed, you will have to wait 11 seconds (the Inter-Digit Time) before the V/IP gateway will make the call.

- Fourth example dialed string: **8 805#**

  The V/IP gateway will make the call immediately.

Some pointers about the **#** symbol:

- Since **#** is used as the fast dialing character, if you need to include **#** in the dialing string, you must dial ***** before **#**. Example:
  - dialed string: **8 805 *#1**
  - digit string sent: **8 805 #1**
- The **#** and ***** characters cannot be used as part of the gateway code.

Fax Calls

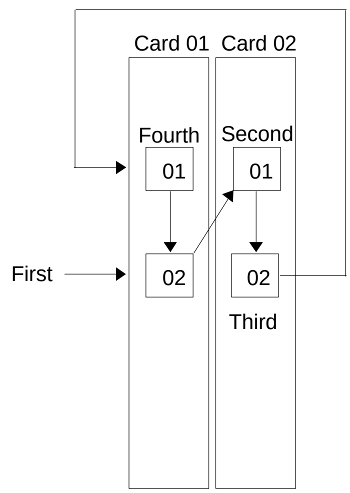
The V/IP gateway can handle fax calls as long as the voice/fax channel configuration Mode parameter is set to *voice/fax* (the default). See page 3-34 for details about setting the Mode parameter.

Dialing of fax calls is the same as for voice calls. The V/IP gateway will automatically detect a fax call and change to the fax mode before making the call.

V.17 fax has the capability of sending faxes at 14,400 bps. However, the maximum rate the V/IP gateway can send faxes is at 9600 bps. When a V.17 fax call is detected, the V/IP gateway will spoof the V.17 mode and send the fax at 9600 bps. The rest of the fax call is handled normally.

Calls to a Hunt Group

When all channels of a V/IP gateway are members of one hunt group (all channel numbers are set to 0), the incoming calls to this gateway are routed in a round-robin fashion. Here is an example:



The ports in the hunt group are handled in this order:

Card 01, Port 02 ↓	The port to connect the first call to after startup of the V/IP gateway.
Card 02, Port 01 ↓	The port to connect the next call to.
Card 02, Port 02 ↓	
. . . ↓	and so on, until the last card is reached, then,
Card 01, Port 01 ↓	
Card 01, Port 02 ↓	
etc.	

An easy way to make all channels of a V/IP gateway as members of one hunt group is to set the *Number of Digits Assigned for Channel* parameter to 0.

Administration 5

Accessing the V/IP Maintenance Program

For the DOS version, the V/IP software operates exclusively – no other programs can be running. Thus, while the V/IP gateway function is in operation on a DOS PC, the V/IP maintenance program is active and is always displayed on the PC's screen.

For the NetWare version, the V/IP call processing software operates in the background. Interaction between the V/IP gateway and an operator at the NetWare server console on a continuous basis is not required. However, you'll need to access the V/IP maintenance program to perform any of the procedures in this section. To access the V/IP maintenance program on the NetWare server, you must invoke the micmcfg NetWare Loadable Module:

TECH_PUBS: **load micmcfg**

The V/IP maintenance program will be displayed:

```
V/IP SERVER MAINTENANCE VERSION 1.1            NetWare Loadable Module

            V/IP SERVER MAIN MENU

  View Configuration
  View Status/Statistics
  Modify Configuration
  Control
  Exit

<CR> = Select, <ESC> = Exit
```

Note: The screens shown throughout the remainder of this manual reflect the DOS version of the V/IP software. The NetWare version does not have entries in the menus for *DVC channels* and *T1/E1 cards*, as these products are not supported in the NetWare version of V/IP.

Viewing Channel Status

You can display the status of each channel, as follows:

1

```
  V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
 Control
 Exit
```

2

```
 MODULE SELECTION
 Analog Channel Status
 Digital Voice (DVC)
 T1/E1
 RSVP
 Statistics
 Phone Directory
 SNMP
 Done
```

3

```
       CARD – PORT
 CARD # 01 – PORT # 01
 CARD # 01 – PORT # 02
 CARD # 02 – PORT # 01
 CARD # 02 – PORT # 02
```

Select the channel you want to view.

```
                       CHANNEL STATUS
 CHANNEL .......................      CARD 01 – PORT 01
 RATE .........................       8000 BPS(G.729)
 STATE .........................      IDLE
 DSP SOFTWARE REVISION .....          2249EnM
 TEST MODE ...................        NONE
 TEST STATUS .................        NORMAL/PASSED
 INPUT LEVEL .................        –25
 MODE .........................       VOICE CONNECTION
```

The following table describes each entry in the channel status display. Press the Esc key to exit from the display.

Parameter	Description
Channel	The card number and port number whose status is displayed.
Rate	The analog to digital conversion rate in bits per second that is currently in effect for the channel. If the channel is in the fax mode, this is the rate at which the channel transmits fax messages. (This is *not* the fax receive rate.)
State	The current state of the channel. This will be one of the following: • `IDLE`: the channel is on-hook and is available for connections. • `DIALING`: the number is being dialed at the remote channel. • `CONNECTION IN PROGRESS`: the channel is off-hook and an attempt is being made to connect to a remote channel. • `CONNECTED – INCOMING`: the channel is off-hook and is connected to a remote channel. The call originated from the remote channel. • `CONNECTED – OUTGOING`: the channel is off-hook and is connected to a remote channel. The call originated from the local channel. • `OFF HOOK`: the channel is off-hook (dialing digits are entered while in this state). • `OUT OF SERVICE`: the channel is out of service due to a failure in hardware, software, or the network.
DSP Software Revision	The revision level of the card's boot up and operating software.
Test Mode	The type of test, if any, that is currently running.
Test Status	When in test mode, the status of the current test.
Input Level	The strength of the input signal, measured in dBm.

Parameter	Description
Mode	The mode within which the channel is currently operating. This will be one of the following: • `IDLE` • `CALL IN PROGRESS` • `VOICE CONNECTION` • `FAX CONNECTION`
Channel Recovery	The number of times the channel was reset by the host software.
Interface Number	The part number and revision letter of the card.
Interface Description	The type of telephone interface selected by the jumpers on the Voice Interface Card for this channel. • `ENHANCED FXS` • `ENHANCED E&M` • `ENHANCED FXO` • `STANDARD FXS` • `STANDARD E&M` • `STANDARD FXO` • `GROUND START FXS` • `GROUND START FXO` • `3–PORT` • `HIGH DENSITY VOICE`
Flash Status	The status of the channel's Flash memory. This will be one of the following: • `VALID – CURRENTLY USED:` loaded and operational • `ERASED`: ready to receive code download • `INVALID:` checksum error, no analog interface driver, security violation, or other fault

Statistics

Viewing Channel Statistics

You can display the operating statistics of each channel, as follows:

```
     V/IP SERVER MAIN MENU
   View Configuration
1  View Status/Statistics
   Modify Configuration
   Control
   Exit
```

```
   MODULE SELECTION
   Analog Channel Status
   Digital Voice (DVC)
   T1/E1
   RSVP
2  Statistics
   Phone Directory
   SNMP
   Done
```

```
       V/IP STATUS/STATISTICS
   System Statistics
3  Channel Statistics
   Done
```

```
          CARD - PORT
4  CARD # 01 - PORT # 01
   CARD # 01 - PORT # 02
   CARD # 02 - PORT # 01
   CARD # 02 - PORT # 02
```

Select the channel you want to view.

```
          CHANNEL STATISTICS : CARD 01 - PORT 01
   TOTAL TIME CONNECTED ...........        0:00
   OUTGOING CALLS ATTEMPTED ......            0
```

The following table describes each entry in the channel statistics display. Press the Esc key to exit from the display.

Parameter	Description
Total Time Connected	The amount of time in minutes and seconds that this channel was connected to a remote channel. This statistic covers incoming and outgoing, phone and fax calls.
Outgoing Calls Attempted	The number of times this channel went off hook and a dialing string entered, whether the call attempt was successful or not.
Outgoing Calls Completed	The number of connection attempts to remote channels that were successful.
Incoming Calls Completed	The number of successful connections to this channel that were made by remote channels.
Total Duration of Voice Calls	The amount of time in minutes and seconds that this channel was connected to a remote channel for incoming and outgoing voice calls.
Average Voice Call Duration	The *Total Duration of Voice Calls* divided by the number of voice calls.
Voice Call %	The *Total Duration of Voice Calls* divided by the *Total Time Connected*, expressed as a percentage.
Total Duration of Fax Calls	The amount of time in minutes and seconds that this channel was connected to a remote channel for incoming and outgoing fax calls.
Average Fax Call Duration	The *Total Duration of Fax Calls* divided by the number of fax calls.
Fax Call %	The *Total Duration of Fax Calls* divided by the *Total Time Connected*, expressed as a percentage.
Voice Packets Sent	The number of packets sent across the LAN by this channel for voice calls. Calls that are internally switched within the local V/IP gateway (from one channel to another within the same PC) do not increment this count. This indicates how busy the V/IP gateway is keeping the LAN for outgoing voice calls.
Voice Packets Received	The number of packets received via the LAN by this channel for voice calls. Calls that are internally switched within the local V/IP gateway (from one channel to another within the same PC) do not increment this count. This indicates how busy the V/IP gateway is keeping the LAN for incoming voice calls.
Voice Bytes Sent	The number of bytes sent across the LAN by this channel for voice calls. The byte count includes the Medium Access Control (MAC), IP, and UDP headers of the voice packet. Calls that are internally switched within the local V/IP gateway (from one channel to another within the same PC) do not increment this count. This indicates how much bandwidth on the LAN is being used for outgoing voice calls.

Parameter	Description
Voice Bytes Received	The number of bytes received via the LAN by this channel for voice calls. The byte count includes the Medium Access Control (MAC), IP, and UDP headers of the voice packet. Calls that are internally switched within the local V/IP gateway (from one channel to another within the same PC) do not increment this count. This indicates how much bandwidth on the LAN is being used for incoming voice calls.
Fax Packets Sent	The number of packets sent across the LAN by this channel for fax calls. Calls that are internally switched within the local V/IP gateway (from one channel to another within the same PC) do not increment this count. This indicates how busy the V/IP gateway is keeping the LAN for outgoing fax calls.
Fax Packets Received	The number of packets received via the LAN by this channel for fax calls. Calls that are internally switched within the local V/IP gateway (from one channel to another within the same PC) do not increment this count. This indicates how busy the V/IP gateway is keeping the LAN for incoming fax calls.
Fax Bytes Sent	The number of bytes sent across the LAN by this channel for fax calls. Calls that are internally switched within the local V/IP gateway (from one channel to another within the same PC) do not increment this count. This indicates how much bandwidth on the LAN is being used for outgoing fax calls.
Fax Bytes Received	The number of bytes received via the LAN by this channel for fax calls. Calls that are internally switched within the local V/IP gateway (from one channel to another within the same PC) do not increment this count. This indicates how much bandwidth on the LAN is being used for incoming fax calls.
Outgoing Packets Discarded	The number of V/IP packets sent across the LAN for this channel that were discarded.
Incoming Packets Out of Sequence	The number of V/IP packets for incoming calls that were not received in the correct sequence.

Resetting Channel Statistics Counters

The channel statistics counters keep incrementing continuously as events occur. In order to determine the amount of activity in a given time interval, you can reset the counters, to have a known starting point.

1

```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2

```
CONTROL
Analog Channel Control
Digital Voice (DVC)
T1/E1
Analog Card Control
Analog Channel Diagnostics
Reset Analog Channel Counters
Reset DVC Counter
Reset System Counters
Ping
Update From Directory Server
Done
```

3

```
CARD – PORT
CARD # 01 – PORT # 01
CARD # 01 – PORT # 02
CARD # 02 – PORT # 01
CARD # 02 – PORT # 02
```

Select the channel whose counters you want to reset.

The V/IP software will confirm that the channel has been reset by presenting a dialog box. The statement RESET COUNTER OK <PRESS ENTER TO CONTINUE> is shown in the box. Press the Enter key to acknowledge the channel reset.

Viewing and Resetting System Statistics

You can display the operating statistics of the V/IP gateway, as follows:

1
```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2
```
MODULE SELECTION
Analog Channel Status
Digital Voice (DVC)
T1/E1
RSVP
Statistics
Phone Directory
SNMP
Done
```

3
```
V/IP STATISTICS
System Statistics
Channel Statistics
Done
```

```
SYSTEM STATISTICS
TOTAL INCOMING CALLS ATTEMPTED .......      123,456
CALL SET-UP PACKETS SENT ..............     1,234,567
CALL SET-UP PACKETS RECEIVED ..........     1,234,567
```

The following table describes each entry in the system statistics display. Press the Esc key to exit from the display.

Parameter	Description
Total Incoming Calls Attempted	The number of call attempts made to all channels of all Voice Interface Cards in this PC.
Call Set-Up Packets Sent	The number of packets transmitted across the network to set up calls to remote V/IP gateways.
Call Set-Up Packets Received	The number of packets received via the network to set up incoming calls to this gateway.
Call Set-Up Bytes Sent	The number of bytes transmitted across the network to set up calls to remote V/IP gateways.
Call Set-Up Bytes Received	The number of bytes received via the network to set up incoming calls to this gateway.
Phone Database Packets Sent	The number of packets that were transmitted across the network for updating the phone directory database.
Phone Database Packets Received	The number of packets received via the network for updating the phone directory database.

To reset the counters, make the following selections:

```
  V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
1 Control
 Exit
```

```
          CONTROL
 Analog Channel Control
 Digital Voice (DVC)
 T1/E1
 Analog Card Control
 Analog Channel Diagnostics
 Reset Analog Channel Counters
 Reset DVC Counter
2 Reset System Counters
 Ping
 Update From Directory Server
 Done
```

After the counters have been reset, the V/IP software displays a dialog box. This box contains the statement RESET SYSTEM COUNTER OK <PRESS ENTER TO CONTINUE>. Press the Enter key to acknowledge the counter reset.

Viewing Configurations

Channel Configuration

You can view the configuration of a voice/fax channel as follows:

1
```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2
```
MODULE SELECTION
Analog Voice/Fax
Digital Voice (DVC)
T1/E1
RSVP
SNMP
V/IP System
Hardware
Done
```

3
```
CARD – PORT
CARD # 01 – PORT # 01
CARD # 01 – PORT # 02
CARD # 02 – PORT # 01
CARD # 02 – PORT # 02
```

Select the channel you want to view.

```
VOICE/FAX Configuration – FXS
CHANNEL NUMBER ..........................    1
UDP PORT NUMBER .........................    65534
MODE ....................................    VOICE/FAX
```

The display shows the channel's parameters in the same format as the menu that is displayed when configuring the channel (for example, as shown on page 3-30). To exit from the display, press the Esc key.

Voice Interface Card Hardware Configuration

You can view the physical configuration of any Voice Interface Card installed in the PC as follows:

1

```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2

```
MODULE SELECTION
Analog Voice/Fax
Digital Voice (DVC)
T1/E1
RSVP
SNMP
V/IP System
Hardware
Done
```

3

```
CARD
CARD # 01
CARD # 02
```

Select the card you want to view.

```
HARDWARE STATUS
CARD NUMBER ..............................  1
NO OF CHANNELS ..........................  2
IRQ .......................................  11
I/O ADDRESS ..............................  200
SHARED MEMORY ADDRESS ..................  d000
INTERFACE TYPE ..........................  ANALOG INTERFACE
```

The display shows the card's parameters as configured from jumpers and from the Installation menu (shown on page 3-11). To exit from the display, press the Esc key.

System Configuration

You can view the configuration of any of the system parameters as follows:

1

```
   V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
 Control
 Exit
```

2

```
 MODULE SELECTION
 Analog Voice/Fax
 Digital Voice (DVC)
 T1/E1
 RSVP
 SNMP
 V/IP System
 Hardware
 Done
```

Select **RSVP** to view the Resource ReSerVation Protocol settings.

Select **SNMP** to view the SNMP network management settings (DOS version only).

Select **V/IP System** to view the overall V/IP gateway configuration.

The displayed screens show the configuration settings in the same format as the menu displays that are presented when configuring these parameters. To exit from any display, press the Esc key.

Viewing the Phone Directory Database

You can view the phone directory database as it exists in the local V/IP gateway, as follows:

```
    V/IP SERVER MAIN MENU
    View Configuration
1   View Status/Statistics
    Modify Configuration
    Control
    Exit
```

```
    MODULE SELECTION
    Analog Channel Status
    Digital Voice (DVC)
    T1/E1
    RSVP
    Statistics
2   Phone Directory
    SNMP
    Done
```

```
                        PHONE DATABASE
GATEWAY CODE    IP ADDRESS       MAX DIGIT    CHAN DIGIT    [1]
-------------------------------------------------------------
  805          199.30.25.223        00            00
  707          199.30.19.11         00            00
  415          199.30.18.15         00            04
  213          199.30.17.10         00            04
```

The entry fields are as follows:

GATEWAY CODE	The gateway code of a V/IP gateway that must be dialed to connect to one of its channels.
IP ADDRESS	The local V/IP gateway will map the dialed gateway code to this IP address to make a connection over the IP network.
MAX DIGIT	The maximum number of digits that the V/IP gateway, specified by the dialed gateway code, can dial on its outgoing channel/phone lines.
CHAN DIGIT	The number of digits that must be dialed to connect directly to a specific channel of the V/IP gateway specified by the dialed gateway code.

Updating the Phone Directory Database

The phone directory database of each V/IP gateway is automatically updated every 24 hours (by default, or whatever interval you have set for the V/IP System Synchronizing Interval parameter). However, you can update the database for an individual V/IP gateway that is not the phone directory database server. The procedure is as follows:

```
        V/IP SERVER MAIN MENU
     View Configuration
     View Status/Statistics
     Modify Configuration
1    Control
     Exit
```

```
             CONTROL
     Analog Channel Control
     Digital Voice (DVC)
     T1/E1
     Analog Card Control
     Analog Channel Diagnostics
     Reset Analog Channel Counters
     Reset DVC Counter
     Reset System Counters
     Ping
2    Update From Directory Server
     Done
```

```
          Are you sure ?
3    Yes
     No
```

The V/IP gateway will request a download of the database from the phone directory database server.

Screen Display Messages

The following table lists messages that may be displayed by the V/IP gateway.

Note: If you encounter a message where the action to be taken is to contact MICOM Customer Service, please record the sequence of events that led up to the displaying of the message prior to calling MICOM Customer Service. Also, note any Code ID or Error Numbers that may be displayed.

Message	Description and/or Action to Remedy
`Unable to initialize DSP port No.` *nn* `in Card No.` *nn*	• Mismatch between the I/O port base address set on the card's jumpers and the address that was entered at the I/O port base address prompt during software installation. • A conflict with another device using the same I/O port base address.
`Incorrect IP address`	Network portion of IP address must match with the network portion of the default router. • In NetWare, the IP address entered at the V/IP software Installation menu must match the IP address assigned to the Network Interface Card that is running an IP stack. To correct the problem, delete the `SYS:\micmvoic.cfg` file and start the software installation procedure from the beginning.
`MICMVOIC.CFG: Unknown error code` *nn* `while saving MICMVOIC.CFG`	• Make sure that the server's hard disk drive has enough free space. • If the problem persists, contact MICOM Customer Service.
`MICMVOIC.NLM: Unable to rename MICMVOIC.CFG to MICMVOIC.B1K`	The V/IP gateway was unable to make a backup copy of the V/IP's configuration file. • Make sure that an existing `micmvoic.b1k` file or the root directory is not write protected.
`Unable to do non-blocking socket call. Code id:` *nn* `errno =` *nn*	• In NetWare, make sure that the server has the latest version of TCPIP.NLM. • In DOS, or in NetWare if the problem persists, contact MICOM Customer Service.

Message	Description and/or Action to Remedy
`Unable to bind to socket. Code id:` *nn* `errno =` *nn*	• In NetWare, make sure that the server has the latest version of TCPIP.NLM. • In DOS, or in NetWare if the problem persists, contact MICOM Customer Service.
`Unable to access MICMPHON.DAT file`	The V/IP gateway cannot access the phone directory database file. • Check the operation of the PC's hard disk drive.
`Unknown call set-up initialization error code` *nn*	Contact MICOM Customer Service.
`Unknown database initialization error code` *nn*	Contact MICOM Customer Service.
`Unknown socket call error(Code id:` *nn* `errno =` *nn*	• In NetWare, make sure that the server has the latest version of TCPIP.NLM. • In DOS, or in NetWare if the problem persists, contact MICOM Customer Service.
`!!! Not enough memory to run MICMVOIC.NLM Code id:` *nn* `!!!`	• In NetWare, try unloading any unnecessary NLMs. • In DOS, make sure there are no terminate and stay resident programs loaded into memory. • Add more memory to the PC.
`Card No.` *nn* `has nn channels, it does not match with the number found in MICMVOIC.CFG which indicates it should have` *nn* `channels.`	• The V/IP gateway configuration file MICMVOIC.CFG is corrupt. Perform the *V/IP Gateway Does Not Operate Correctly After Configuration Change* procedure on page 5-39. • You have added new hardware. Put back the old hardware and perform the *Adding A New Card* procedure on page 5-28.
`Card No.` *nn* `indicates it has` *xxx* `interface. However MICMVOIC.CFG indicates it should have` *xxx* `interface.`	• The V/IP gateway configuration file MICMVOIC.CFG is corrupt. Perform the *V/IP Gateway Does Not Operate Correctly After Configuration Change* procedure on page 5-39. • You have added new hardware. Put back the old hardware and perform the *Adding A New Card* procedure on page 5-28.

Message	Description and/or Action to Remedy
`Unable to initialize card no.` *nn*`. The card parameters were: I/O address =` *nnn*`H Interrupt =` *nn* `Shared ram =` *nnnn*`0H.`	Make sure that the I/O port address, the interrupt, and the shared RAM used by the V/IP card is not in conflict with other adapters or ports in the PC. • In NetWare, type `config` in the system console screen to find out what interrupt, I/O ports and shared RAM are being used. • If the problem persists, the Voice Interface Card may be malfunctioning. Contact MI-COM Customer Service.
`The code discovered there are` *nn* `channels in the server. However MICMVOIC.CFG file indicate there are` *nn* `channels in the server`	• The V/IP gateway configuration file MICM-VOIC.CFG is corrupt. Perform the *V/IP Gateway Does Not Operate Correctly After Configuration Change* procedure on page 5-39. • You have added new hardware. Put back the old hardware and perform the *Adding A New Card* procedure on page 5-28.
`Checksum error in MICMVOIC.CFG parameters file`	The V/IP gateway configuration file MICMVOIC.CFG is corrupt. Perform the *V/IP Gateway Does Not Operate Correctly After Configuration Change* procedure on page 5-39.
`Incorrect length in MICMVOIC.CFG parameters file`	The V/IP gateway configuration file MICMVOIC.CFG is corrupt. Perform the *V/IP Gateway Does Not Operate Correctly After Configuration Change* procedure on page 5-39.
`Unable to initialize RSVP or SNMP code.`	• In NetWare, make sure that the server has the latest version of TCPIP.NLM and SNMP.NLM. • In DOS, or in NetWare if the problem per-sists, contact MICOM Customer Service.
`Incorrect number of channels found.`	Contact MICOM Customer Service.
`Revision mismatch between MICMVOIC.NLM and MICMDRV.NLM. MICMDRV.NLM revision is` *nn* `and MICMVOIC.NLM is` *nn*	Make sure you have the latest versions of the MICMDRV.NLM and MICMVOIC.NLM files copied to the `SYS:\system` directory of NetWare or the `C:\` directory of DOS.
`Unknown driver error code:` *nn*	Contact MICOM Customer Service.

Message	Description and/or Action to Remedy
`Unable to bind Pre-Scan MICMVOIC to the MLID.`	Too many protocols have been bound to the LAN driver in the NetWare server. You must unbind any unnecessary protocols from the LAN driver used by V/IP and bind `micmvoic` to the LAN driver once again.
`Unable to obtain the MLID's control routine.`	A non-standard LAN driver is used in NetWare.
`Unable to get the MLID's configuration structure.`	A non-standard LAN driver is used in NetWare.
`MLID is unable to handle the maximum voice packet size.`	The LAN driver could not send the maximum size of the voice packet that V/IP must send. • Update the LAN driver or use a different Network Interface Card.
`MLID does not have a raw send capability.`	The LAN driver could not provide the raw-send function needed by V/IP. • Update the LAN driver or use a different Network Interface Card.
`Hashing error at the Pre-Scan stack.`	Contact MICOM Customer Service.
`Fatal parameters error has occurred.`	The V/IP gateway configuration file MICMVOIC.CFG is corrupt. Perform the *V/IP Gateway Does Not Operate Correctly After Configuration Change* procedure on page 5-39.
`The reserved 0x01EA UDP port number is being used by other code.`	Another application in NetWare is using the UDP port number assigned to MICOM by IETF. • Unload the offending application.
`Unknown DSP packet`	The Voice Interface Card has generated a packet that is unexpected by the V/IP host software. • Contact MICOM Customer Service.
`Unable to communicate with default router.`	DOS version only. The V/IP gateway is unable to communicate with the default router. • Make sure the IP address for the default router is entered correctly and that the default router is operational.

Message	Description and/or Action to Remedy
`Registering Shutdown Routine Failed`	DOS version only. The V/IP gateway is unable to properly shutdown during the unloading process. • Turn off the PC prior to re-starting MICMVOIC.EXE.
`Scheduler returned code = xx.`	Contact MICOM Customer Service.
`UDP port number conflict with other application in NetWare.`	• Each channel must have a unique UDP port number. The default is 65534 for the first channel, 65533 for the second channel, and so on. • Another application is using the same UDP port assignment as a V/IP gateway channel.
`Directory server has a damaged database file.`	You can specify another V/IP gateway to supply a phone directory database, or you can continue the startup using a single entry database generated locally.
`Unable to load directory database from other V/IP.`	The phone directory database could not be loaded from the specified phone directory database server. (This would have been as a result of entering another V/IP gateway after receiving the message: Directory server has a damaged database file.) You can specify another V/IP gateway to supply a phone directory database, or you can continue the startup using a single entry database generated locally.
`The assigned server code for this PFS is in conflict with the directory database data.`	You can change the gateway code for this V/IP gateway, or de-register the incorrect entry from the phone directory database server.
`IP address is in conflict with the directory database data.`	You can de-register the incorrect entry from the phone directory database server.
`INTERRUPT CONFLICT WITH THE NIC CARD`	A V/IP Voice Interface Card has been set to the same IRQ as the Network Interface Card.
`I/O ADDRESS CONFLICT WITH THE NIC CARD`	A V/IP Voice Interface Card has been set to the same I/O port address range as the Network Interface Card.

Message	Description and/or Action to Remedy
`SHARE MEMORY CONVFLICT WITH THE NIC CARD`	A V/IP Voice Interface Card has been set to the same shared memory address range as the Network Interface Card.
`THE CONFIGURED I/O ADDRESS DOES NOT MATCH THE I/O HARDWARE STRAP ON THE CARD`	The setting of the I/O port base address jumpers on the Voice Interface Card does not match the I/O port base address that was entered in the Hardware Installation menu.

Channel Profiles

The V/IP gateway allows you to store and load profiles containing the configuration settings of Voice Interface Card channels. Each channel profile is specific to one channel on one Voice Interface Card. You can store multiple profiles for each channel.

You should store profiles for all the channels in each V/IP gateway. These profiles can then be used to quickly configure the channels after a software upgrade or other major change to the overall V/IP gateway configuration.

Creating A Channel Profile

1. Make sure the channel is configured as you require. (See page 3-30 if you want information on how to configure a channel.)

2. Select Save Profile from the Channel Control menu:

```
     V/IP SERVER MAIN MENU
   View Configuration
   View Status/Statistics
   Modify Configuration
1  Control
   Exit
```

```
             CONTROL
2  Analog Channel Control
   Digital Voice (DVC)
   T1/E1
   Analog Card Control
   Analog Channel Diagnostics
   Reset Analog Channel Counters
   Reset DVC Counter
   Reset System Counters
   Ping
   Update From Directory Server
   Done
```

```
      CHANNEL CONTROL
   Enable Channel
   Disable Channel
   Reset Channel
   Load Profile
3  Save Profile
   Delete Profile
   Download
   Done
```

```
          CARD – PORT
4  CARD # 01 – PORT # 01
   CARD # 01 – PORT # 02
   CARD # 02 – PORT # 01
   CARD # 02 – PORT # 02
```

Select the channel whose profile you want to save.

3. Enter the profile name in the pop up window:

```
Save CARD:01, PORT:01 Configuration
-----------------------------------
File Name :
```

Enter a name for this profile.

You can save the profile by any name that meets the DOS file naming conventions, without the extension (V/IP will insert the correct extension for you).

The V/IP gateway will confirm that the profile was saved.

Loading A Channel Profile

1. Select Load Profile from the Channel Control menu:

```
    V/IP SERVER MAIN MENU
    View Configuration
    View Status/Statistics
    Modify Configuration
1   Control
    Exit
```

```
                CONTROL
2   Analog Channel Control
    Digital Voice (DVC)
    T1/E1
    Analog Card Control
    Analog Channel Diagnostics
    Reset Analog Channel Counters
    Reset DVC Counter
    Reset System Counters
    Ping
    Update From Directory Server
    Done
```

```
      CHANNEL CONTROL
    Enable Channel
    Disable Channel
    Reset Channel
3   Load Profile
    Save Profile
    Delete Profile
    Download
    Done
```

```
          CARD – PORT
4   CARD # 01 – PORT # 01
    CARD # 01 – PORT # 02
    CARD # 02 – PORT # 01
    CARD # 02 – PORT # 02
```

Select the channel whose profile you want to load.

2. Select the profile name from the pop up window:

```
      PROFILE LIST
    FIRST01.FXS
    SECND01.FXS
```

Select the name of the desired profile.

Deleting A Channel Profile

1. Select Delete Profile from the Channel Control menu:

```
   V/IP SERVER MAIN MENU
  View Configuration
  View Status/Statistics
  Modify Configuration
1 Control
  Exit
```

```
          CONTROL
2 Analog Channel Control
  Digital Voice (DVC)
  T1/E1
  Analog Card Control
  Analog Channel Diagnostics
  Reset Analog Channel Counters
  Reset DVC Counter
  Reset System Counters
  Ping
  Update From Directory Server
  Done
```

```
    CHANNEL CONTROL
  Enable Channel
  Disable Channel
  Reset Channel
  Load Profile
  Save Profile
3 Delete Profile
  Download
  Done
```

2. Select the profile you want to delete from the Profile List:

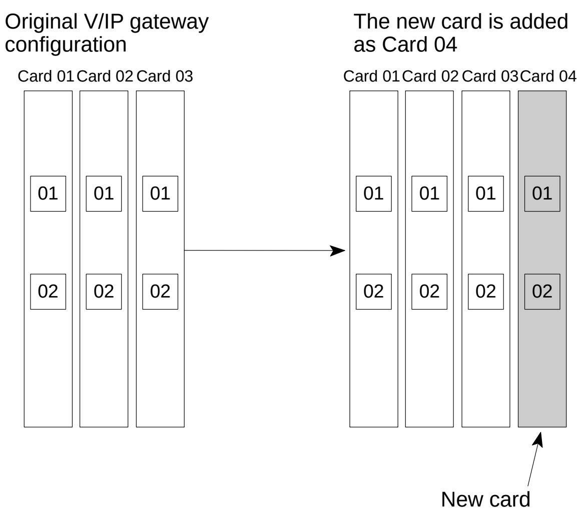
```
     PROFILE LIST
  FIRST01.FXS
  SECND01.FXS
  BUSYOUT.FXS
  BUSYOUT.ENM
```

3. Select Yes from the *Are you sure* prompt:

```
   Are you sure ?
  Yes
  No
```

Adding A New Voice Interface Card

You can add a new Voice Interface Card as the next higher card number. For example, if there are three Voice Interface Cards in the PC, the new card can be installed as Card 04:



To add a new card, proceed as follows:

1. Install the new Voice Interface Card into the PC. The basic steps are:
   a. Shut down the PC/NetWare server.
   b. Configure the jumpers on the Voice Interface Card (see pages 2-9 through 2-20).
   c. Install the card into the PC.
   d. Start the PC/NetWare server.
   e. Start the V/IP gateway.

   For NetWare:

   Enter the following commands at the server prompt to start the V/IP gateway:

```
TECH_PUBS: load micmvoic
TECH_PUBS: bind micmvoic to vip
TECH_PUBS: load micmcfg
```

For DOS:

Start the V/IP gateway:

C:\MICMVOIC> **micmvoic** *nic*.**lan**

2. Enter the new card into the V/IP gateway configuration:

1

```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2

```
MODULE SELECTION
Analog Voice/Fax
Digital Voice (DVC)
T1/E1
RSVP
SNMP
V/IP System
Hardware
Done
```

3

```
CARD
CARD # 01
CARD # 02
CARD # 03
```

Press the Ins (Insert) key to insert a new card number.

4

```
CARD
CARD # 01
CARD # 02
CARD # 03
CARD # 04
```

Select the new card.

5

```
Is it a T1/E1 expansion card?
Yes
No
```

Select No to this prompt.

3. Configure the new card's hardware parameters from the Hardware Configuration menu:

```
                 HARDWARE CONFIGURATION
 IRQ                                         [11]
 I/O Address                                 [200H]
 Shared Memory Address                       [D0000H]
 Done
```

4. Select **Done** from the Hardware Configuration menu. At this point, the V/IP console will automatically shut down and you will see the display return to the system console (NetWare) or the DOS prompt (DOS).

5. Restart the V/IP gateway.

   For NetWare:

   – Enter the following commands at the server prompt to shut down the V/IP gateway:

   TECH_PUBS: **unload micmvoic**
   TECH_PUBS: **unload micmdrv**

   – Enter the following commands at the server prompt to restart the V/IP gateway:

   TECH_PUBS: **load micmvoic**
   TECH_PUBS: **bind micmvoic to** *vip*
   TECH_PUBS: **load micmcfg**

   For DOS:

   – Restart the V/IP gateway:

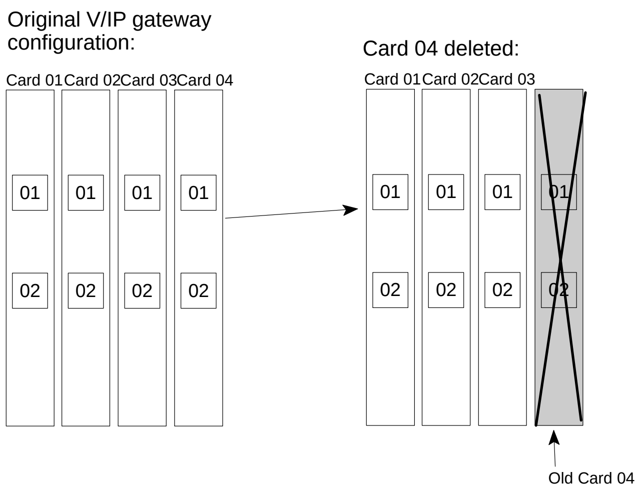
   C:\MICMVOIC> **micmvoic** *nic*.**lan**

6. Configure the voice/fax parameters for the channels of the new card. See page 3-30 for details.

Deleting A Voice Interface Card

You can delete only the highest numbered card in a V/IP gateway.

Here is the effect on the V/IP gateway when a card is deleted:



To delete a card, proceed as follows:

1. Delete the card from the V/IP gateway:

```
1   V/IP SERVER MAIN MENU
    View Configuration
    View Status/Statistics
    Modify Configuration
    Control
    Exit
```

```
2   MODULE SELECTION
    Analog Voice/Fax
    Digital Voice (DVC)
    T1/E1
    RSVP
    SNMP
    V/IP System
    Hardware
    Done
```

```
3   CARD
    CARD # 01
    CARD # 02
    CARD # 03
    CARD # 04
```

Select the highest numbered card and press the `Del` (Delete) key.

```
4   Are you sure ?
    No
    Yes
```

2. Select **Done** from the Module Selection menu. At this point, the V/IP console will automatically shut down and you will see the display return to the system console (NetWare) or the DOS prompt (DOS).
3. Shutdown the PC.
4. Remove the deleted card.
5. Restart the PC/server and V/IP gateway.

Channel Code Download

Each channel of a Voice Interface Card maintains its own copy of the operating code in firmware. The original code your Voice Interface Cards shipped with is located in the Utilities diskette. For analog cards, the file is located in the directory `\voice\analog`. You can command the V/IP gateway to download either the original code or new code (for upgrade purposes) to a selected channel.

1. Place the operating code file into either of the following locations:
   - For NetWare, place the file into the root directory of the server hosting the V/IP gateway.
   - For DOS, place the file into the `C:\micmvoic\vip_file` directory of the V/IP gateway PC.

2. Command the code download from the V/IP console:

```
   V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
1 Control
 Exit
```

```
                CONTROL
2  Analog Channel Control
   Digital Voice (DVC)
   T1/E1
   Analog Card Control
   Analog Channel Diagnostics
   Reset Analog Channel Counters
   Reset DVC Counter
   Reset System Counters
   Ping
   Update From Directory Server
   Done
```

```
     CHANNEL CONTROL
   Enable Channel
   Disable Channel
   Reset Channel
   Load Profile
   Save Profile
   Delete Profile
3  Download
   Done
```

3. Select the file containing the operating code:

```
     Download File
 22600H01.VOC
```

4. Select the channel that is to receive the operating code:

```
      CARD – PORT
 CARD # 01 – PORT # 01
 CARD # 01 – PORT # 02
 CARD # 02 – PORT # 01
 CARD # 02 – PORT # 02
```

The V/IP gateway will inform you on the progress of the download.

> ⚠ ***WARNING***
>
> Do not stop the channel code download while in process. Early termination could result in corrupt voice code on the card.

Description of Files

NetWare Files

`SYS:\system\micmvoic.nlm`

The V/IP host operating code for the NetWare environment.

`SYS:\system\micmdrv.nlm`

The device driver that handles:

- the V/IP digital signal processor packets from each Voice Interface Card channel,
- the Voice Interface Card channel resets,
- the Voice Interface Card resets.

`SYS:\system\micmcfg.nlm`

The V/IP console (the user interface).

`SYS:\micmphon.dat`

This is the phone directory database as known by this V/IP gateway.

`SYS:\micmvoic.cfg`

The parameter settings (values) for all Voice Interface Cards are stored in this file.

`SYS:\micmvoic.b1k`

The backup version of the micmvoic.cfg file. When you make changes to the parameters settings of any Voice Interface Card, a new micmvoic.cfg file is created and the old file is renamed micmvoic.b1k.

`SYS:\22600h01.voc`

A copy of the firmware that can be downloaded to the channel of an analog Voice Interface Card.

DOS Files

The V/IP for DOS files are located within a directory that was specified by the installer during initial installation. The following file list assumes that the directory specified was *C:\micmvoic*.

`C:\micmvoic\micmvoic.exe`

The V/IP host operating code.

`C:\micmvoic\dos4gw.exe`

Extensions to DOS required to run V/IP.

`C:\micmvoic\d4grun.exe`

Extensions to DOS required to run V/IP.

`C:\micmvoic\lan.cfg`

The Network Interface Card parameters that were created by the setup.exe program.

`C:\micmvoic\vip_file\22600h01.voc`

A copy of the firmware that can be downloaded to the channel of a analog Voice Interface Card.

`C:\micmvoic\vip_file\micmvoic.cfg`

The parameter settings (values) for all Voice Interface Cards are stored in this file.

`C:\micmvoic\vip_file\micmphon.dat`

This is the phone directory database as known by this V/IP gateway.

`C:\micmvoic\vip_file\micmvoic.b1k`

The backup version of the micmvoic.cfg file. When you make changes to the parameters settings of any Voice Interface Card, a new micmvoic.cfg file is created and the old file is renamed micmvoic.b1k.

MIB File

The `micmv11c.mib` file in the Utilities diskette contains the Enterprise MIB Extensions for MICOM V/IP products. See Appendix C for a detailed description about the Enterprise MIB Extensions.

Troubleshooting Procedures

Channel Not Working

If a channel does not appear to be working, try the following procedure:

1. Do a channel reset:

```
   V/IP SERVER MAIN MENU
  View Configuration
  View Status/Statistics
  Modify Configuration
1 Control
  Exit
```

```
          CONTROL
2 Analog Channel Control
  Digital Voice (DVC)
  T1/E1
  Analog Card Control
  Analog Channel Diagnostics
  Reset Analog Channel Counters
  Reset DVC Counter
  Reset System Counters
  Ping
  Update From Directory Server
  Done
```

```
    CHANNEL CONTROL
  Enable Channel
  Disable Channel
3 Reset Channel
  Load Profile
  Save Profile
  Delete Profile
  Download
  Done
```

```
       CARD – PORT
4 CARD # 01 – PORT # 01
  CARD # 01 – PORT # 02
  CARD # 02 – PORT # 01
  CARD # 02 – PORT # 02
```

Select the channel you want to reset.

The selected channel will perform a hardware reset, followed by initialization of that channel.

Test the channel by attempting to place a call. If the channel has not recovered to normal operation, proceed to step 2.

2. Reset the card:

1

```
  V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2

```
           CONTROL
Analog Channel Control
Digital Voice (DVC)
T1/E1
Analog Card Control
Analog Channel Diagnostics
Reset Analog Channel Counters
Reset DVC Counter
Reset System Counters
Ping
Upload From Directory Server
Done
```

3

```
     CARD CONTROL
Reset
Done
```

4

```
   CARD
CARD # 01
CARD # 02
```

Select the card you want to reset.

The selected card will perform a hardware reset, followed by card initialization and then initialization of all channels on that card.

Test the channel by attempting to place a call. If the channel has not recovered to normal operation, contact your MICOM Certified Distributor.

V/IP Gateway Does Not Operate Correctly After Configuration Change

If you encounter any operational problems after making a configuration change, the following procedure will recover the gateway to the previous configuration by restoring from the backup configuration file.

For NetWare:

1. Login as supervisor (admin, NetWare 4.x) on the server hosting the V/IP gateway.
2. Delete the `SYS:\micmvoic.cfg` file.
3. Rename the `SYS:\micmvoic.b1k` file to `micmvoic.cfg`.
4. Restart the V/IP gateway:
   a. Select **Exit** from the V/IP gateway console, select **Yes** at the *Exit the program* pop up window, and press any key to return to the system console.
   b. Unload the micmvoic NLM:
   ```
   TECH_PUBS: unload micmvoic
   TECH_PUBS: unload micmdrv
   ```
   c. Load the micmvoic NLM and bind it to the IP stack of the Network Interface Card:
   ```
   TECH_PUBS: load micmvoic
   TECH_PUBS: bind micmvoic to vip
   ```
5. Load the V/IP gateway console and verify the operation of the gateway.
   ```
   TECH_PUBS: load micmcfg
   ```

For DOS:

1. Shut down the V/IP gateway: Select **Exit** from the V/IP gateway console, select **Yes** at the *Exit the program* pop up window, and press any key to return to the DOS prompt.
2. Delete the `C:\micmvoic\vip_file\micmvoic.cfg` file.
3. Rename the `C:\micmvoic\vip_file\micmvoic.b1k` file to `micmvoic.cfg`.
4. Restart the V/IP gateway:
   ```
   C:\MICMVOIC> micmvoic nic.lan
   ```
5. Verify the operation of the gateway.

Changing the IP Address

If you need to change the IP address of the V/IP gateway, either as a network change or if you encounter the message `Incorrect IP Address`, proceed as follows:

For NetWare:

1. Login as supervisor (admin, NetWare 4.x) on the server hosting the V/IP gateway.

2. Delete the `SYS:\micmvoic.cfg` file.

3. Shut down the V/IP gateway:

   a. Select **Exit** from the V/IP gateway console, select **Yes** at the *Exit the program* pop up window, and press any key to return to the system console.

   b. Unload the micmvoic NLM:

   ```
   TECH_PUBS: unload micmvoic
   TECH_PUBS: unload micmdrv
   ```

4. Startup the V/IP gateway and enter the system parameters.

   a. Load the micmvoic NLM:

   ```
   TECH_PUBS: load micmvoic
   ```

   You will see the menu shown on page 3-11. These are the system parameters. Enter the correct information as detailed on page 3-11 through 3-14.

   b. Bind the micmvoic NLM to the IP stack of the Network Interface Card:

   ```
   TECH_PUBS: bind micmvoic to vip
   ```

5. Load the V/IP gateway console and verify the operation of the gateway.

   ```
   TECH_PUBS: load micmcfg
   ```

For DOS:

1. Shut down the V/IP gateway: Select **Exit** from the V/IP gateway console, select **Yes** at the *Exit the program* pop up window, and press any key to return to the DOS prompt.

2. Delete the `C:\micmvoic\vip_file\micmvoic.cfg` file.

3. Restart the V/IP gateway:

   ```
   C:\MICMVOIC> micmvoic nic.lan
   ```

   You will see the NIC Configuration Utility screen, as shown on page 3-19.

4. Enter the Network Interface Card information. When finished, you will see the Installation screen, as shown on page 3-20.

5. Enter the system information as detailed on pages 3-20 through 3-23.

6. Verify the operation of the gateway.

Gateway Not Found, Defining Autocall

If there is a number in the Autocall field that contains a gateway code not in the current phone directory, you will receive the following display:

```
Gateway not found. Please enter the following information...
<press ENTER to continue>
```

Enter the number of channel digits in the field provided and press Enter.

```
      Gateway Code
Number of Channel Digits:
```

Diagnostics

The following diagnostics functions are built into the V/IP software for testing individual channels:

- Input Level Display
- Local Self Test
- Local Loopback
- LED Test

Input Level Display

To view the input level of a voice/fax channel within a dynamic display, proceed as follows:

1

```
  V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
 Control
 Exit
```

2

```
        CONTROL
 Analog Channel Control
 Digital Voice (DVC)
 T1/E1
 Analog Card Control
 Analog Channel Diagnostics
 Reset Analog Channel Counters
 Reset DVC Counter
 Reset System Counters
 Ping
 Update From Directory Server
 Done
```

3

```
        DIAGNOSTICS
 Input Level Display
 Start the Local Self Test
 Local Channel Loopback Test
 LED Test
 Terminate Test
 Done
```

4

```
      CARD – PORT
 CARD # 01 – PORT # 01
 CARD # 01 – PORT # 02
 CARD # 02 – PORT # 01
 CARD # 02 – PORT # 02
```

Select the channel whose input level you want to view.

You will see the input level display as a graph on screen, as follows:

```
            INPUT LEVEL DISPLAY
–25–24–21–18–15–12–10–08–06–04–03–02–01 00

              INPUT LEVEL DISPLAY : –24 dBm
```

Press the Esc key to exit from the input level display.

Local Self Test

To perform a quick, standalone test of a voice/fax channel, proceed as follows:

1

```
   V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
 Control
 Exit
```

2

```
         CONTROL
 Analog Channel Control
 Digital Voice (DVC)
 T1/E1
 Analog Card Control
 Analog Channel Diagnostics
 Reset Analog Channel Counters
 Reset DVC Counter
 Reset System Counters
 Ping
 Update From Directory Server
 Done
```

3

```
          DIAGNOSTICS
 Input Level Display
 Start the Local Self Test
 Local Channel Loopback Test
 LED Test
 Terminate Test
 Done
```

4

```
       CARD – PORT
 CARD # 01 – PORT # 01
 CARD # 01 – PORT # 02
 CARD # 02 – PORT # 01
 CARD # 02 – PORT # 02
```

Select the channel you want to test.

The V/IP gateway will perform a self test of the channel and display the results, like the following successful test:

```
Local Self Test OK
<press ENTER to continue>
```

Press the Enter key to exit from the local self test.

Local Channel Loopback Test

This test loops the transmit signal to the receive signal at a local channel. You can then connect an analog telephone to the channel, pick up the receiver, and then talk into the receiver. You should then hear an echo of your voice if the channel is operating correctly.

1

```
  V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2

```
        CONTROL
Analog Channel Control
Digital Voice (DVC)
T1/E1
Analog Card Control
Analog Channel Diagnostics
Reset Analog Channel Counters
Reset DVC Counter
Reset System Counters
Ping
Update From Directory Server
Done
```

3

```
          DIAGNOSTICS
Input Level Display
Start the Local Self Test
Local Channel Loopback Test
LED Test
Terminate Test
Done
```

4

```
      CARD – PORT
CARD # 01 – PORT # 01
CARD # 01 – PORT # 02
CARD # 02 – PORT # 01
CARD # 02 – PORT # 02
```

Select the channel you want to test.

The V/IP gateway will set up a loopback within the channel and display the following:

```
Local Channel Loopback Test OK
<press ENTER to continue>
```

Press the Enter key to acknowledge the prompt.

At this point, if you pick up the receiver of an analog telephone connected to the channel and speak into it, you should hear your voice echoed back if the channel is operating properly.

To return the channel to normal operation, proceed as follows:

1

```
   V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
 Control
 Exit
```

2

```
        CONTROL
 Analog Channel Control
 Digital Voice (DVC)
 T1/E1
 Analog Card Control
 Analog Channel Diagnostics
 Reset Analog Channel Counters
 Reset DVC Counter
 Reset System Counters
 Ping
 Update From Directory Server
 Done
```

3

```
        DIAGNOSTICS
 Input Level Display
 Start the Local Self Test
 Local Channel Loopback Test
 LED Test
 Terminate Test
 Done
```

4

```
      CARD – PORT
 CARD # 01 – PORT # 01
 CARD # 01 – PORT # 02
 CARD # 02 – PORT # 01
 CARD # 02 – PORT # 02
```

Select the channel that was being tested.

The V/IP gateway will remove the loopback and display the following:

```
Terminate Test OK
<press ENTER to continue>
```

LED Test

This item is reserved for future use.

Specifications A

General Card Specifications

PC Interface:	ISA-compatible
Size:	Full length ISA-bus
Temperature	
Operating:	32°F to 113°F (0°C to 45°C) free air temperature inside the PC enclosure
Storage:	-4°F to 149°F (-20°C to 65°C)
Humidity:	95% maximum, noncondensing
Typical Power Consumption:	FXS/E&M card: +5V, 1 ampere -5V, 0.03 ampere +12V, 1.5 amperes during startup, 0.4 ampere sustained -12V, 0.03 ampere FXO card: +5V, 1 ampere -5V, 0.03 ampere +12V, 0.03 ampere -12V, 0.03 ampere
Supply Voltage Tolerance:	+5V ± 5% -5V ± 5% +12V ± 5% -12V ± 5%

Standards

Electromagnetic Interference (radiated and conducted):	FCC Part 15 Level A C.R.C., c, 1374 EN 55022 (CISPR 22)

Susceptibility to external radiation and electrostatic discharge:	EN 50082-1:1992 IEC 801-2:1991/prEN55024-2:1992, 3KV CD, 8KV AD IEC801-3:1984/prEN55024-3:1991, 3 V/m IEC801-4:1988/prEN55024-4:1993, 0.5KV Signal, 1KV Power EN60555 Power Harmonics (June 1996 and later)
PTT/PSTN	
Canada:	DOC CS-03
European Union:	NET 4 (each country may have its own PTT requirements)
U.S.A.:	FCC Part 68
Safety	
Canada:	CSA Standard C22.2 No. 950:1993
European Union:	IEC 950:1991+A1:1992 TÜV: EN 60950:1992 EN 41003:1993 VDE 0805 BABT 340: Sixth Edition (Manufacturing)
U.S.A.:	UL 1950:1993, UL 1459
EU Declaration of Conformity	
Safety:	Directive: 73/23/EEC
EMC:	Directive: 89/336/EEC
CE-compliant:	The card meets the European EMC directive 89/336/EEC when installed in a CE-compliant chassis.

General Voice/Fax Interface Specifications

Signals Supported:	Analog voice and Group 3 facsimile (fax)
Fax Signal Types:	V.27 ter 2400, V.27 ter 4800, V.29 7200, and V.29 9600

Telephone Interfaces

PBX Tie Trunk:	E&M types I, II, and V, 2-wire or 4-wire
PBX Station or Central Office/PSTN:	FXO loop start, 2-wire
Key Telephone Systems or Telephone Set:	FXS loop start, 2-wire
Interface Connectors:	One RJ1CX 8-pin modular jack per channel, FXS and E&M interface types
	One RJ11C/W 6-pin modular jack per channel, FXO interface

FXS Telephone Interface Analog Specifications

Input Impedance	
Default:	600 Ω in series with a 2.15 µf capacitor
BSI Complex:	370 Ω + 0.31 µf in parallel with 620 Ω or 220 Ω + 0.12 µf in parallel with 820 Ω
Insertion Loss (End to End):	2 dB minimum @ 1004 Hz between Transmit and Receive of Port 1 and Transmit and Receive of Port 2
Frequency Response:	304 Hz to 3404 Hz, +1 dB/-2 dB with respect to 1004 Hz
Return Losses	
Echo:	≥ 22 dB
Singing:	≥ 18 dB
204 to 3404 Hz:	≥ 16 dB
Input Level Gain:	Adjustable from -6 dB to +6 dB in 1 dB increments
Output Level Attenuation:	Adjustable from 0 dB to 19 dB in 1 dB increments
Input and Output Level Adjustment Restriction:	If your equipment is operated in the U.S.A. or Canada, input and output levels must be set to 0 dBm.
Longitudinal Balance	
200 to 1004 Hz:	≥ 58 dB
1004 to 3404 Hz:	≥ 48 dB
Non-Linear Distortion (Multitone Signal) Second and Third Harmonics:	≥ 40 dB below signal level

Signal to Noise Ratio at 1004 Hz:	≥ 37 dB
Echo Suppression:	≥ 35 dB
Echo Canceling:	≥ 16 milliseconds (≤ 1000 kilometers or 600 miles)
Crosstalk (Near/Far End) Between Channels:	≤ 75 dB
Signaling Formats	
AC:	DTMF
DC:	Pulsed
DTMF Distortion:	≤ 1.5%, transparently passed
Pulse Distortion:	≤ 3%
Battery Current:	≥ 35 milliamperes into 130 Ω load 20 milliamperes minimum into 800 Ω total loop
Ringing Tone:	25 Hz (default) or 50 Hz
Ringing Voltage:	With AC load of 2 ringers (4000 Ω) and zero line: 25 Hz tone: ≥ 54 Vrms 50 Hz tone: ≥ 50 Vrms
Ringing Current:	15 milliamperes into 5000 Ω load

E&M Telephone Interface Analog Specifications

AC15 and Pulsed DC are not supported.

Input Impedance	
4-wire:	600 Ω ±10% resistive only
2-wire	
Default:	600 Ω ±10%
BSI Complex:	370 Ω + 0.31 µf in parallel with 620 Ω or 220 Ω + 0.12 µf in parallel with 820 Ω
Input Level Gain:	Adjustable from -6 dB to +6 dB in 1 dB increments
Output Level Attenuation:	Adjustable from 0 dB to 19 dB in 1 dB increments
Maximum Output Level, 4-wire only:	0 dB or +7 dB, software selectable
Input and Output Level Adjustment Restriction:	If your equipment is operated in the U.S.A. or Canada, input and output levels must be set to 0 dBm.
Longitudinal Balance	
204 to 1004 Hz:	≥ 58 dB
1005 Hz to 3404 Hz	≥ 48 dB
Insertion Loss	
2-wire Transmit/Receive of Port 1 to 2-wire Transmit/Receive of Port 2:	2 dB nominal at 1004 Hz
4-wire Transmit/Receive of Port 1 to 4-wire Transmit1/Receive1 of Port 2:	2 dB nominal at 1004 Hz
Return Losses	
Echo:	≥ 22 dB
Singing:	≥ 18 dB
204 to 3404 Hz:	≥ 16 dB

Frequency Response:	304 Hz to 3404 Hz, +1 dB/-2 dB with respect to 1004 Hz
Idle Channel Noise	
In band	
"C" Message:	$\leq$ 20 dBrnc
Psofometric:	$\leq$ -65 dBmpo
Out of band, 10 KHz to 10 MHz, Transverse or Metallic Noise:	$\leq$ -70 dBm
Longitudinal Noise:	$\leq$ -65 dBm
Non-Linear Distortion (Multitone Signal) Second and Third Harmonics:	$\leq$ 40 dB below signal level
Signal to Noise Ratio at 1004 Hz:	$\geq$ 37 dB
Echo Suppression, 2-wire:	$\geq$ 35 dB
Echo Canceling, 2-wire:	$\leq$ 16 milliseconds ($\leq$ 1000 kilometers or 600 miles)
Signaling	
E-Lead Current Limit:	$\leq$ 25 milliamperes
M-Lead Sensitivity:	48 V in series with $\leq$ 1400 Ω
DC Pulse Distortion:	$\leq$ 3% at 10 pps
Signaling Formats	
AC:	DTMF
DC:	Steady (supports dial pulse)
Signaling Types:	I, II, and V, as strapped

FXO Telephone Interface Analog Specifications

Input Impedance	
Default:	600 Ω in series with a 2.15 μf capacitor
BSI Complex:	370 Ω + 0.31 μf in parallel with 620 Ω or 220 Ω + 0.12 μf in parallel with 820 Ω
Insertion Loss (End to End):	2 dB minimum @ 1004 Hz between Transmit and Receive of Port 1 and Transmit and Receive of Port 2
Input Level Gain:	Adjustable from -6 dB to +6 dB in 1 dB increments
Output Level Attenuation:	Adjustable from 0 dB to 25 dB in 1 dB increments
Input and Output Level Adjustment Restriction:	If your equipment is operated in the U.S.A. or Canada, input and output levels must be set to 0 dBm.
Return Losses	
Echo:	≥ 22 dB
Singing:	≥ 18 dB
204 to 3404 Hz:	≥ 16 dB
Longitudinal Balance	
200 to 1004 Hz:	≥ 58 dB
1004 to 3404 Hz:	≥ 48 dB
Frequency Response Over the Range of 304 Hz to 3404 Hz:	+1 dB/-2 dB with respect to 1004 Hz
Idle Channel Noise	
In band	
"C" Message:	≤ 20 dBrnc
Psofometric:	≤ -65 dBmpo
Out of band, 10 KHz to 10 MHz, Transverse or Metallic Noise:	≤ -70 dBm
Longitudinal Noise:	≤ -65 dBm

Non-Linear Distortion (Multitone Signal) Second and Third Harmonics:	≤ 40 dB below signal level
Signal to Noise Ratio at 1004 Hz:	≥ 37 dB
Echo Suppression:	≥ 35 dB
Echo Canceling:	≤ 16 milliseconds (≤ 1000 kilometers or 600 miles)
Crosstalk (Near/Far End) Between Channels:	≤ 75 dB
Signaling Formats	
AC:	DTMF
DC:	Pulsed
DTMF Distortion:	≤ 1.15% (transparently passed)
Pulse Distortion:	≤ 3% at 10 PPS
DC Loop Range 48 V Battery:	≤ 1200 Ω
Disconnect Supervision	
Tone:	Voice/fax transmission will be disconnected in response to a call progress tone of less than 600 Hz.
Power Interrupt:	Voice/fax transmission will be disconnected in response to a pulse of 600 milliseconds minimum
Off-Hook DC V/I Characteristics at Tip-Ring:	≤ 300 Ω
Ringing Voltage Input, 18 to 53 Hz:	25 Vrms to 105 Vrms
Ringing Cadence Repeat Distortion:	≤ 2%

Analog Interface Descriptions B

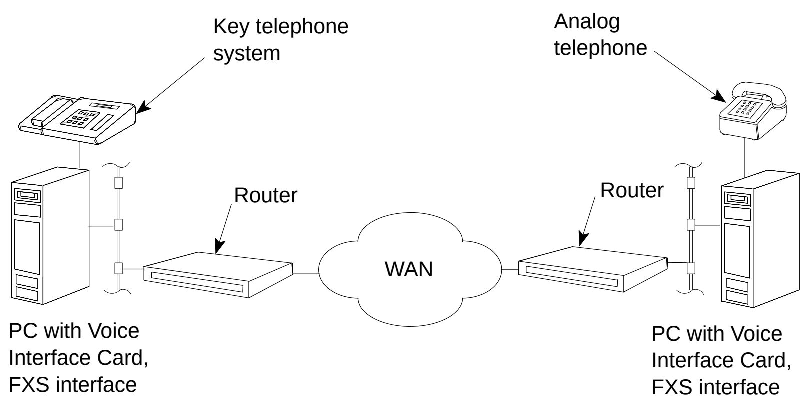
This appendix describes the FXS, E&M, and FXO interfaces.

General Description

FXS Interface

The FXS interface provides ringing voltage and battery to the attached telephone equipment. It is normally used to connect the voice/fax channel directly to a telephone instrument or to the trunk side of a key telephone system. *It must never be connected to a telephone interface that also provides ringing voltage*, such as a Central Office switch, a PBX station, or PBX off-premise circuit.

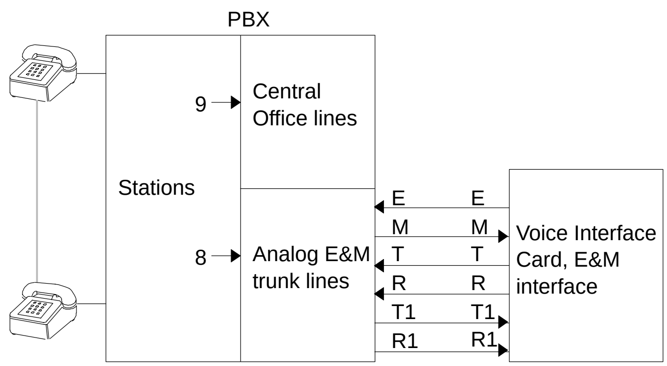
Here is a typical FXS application:



Note: The FXS interface is ***not*** intended for connection to the Public Switched Telephone Network.

E&M Interface

The E&M interface is designed to connect the card with the trunk side of a Private Branch Exchange (PBX) as shown in the illustration below. This type of telephone interface breaks out the audio signal and status control (or signaling) on separate wires.



Audio is implemented on either a single pair of wires (2-wire) labeled T/R, or on two pairs of wires (4-wire) labeled T/R and T1/R1, as configured. When configured for 2-wire operation, the T/R pair is used for both transmit and receive. When configured for 4-wire operation, the T/R pair is used for transmit (output from the Voice Interface Card) and the T1/R1 pair is used for receive (input to the Voice Interface Card).

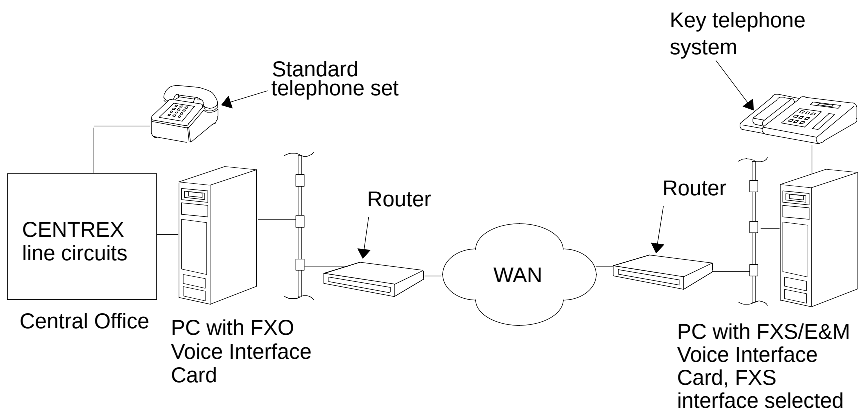
Signaling consists of an M lead and an E lead. In the E&M interface, the M lead is defined as the input, or receive, lead. The E lead is defined as the output, or transmit, lead. In some cases, an added return lead, SB, is used.

There are five E&M signaling formats that exist in the industry: types I, II, III, IV, and V. However, V/IP specifically supports the three formats commonly used today: types I, II, and V. Type I is most commonly used in the United States. Type V is common throughout the rest of the world. Type II was designed for compatibility with some AT&T PBX types. Type II can be used for most type IV applications, as well.

FXO Interface

The FXO interface normally attaches to a station side of a PBX or Central Office loop start equipment. As such, it allows the PBX station or Central Office to be extended over the network to a remote site.

Here is a typical FXO application:



Telephone Interface Connectors

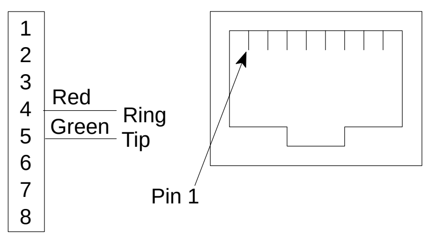
! WARNING

Remove the PC's or server's power plug from the power socket before making any connections to the telephone interfaces on the Voice Interface Cards.

FXS Interface

The FXS interface, when selected on the FXS/E&M combination card, uses an RJ1CX 8-pin modular jack. The card may include one or two telephone interface connectors, depending on the model.

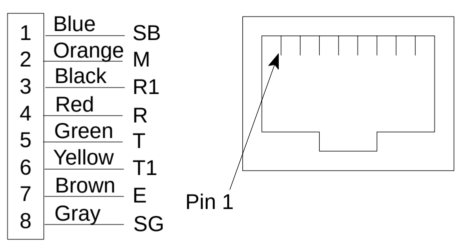
Here is the schematic diagram of the FXS connector:



E&M Interface

The E&M interface, when selected on the FXS/E&M combination card, uses an RJ1CX 8-pin modular jack. The card may include one or two of these connectors, depending on the model.

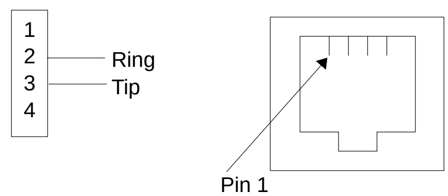
Here is the schematic diagram of the E&M connector:



FXO Interface

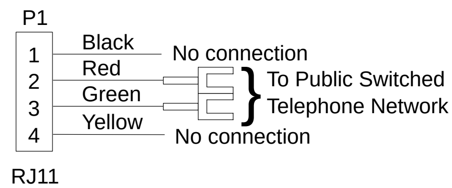
The FXO Voice Interface Card has RJ11C/W modular jacks. The card may include one or two of these connectors, depending on the model.

Here is the schematic diagram of the FXO connector:

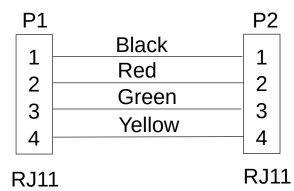


Caution

If you are going to connect the FXO Voice Interface Card to the Public Switched Telephone Network, you ***must*** use the supplied cables (part number 345-5468-014 or 345-5495-014):



MICOM cable, part number 345-5468-014

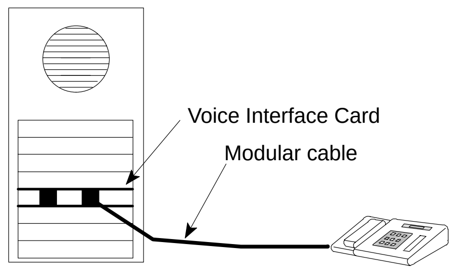


MICOM cable, part number 345-5495-014

Telephone Interface Cable Connections

FXS Interface

In most FXS interface applications, we recommend you use the modular cable(s) supplied with the Voice Interface Card to connect the channel(s) to the telephone equipment:

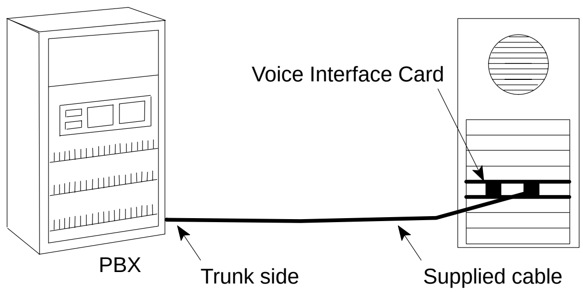


⚠ *Caution*

The RJ1CX connectors on the FXS/E&M card are 8-pin modular jacks. If you attempt to plug in 6-pin RJ11-type connectors, the outer pins of the RJ1CX connectors may be bent beyond the point where they can be used. The connectors will work for FXS interface, but might no longer function for E&M interface.

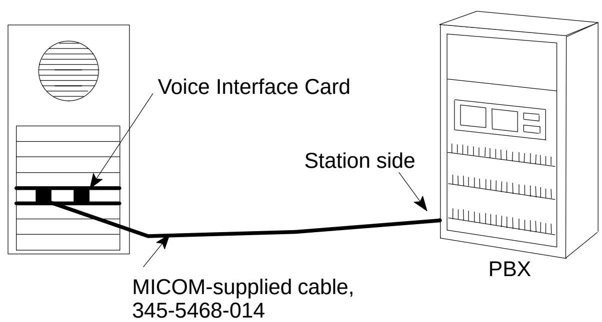
E&M Interface

For E&M interface applications, you should use the cable(s) supplied with your Voice Interface Card. The supplied cable(s) has an RJ1CX connector at one end to connect to the Voice Interface Card, and lugs at the other end to connect to the trunk side of the PBX:



FXO Interface

For FXO interface applications, you *must* use the cables supplied with your Voice Interface Card. One of the supplied cables has an RJ11 connector at one end to connect to the Voice Interface Card, and lugs to connect to the Public Switched Telephone Network or to the station side of a PBX:

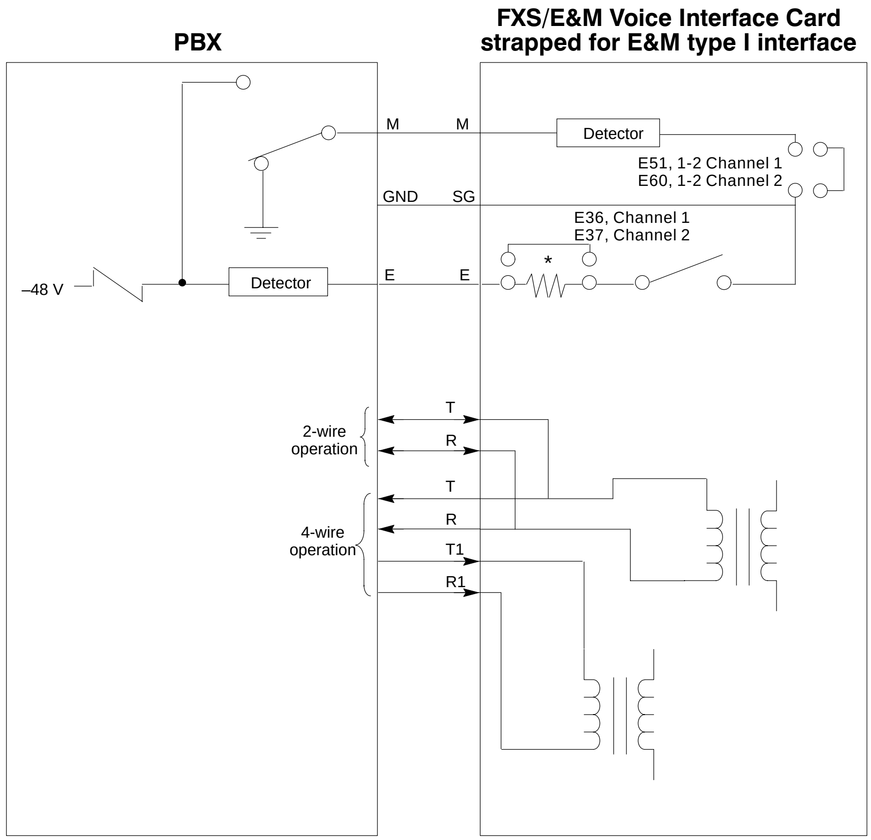


The other supplied cable has RJ11 connectors at both ends.

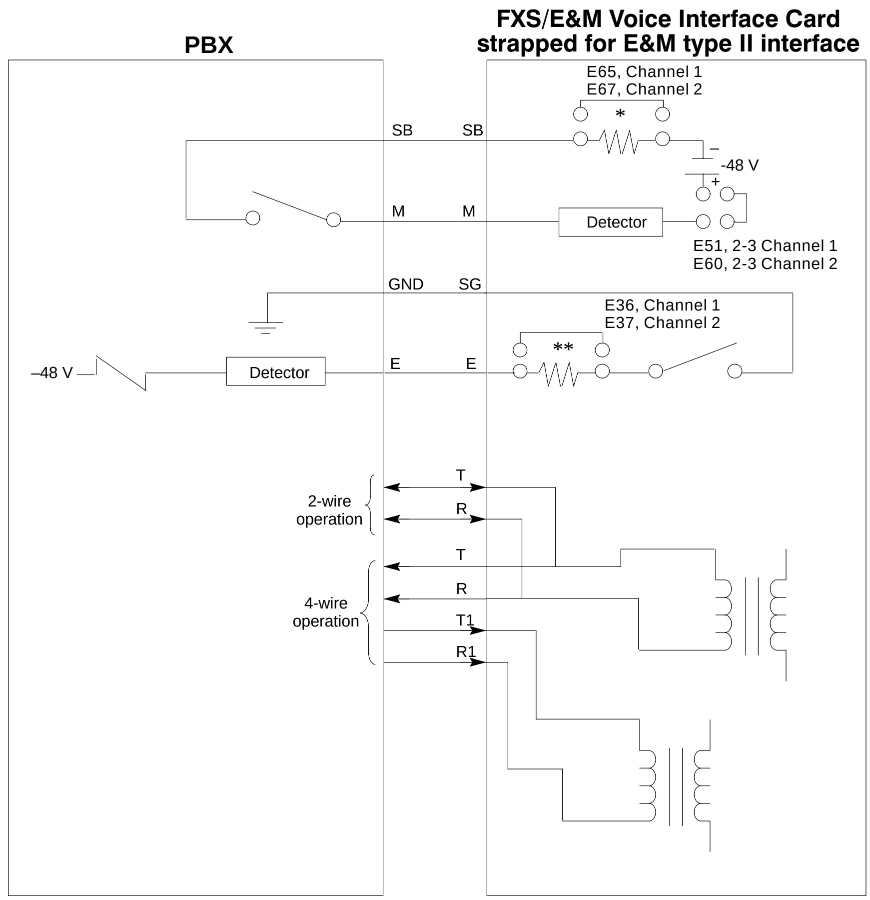
E&M Connection Diagrams

The following are simplified connection diagrams between the card and the PBX for signaling types I, II, and V.

> **Notes on T/R and T1/R1 labeling.** Four wire E&M circuits use two audio pairs: a transmit pair and a receive pair. The labeling convention shown on these pages is such that the transmit pair is designated as T/R, with arrows pointing to the receive circuits of the PBX. Conversely, the receive pair is designated as T1/R1, with the arrows coming from the transmit circuits of the PBX. Before making a connection, identify the matching pins on the interfacing PBX.



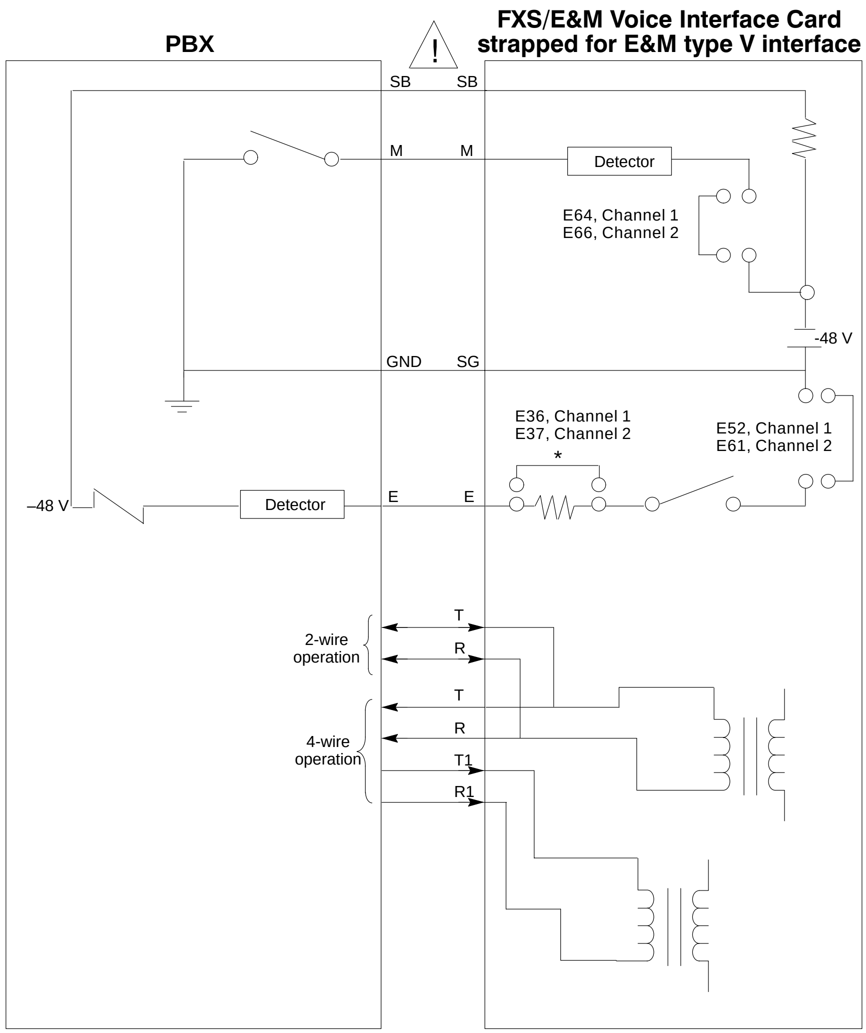
* This jumper is installed only when E lead loop resistance is too high due to long cable lengths.



* This jumper is installed only when SB lead loop resistance is too high due to long cable lengths.

** This jumper is installed only when E lead loop resistance is too high due to long cable lengths.

Note: This interface should work for most type IV interfaces. The signal designations on the PBX may differ. If the *M relay* in the PBX is polarity sensitive, try reversing the M and SB leads.



* This jumper is installed only when E lead loop resistance is too high due to long cable lengths.

Caution

PBX SB and card SB are to be connected only when -48 volt supply is not available on the PBX side.

Description of Voice/Fax Channel Characteristics Options

The following are technical descriptions of the voice/fax channel characteristics options. These options are presented when you modify the configuration of a voice/fax channel, as follows:

```
        V/IP SERVER MAIN MENU
   View Configuration
   View Status/Statistics
1  Modify Configuration
   Control
   Exit
```

```
        MODULE SELECTION
2    Analog Voice/Fax
     Digital Voice (DVC)
     T1/E1
     RSVP
     SNMP
     V/IP System
     Hardware
     Done
```

```
            CARD – PORT
3     CARD # 01 – PORT # 01
      CARD # 02 – PORT # 01
      CARD # 02 – PORT # 02
```

Select the channel you want to configure.

```
              VOICE/FAX CONFIGURATION – FXS
   Channel Number                     [1]
   UDP Port Number                    [65534]
   Mode                               [VOICE/FAX]
   Input Level Gain                   [ 0 dB]
   Output Level Attenuation           [ 0 dB]
```

Channel Number

The Channel Number is part of the overall V/IP gateway dialing plan. The Channel Number is dialed by the caller when a call is to be placed to a specific channel. The caller dials the gateway code plus the Channel Number. The call will be completed if the channel is available (on-hook and not in a busyout state). Otherwise, the caller will get a busy signal.

The Number of Digits Assigned for Channel parameter in the System Configuration menu specifies how many digits must be dialed to call to a specific channel.

If you change the Number of Digits Assigned for Channel from 1 to 2, you do not have to assign new channels. You can connect to the channels simply by dialing a 0 before the channel number.

If the Number of Digits Assigned for Channel parameter is set to 0, the V/IP gateway will assign a Channel Number of *0* to all channels. Then, all channels of all Voice Interface Cards in that gateway become one hunt group. When an incoming call comes in, it will be routed to an available channel of that gateway. The caller dials only the gateway code. The caller will get a busy signal only if all channels in the hunt group are off-hook or in a busyout state.

If you want selected channels within a V/IP gateway to be part of a hunt group, just configure those channels with the same Channel Number.

UDP Port Number

A UDP port is used by the calling side to initially set up a call. The default port number is *65534* for the first channel of the first card, *65533* for the second channel, and so on. The UDP Port Number must be unique for each channel and must not interfere with any other UDP port number used by other applications being run on the PC.

For those networks having a firewall, it is usually necessary to change the UDP port numbers. This can be done for the V/IP gateway channels by way of this parameter. However, you may find it more convenient to make the required changes to the firewall instead of the V/IP gateway channels.

Mode

The *voice/fax* mode sets the channel to process both voice and fax calls. Normally, the channel operates in the voice mode. However, if a fax signal is detected and there is no active call in progress over that channel, the channel will automatically switch to the fax mode. The channel will stay in fax mode until fax signals are no longer detected; then the channel will return to the voice mode.

The *voice only* mode should be selected when the channel is to be dedicated to voice traffic. The sensitivity of the voice/fax channels is such that they could detect a background fax connection. If this happens, the voice/fax channel could disrupt the on-going voice connection and switch to fax mode. If this happens, simply select the *voice only* mode.

Input Level Gain

You can select one of 13 input signal gain values. The range is from *-6 dB* to *+6 dB*, in 1 dB increments. *0 dB*, the default, produces no gain or attenuation. For U.S.A. and Canada networks, you must set this option to *0 dB*.

Output Level Attenuation

You can select one of 20 output level attenuation values. The range is from *0 dB* to *19 dB* of attenuation. *0 dB*, the default, produces no attenuation. For U.S.A. and Canada networks, you must set this option to *0 dB*.

Busyout Mode

If you select *system controlled*, the channel will be placed into the busyout state when there is no link bandwidth available to support a call on that channel.

If you select *forced on*, the channel will immediately go into the busyout state. This prevents any attempt to connect to the channel for an incoming call. A caller to this channel will get a busy signal. The busyout state also prevents outgoing calls from originating at this channel (the caller will get a busy signal). The busyout state is useful to disable an intermittent or defective channel until it can be repaired.

Background

If you select *regenerated*, the background sound is reproduced locally and heard by the local telephone user (during gaps in speech).

If you select *silence*, the gaps in the speech are filled by silence. This option uses less link bandwidth. Try both options and choose the setting that is most acceptable to you.

Regeneration Type

This option allows the voice/fax channel to match the outgoing dialing digits with that of the interfacing PBX.

You should select *dial pulse* for E&M and FXO interfaces if the interfacing PBX requires the dial pulse format.

You should select *DTMF* for E&M, FXS, and FXO interfaces if the interfacing PBX requires the DTMF format.

Fax Digitizing Rate

This option allows you to select from four fax digitizing rates. The available rates are *9600*, *7200*, *4800*, and *2400* bits per second.

You cannot send a fax message while a voice call is in progress. You must set the fax machine to auto answer and then place a new call. The channel's fax detector relies on the answer tone of the called fax machine. The answer tone is provided automatically when the called fax machine is set for auto answer mode.

When a V.17 fax call is detected, the Voice Interface Card will spoof the V.17 mode and send the fax at 9600 bits per second.

Regeneration Delay

Regeneration Delay compensates for the time it takes for the remote PBX to go off-hook and be ready to accept dialing digits from the voice/fax channel to complete the call setup.

Automatic Level Enhancement

Automatic Level Enhancement compensates for varying audio levels from site to site (phone system to phone system). When enabled, it boosts weak input DTMF and voice signals above the noise threshold to a minimum usable level.

You should select *disabled* for most applications.

You should select *enabled* if the signal level falls to a level too low for normal conversation.

Note: Automatic Level Enhancement is not a substitute for level adjustment for voice/fax channels.

If level adjustment is required, be sure to do the adjustment with the Automatic Level Enhancement option set to *disabled*.

Call Inhibit

If you set Call Inhibit to *enabled*, the channel will not be able to originate calls. This sets the channel to an answer only mode, if Receive Inhibit is *disabled*.

Receive Inhibit

If you set Receive Inhibit to *enabled*, the channel will not be able to answer calls. This sets the channel to an originate only mode, if Call Inhibit is *disabled*.

Note: Setting Receive Inhibit to *enabled* does not prevent the PBX from attempting to connect to the channel. You should busyout the channel if you want to take the channel off the network and stop any connections from being made to it (see page B-14).

Autocall Extension Number

This option allows you to preselect a fixed destination (extension number) within the network. When the equipment connected to this channel goes off-hook, it automatically is connected to that extension number. The autocall feature can place calls only from V/IP gateway to V/IP gateway. There should not be a PBX in front of the channel that is to dial an autocall extension number.

When autocall extension number is configured, it affects outbound calls only. When the equipment connected to this channel is on-hook, it is free to accept calls from any other extensions in the network.

Autocall extension number components are: gateway code + channel number + forwarded digits.

If you enter the gateway code of a V/IP gateway that is not listed in the phone directory database, you will see the following prompt:

```
Gateway not found: Please enter the following information...
<press ENTER to continue>
```

Press the Enter key to this prompt. You will see the following form:

```
        Gateway Code
Number of channel digits:
```

Since the V/IP gateway has no record of the gateway code, it will not know the number of channel digits for the extension number of a channel on that gateway. You must enter the number of channel digits in the field provided and press the Enter key.

Jitter Buffer

The Jitter Buffer compensates for varying delays of V/IP voice packets transiting across the WAN. The jitter buffer takes effect on the receiving side. For this release, only *static* Jitter Buffer operation is operational. The Jitter Buffer Size controls the amount of jitter buffering to be used.

Jitter Buffer Size

Jitter Buffer Size specifies how much jitter buffering to be used (how much time for which the jitter buffer should compensate). This feature improves V/IP performance on networks that exhibit variable delays of packet transit times. Also, the configurable jitter buffer compensates for a higher jitter that may be present in multinational networks that utilize multiple network services. An easy way to determine how much jitter buffering to use is to perform a "ping" of the remote host.

For NetWare, the command to perform a ping is:

TECH_PUBS: **load ping *host***

For DOS, the V/IP software provides a ping function:

```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

1

```
CONTROL
Analog Channel Control
Digital Voice (DVC)
T1/E1
Analog Card Control
Analog Channel Diagnostics
Reset Analog Channel Counters
Reset DVC Counter
Reset System Counters
Ping
Update From Directory Server
Done
```

2

```
Host IP ADDRESS
IP ADDRESS:
```

3

Enter the IP address of the computer you want to ping.

You will see a display on the status of the pings, like the following example:

```
  Ping Host: 199.30.18.10

Sent:                   0
Received:               0
Percent:                0.0%
Min Time:               0.0 ms
Max Time:               0.0 ms
Ave Time:               0.0 ms
```

The ping display will tell you the amount of time it took for a ping packet to travel from the local node to the remote node, to be responded to by the remote node, and the response packet to travel back to the local node. The ping time varies, depending on the traffic load on the network at the instant of the ping. This ping time is the value you should set for jitter buffer size.

Forward Error Correction

V/IP uses Forward Error Correction to ensure quality voice over congested networks. When the error correction is *enabled*, there is additional data included in the voice packet. This additional data is error correction information that allows the remote end to reconstruct the previously sent packet, in case it was dropped or received in error. When error correction is *enabled*, the typical peak bandwidth for a call will be 21.5 Kbps. If Background is set to silence (see page B-14), this bandwidth will be significantly reduced.

If your network does not have congestion occurring frequently to cause dropped packets, or otherwise cause packets to be received in error, you can disable the error correction. When error correction is *disabled*, the typical peak bandwidth for a call will be 17.5 Kbps. If Background is set to silence, this bandwidth will be significantly reduced.

Fax connections do not have any error correction. Their typical peak bandwidth will be 19.2 Kbps or less.

Echo Canceller

The Echo Canceller is used in selected loopback test applications. You should select *enabled* during normal operation. You must select *disabled* when performing external loopback tests or when using an external echo canceller.

Line Impedance

The line impedance values you can set the channel for are shown in the following table. The line impedance for E&M 4-wire operation is fixed at *600 ohms*.

Setting	Impedance	FXS	E&M 2-wire	E&M 4-wire	FXO
600 ohms	600 ohms, resistive	No	Yes	Yes	No
	600 ohms + 2.2 µf	Yes	No	No	Yes
BSI *Complex*	370 ohms + 0.31 µf in parallel with 620 ohms, or, 220 ohms + 0.12 µf in parallel with 820 ohms	Yes	Yes	No	Yes

Maximum Output Level

This applies to E&M 4-wire interface. *+7 dBm* should be used when the interfacing tie trunk equipment includes a pad. *0 dBm* should be used for all other applications. This option works in combination with the Output Level Attenuation to set the final output level. For example, if you set the Maximum Output Level to *+7 dBm* and you set the Output Level Attenuation to 5 dB, the resultant output level is a net gain of 2 dB.

Port Emulation

This option controls the operation of the E&M interface. The values are *DC* and *Wink Start*. Choose whichever value is appropriate for your PBX.

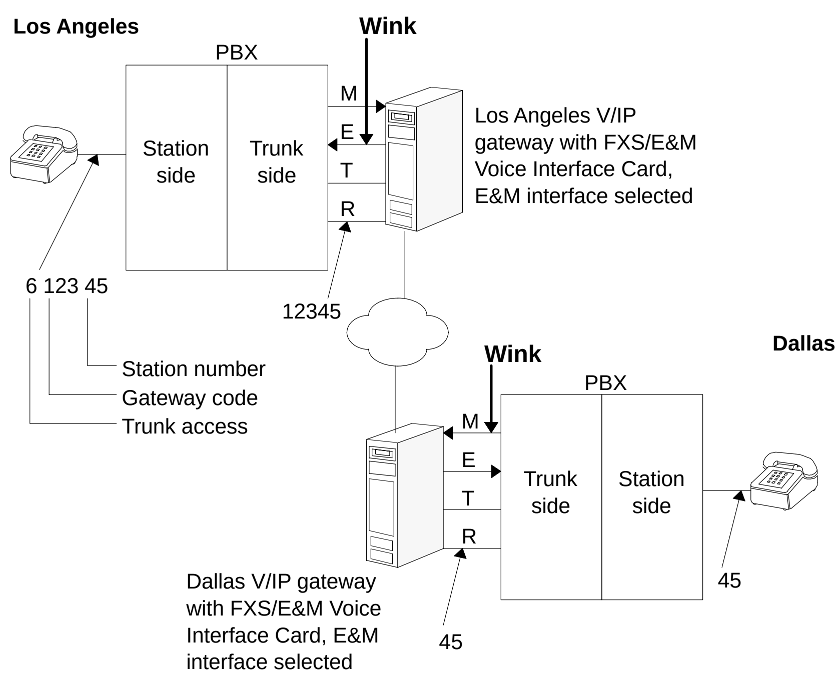
DC: the channel senses the idle/active state by the DC condition of the M lead. The channel sends this idle/active state to the remote channel as a DC condition of the E lead.

Wink Start: supports voice/fax switching between selected PBXs, such as the AT&T Legend. A caller at the originating PBX station enters a string of digits to call a remote PBX station. The originating PBX seizes the M lead and waits for a wink signal response from the local V/IP gateway channel. The PBX then forwards the dialed digits to the V/IP gateway. The remote V/IP gateway channel seizes the E lead and starts a regeneration delay timer. When the delay time expires, the remaining digits are forwarded to the destination PBX.

Example of Wink Start Operation

As shown in the illustration below, a caller at the Los Angeles PBX picks up the phone and dials the number string 612345. The trunk access code 6 causes the Los Angeles PBX to store digits 12345, seize the trunk M lead, and wait for a Wink signal from the Los Angeles V/IP gateway E&M channel E lead. When the Wink signal is received, the Los Angeles PBX forwards digits 12345 on the T and R leads to the V/IP gateway. The V/IP gateway notes that 123 is the gateway code for the Dallas site, makes a call connection to the Dallas V/IP gateway, and forwards 45 to it.

The Dallas V/IP gateway E&M channel goes off-hook by seizing the E lead and starts the regeneration delay timer. During the regeneration delay time, the Dallas PBX must respond with a Wink signal on the M lead. When the regeneration delay time expires, the V/IP gateway forwards digits 45 in DTMF or pulse form (as configured) to the Dallas PBX. The Dallas PBX rings station 45. The call connection is established when someone answers the phone.

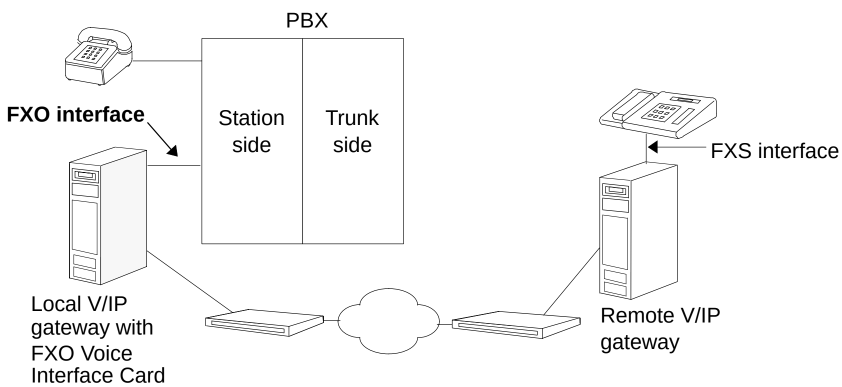


Ringing Frequency

This option controls the signal that rings the bell on the called telephone to indicate an incoming call. *25 Hz* is the normal setting. Some European telephone systems require a *50 Hz* Ringing Frequency.

Number of Ring Cycles

The FXO interface operates like a Direct Inward System Access (DISA) trunk on a PBX. It automatically answers an incoming ringing signal. This option controls the number of cycles before an FXO channel answers and provides a dial tone. Here is an example application:



In the above example, at the local side, you would dial the local FXO station number and wait for a second dial tone. The local server's FXO channel will wait the number of cycles selected for the Number of Ring Cycles option, before providing a dial tone. This applies only to calls originating at the FXO side of the WAN; it does not affect calls received by the FXO channel.

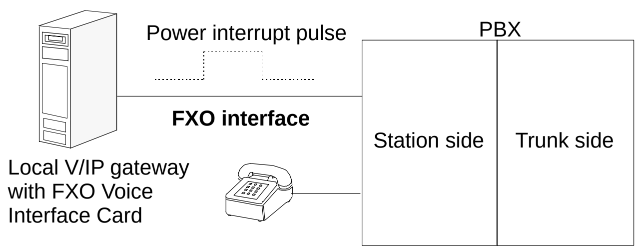
Note: If Number of Ring Cycles is configured for *8* or *9*, you may hear 9 or more actual rings.

Disconnect Supervision

Disconnect supervision is a method used to disconnect a telephone connection when the telephone equipment at one end goes from an off-hook (active) state to an on-hook (idle) state. This is required only for FXO channels.

You should select *tone* if the attached PBX or Central Office cannot supply a power interrupt pulse to disconnect the connection. The FXO channel will then look for a call progress tone of 600 Hz or less to disconnect the connection. This is effective only during call setup, until the remote interface answers the call.

You should select *power interrupt* if the attached PBX or Central Office can provide a 600 millisecond pulse when the telephone set on the remote side goes on hook. This setting is recommended for CENTREX® (Bell System) or CENTRANET® (GTE) station lines from the Central Office. Here is an example application:



DTMF Detector

When *enabled*, this function ensures that DTMF signals at the remote end are regenerated with a uniform on/off time of 100 milliseconds.

SNMP Management C

SNMP Configuration

The following configuration is required only for DOS PCs.

```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

1 — **Modify Configuration**

```
MODULE SELECTION
Analog Voice/Fax
Digital Voice (DVC)
T1/E1
RSVP
SNMP
V/IP System
Hardware
Done
```

2 — **SNMP**

You will see the SNMP Configuration menu. The menu items and their function are described in the table on the next page.

Option	Default	Description
Functionality	Enable	Enables or disables the SNMP function.
Set	Enable	Enables or disables the ability of the V/IP gateway to respond to SNMP set commands.
Trap	Enable	Enables or disables the sending of SNMP traps. Traps are not supported in this release.
Set Command Community String	private	The community string that must accompany SNMP set commands to the V/IP gateway.
Get Command Community String	public	The community string that must accompany SNMP get commands to the V/IP gateway.
Trap Community String	private	The community string that will accompany SNMP traps sent by the V/IP gateway. Traps are not supported in this release.
Management Stations – Allow		The IP addresses of the SNMP management stations allowed to access the V/IP gateway.
Management Stations – Trap Receivers		The IP addresses of the SNMP management stations that will receive traps sent by the V/IP gateway. Traps are not supported in this release.

MIB-II

MIB-II Support In NetWare

The V/IP gateway in the NetWare environment supports MIB-II via the Novell SNMP stack.

MIB-II Support In DOS

The V/IP gateway in the DOS environment supports MIB-II as follows:

MIB-II Group	Functions and Objects Supported
System Group {mib-2 1}	GET, GETNEXT, and SET are supported for all objects.
Interface Group {mib-2 2}	`ifTable` (Interfaces Table): read-only objects (such as all counters) are supported. Read/write fields (such as `ifAdminStatus`) are not supported.
Address Translation Group {mib-2 3}	`atTable` (Address Translation Table): GET and GETNEXT are supported. (Cannot configure static ARP table entries.)
IP Group {mib-2 4}	• Counters: GET and GETNEXT are supported • `ipAddrTable` (IP Address Table): GET, GETNEXT, and SET are supported. – Only one interface is available. – New interfaces cannot be added. – Can change IP address (`ipAdEntAddr`) and network mask (`ipAdEntNetMask`). – Cannot change reassembly size (`ipAdEntReasmMaxSize`) and IP broadcast address (`ipAdEntBcastAddr`). • `ipRoutingTable` (IP Routing Table): GET and GETNEXT are supported for all objects. SET can configure only the default route. • `ipNetToMediaTable` (IP Address Translation Table): GET and GETNEXT are supported. Cannot configure static ARP table entries.

MIB-II Group (Cont.)	**Functions and Objects Supported** (Cont.)
ICMP Group {mib-2 5}	GET and GETNEXT are supported.
TCP Group {mib-2 6}	Not supported.
UDP Group {mib-2 7}	GET and GETNEXT are supported, including counters and `udpTable` (UDP Port Table).
EGP Group {mib-2 8}	Not supported.
Transmission Group {mib-2 10}	Not supported.
SNMP Group {mib-2 11}	GET and GETNEXT are supported for all counters.

Enterprise MIB Extensions

The Utilities diskette contains the file `micmv11c.mib`. This file contains the enterprise MIB extensions for MICOM V/IP products.

The `micmv11c.mib` file can be compiled into your network management system to allow you to manage all V/IP gateways in your network and their Voice Interface Cards. The overall architecture of the V/IP Enterprise MIB Extensions is shown in the paragraphs that follow.

MICOMVIP.MIB, the Enterprise MIB Extension for MICOM V/IP Products

VIP-VOICE

iso.org.dod.internet.private.enterprises.micom.products.
micom-vip.vip-voice
1.3.6.1.4.1.335.1.3.1

Voice/Fax Channel Diagnostics Table

```
voiceChannelDiagTable {vip–voice 1}
     voiceChannelDiagEntry (1)
          voiceChannelDiagCardNumber (1)
          voiceChannelDiagChannelNumber (2)
          voiceChannelDiagCommand (3)
```

Voice/Fax Channel Physical Parameters Table

```
voiceFaxChPhyParaTable {vip–voice 2}
     voiceFaxChPhyParaEntry (1)
          voiceFaxChPhyParaCardNumber (1)
          voiceFaxChPhyParaChannelNumber (2)
          voiceFaxChPhyParaEnMWireOperation (3)
          voiceFaxChPhyParaViewSignalType (4)
          voiceFaxChPhyParaBergStrapType (5)
          voiceFaxChPhyParaFXONumberOfRings (6)
          voiceFaxChPhyParaDisconnectSupervision (7)
          voiceFaxChPhyParaRingFrequency (8)
          voiceFaxChPhyParaForwardErrCorrection (9)
          voiceFaxChPhyParaJitterBufferType (10)
          voiceFaxChPhyParaJitterSize (11)
          voiceFaxChPhyParaAutoLevelEnhancement (12)
          voiceFaxChPhyParaBackground (13)
          voiceFaxChPhyParaDigitizingRate (14)
          voiceFaxChPhyParaBusyoutMode (15)
          voiceFaxChPhyParaVoiceFaxMode (16)
          voiceFaxChPhyParaInputGain (17)
          voiceFaxChPhyParaEMTypeStrapping (18)
          voiceFaxChPhyParaOutputAttenuation (19)
          voiceFaxChPhyParaFaxDigitizingRate (20)
          voiceFaxChPhyParaLineImpedance (21)
          voiceFaxChPhyParaMaxOutputLevel (22)
          voiceFaxChPhyParaRegenerationDelay (23)
          voiceFaxChPhyParaRegenerationType (24)
          voiceFaxChPhyParaDtmfDetect (25)
          voiceFaxChPhyParaEchoCanceller (26)
          voiceFaxChPhyParaEMSignalFormat (27)
```

Voice/Fax Channel Switching Parameters Table

```
voiceFaxChSwParaTable {vip–voice 3}
     voiceFaxChSwParaEntry (1)
          voiceFaxChSwParaCardNumber (1)
          voiceFaxChSwParaChannelNumber (2)
          voiceFaxChSwParaAutoCallNumber (3)
          voiceFaxChSwParaReceiveInhibit (4)
          voiceFaxChSwParaCallInhibit (5)
          voiceFaxChSwParaChannelEnabled (6)
```

Digital Voice Channel Physical Parameters Table

```
digitalVoiceChPhyParaTable {vip–voice 5}
     digitalVoiceChPhyParaEntry (1)
          digitalVoiceChPhyParaCardNumber (1)
          digitalVoiceChPhyParaChannelNumber (2)
          digitalVoiceChPhyParaViewSignalType (3)
          digitalVoiceChPhyParaForwardErrCorrection (4)
          digitalVoiceChPhyParaPortEmulation (5)
          digitalVoiceChPhyParaJitterBufferType (6)
          digitalVoiceChPhyParaJitterSize (7)
          digitalVoiceChPhyParaAutoLevelEnhancement (8)
          digitalVoiceChPhyParaBackground (9)
          digitalVoiceChPhyParaDigitizingRate (10)
          digitalVoiceChPhyParaBusyoutMode (11)
          digitalVoiceChPhyParaInputGain (12)
          digitalVoiceChPhyParaOutputAttenuation (13)
          digitalVoiceChPhyParaFaxDigitizingRate (14)
          digitalVoiceChPhyParaRegenerationDelay (15)
          digitalVoiceChPhyParaRegenerationType (16)
          digitalVoiceChPhyParaCompander (17)
          digitalVoiceChPhyParaModuleIdentification (18)
          digitalVoiceChPhyParaDtmfDetect (19)
          digitalVoiceChPhyParaEchoCanceller (20)
          digitalVoiceChPhyParaVoiceFaxMode (21)
```

Digital Voice Channel Switching Parameters Table

```
digitalVoiceChSwParaTable {vip–voice 6}
     digitalVoiceChSwParaEntry (1)
          digitalVoiceChSwParaCardNumber (1)
          digitalVoiceChSwParaChannelNumber (2)
          digitalVoiceChSwParaDvcNumber (3)
          digitalVoiceChSwParaDs0Number (4)
          digitalVoiceChSwParaDvcDs0Result (5)
          digitalVoiceChSwParaAutoCallNumber (6)
          digitalVoiceChSwParaReceiveInhibit (7)
          digitalVoiceChSwParaCallInhibit (8)
```

Reset Port (Channel) Table

```
voiceResetPortTable {vip-voice 7}
     voiceResetPortEntry (1)
          voiceResetPortCardNumber (1)
          voiceResetPortChannelNumber (2)
          voiceResetPortCommand (3)
```

Reset Card Table

```
voiceResetCardTable {vip-voice 8}
     voiceResetCardEntry (1)
          voiceResetCardCardNumber (1)
          voiceResetCardCommand (2)
```

Channel Status Table

```
voiceChStatusTable {vip-voice 9}
     voiceChStatusEntry (1)
          voiceChStatusCardNumber (1)
          voiceChStatusChannelNumber (2)
          voiceChStatusInputLevel (3)
          voiceChStatusStatus (4)
          voiceChStatusDspSoftwareRevision (5)
          voiceChStatusHardwareInterfaceRev (6)
          voiceChStatusTestMode (7)
          voiceChStatusTestResult (8)
          voiceChStatusFlashState (9)
          voiceChStatusVoiceFaxMode (10)
```

DSP Download Table

```
voiceDspDownloadTable {vip-voice 10}
     voiceDspDownloadEntry (1)
          voiceDspDownloadCardNumber (1)
          voiceDspDownloadPortNumber (2)
          voiceDspDownloadChannelIndex (3)
          voiceDspDownloadCommand (4)
          voiceDspDownloadStatus (5)
          voiceDspDownloadFailedReason (6)
          voiceDspCividNumber (7)
```

DSP Download Group

```
voiceDspDownloadGroup {vip-voice 11}
     voiceDspDownloadImageName (1)
     voiceDspDownloadAction (2)
     voiceDspDownloadLastActionResult (3)
```

VIP-SNMP

iso.org.dod.internet.private.enterprises.micom.products.
micom-vip.vip-snmp
1.3.6.1.4.1.335.1.3.2

SNMP Configuration Group

```
vipSnmpConfigGroup {vip–snmp 1}
     vipSnmpEnable (1)
     vipSnmpSetEnable (2)
     vipSnmpTrapEnable (3)
     vipSnmpGetCommunityString (4)
     vipSnmpSetCommunityString (5)
     vipSnmpTrapCommunityString (6)
```

SNMP Allowed Managers Table

SNMP managers that are allowed to do GET/SET operations on the local gateway.

```
vipSnmpAllowedManagersTable {vip–snmp 2}
     vipSnmpAllowedManagersEntry (1)
          vipSnmpAllowedManagersIpAddress (1)
          vipSnmpAllowedManagersStatus (2)
```

SNMP Trap Receivers Table

SNMP managers to whom SNMP traps will be sent by the local gateway.

```
vipSnmpTrapReceiversTable {vip–snmp 3}
     vipSnmpTrapReceiversEntry (1)
          vipSnmpTrapReceiversIndex (1)
          vipSnmpTrapReceiversIpAddress (2)
          vipSnmpTrapReceiversStatus (3)
```

VIP-RSVP

iso.org.dod.internet.private.enterprises.micom.products.
micom-vip.vip-rsvp
1.3.6.1.4.1.335.1.3.3

RSVP Configuration Group

```
vipRsvpConfigGroup {vip–rsvp 1}
     vipRsvpEnable (1)
     vipRsvpOptional (2)
     vipRsvpPathRefreshTime (3)
     vipRsvpResvRefreshTime (4)
     vipRsvpAgeoutFactor (5)
     vipRsvpPathAdjustmentFactor (6)
     vipRsvpFirstRouterAddress (7)
     vipRsvpFirstRouterSubnet (8)
```

VIP-SYSTEM

iso.org.dod.internet.private.enterprises.micom.products.
micom-vip.vip-sys
1.3.6.1.4.1.335.1.3.4

General System Configuration Parameters Group

```
vipSysConfigGroup {vip–sys 1}
     vipGatewayCode (1)
     vipDirectoryServer (2)
     vipDatabaseSyncInterval (3)
     vipChannelDigits (4)
     vipMaxDialDigits (5)
     vipDefDigits (6)
     vipDefInterDigitTimer (7)
     vipInterDigitTime (8)
     vipCallProgressTone (9)
     vipUdpCtrlPort (10)
     vipLoadDirDbase (11)
     vipUpdateDirDbase (12)
     vipDisconnectTimeout (13)
```

Voice Channel to UDP Port Mapping Table

```
vipChannelToUdpPortNumberMapTable {vip–sys 2}
     vipChannelToUdpPortNumberMapEntry (1)
          vipChannelToUdpPortNumberMapCardNum (1)
          vipChannelToUdpPortNumberMapPortNum (2)
          vipChannelToUdpPortNumberMapVoiceChannel (3)
          vipChannelToUdpPortNumberMapUdpPortNumber (4)
```

Gateway Code to IP Address Mapping Table

```
vipGatewayCodeToIpAddrMapTable {vip–sys 3}
     vipGatewayCodeToIpAddrMapEntry (1)
          vipGatewayCodeToIpAddrMapGatewayCode (1)
          vipGatewayCodeToIpAddrMapGatewayIpAddr (2)
          vipGatewayCodeToIpAddrMapNumOfDialDigits (3)
          vipGatewayCodeToIpAddrMapNumOfChnlDigits (4)
```

System Statistics Group

```
vipSysStatisticsGroup {vip–sys 4}
     vipDatabasePacketsSent (1)
     vipDatabasePacketsRcvd (2)
     vipCallMgmtPacketsSent (3)
     vipCallMgmtBytesSent (4)
     vipCallMgmtPacketsRcvd (5)
     vipCallMgmtBytesRcvd (6)
     vipIncomingCallAttempts (7)
```

System Counters Reset Action Group

```
vipSysCounterResetGroup {vip–sys 5}
    vipSysCounterResetCommand (1)
```

Channel Statistics Table

```
vipChanStatisticsTable {vip–sys 6}
    vipChanStatisticsEntry (1)
        vipChanStatisticsCardNum (1)
        vipChanStatisticsPortNum (2)
        vipChanStatisticsIncomingConnections (3)
        vipChanStatisticsOutgoingConnections (4)
        vipChanStatisticsOutgoingAttempts (5)
        vipChanStatisticsNumVoiceConn (6)
        vipChanStatisticsTotalVoiceCallDuration (7)
        vipChanStatisticsAveVoiceCallDuration (8)
        vipChanStatisticsCurrentVoicePktsRcvd (9)
        vipChanStatisticsCurrentVoicePktsSent (10)
        vipChanStatisticsTotalVoicePktsRcvd (11)
        vipChanStatisticsTotalVoicePktsSent (12)
        vipChanStatisticsTotalVoiceBytesRcvd (13)
        vipChanStatisticsTotalVoiceBytesSent (14)
        vipChanStatisticsNumFaxConn (15)
        vipChanStatisticsTotalFaxCallDuration (16)
        vipChanStatisticsAveFaxCallDuration (17)
        vipChanStatisticsCurrentFaxPktsRcvd (18)
        vipChanStatisticsCurrentFaxPktsSent (19)
        vipChanStatisticsTotalFaxPktsRcvd (20)
        vipChanStatisticsTotalFaxPktsSent (21)
        vipChanStatisticsTotalFaxBytesRcvd (22)
        vipChanStatisticsTotalFaxBytesSent (23)
        vipChanStatisticsOutgoingPktsDiscarded (24)
        vipChanStatisticsPktsOutOfSequence (25)
```

Channel Counters Reset Table

```
vipChanCounterResetTable {vip–sys 7}
    vipChanCounterResetEntry (1)
        vipChanCounterResetCardNumber (1)
        vipChanCounterResetPortNumber (2)
        vipChanCounterResetCommand (3)
```

VIP-HARDWARE

iso.org.dod.internet.private.enterprises.micom.products.
micom-vip.vip-hardware
1.3.6.1.4.1.335.1.3.5

```
vipHardwareCardTable {vip–hardware 1}
    vipHardwareCardEntry (1)
        vipHardwareCardNum (1)
        vipHardwareNumOfVipChannels (2)
        vipHardwareIRQAddress (3)
        vipHardwareIOAddress (4)
        vipHardwareSharedRamAddress (5)
        vipHardwareInterfaceType (6)
```

VIP-TRAP

iso.org.dod.internet.private.enterprises.micom.products.
micom-vip.vip-trap
1.3.6.1.4.1.335.1.3.6

Trap Identification Group

```
trapGatewayCode {vip–trap 1}
trapGatewayIpAddress {vip–trap 2}
trapDirServerIpAddress {vip–trap 3}
trapGatewayCodeDigits {vip–trap 4}
trapHardwareErrorCode {vip–trap 5}
trapCardNum {vip–trap 6}
trapChanNum {vip–trap 7}
```

VIP Trap Messages

```
vipEventGatewayUp 1001
vipEventLocateDirServerFailed 1002
vipEventDirDbaseSyncFailed 1003
vipEventDirDbaseDeRegFailed 1004
vipEventGatewayCodeDigitMismatch 1005
vipEventDuplicateGatewayCode 1006
vipEventInvalidPassword 1007
vipEventDirectoryServerBusy 1008
vipEventHardwareInitFail 1009
vipEventChannelInitFail 1010
vipEventGatewayDown 1011
vipEventDspDownloadOk 1012
vipEventDspDownloadFailed 1013
```

VIP-DSX1

iso.org.dod.internet.private.enterprises.micom.products.
micom-vip.vip-dsx1
1.3.6.1.4.1.335.1.3.7

Digital Card Configuration

```
vipDsx1ConfigTable {vip–dsx1 1}
    vipDsx1ConfigEntry (1)
        vipDsx1LineIndex (1)
        vipDsx1LineType (2)
        vipDsx1LineCode (3)
        vipDsx1LoopbackConfig (4)
        vipDsx1LineStatus (5)
        vipDsx1SystemClock (6)
        vipDsx1LineLength (7)
        vipDsx1IdleCode (8)
```

Digital Card Status

```
vipDsx1StatusTable {vip–dsx1 2}
    vipDsx1StatusEntry (1)
        vipDsx1StatusLineIndex (1)
        vipDsx1StatusTS16AISAlarm (2)
        vipDsx1StatusAISDetected (3)
        vipDsx1StatusRedAlarm (4)
        vipDsx1StatusT1YellowOrE1TS0Alarm (5)
        vipDsx1StatusE100SMFAlarm (6)
        vipDsx1StatusE1RemoteYBitAlarm (7)
        vipDsx1StatusE1RemoteABitAlarm (8)
        vipDsx1StatusE100CMFAlarm (9)
```

Agency Requirements D

Agency Requirements

Information provided in the following paragraphs is applicable only to cards used in networks within the U.S.A., Canada, and United Kingdom. If your network is located outside of these countries, you must meet the requirements of the country wherein the equipment is operated.

FCC Requirements

The Federal Communications Commission (FCC) has approved this equipment for operation in the U.S.A. In so doing, with the exception of the FXS interface, the FCC approves this equipment as not being harmful to the telephone network when this equipment is connected directly to the telephone lines.

Connecting to Public Switched Telephone Network

- FXS interface: must ***not*** be connected to the Public Switched Telephone Network.
- E&M interface: may be connected to the Public Switched Telephone Network.
- FXO interface: may be connected to the Public Switched Telephone Network only if the MICOM-supplied cable is used.

Telephone Company Requirements

User's Responsibility

If a need arises in the future, the telephone company will call you and request the following information:

- Manufacturer of the device: MICOM Communications Corp.
- Model number of the device: VIP-1-ISA-FXE (single channel FXS/E&M card)
  VIP-2-ISA-FXE (dual channel FXS/E&M card)
  VIP-1-ISA-FXO (single channel FXO card)
  VIP-2-ISA-FXO (dual channel FXO card)
  VIP-2002-ISA-FXO-UK (BABT version of dual channel FXO card)
  VIP-2001-ISA-FXO-UK (BABT version of single channel FXO card)
- FCC Part 68 (U.S.A.): See label on card for registration number.
- DOC CS-03 (Canada): See label on card for registration number.
- Type of interface: FXS, E&M, or FXO
- Jacks, Facility Interface Codes, Service Order Codes, and Ringer Equivalence; *see table below.*

Interface type	USO jack	Facility Interface Code	Service Order Code	Ringer Equivalence
E&M	RJ1CX	TC11E 2-wire TC32E 4-wire	9.0 N 9.0 N	– –
FXO	RJ11C/W	O2LS2 2-wire	9.0 F	1.0

Registration

The FXO interface card, and the FXS/E&M interface card with E&M interface selected, are registered with the FCC based upon compliance with Part 68 of its rules. The FXS/E&M card with FXS interface selected is not registered with the FCC. Therefore, the FXS interface cannot be connected to the Public Switched Telephone Network.

Note: FCC registration does not constitute an expressed or implied guarantee of performance. Only the warranty set forth in this manual covers the performance of these cards.

Telephone Company Rights and Responsibilities

If your equipment causes harm to the telephone network, the telephone company may discontinue your service temporarily. If possible, they will notify you in advance. But, if advance notice is not practical, you will be notified as soon as possible. You will be given the opportunity to correct the situation and you will be informed of your right to file a complaint with the regulatory agency.

Your telephone company may make changes in its facilities, equipment, operation, or procedures that could affect the proper functioning of your equipment. If they do, you will be notified in advance to give you an opportunity to maintain uninterrupted service.

Repair Instructions

If you experience any operational problems while using your equipment, determine if the problem is due to a malfunction in your equipment or in the telephone interface.

! ***WARNING***

Do not attempt to repair the cards. Attempts to repair the cards may cause injury and may also damage equipment on the telephone network. Attempts to repair the cards are violations of FCC rules. Repairs to the cards can be made only by the manufacturer, its authorized agents, and by others who may be authorized by the FCC. Please contact your Certified Distributor.

1. The problem may be in the telephone service.

Verify the integrity of the telephone line. If the line is not functioning properly, disconnect your equipment from the telephone interface and notify the telephone company of the problem.

2. The problem may be in your equipment.

If the telephone line is operational, or if the telephone line is known to work with other equipment, then the problem is most likely in the card.

Refer to the following guidelines for obtaining service:

a. Verify that the card is configured as required and you are using the proper cables.

b. If the card is covered by warranty, follow the procedure set forth in the Warranty page for obtaining repair or replacement of the card.

c. If the card is no longer covered by warranty, contact your Certified Distributor or write to:

MICOM Communications Corp.

4100 Los Angeles Avenue

Simi Valley, CA 93063-3397 U.S.A.

BABT Notes

Ringer Equivalence Number (REN, U.K.)

The Ringer Equivalence Number (REN) for the FXO interface port is 1.0.

Satisfactory operation of this equipment will be achieved if the sum of RENs of terminal equipment connected in parallel does not exceed 4. For the purposes of this calculation, the REN of a terminal equipment rented or bought from plc should be assumed to be REN = 1, unless otherwise specified.

Loop Disconnect Dialing

Although this equipment can use either loop disconnect or DTMF dialing, only the performance of the DTMF signaling is subject to regulatory requirements for the correct operation. It is therefore strongly recommended that the equipment be set to use DTMF signaling for access to public or private emergency services. DTMF signaling also provides faster call setup.

Equipment Attachment Limitations for Operation in Canada

CP-01, Part I, Section 10.1

NOTICE: The Canadian Department of Communications label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. In some cases, the company's inside wiring associated with a single line individual service may be extended by means of a certified connector assembly (telephone extension cord). The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

! *Caution*

Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

CP-01, Part I, Section 10.2

NOTICE: The **Load Number** (LN) assigned to each terminal device denotes the percentage of the total load to be connected to a telephone loop which is used by the device, to prevent overloading. The termination on a loop may consist of any combination of devices subject only to the requirement that the total of the Load Numbers of all the devices does not exceed 100.

Input/Output Level Adjustments E

This appendix contains procedures for adjusting the input/output levels of a V/IP Voice Interface Card.

These procedures are intended for equipment operated outside of the U.S.A. and Canada, and must meet the standards of the country wherein the equipment is used.

Overview

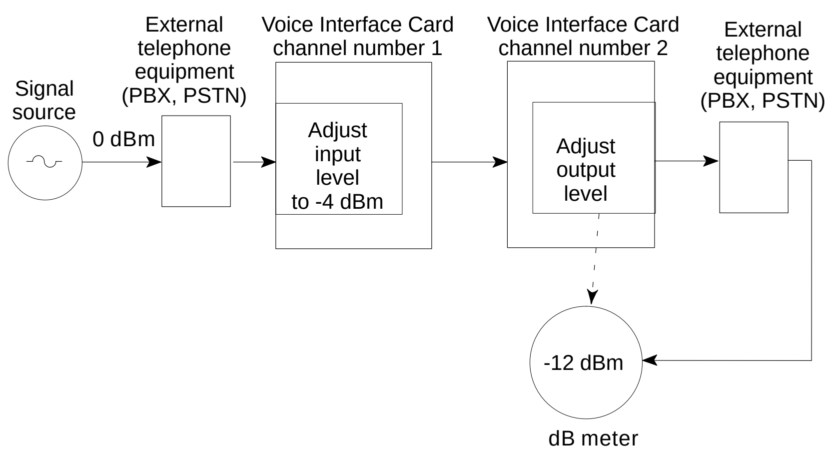
Voice Interface Card input/output levels are set at the factory for a default value of 0/0 dB. At some installations, the input/output levels need to be adjusted to compensate for external network losses and variations, to optimize the voice compression operation, and to provide a consistent communication level throughout the network. However, no Voice Interface Card adjustments can correct for voice distortions prior to the arrival at the input of the Voice Interface Card channels, or for unusual impedances inherent in the external network.

The objective of these procedures is to obtain a Voice Interface Card input level of -4 dBm, and achieve an overall circuit loss ranging from 8 to 16 dB without introducing an input gain setting of more than 6 dB to achieve it. An excessive input gain setting may cause echoes or other voice impairments such as singing or repeated DTMF signaling digits.

Adjust all Voice Channels at One Location

If the local V/IP gateway includes at least two voice channels (these channels may be on the same card), all voice channels at that location may be adjusted by a single technician without going across the network. This is accomplished by connecting any two compatible voice channels at the same location.

First, a signal from a fixed dBm source is applied in one direction through the connected pair, with the voice channel nearest the signal source serving as the sending channel, and the other voice channel serving as the receiving channel. Adjustment involves setting the sending channel's input level to a Transmission Level Point (TLP) of -4 dBm at the input level display, and the receiving channel's output level to a reading of -12 dBm at the associated dB meter:



Next, the two voice channels remain connected as before, but the signal flow is reversed. What was previously the receiving voice channel now becomes the sending channel, and vice versa, and the adjustments are made in a reverse order.

If your location has but one voice channel, you must pair up this channel with another, compatible voice channel across the network. In this case the adjustment procedure requires two technicians, one at each location.

Private Network or PSTN

V/IP Voice Interface Card channels were designed for communication among multiple offices of a single company. As such, the levels obtained from the various telephone instruments of that private network will normally be consistent within a few dBs. Under these conditions, the input level adjustment procedure is based on a signal source (speaker) of a 1004 Hz tone at 0 dBm.

Where regulations allow, certain Voice Interface Card channels may connect directly or indirectly to the Public Switched Telephone Network (PSTN), presenting a different set of conditions. Here, the calls may originate from widely different sources with extreme variations in signal levels (up to 20 dB). Further, the levels from calls within an office will often be different from the PSTN levels.

Therefore, it is strongly recommended that the channels be dedicated *either* for private network use *or* PSTN use, but not both.

If the planned usage for a single voice channel is both private network and PSTN operation, then one of the applications should be selected as the primary use and the voice channel aligned for that purpose. The secondary application should be validated to determine if a compromise setting is required.

Equipment Required

- Two MetroTel Voice Network Analyzers (VNA-70A) or equivalent, capable of generating a 1004 Hz tone at 0 dBm and also measuring a receive tone level in dBm.
- One telephone lineman's test set. If not available, a standard single-line analog telephone instrument may be substituted. A second phone or test set may be helpful.
- One FXS/E&M Voice Interface Card with FXS interface selected on one of the channels. The FXS channel is required as a companion channel for adjusting FXO channels in a V/IP gateway that has only FXO cards.

Preliminary Considerations and Connections

The telephone equipment to which a Voice Interface Card channel is attached should be installed and made functional. If the telephone equipment is not installed, then the V/IP installation should be scheduled after the telephone equipment is installed and configured. Or, a joint installation should be scheduled (as a last resort).

Interface Pairings

As described previously, every adjustment procedure described here involves a local pair of voice channels. The voice pairings treated here follow the most common applications, as follows:

- PBX trunk application (E&M to E&M or FXS to FXS)
- PBX or PSTN station application (FXO to FXS)
- Key telephone system application (FXS to FXS)

Preliminary Settings

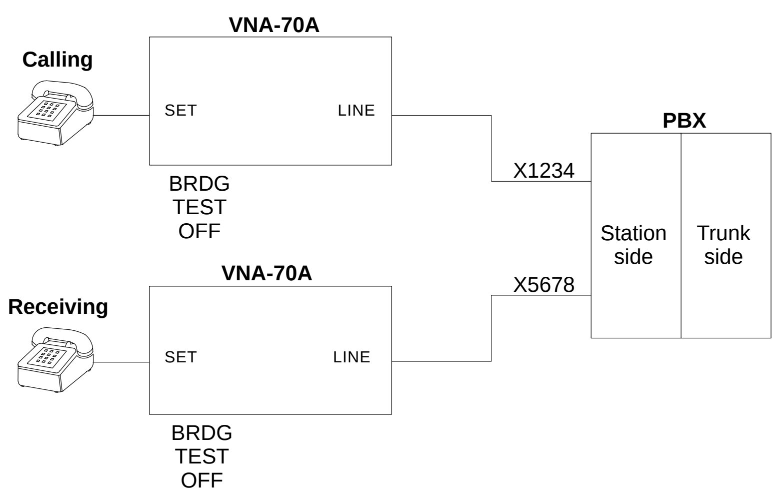
Before attempting to adjust the input output levels, configure them to the following settings:

Application	Input level setting	Output level setting
PBX trunk (E&M)	-2	2
PBX station (FXO)	2	0
Key telephone system (FXS)	-3	2

Checking the Operation of the PBX Station

This procedure is used to verify that the losses across the PBX, in a PBX station application, are within acceptable limits.

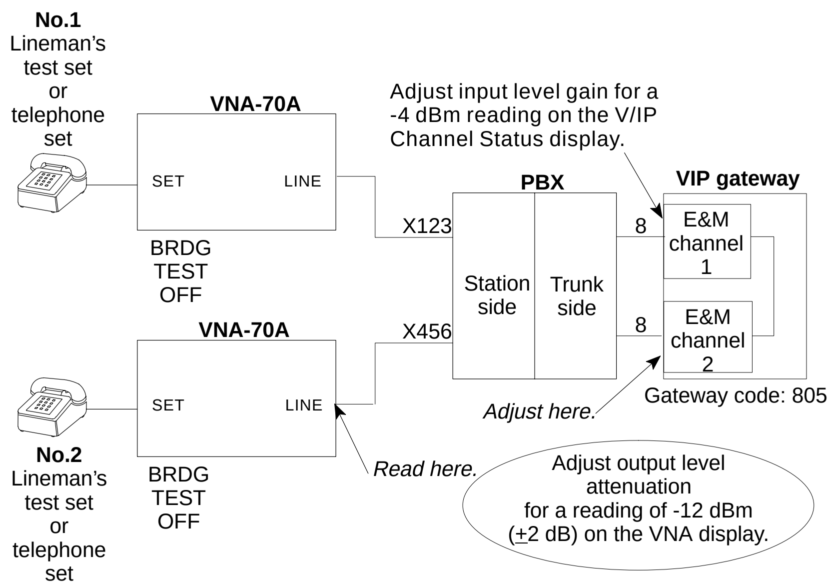
1. Use the test setup shown below. Initially, set both VNAs (Voice Network Analyzer) to BRDG, TEST and OFF. The extension numbers shown are fictitious, used here for reference only.
2. Lift the CALLING telephone set off-hook, wait for the dial tone, and dial station 5678. The RECEIVING telephone will ring.
3. Place the SETUP switch on the RECEIVING VNA to TERM. The telephone will stop ringing.
4. Place the SETUP switch on the CALLING VNA to TONE and disconnect the CALLING telephone from the VNA.
5. Observe the displayed value on the receiving VNA. It should read between -5 dBm and -8 dBm. If the level is outside of this range, the PBX should be checked for proper operation by authorized service personnel.



PBX Tie Trunk Application Adjustments

Many PBXs have selectable input/output level combinations for tie trunks, e.g., +7/-16, -16/+7 and 0/0 dB. The 0/0 dB level option should be selected. Also, PAD (2 dB) switching option, if available, should be selected.

1. Use the test setup shown below. Initially, set both VNAs to BRDG, TEST and OFF. The extension numbers shown are fictitious, used here for reference only.



2. Select the first two E&M voice channels within the V/IP gateway to be adjusted. Set any additional voice channels to the busyout mode to insure that the test calls operate only through the channels under test.
3. Place a call from the No.1 telephone to the No.2 telephone. To do that, lift the No.1 telephone off-hook, wait for the dial tone, and dial the tie trunk access code (typically 8). A second dial tone should be heard from the V/IP gatway.
4. Dial the gateway code and the channel number of the other E&M channel. In this example, you would dial 8052. A third dial tone should be heard from the PBX.

5. Dial extension 456. The No.2 telephone set will ring.
6. Place the SETUP switch on the No.2 VNA to TERM. The telephone will stop ringing.
7. Place the SETUP switch on the No.1 VNA to TONE and disconnect the No.1 telephone from the VNA.
8. Enter the Channel Status display:

1

```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2

```
MODULE SELECTION
Analog Channel Status
Digital Voice (DVC)
T1/E1
RSVP
Statistics
Phone Directory
SNMP
Done
```

3

```
CARD – PORT
CARD # 01 – PORT # 01
CARD # 01 – PORT # 02
CARD # 02 – PORT # 01
CARD # 02 – PORT # 02
```

```
CHANNEL STATUS
CHANNEL ......................        CARD 01 – PORT 01
RATE .........................        8000 BPS(G.729)
STATE ........................        IDLE
DSP SOFTWARE REVISION .....           2249EnM
TEST MODE ...................         NONE
TEST STATUS .................         NORMAL/PASSED
INPUT LEVEL .................         -4
MODE .........................        VOICE CONNECTION
```

9. Observe the display on the PC screen and note the Input Level. If the reading is more positive than -4 dBm, apply negative gain (input attenuation) in the amount that this reading is above -4 dBm.

Example: The reading is -1 dBm.

The Input Level Gain for this channel is set to -2 dB.

Add three increments of negative gain by changing the setting to -5 dB.

To adjust the input level, enter the Voice/Fax Configuration menu:

1

```
  V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
 Control
 Exit
```

2

```
 MODULE SELECTION
 Analog Voice/Fax
 Digital Voice (DVC)
 T1/E1
 RSVP
 SNMP
 V/IP System
 Hardware
 Done
```

3

```
      CARD – PORT
 CARD # 01 – PORT # 01
 CARD # 01 – PORT # 02
 CARD # 02 – PORT # 01
 CARD # 02 – PORT # 02
```

4

```
                  VOICE/FAX CONFIGURATION – E&M
 Channel Number                         [1]
 UDP Port Number                        [65534]
 Mode                                   [Voice/Fax]
 Input Level Gain                       [ 0 dB]
 Output Level Attenuation               [ 0 dB]
 Busyout Mode                           [SYSTEM CONTROLLED]
 Background                             [REGENERATED]
```

After the input level gain adjustment is completed, note the setting used to bring the input level to -4 dBm. You will need it for comparison when setting other E&M channels.

10. Move to the voice/fax Output Level Attenuation parameter for E&M channel No.2 (the channel that is connected to the channel just adjusted):

1
```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2
```
MODULE SELECTION
Analog Voice/Fax
Digital Voice (DVC)
T1/E1
RSVP
SNMP
V/IP System
Hardware
Done
```

3
```
CARD – PORT
CARD # 01 – PORT # 01
CARD # 01 – PORT # 02
CARD # 02 – PORT # 01
CARD # 02 – PORT # 02
```

4
```
VOICE/FAX CONFIGURATION – E&M
Channel Number                        [2]
UDP Port Number                       [65534]
Mode                                  [Voice/Fax]
Input Level Gain                      [ 0 dB]
Output Level Attenuation              [ 0 dB]
Busyout Mode                          [SYSTEM CONTROLLED]
Background                            [REGENERATED]
```

11. Read the value of the receive level on the No.2 VNA display. The receive level should be approximately -12 dBm (±2 dB).

    You have now established the input level gain setting on the No.1 channel and the output level attenuation setting on the No.2 channel. Next, adjust the input level gain setting on the No.2 channel and the output level attenuation setting on the No.1 channel.

12. Reconnect the No.1 telephone set to its VNA and take it off-hook.

13. Set the SETUP switch on No.1 VNA to TERM, then disconnect No.1 telephone.

14. Take the No.2 telephone off-hook.

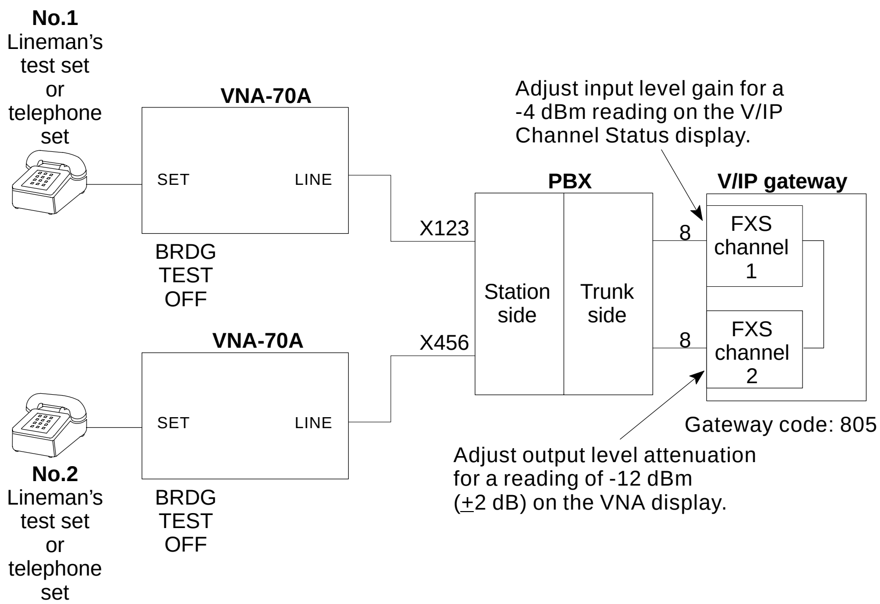
15. Place the SETUP switch on the No.2 VNA to TONE.
16. Disconnect the No.2 telephone from the VNA.
17. Enter the Channel Status display as in step 8, but for channel number 2.
18. Observe the display on the PC screen and note the Input Level Display. If the reading is more positive than -4 dBm, apply attenuation (negative gain) in the amount that this reading is above -4 dBm.

    To adjust the input level, enter the Voice/Fax Configuration menu for the appropriate channel, as in step 9. You will find that the Input Level Gain setting will need to be adjusted similar to that established in step 10.
19. Move to the voice/fax Output Level Attenuation parameter for E&M channel No.1, as in step 10.
20. Read the value of level on the No.1 VNA display. The receive level should be approximately -12 dBm (±2 dB).

    Now that both channels are adjusted, test the voice quality. To do that, place both VNA SETUP switches to BRDG, take one telephone off-hook, dial access code 8, then dial the gateway code and channel number of the opposite channel, and then dial the extension number of the other phone. Talk to verify that there are no echoes. If there are echoes, the output attenuation of the distant channel is too low. You must increase the output attenuation setting of the distant channel.
21. Adjust the input/output levels of the remaining E&M channels at each V/IP gateway, two at a time. The input gain and output attenuation settings and the output level reading should be within 1 dB of the original pair. If any channel exhibits a greater deviation, check the Voice Interface Card channel, the cabling between the channel and the PBX, and the PBX circuit.
22. Clear all connections and restore all channels to System Controlled Busy.
23. Remove any and all test equipment, such as VNAs and telephone sets, and restore the system to normal service.

PBX Central Office Trunk Application Adjustments

If your application involves Central Office trunks connecting to Voice Interface Card channels, the channels will be configured with FXS rather than E&M type interfaces. In all other respects the adjustment procedure is the same as that given for the PBX tie trunk. Here are the test setup details:

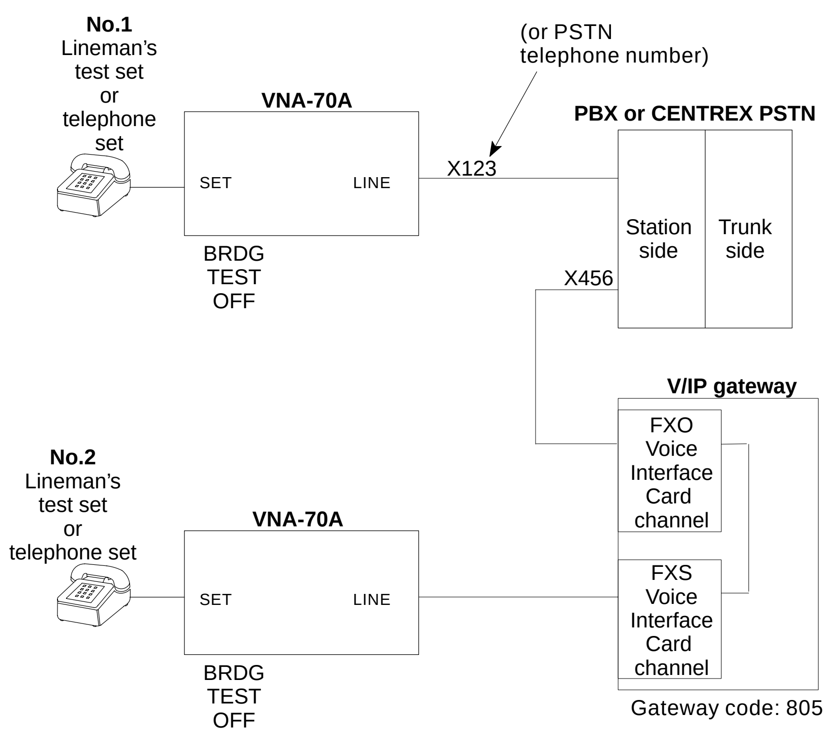


PBX Station or CENTREX PSTN Line Application Adjustments

For this procedure, one interface will have to be temporarily converted for FXS operation. To do that, use a spare FXS interface Voice Interface Card, if available, as listed in the Equipment Required paragraph. If none is available, the local FXO card will have to be connected to a companion FXS card across the network. In that case, two technicians will be required to perform the adjustment procedure, one at each end.

Note: If there are no single-line telephone circuits available from the PBX or CENTREX/PSTN, then the line assigned to the channel that is temporarily converted to FXS can be used for test purposes.

1. Use the test setup shown below. Initially, set both VNAs to BRDG, TEST, and OFF. The extension numbers shown are fictitious, used here for reference only.



2. Place a call from PBX extension 123 to extension 456. To do that, lift the No. 1 telephone off-hook, wait for the dial tone from the PBX, then dial extension number 456. (In CENTREX PSTN applications, dial the published PSTN number.)
3. You should hear the dial tone from the V/IP gateway. Dial the gateway code and FXS channel number. The No. 2 telephone set will ring.
4. Place the SETUP switch on the No.2 VNA to TERM. The telephone will stop ringing.
5. Place the SETUP switch on the No.1 VNA to TONE and disconnect the No.1 telephone from the VNA.
6. Enter the Channel Status display:

1

```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2

```
MODULE SELECTION
Analog Channel Status
Digital Voice (DVC)
T1/E1
RSVP
Statistics
Phone Directory
SNMP
Done
```

3

```
CARD – PORT
CARD # 01 – PORT # 01
CARD # 01 – PORT # 02
CARD # 02 – PORT # 01
CARD # 02 – PORT # 02
```

```
CHANNEL STATUS
CHANNEL .......................    CARD 01 – PORT 01
RATE ..........................    8000 BPS(G.729)
STATE .........................    IDLE
DSP SOFTWARE REVISION .....        2249EnM
TEST MODE ...................      NONE
TEST STATUS .................      NORMAL/PASSED
INPUT LEVEL ................       -4
MODE ..........................    VOICE CONNECTION
```

7. Observe the display on the PC screen and note the Input Level. If the reading is more positive than -4 dBm, apply attenuation (negative gain) in the amount that this reading is above -4 dBm. If the reading is more negative than -4 dBm, add gain (6 dB maximum) in the amount that the reading is below -4 dBm. Normally, station circuits require an Input Level Gain setting of between +2 and +4 dB in order to obtain the -4 dBm reading. In most cases, you will not have to use a negative Input Level Gain setting (-1 dB or lower). CENTREX or PSTN lines normally require an Input Level Gain setting of +2 to +6 dB.

   To adjust the input level for the FXO channel, enter the Voice/Fax Configuration menu:

1

```
   V/IP SERVER MAIN MENU
 View Configuration
 View Status/Statistics
 Modify Configuration
 Control
 Exit
```

2

```
   MODULE SELECTION
 Analog Voice/Fax
 Digital Voice (DVC)
 T1/E1
 RSVP
 SNMP
 V/IP System
 Hardware
 Done
```

3

```
        CARD – PORT
 CARD # 01 – PORT # 01
 CARD # 01 – PORT # 02
 CARD # 02 – PORT # 01
 CARD # 02 – PORT # 02
```

4

```
                 VOICE/FAX CONFIGURATION – FXO
 Channel Number                        [2]
 UDP Port Number                       [65534]
 Mode                                  [Voice/Fax]
 Input Level Gain                      [ 0 dB]
 Output Level Attenuation              [ 0 dB]
 Busyout Mode                          [SYSTEM CONTROLLED]
 Background                            [REGENERATED]
```

After the input level gain adjustment is completed, note the setting used to bring the input level to -4 dBm. You will need it for comparison when setting other FXO channels.

8. Read the value of the receive level on the No.2 VNA display. Note down this value for comparison when adjusting other FXO channels.

   You have now established the input level gain setting on the FXO channel. In the following steps, you will adjust the output level attenuation setting on the FXO channel.

9. Reconnect the No.1 telephone set to its VNA.
10. Set the SETUP switch on both VNAs to BRDG.
11. Take the No.2 telephone set off-hook. You will receive a dial tone from the V/IP gateway. Dial the gateway code and the first channel's number. You will receive a dial tone from the PBX across the connected channels. Dial extension 123 (or PSTN telephone number). The No.1 telephone set will ring.
12. Place the SETUP switch on the No.1 VNA to TERM. The telephone will stop ringing.
13. Place the SETUP switch on the No.2 VNA to TONE and disconnect the No.2 telephone from the VNA.
14. Enter the Channel Status display as in step 6.
15. Observe the display on the PC screen and note the input level of the FXS channel. If the reading is more positive than -4 dBm, apply attenuation (negative gain) in the amount that this reading is above -4 dBm. If the reading is more negative than -4 dBm, add gain (6 dB maximum) in the amount that the reading is below -4 dBm. To adjust the input level, enter the Voice/Fax Configuration menu for the appropriate channel number, as in step 7.

16. Read the value of the receive level on the No. 1 VNA display. The receive level should be -12 dBm or lower. In this case there are no adjustments to be made on the output level attenuation, since the 0 dB setting (no attenuation) is the maximum obtainable level. If it is necessary to adjust the output attenuation, follow these steps:

1

```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2

```
MODULE SELECTION
Analog Voice/Fax
Digital Voice (DVC)
T1/E1
RSVP
SNMP
V/IP System
Hardware
Done
```

3

```
CARD – PORT
CARD # 01 – PORT # 01
CARD # 01 – PORT # 02
CARD # 02 – PORT # 01
CARD # 02 – PORT # 02
```

4

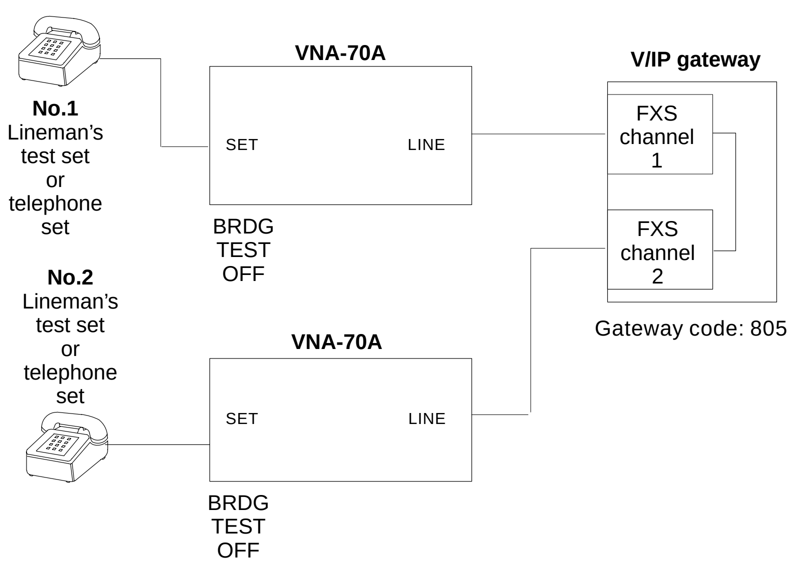
```
VOICE/FAX CONFIGURATION – FXO
Channel Number                  [2]
UDP Port Number                 [65534]
Mode                            [Voice/Fax]
Input Level Gain                [ 0 dB]
Output Level Attenuation        [ 0 dB]
Busyout Mode                    [SYSTEM CONTROLLED]
Background                      [REGENERATED]
```

Set the output level attenuation until the Ch.1 receive level is approximately -12 dBm ($\pm$2 dB).

17. Now that both channels are adjusted, test the voice quality. To do that, place both VNA SETUP switches to BRDG, take the No. 2 telephone off-hook, wait for the dial tone, dial the gateway code and FXO channel number, and dial extension 123. Talk to verify that there are no echoes. If there are echoes, the output level attenuation of the FXO channel is too low. You must increase the output attenuation setting of the FXO channel.
18. Adjust the remaining FXO channels at your installation by connecting them with the FXS channel used in this procedure. The input level gain and output level attenuation settings should be within 1 dB of this channel. If any FXO channel exhibits a greater deviation, check the channel, the cabling between the channel and the PBX, and the PBX circuit.
19. Restore the FXS card to an FXO one; it was used temporarily as a companion device with the FXO channel under test. Clear all local connects and restore all channels to System Controlled Busy. Remove any test equipment installed and restore the system to normal service.

Single Line Telephone Application

1. Use the test setup shown below. Initially, set both VNAs to BRDG, TEST, and OFF.



2. Lift the No.1 telephone set off-hook. Dial the gateway code and the No.2 channel number. The No.2 telephone set will ring.

3. Place the SETUP switch on the No.2 VNA to TERM. The telephone will stop ringing.

4. Place the SETUP switch on the No.1 VNA to TONE and disconnect the No.1 telephone from the VNA.

5. Enter the Channel Status display:

```
   V/IP SERVER MAIN MENU
   View Configuration
1  View Status/Statistics
   Modify Configuration
   Control
   Exit
```

```
   MODULE SELECTION
2  Analog Channel Status
   Digital Voice (DVC)
   T1/E1
   RSVP
   Statistics
   Phone Directory
   SNMP
   Done
```

```
   CARD – PORT
3  CARD # 01 – PORT # 01
   CARD # 01 – PORT # 02
   CARD # 02 – PORT # 01
   CARD # 02 – PORT # 02
```

```
CHANNEL STATUS
CHANNEL ......................     CARD 01 – PORT 01
RATE .........................     8000 BPS(G.729)
STATE ........................     IDLE
DSP SOFTWARE REVISION .....        2249EnM
TEST MODE ..................       NONE
TEST STATUS ................       NORMAL/PASSED
INPUT LEVEL ................       -4
MODE .........................     VOICE CONNECTION
```

6. Observe the display on the PC screen and note the Input Level. If the reading is more positive than -4 dBm, apply attenuation (negative gain) in the amount that this reading is above -4 dBm.

   Normally, this will require the Input Level Gain to be set between -2 and -4 dBm depending on cabling loss. (Note: if the Input Level Gain is currently set for 0 and the input level is below -10 dBm, then the FXS channel should be evaluated for a problem.)

To adjust the input level for the No.1 FXS channel, enter the Voice/Fax Configuration menu as in step 7 of the PBX Station Application procedure.

7. After the input level gain adjustment is completed, note the setting used to bring the input level to -4 dBm. You will need it for comparision when setting other FXS channels.

8. Move to the voice/fax Output Level Attenuation parameter for the FXS channel No.2:

1

```
V/IP SERVER MAIN MENU
View Configuration
View Status/Statistics
Modify Configuration
Control
Exit
```

2

```
MODULE SELECTION
Analog Voice/Fax
Digital Voice (DVC)
T1/E1
RSVP
SNMP
V/IP System
Hardware
Done
```

3

```
CARD – PORT
CARD # 01 – PORT # 01
CARD # 01 – PORT # 02
CARD # 02 – PORT # 01
CARD # 02 – PORT # 02
```

4

```
VOICE/FAX CONFIGURATION – FXS
Channel Number                [2]
UDP Port Number               [65534]
Mode                          [Voice/Fax]
Input Level Gain              [ 0 dB]
Output Level Attenuation      [ 0 dB]
Busyout Mode                  [SYSTEM CONTROLLED]
Background                    [REGENERATED]
```

9. Read the value of the receive level on the No.2 VNA display. The receive level should fall between -6 and -8 dBm. Set the output attenuation until the receive level display is -8 dBm. If the receive level is below (more negative than) -8 dBm, reduce the output attenuation until no further adjustment is possible (0 attenuation). Note the setting. You will need it for comparison when setting other FXS channels.
10. You have now established the input level gain setting on the No.1 channel and the output level attenuation setting on the No.2 channel. Next, adjust the input level gain setting on the No.2 channel and the output level attenuation setting on the No.1 channel.
11. Reconnect the No.1 telephone set to its VNA.
12. Set the SETUP switch on both VNAs to BRDG.
13. Take the No.2 telephone set off-hook. Dial the gateway code and the No.1 channel number. The No.1 telephone set will ring.
14. Place the SETUP switch on the No.1 VNA to TERM. The telephone will stop ringing.
15. Place the SETUP switch on the No.2 VNA to TONE and disconnect the No.2 telephone from the VNA.
16. Enter the Channel Status display as in step 5.
17. Observe the display on the PC screen and note the Input Level Display. If the reading is more positive than -4 dBm, apply attenuation (negative gain) in the amount that this reading is more positive than -4 dBm.

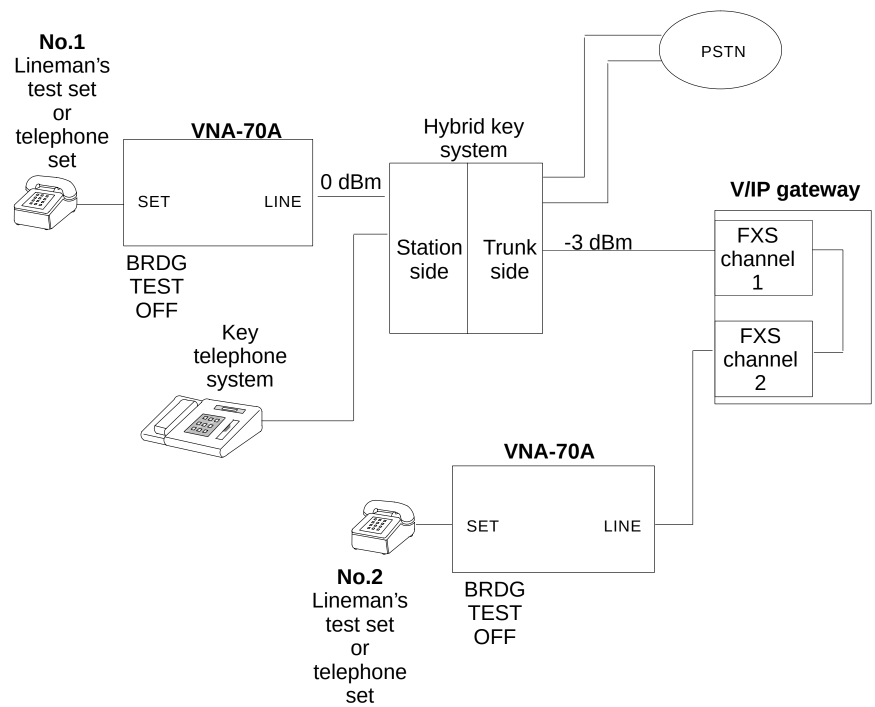
    To adjust the input level, follow the procedure of step 6 above. You will find that the setting is similar to that established in step 7.
18. Move to the voice/fax Output Level Attenuation parameter for FXS channel No.1, as in step 8 above.
19. Read the value of the receive level on the No.1 VNA display. The receive level should fall between -6 and -8 dBm. Set the output attenuation until the receive level display is -8 dBm. If the receive level is below (more negative than) -8 dBm, reduce the output attenuation until no further adjustment is possible (0 attenuation).

20. Now that both channels are adjusted, test the voice quality. To do that, place both VNA SETUP switches to BRDG, take one telephone off-hook, dial the gateway code and appropriate channel number, and lift the other telephone off-hook when it rings. Talk to verify that there are no echoes. If there are echoes, increase the output level attenuation setting of the distant channel.
21. Adjust the input/output levels of the remaining FXS channels at your location, two at a time, in pairs. The input level gain and output level attenuation settings should be within 1 dB of the original pair. If any channel exhibits a greater deviation, check the channel and the cabling, and, if applicable, the key telephone system.
22. Clear all connections and restore all channels to System Controlled Busy.
23. Remove any and all test equipment, such as VNAs and telephone sets, and restore the system to normal service.

Hybrid Key System Adjustment

If your installation includes a hybrid key system instead of a single-line telephone, there may be a loss of up to 3 dB across that system in each direction. To compensate for that loss in any application, you will have to adjust the input/output levels accordingly.

In the test setup shown below, you will need only 1 dB attenuation on channel No.1 for an input level value of -4 dBm. Similarly, when adjusting the output level on channel No.1, you will most likely require little or no attenuation in order to maintain the output level at -12 dBm. In all other respects, this adjustment procedure is the same as the single-line telephone adjustment procedure described in the previous paragraph.



Glossary

busyout

A configuration option that places a voice/fax channel into a busy state, effectively disabling the channel.

ClearVoice Technology

A group of MICOM voice technologies centered around the ITU G.729 voice algorithm. The technologies include voice switching, fax/modem demodulation, silence suppression, and background noise regeneration. *See* G.729.

DID, Direct Inward Dialing

A service offered by the telephone company. DID allows an outside caller to dial an internal extension without operator intervention. Billing does not start until the extension answers.

DISA, Direct Inward System Access

A service provided by a PBX that allows incoming calls to the PBX to have dialing access within the private network. This access can be protected by a dialed password. Billing starts at the time when the PBX provides the dial tone.

disconnect supervision

A voice communications protocol that indicates to the local user that the remote end has gone on-hook. This protocol is significant in loop start, where disconnect is denoted by removal of power to the station equipment.

DIT, Direct In Termination

A service provided by a PBX that allows incoming calls to the PBX to be routed directly to a selected telephone or group of telephones without operator intervention. Billing does not start until the telephone answers.

DTMF, Dual Tone MultiFrequency

A system of audio signal combinations used for call addressing in pushbutton telephones. Also known as MultiFrequency Push-Button (MFPB).

E&M, Ear and Mouth

A signaling convention between voice PBXs to set up and tear down calls.

fax, facsimile

Transmission of images (written, typed, or drawn material) through telephone lines.

FXS, foreign exchange station

Telephone extension that has been extended off-premises. The end of the circuit that connects to the subscriber's foreign exchange station; the other end of the circuit is called foreign exchange office (at the office).

G.729

A voice compression algorithm developed for transporting high quality voice at 8 Kbps. Also defined as the Conjugate-Structured Algebraic-Code-Excited Linear-Predictive Algorithm.

gateway

A server that can access two or more different networks.

IP address

See network address.

jumper

A miniature connector that fits over and electrically connects two pins.

key telephone system

A system in which the telephones have multiple pushbuttons to allow users to select outgoing/incoming calls directly. Also known as FXS.

latency

The amount of time it takes for a discrete event to occur.

link

A communications circuit or transmission path connecting multiple points in a network.

loopback

A diagnostic function used to test a voice/fax channel. The transmit line is looped back to the receive line at some point either inside the local device, along the phone line, or inside the remote device.

MIB, Management Information Base

A tree-structured database of management information stored within the memory of a network device. The database can be accessed to monitor device operation and to change the device's configuration.

NetWare

A Novell-developed Network Operating System. NetWare provides file and printer sharing among networks of personal computers. Each NetWare network must have at least one file server. Access to other resources is dependent on connecting to and logging into the file server.

network address

Every node in a network has one or more addresses associated with it.

Every node has what is called a *hardware address* that is unique across every network everywhere, at any time. If you know a node's hardware address, you should be able to identify the exact piece of equipment it goes with. Hardware addresses are generally setup by the company that manufactured the equipment and should never change. This address is usually specified as a set of six hexadecimal digits separated by dashes, such as `04-34-2c-1d-96-f1`.

In the case of TCP/IP networks, each node also has a *software* or *IP* address. This address is configurable by the network administrator of the nodes. The software address is usually specified as four decimal numbers separated by periods (for example, 192.30.18.11). Each number must be between 0 and 255. The network or sub-network portion of the address varies, depending on the class of address and the subnetting established for the network.

node

Any intelligent device physically connected to the network.

off-hook

A line condition caused when a telephone handset is removed from its cradle.

on-hook

A line condition that exists when a telephone handset is resting in its cradle.

PBX, private branch exchange

A privately owned phone system installed within the premises of an organization. It allows communication among users within the organization, as well as between those users and the outside world. It differs from a key telephone system in that the user must dial an access number (such as 9) to get an outside line.

PING, Packet InterNet Groper

A method of testing the accessibility of a destination by sending an echo request across the network (LAN or WAN) and then waiting for a reply.

pulse dialing

Method used for call addressing by rotary telephones that consists of short pulses of on-hook/off-hook. Also known as decadic pulsing.

router

A device that looks at a packet's destination address to determine which network is its destination. The router will then find the best path to use to send the packet across the network(s).

RSVP, Resource ReSerVation Protocol

A protocol that allows a host to request a specific Quality of Service (QoS) from the network to support an application's data stream. The request is carried across the network and an attempt is made to reserve the desired QoS at each visited node that the data stream must transit.

server

A network node that provides services to other nodes in the network.

signaling

A handshaking protocol used between telephone equipment. This includes supervising (off-hook/on-hook line status), alerting (ringing), and call addressing (dialing) for switched services.

SNMP, Simple Network Management Protocol

A widely used network management protocol that allows network administrators to monitor, troubleshoot, and control other SNMP-compliant devices attached to the network.

subnet

A means of splitting IP addresses into two fields to separate packets for local destinations from packets intended for remote destinations.

subnet mask

When looking at an IP packet, a router must decide whether the packet's destination is for a node on the local network, or whether the destination is a node on a remote network and must be accessed through a gateway. The router does this using the subnet mask configured by the network administrator. The router uses the subnet mask as a filter; if the router's IP address and the destination IP address appear the same after the subnet filter, the destination node is assumed to be on the same local network. Otherwise, the packet is sent to the gateway.

TCP/IP, Transmission Control Protocol/Internet Protocol

This is a network protocol set whose major components are IP, UDP, and TCP. TCP/IP support may be integral to a computer's operating system, as in UNIX, or it can be separate product.

IP is the low level protocol. IP provides packet delivery services between nodes.

UDP (User Datagram Protocol) is an "unreliable" connectionless protocol. "Unreliable" simply means that there is no verification that packets have reached their destination. However, the process is sufficient to allow an application on one node to communicate with a process on another machine. V/IP uses UDP for voice and fax packet transmission and administration between V/IP gateways.

TCP is a reliable stream-delivery, virtual circuit connection-oriented protocol that runs on top of IP. Usually included with TCP are telnet (a terminal emulation program that allows an operator session with a host computer) and FTP (a file transfer program).

2-wire/4-wire

In a 2-wire system, the same pair of wires is used for both transmit and receive of audio signals. In a 4-wire system, one pair of wires is used for transmit and the other pair for receive.

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MICOM welcomes your evaluation of this manual and any suggestions you may have. These help us to improve the quality and usefulness of our publications.

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	Excellent	Good	Fair	Poor
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Are the *installation instructions* effective?	❑	❑	❑	❑
Are the *operating instructions* clear and complete?	❑	❑	❑	❑
Is the manual properly *organized*?	❑	❑	❑	❑
Is the *artwork* clear and easy to understand?	❑	❑	❑	❑
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