V/IP Phone/Fax IP Gateway

Analog Models

Microsoft® Windows® 95

User's Manual

Part Number 800-1887-11, 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.

MICOM Communications Corp.
4100 Los Angeles Avenue
Simi Valley, California 93063-3397
U.S.A.
(805) 583-8600

MICOM Communications Corp. (Europe) Ltd.
The Granary
Grange Court
Grange Road
Tongham, Surrey GU10 1DW
England, UK
44 1252 781 777

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

Notice

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

Trademark Notice

MICOM®, Marathon®, NetRunner®, FrameRunner™, STADIA®, ClearVoice™, NETMan™, ®FEATUREPAK, FlashPak™, ™*MicroBand* ATM, Power Plus™, SNAPS™, MICOM®BOX, ™Val-U-Mux, V/IP™, , and MICOM*Vision*™ are trademarks or registered trademarks of MICOM Communications Corp.

Microsoft and Windows are registered trademarks of Microsoft Corporation.

RSVPD –– ReSerVation Protocol Daemon was developed in part by the University of Southern California, Information Sciences Institute and is copyright © 1996 by the University of Southern California.

All other names or trademarks are the true property of their respective companies.

Notification of FCC Requirements

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.

To the extent the equipment is or contains software or firmware (collectively "Software"), MICOM warrants that for a period of one (1) year from the date of shipment, the Software shall be free from defects in material and workmanship under normal use and that the programs will perform according to the specifications contained in MICOM's user manual. MICOM does not warrant that the functions contained in the Software will meet a specific requirement or that the operation will be uninterrupted or error free.

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.

SERVICE INFORMATION

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.

MICOM Communications Corp.
4100 Los Angeles Avenue
Simi Valley, CA 93063-3397
(805) 583-8600

Contents

Description Of Operation – 1

Installation and Initial Configuration – 2

Operation – 3

Administration – 4

Contents

Administration – 4 (cont'd)

Input/Output Level Adjustments – 5

Specifications – A

Contents

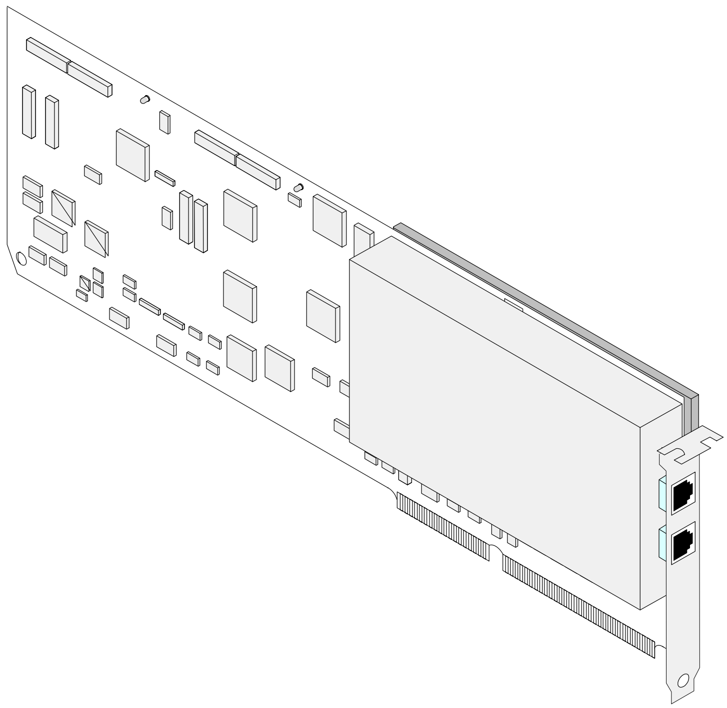
Analog Interface Descriptions – B

SNMP Management – C

Agency Requirements – D

Glossary

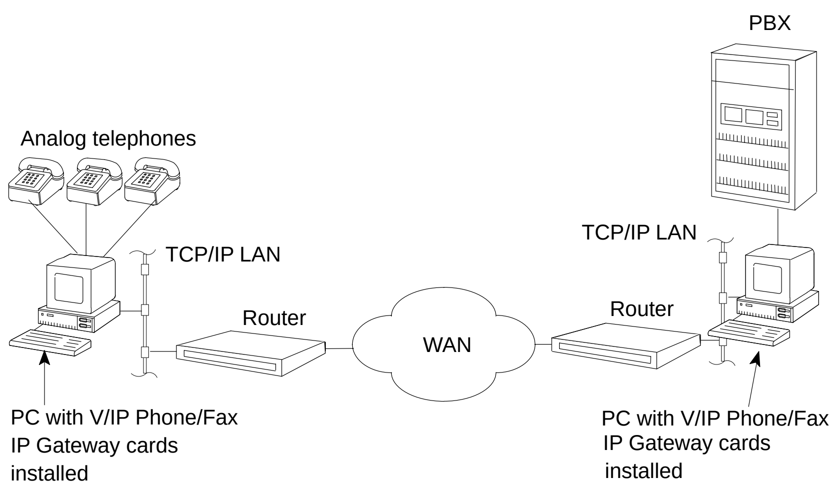
Index



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 IP 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 PCs.



This manual describes the V/IP Phone/Fax IP Gateway Analog Voice Interface Cards operating under Microsoft® Windows® 95. 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 Windows 95, NetWare, and DOS.
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 Windows 95, NetWare, and DOS.
VIP-2001-ISA-FXO	• Single channel, ISA bus, FXO analog interface card, model number VIP-1-ISA-FXO. • V/IP software for Windows 95, NetWare, and DOS.
VIP-2002-ISA-FXO	• Dual channel, ISA bus, FXO analog interface card, model number VIP-2-ISA-FXO. • V/IP software for Windows 95, NetWare, and DOS.
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 Windows 95, NetWare, and DOS.
VIP-2001-ISA-FXO-UK	• Single channel, ISA bus, FXO analog interface card, model number VIP-2-ISA-FXO, tested for BABT compliance. • V/IP software for Windows 95, NetWare, and DOS.

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 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.

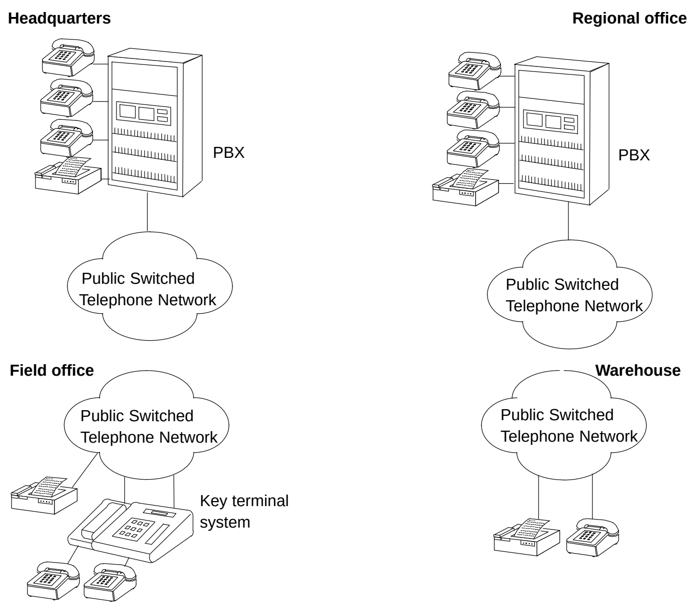
The cards use MICOM's ClearVoice technology to deliver the best quality voice using very little bandwidth. ClearVoice is based on the ITU-standard G.729 compression algorithm.

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

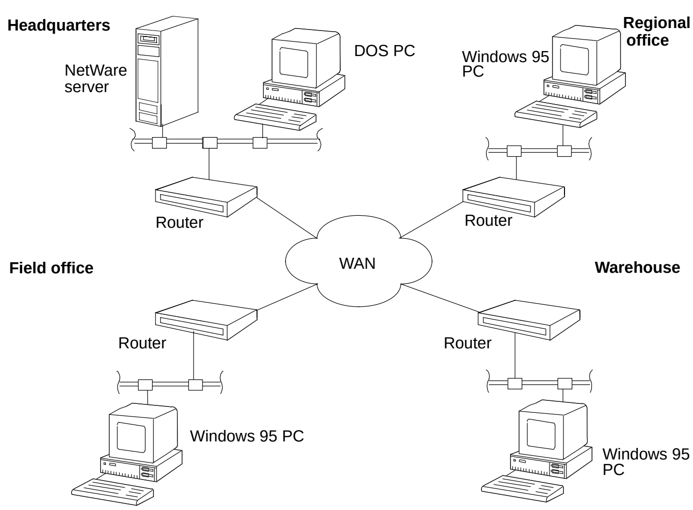
SNMP network management is supported, including GETs, GETNEXTs, SETs, and Traps.

Application and Benefit

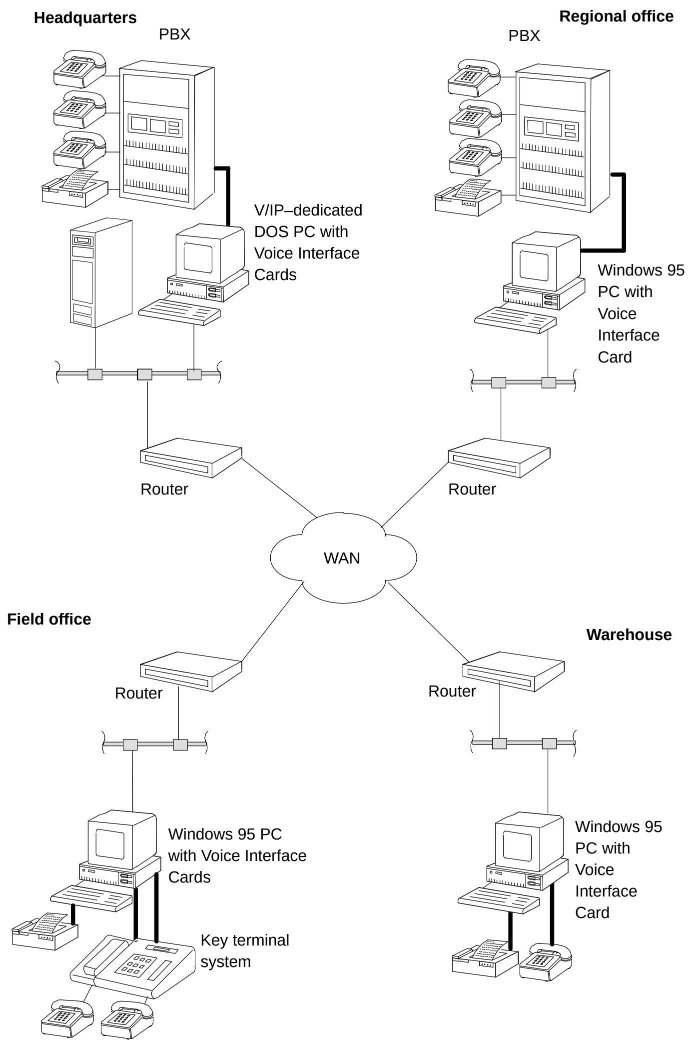
If your company's phone system is similar to this:



And your computer networks are interconnected, like this:



You can install Voice Interface Cards in your PCs 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.

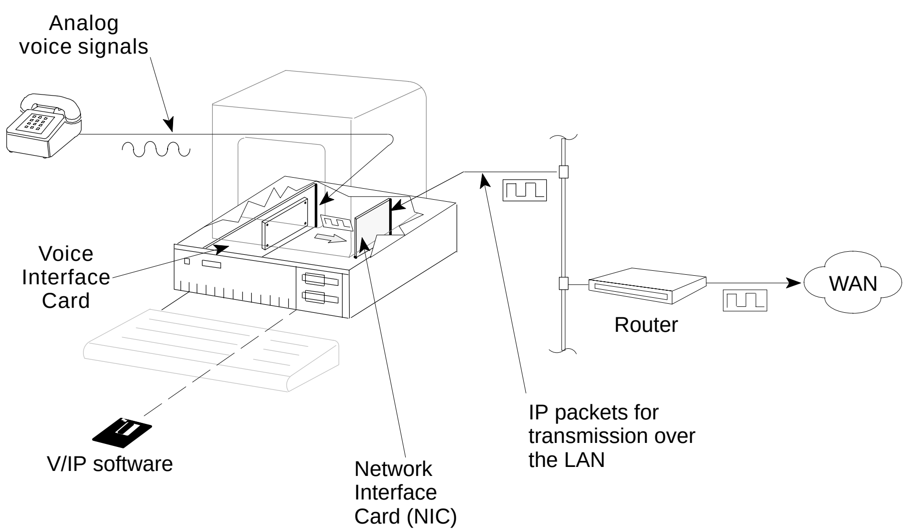
2. The V/IP system software handles the:
   – management of calls
   – encapsulation of digital voice into IP packets
   – phone directory database

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.

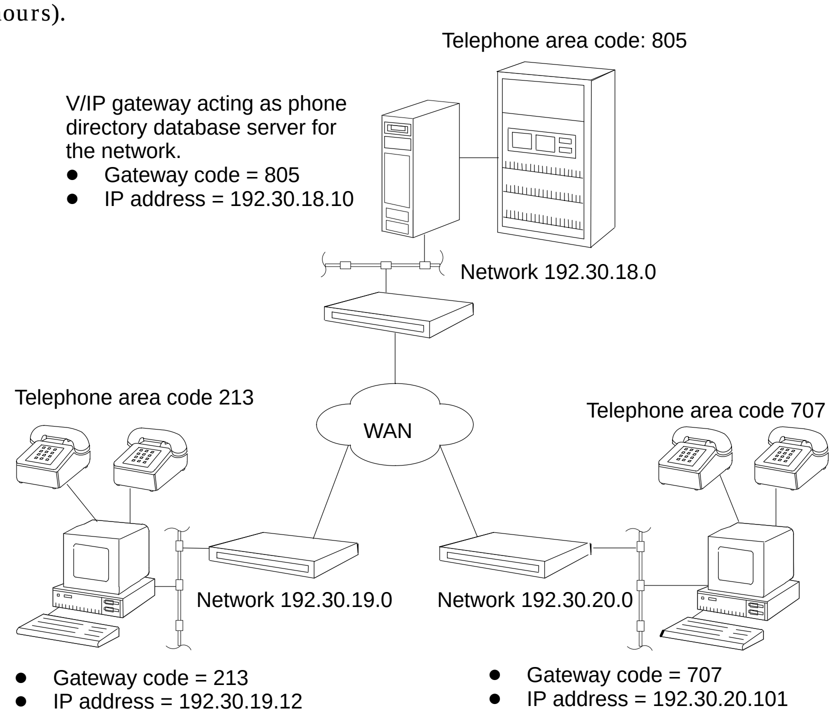
5. A router attached to the LAN forwards the IP packets across the WAN, where they will be received by another V/IP gateway PC at the remote end.

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).



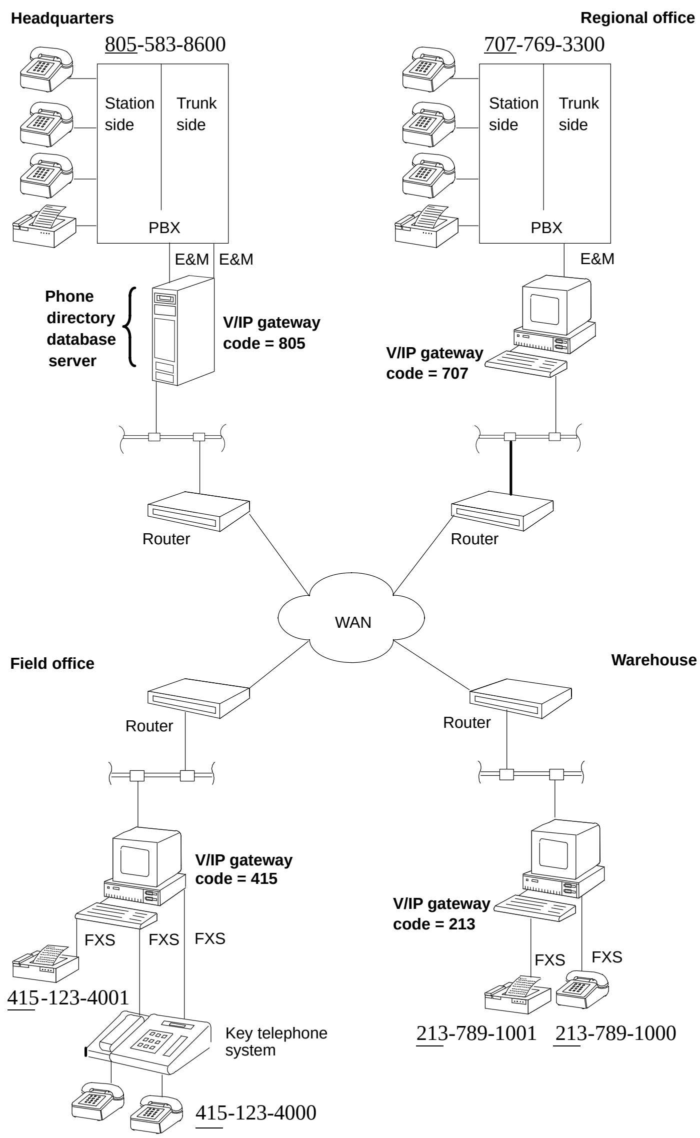
The phone directory database for the above network would look like this:

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:

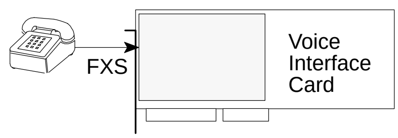
To call from . . .	using the public telephone network, dial . . .	using the V/IP network, dial . . .
Headquarters to the regional office	9 1 (707) 769-3300, extension 3302 You dial 9 for an outside line, then 1 707 769 3300 to get the PBX operator at the regional office and then ask for extension 3302.	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.

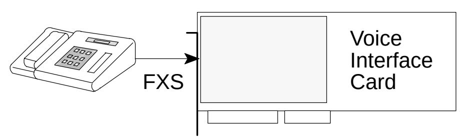
Telephone Interface Types

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



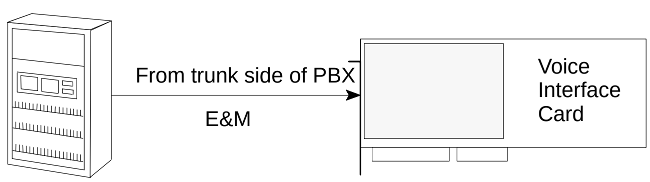
  - the line circuit of a key telephone system



  - 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.

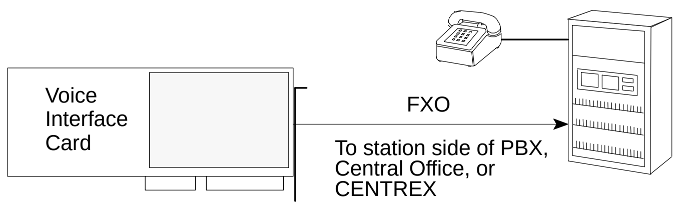
- E&M (Ear and Mouth) – a tie line trunk circuit used to connect between PBXs or other voice switching system.



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®
  - PBX stations

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

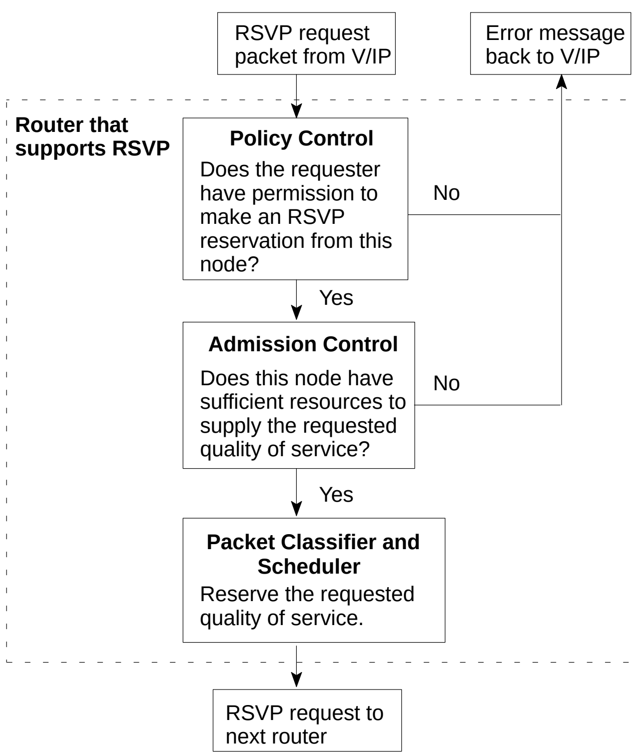


Resource ReSerVation Protocol (RSVP)

The V/IP gateway can use RSVP to request a specific Quality of Service (QoS) from the network to ensure the timely arrival of voice packets to their destination. This is particularly important at WAN data rates less than 128 Kbps and networks that experience saturation with data traffic regardless of speed. The available bandwidth can narrow to the point of delaying the transit of IP packets. For voice, these delays can reduce the quality of the voice heard at the remote end. With RSVP, a portion of the available bandwidth can be held in reserve for a V/IP call to maintain high quality voice.

RSVP will function only with those routers that support it. If any router along the route does not support RSVP, the RSVP packet will tunnel through and otherwise be transparent to that router.

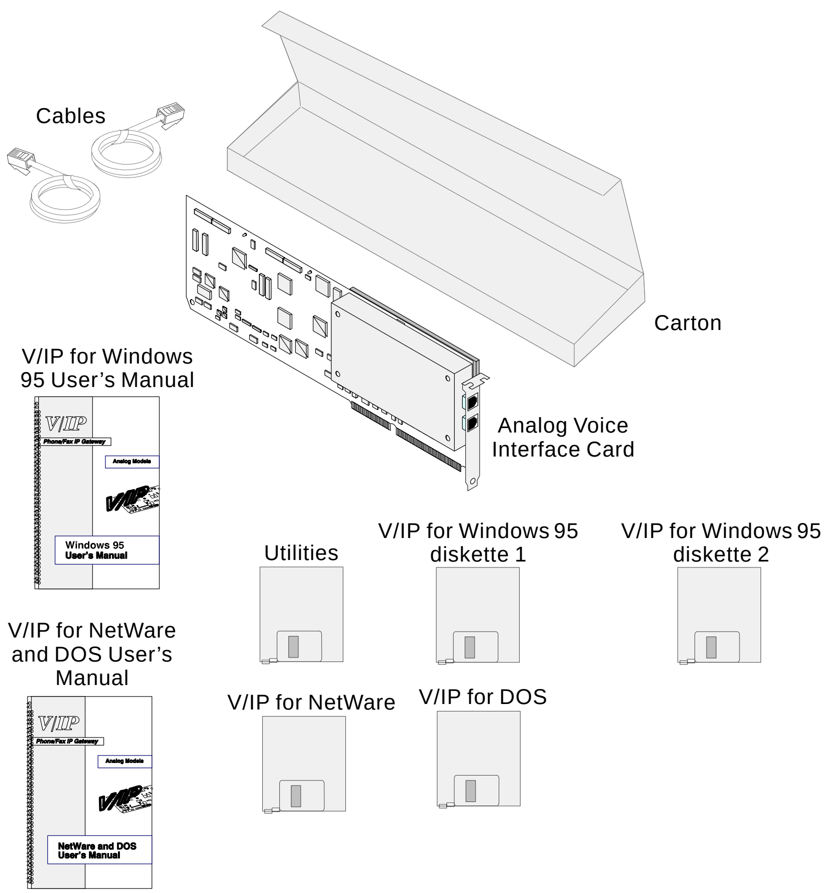
RSVP works like this:



Installation and Initial Configuration 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. There is also a file called `readme.txt` on the V/IP for Windows 95 Diskette 1. You should read this file as well for information specific to installing the V/IP software on a Windows 95 PC.

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 size and activity level of the network. However, the specific processor requirements for running V/IP for Windows 95 are as follows:

- processor type and speed: as specified for Windows 95
- $\leq$ 8 MHz ISA bus
- 8 MB minimum system memory, 16 MB recommended
- one full length, 16-bit ISA bus card slot per Voice Interface Card
- 5 MB available hard disk space
- 3½-inch diskette drive
- VGA display

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.

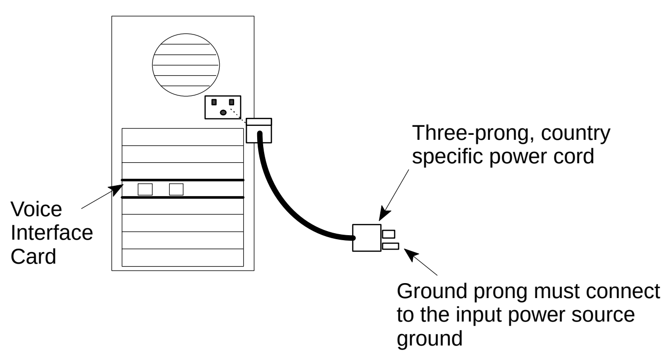
Caution

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

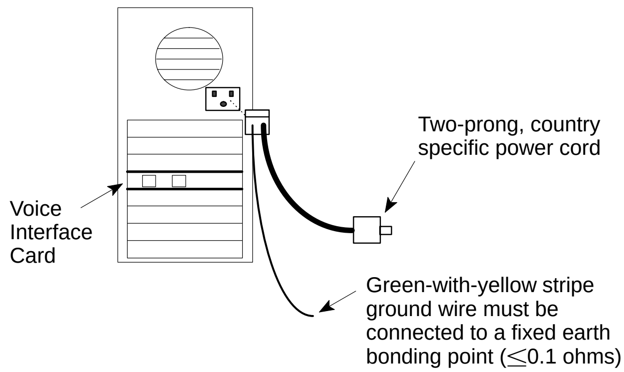
The following is special information about earth grounding that is required before connecting the card to the network.

WARNING

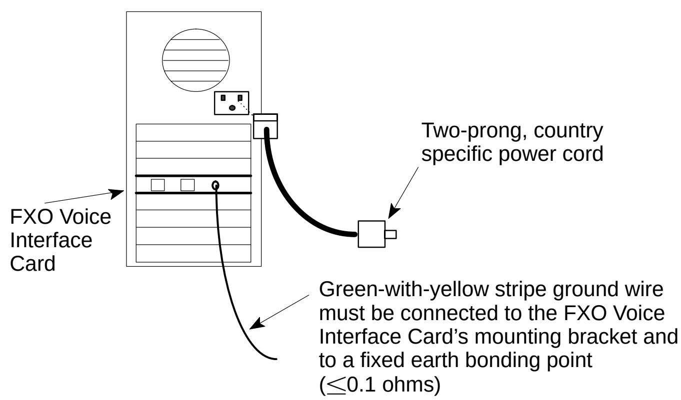
There must be a separate ground wire connection between the PC chassis and the input power source ground. A proper ground connection can be provided in one of three ways, as shown in the following three illustrations:



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 Voice Interface 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

Microsoft Windows 95

Network Interface Card (NIC)

You can use any Ethernet or Token Ring Network Interface Card supported by Windows 95. See the `readme.txt` file in the Utilities diskette for a list of Network Interface Cards that have been tested with V/IP.

Installation

This procedure will take you through the steps you must perform to get your Voice Interface Cards operating on your network. The installation sequence is:

- Planning Your V/IP Network
- Installing the Software
- Configuring and Installing the Voice Interface Cards
- Configuring the System Parameters
- Configuring the Voice/Fax Channels
- Configuring the Routing Priority

Planning Your V/IP Network

There is a first time V/IP gateway configuration that is required during the software installation. 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 numeric 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:	_____	Enter a value from 0 to 7. 0 defines all V/IP channels in this PC as one hunt group.
Maximum number of dial digits:	_____	Enter a value from 0 to 16. This is 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 numeric 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:	Channel 01:	Channel 02:
Card 02 I/O address:	Channel 01:	Channel 02:
Card 03 I/O address:	Channel 01:	Channel 02:
Card 04 I/O address:	Channel 01:	Channel 02:
Card 05 I/O address:	Channel 01:	Channel 02:
Card 06 I/O address:	Channel 01:	Channel 02:
Card 07 I/O address:	Channel 01:	Channel 02:
Card 08 I/O address:	Channel 01:	Channel 02:

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 numeric 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***	Enter a value from 0 to 7. 0 defines all V/IP channels in this PC as one hunt group.
Maximum number of dial digits:	***4***	Enter a value from 0 to 16. This is 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 numeric 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***	Channel 01: ***0***	Channel 02: ***0***
Card 02 I/O address: ***220***	Channel 01: ***0***	Channel 02: ***0***
Card 03 I/O address: ***300***	Channel 01: ***0***	Channel 02: ***0***
Card 04 I/O address: ***320***	Channel 01: ***0***	Channel 02: ***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 numeric 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***	Enter a value from 0 to 7. 0 defines all V/IP channels in this PC as one hunt group.
Maximum number of dial digits:	***0***	Enter a value from 0 to 16. This is 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 numeric 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***	Channel 01: ***4000***	Channel 02: ***4000***
Card 02 I/O address: ***220***	Channel 01: ***4001***	Channel 02:
Card 03 I/O address:	Channel 01:	Channel 02:
Card 04 I/O address:	Channel 01:	Channel 02:

Installing the Software

The following are the steps to install the software:

- Set the TCP/IP properties.
- Load Microsoft Winsock 2 (optional).
- Load Intel PC-RSVP Service Provider (optional).
- Load Microsoft SNMP Agent (optional).
- Load the V/IP software.
- Load the V/IP device driver.

Set the TCP/IP Properties

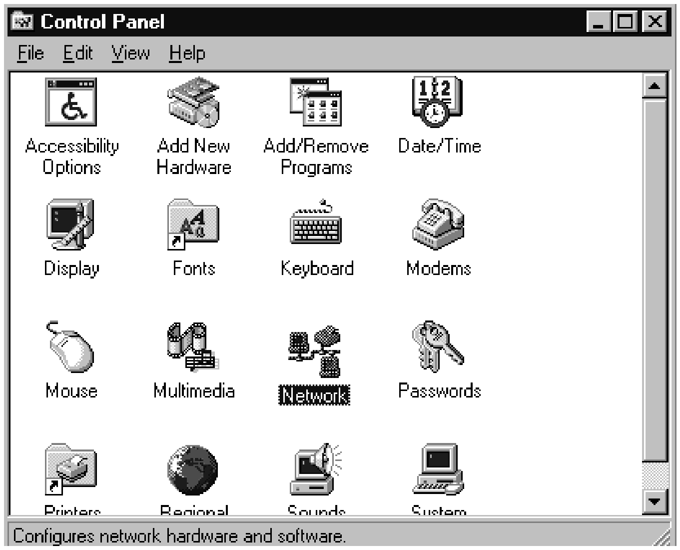
You must have the TCP/IP protocol installed in Windows 95. This is accomplished from Control Panel ➧ Network ➧ Add ➧ Protocol ➧ Microsoft ➧ TCP/IP. The PC will have to be restarted. The installing of the TCP/IP protocol also installs Winsock 1.1.

Now, you must enter the IP address of the PC and the IP address of the gateway/router that will route the V/IP LAN traffic to the network in the Windows 95 TCP/IP Properties Gateway tab, as follows:

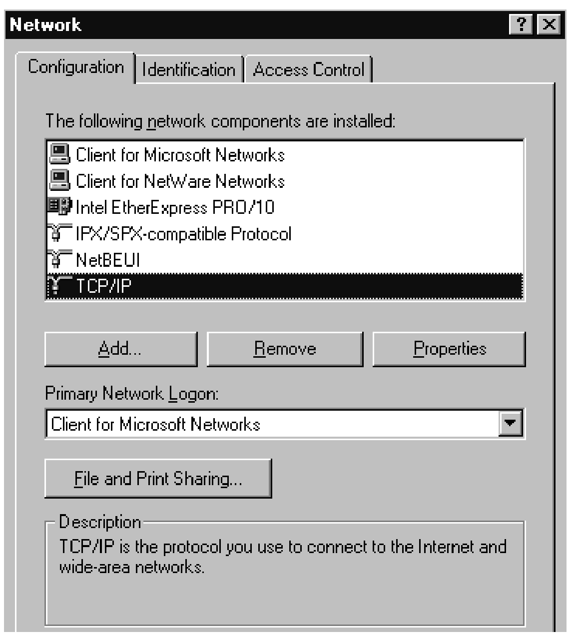
1. Double-click on the My Computer icon on the desktop.

2. Double-click on the Control Panel icon.

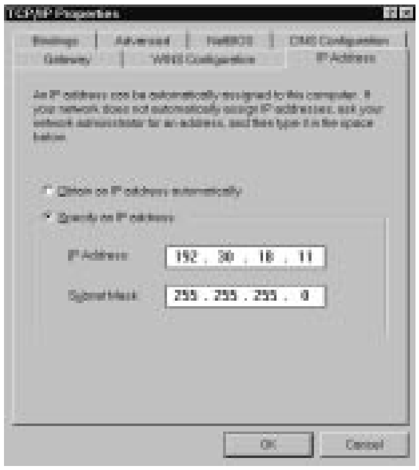
3. Double-click on the Network icon.



4. Select the Configuration tab and Double-click on TCP/IP.

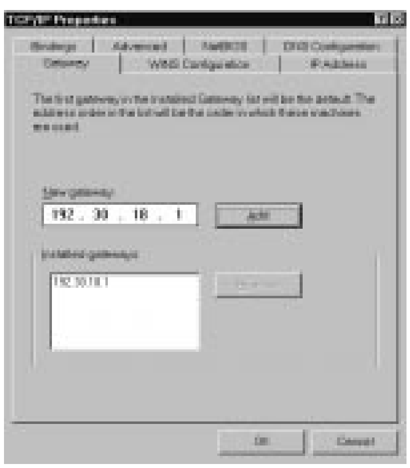


5. Select the IP Address tab and fill in the IP address and the subnet mask for this PC:



6. Select the Gateway tab and enter the IP address of the gateway/router in the New gateway field, click on the Add >> button, and click on OK to save the address. Windows 95 will have to be restarted.

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 in the gateway field.



7. Continue on to *Load Microsoft Winsock 2*.

Load Microsoft Winsock 2

This is an optional step. If you want to use the Resource ReServation Protocol (RSVP) to prioritize V/IP packets across the network, you must install Microsoft Winsock 2 at this time (before installing the V/IP software). You can get Winsock 2 from:

- MICOM's Web site, at:
  `http://www.micom.com/`
- Microsoft's Web site, at:
  `http://www.microsoft.com/win32dev/netwrk/winsock2/ws295sdk.html`

Continue on to *Load Intel PC-RSVP Service Provider*.

Load Intel PC-RSVP Service Provider

This is an optional step. If you want to use the Resource ReServation Protocol (RSVP) to prioritize V/IP packets across the network, you must install Intel PC-RSVP Service Provider at this time (before installing the V/IP software). See the `readme.txt` file in the V/IP for Windows 95 diskette 1 for information about where you can obtain this software.

Continue on to *Load Microsoft SNMP Agent*.

Load Microsoft SNMP Agent

This is an optional step. If you plan on using SNMP to manage the V/IP gateway PCs, you must install the Microsoft SNMP Agent at this time. You can get the latest version from:

- Microsoft's Web site, at:
  `http://www.microsoft.com/windows95/info/admintools.htm`

You will need to download the *SNMP Agent and related files*. Load the agent in accordance with the information provided in the text files included with the software. After installing the agent, Windows 95 will have to be restarted.

Continue on to *Load the V/IP Software*.

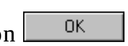
Load the V/IP Software

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

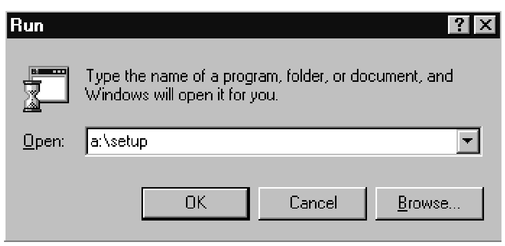
Important: If the PCs used as V/IP gateways are to be taken down for any reason, you must bring up the PC that is the V/IP phone directory database server first. The rest of the PCs will download the phone directory database from that PC when they load the V/IP software.

1. Insert the diskette labeled *V/IP For Windows 95 Diskette 1* into the `a:` drive of your PC.

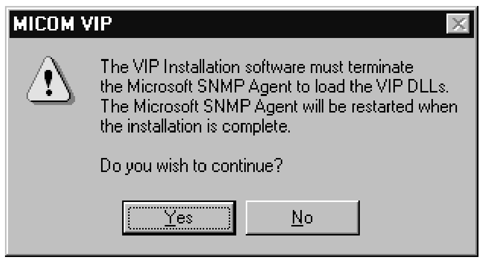
2. Click the Start button and click on Run.

3. Type `a:\setup` in the Run window and click on 

.



4. You will see the following message about SNMP. This means that the V/IP installation program needs to temporarily shut down the Microsoft SNMP Agent in order to install the V/IP SNMP extensions. Click on Yes .

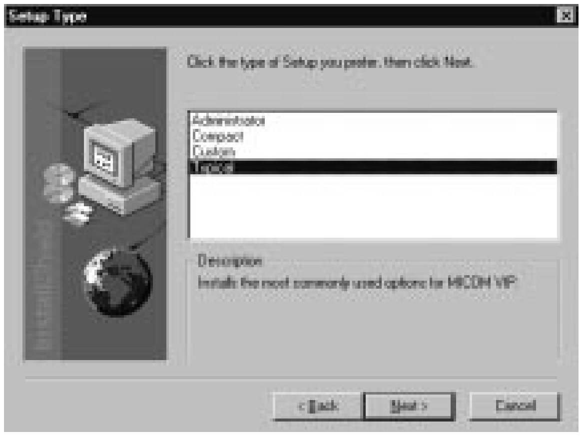


You will see a message that Setup is preparing the installation wizard. When this is complete, the V/IP Installation screen will appear:

5. Click on Next >. You will see the Choose Destination Location dialog box:

If the default destination for the V/IP software is acceptable to you, then click on Next >. Otherwise, change the destination to suit your requirements and click on Next >.

6. You will then see the Setup Type dialog box:

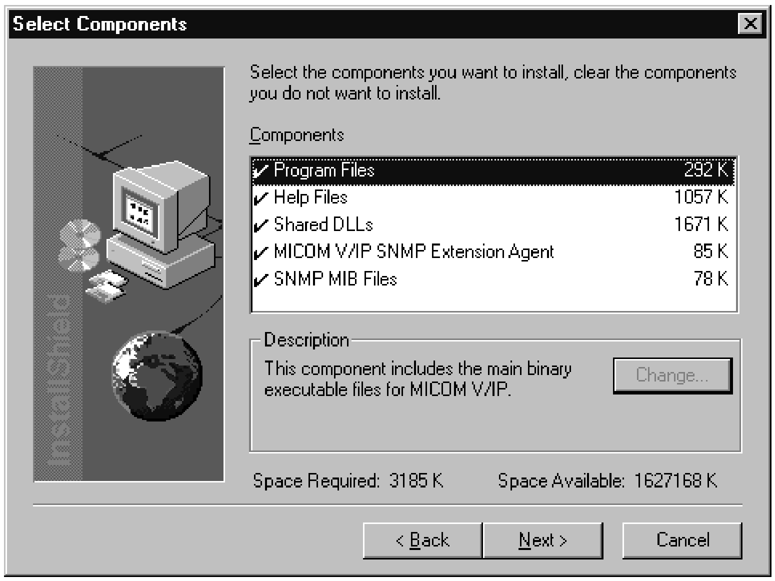


You can select from the following types:

- Administrator. Installs all files, which includes those needed to allow you to administer to other V/IP gateways on your network from this PC.
- Compact. This is a minimum installation. Just enough of the V/IP software is installed to make this PC into a V/IP gateway. You can use this type of setup for those PCs that have limited hard disk capacity.
- Custom. This will allow you to select the groups of files you want installed.
- Typical. This setup is appropriate for most PCs.

Select the setup type that you require and click on Next > .

If you selected the Custom setup type, you will see the Select Components dialog box:



The components you can select from are:

- Program Files. These files are required for V/IP.
- Help Files. These files provide the online help function for the V/IP Configuration program.
- Shared DLLs. These files are required for V/IP.
- MICOM V/IP SNMP Extension Agent. These files allow this V/IP gateway PC to be remotely administered from another computer. You must have installed the Microsoft SNMP Agent in this PC in order to use this feature.
- SNMP MIB Files. These are the enterprise MIB extensions that can be used by a network management system to manage the V/IP gateway PCs.

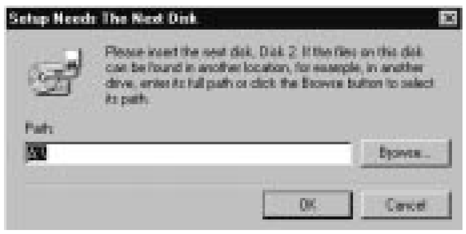
Select the components you require for this PC and click on 

.

7. The setup program will install the V/IP software in accordance with your selections to the previous dialog boxes.

   At some point in the setup, you will be prompted to insert the *V/IP For Windows 95 Diskette 2*. Insert the diskette and click on 

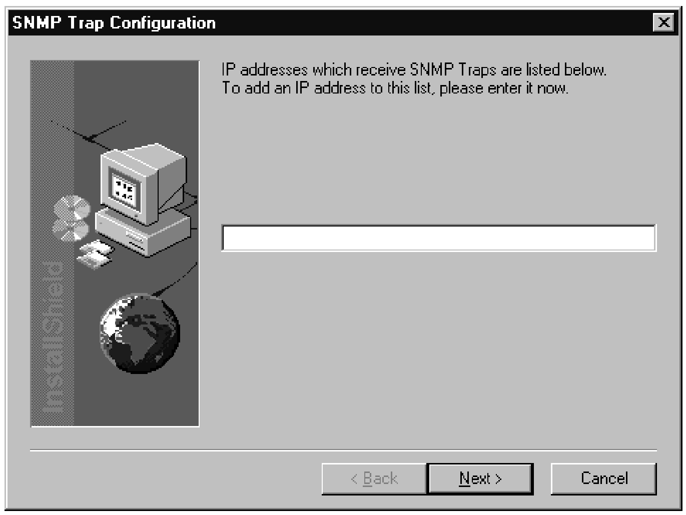
.



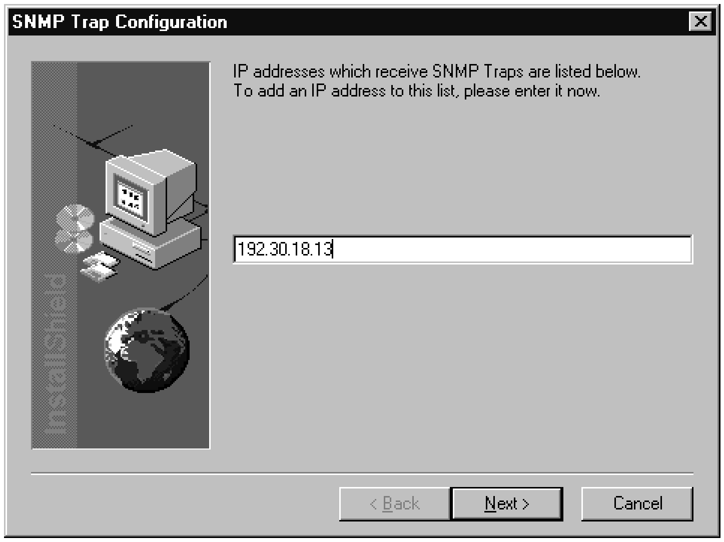
8. The next prompt will ask you whether you want to automatically start the V/IP function upon powerup of the PC:

   You should normally answer Yes to this prompt. That way, whenever the PC boots up, the V/IP function will start automatically after you have gone past the Windows 95 login prompt.

9. If you have selected the MICOM V/IP SNMP Extension Agent software to be installed (for SNMP management of the V/IP gateway), you will see this prompt:



You are provided with an opportunity to enter the IP address of a node to which the V/IP gateway is to send trap messages. A reboot is required to make any changes to the IP stack, so by entering an IP address now, the PC will be ready to start sending trap messages after being restarted following the installation of the V/IP software (which also requires rebooting the PC). You should enter the IP address as decimal digits, like the following example:



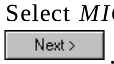
10. The next prompt will ask you if you want to view the Release Notes. You should answer Yes to this prompt. The Release Notes contain the latest information about the V/IP software and also include additional important information about installing the software.

11. The next prompt asks you whether you want the setup program to help you install the hardware (the Voice Interface Cards):

For most installations, you should answer Yes to this prompt and continue on to step 12.

Answer No to the prompt if you have special circumstances (or, if advised to do so by MICOM Customer Service). Then, follow the *Load the V/IP Device Driver* procedure beginning on page 2-26.

12. The setup program will call up the Windows 95 Add New Hardware wizard, automatically make some selections, and then stop at the following dialog box:

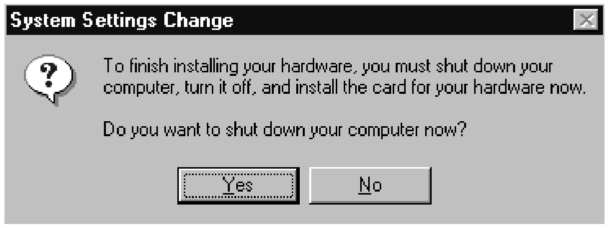
Select *MICOM V/IP Analog Card* from the dialog box and click on 

.

13. You will see a Resource dialog box. Record this information, as you will need it to configure the card and enter the settings during initial system configuration. Click on Next >.

14. After the required files are copied from the diskette to your PC, the driver installation will be complete. Click on Finish.

15. The next prompt will inform you that you must restart the PC:



    If you have only one Voice Interface Card to install in this PC, select Yes to the prompt. This will shut down Windows 95. Continue on to step 16.

    If you have additional Voice Interface Cards to install in this PC, select No to the prompt. You will then see the following prompt:

    Answer Yes. You will be returned to step 11 to use the Add New Hardware Wizard to install another card. Continue in this way until all cards have been added to Windows 95. Then, select Yes to the *Do you want to shut down the computer now* prompt. This will shut down Windows 95. Continue on to step 16.

16. Turn off the power to the PC and remove any diskettes in the drives.

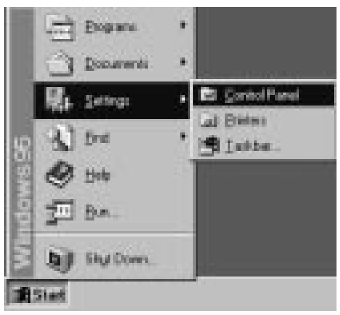
17. Continue on to *Configuring and Installing the Voice Interface Card(s), on page 2-33.*

Load the V/IP Device Driver

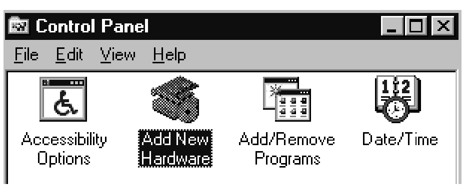
This is an optional procedure. You need to perform this procedure only if you did not use the V/IP setup wizard to add the Voice Interface Cards to Windows 95. Or, perform this procedure if you wish to make changes to the values assigned by Windows 95 to the card's resources.

1. Make sure the diskette labeled *V/IP For Windows 95 Diskette 2* is inserted into the diskette drive of your PC.

2. Click the Start button, point to Settings, and click on Control Panel.

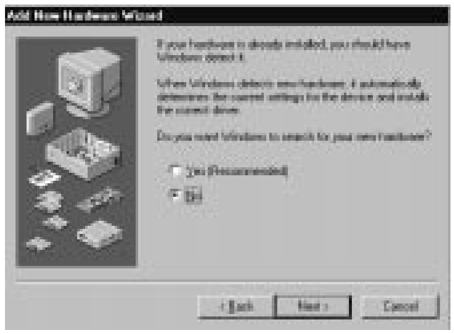


3. Double-click on Add New Hardware.



The Add New Hardware Wizard will be displayed. Click on Next > .

4. Click on No to the prompt to have Windows search for new hardware and click on Next > .

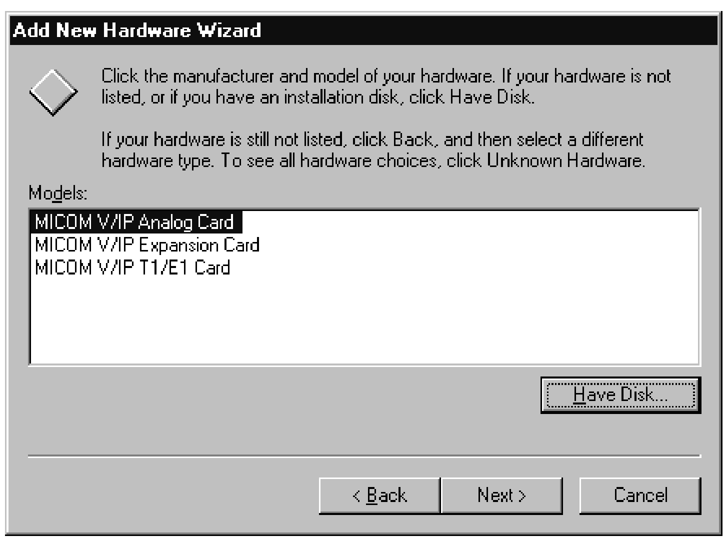


5. Click on Other devices in the Hardware type selection dialog box and click on Next > .

6. Click on Have Disk in the Manufacturer/Model dialog box.

7. Select the diskette drive letter and click on OK in the Install From Disk dialog box.

8. The Models dialog box should display *MICOM V/IP Analog Card*, *MICOM Expansion Card* and *MICOM V/IP T1/E1 Card*. Select ***MICOM V/IP Analog Card*** and click on Next > .

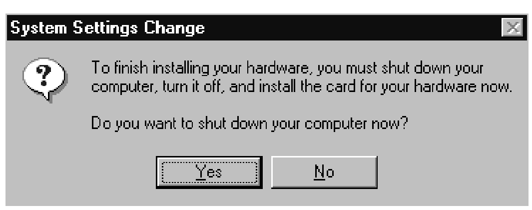


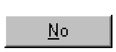
9. You will see a Resource dialog box. Record this information, as you will need it to configure the card(s) and enter the settings during initial system configuration.

10. After the required files are copied from the diskette to your PC, the driver installation will be complete. Click on 

.

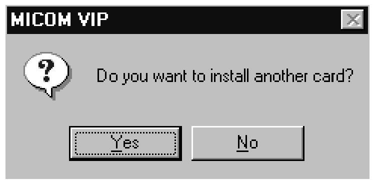
11. The next prompt will inform you that you must restart the PC:



Select 

to this prompt.

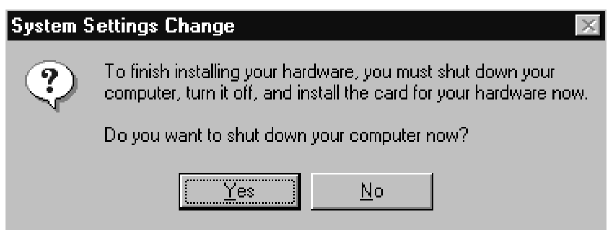
You will then see the following prompt:



If you do not have additional cards to install, answer No and continue on to step 12.

If you have additional Voice Interface Cards to install in this PC, answer Yes. You will be returned to step 4 to use the Add New Hardware Wizard to install another card. Continue in this way until all cards have been added to Windows 95. Then, select No to the *Do you want to install another card* prompt. Continue on to step 12.

12. You will see the following restart reminder:



- **If the card settings that Windows 95 came up with are acceptable to you**, then:

  a. Click on the Yes button. This will shut down Windows 95.

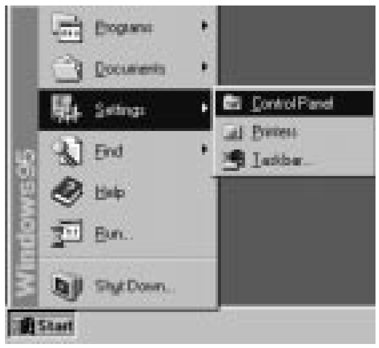
  b. Turn off the power to the PC and remove any diskettes from the drives.

  c. Continue on to *Configuring and Installing the Voice Interface Card(s) (page 2-33).*

- **If you want to use different settings for the Voice Interface Cards than Windows 95 suggests** (i.e., you have a set of non-conflicting card settings that you know will work the best), then:

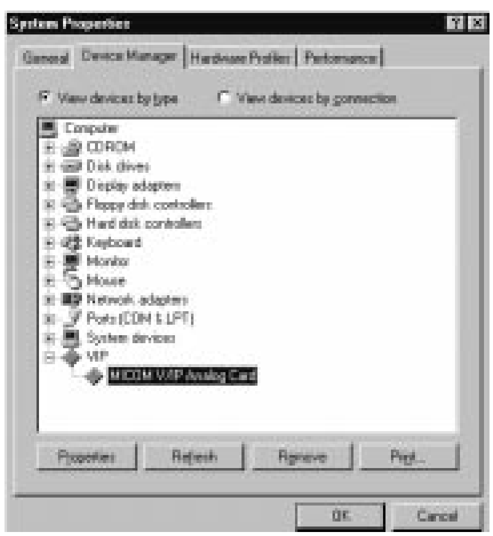
  a. Click on the No button.

b. Select Start ➧ Settings ➧ Control Panel:

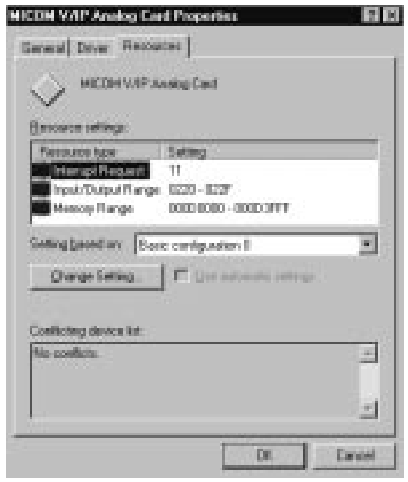


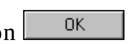
c. Double-click on the System icon:

d. Click on the Device Manager tab, go down the tree view of the hardware list, and double-click on V/IP:

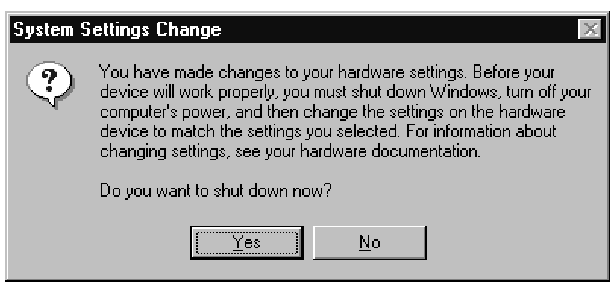


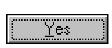
e. Click on MICOM V/IP Analog Card, click on the Properties button, and click on the Resources tab:



Change the settings as required, then click on  .

f. You will be reminded to shut down the PC.



Click on the  button. This will shut down Windows 95.

g. Turn off the power to the PC and remove any diskettes from the drives.

h. Continue on to *Configuring and Installing the Voice Interface Card(s) (page 2-33).*

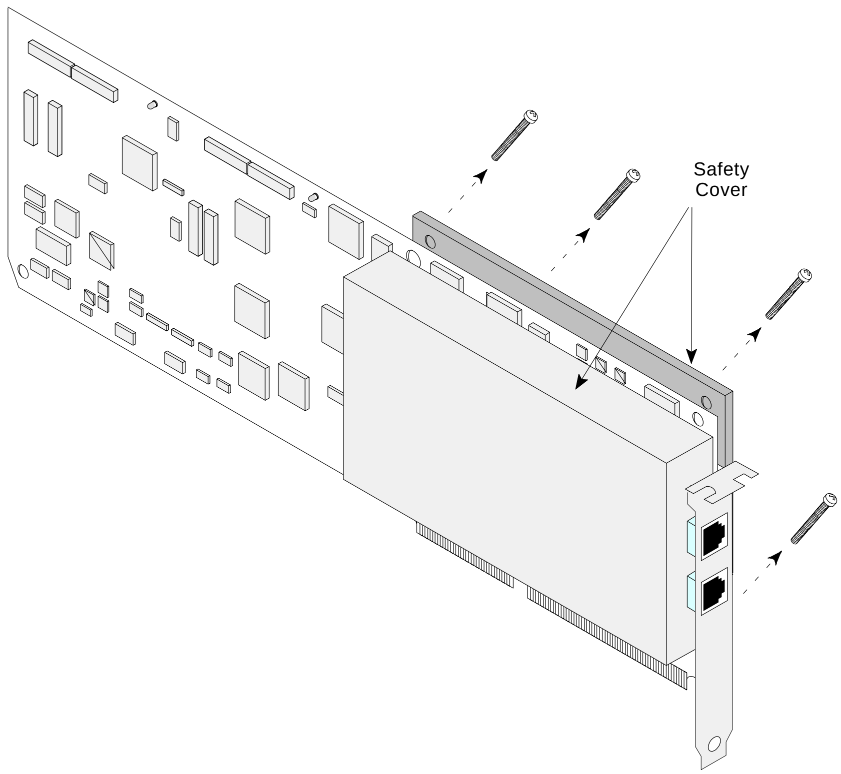
Configuring and Installing the Voice Interface Card(s)

! ***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*).**

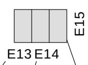
   Each channel can be configured individually for the type of analog interface you require: FXS, E&M type I, E&M type II, or E&M type V. The card is shipped from the factory preconfigured for FXS interface on all channels. However, even if you do plan on using the FXS interface, you should still check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

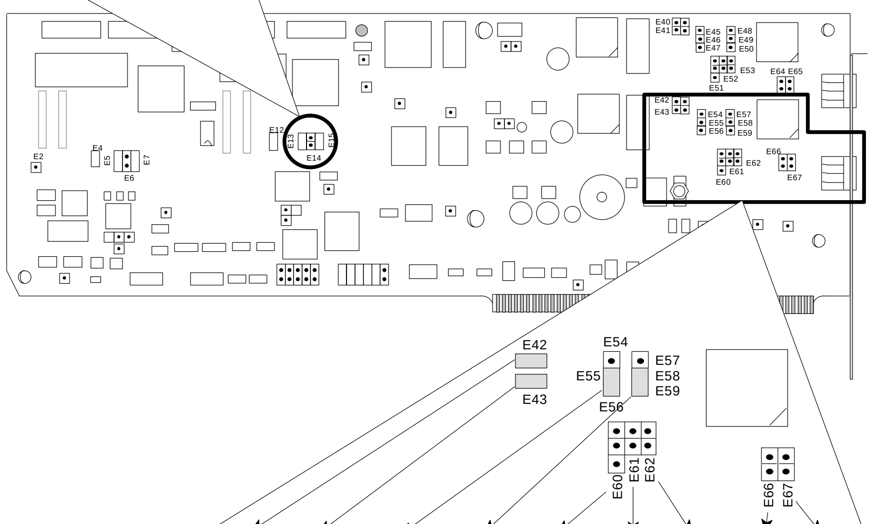
Note: The following diagrams show how to set the voice signaling for each channel of a Voice Interface Card. Note that Channel 2 of a 2-channel Voice Interface Card is in the same physical location as Channel 1 of a 1-channel card. Pages 2-40 to 2-43 show the settings for Channel 1 of a 2-channel card.

FXS Interface

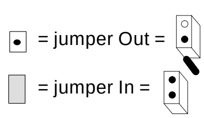
Channel 2 of a 2-channel card
Channel 1 of a 1-channel card



E13	E14	E15
In	In	In

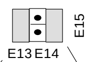


E42	E43	E54 E55 E56	E57 E58 E59	E60	E61	E62	E66	E67
In	In	E55 to E56	E58 to E59	Out	Out	Out	Out	Out

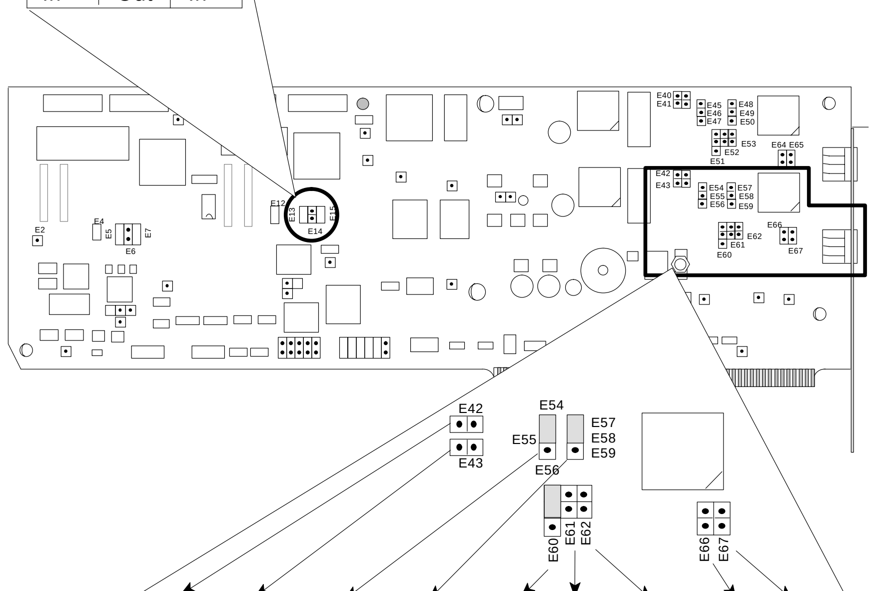


E&M Type I Interface

Channel 2 of a 2-channel card
Channel 1 of a 1-channel card



E13	E14	E15
In	Out	In



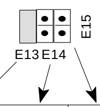
E42	E43	E54 E55 E56	E57 E58 E59	E60	E61	E62	E66	E67
Out	Out	E54 to E55	E57 to E58	1 to 2	Out	Out	Out	Out

= jumper Out =

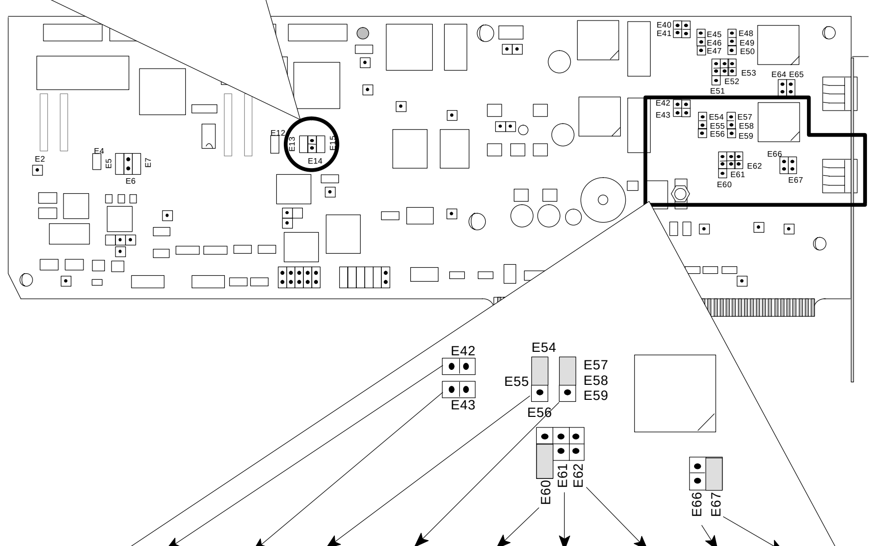
= jumper In =

E&M Type II Interface

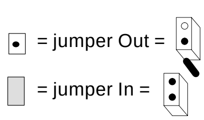
Channel 2 of a 2-channel card
Channel 1 of a 1-channel card



E13	E14	E15
In	Out	Out

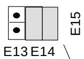


E42	E43	E54 E55 E56	E57 E58 E59	E60	E61	E62	E66	E67
Out	Out	E54 to E55	E57 to E58	2 to 3	Out	Out	Out	In

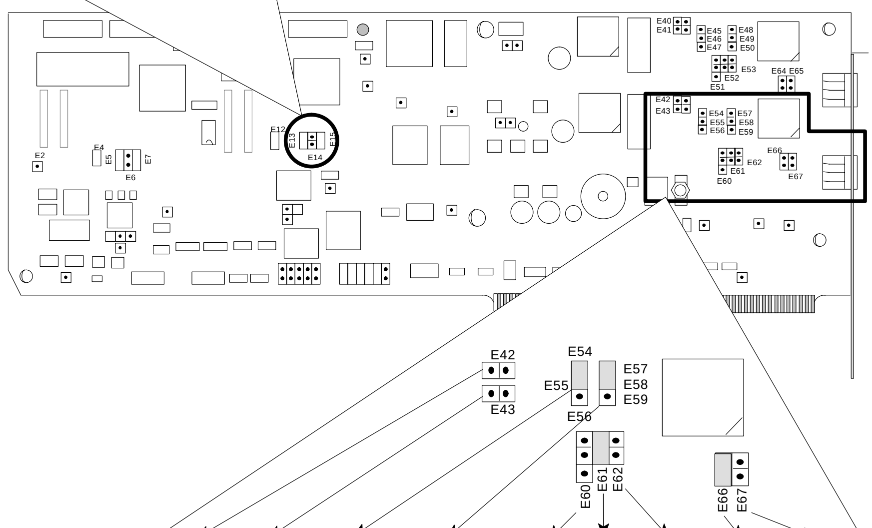


E&M Type V Interface

Channel 2 of a 2-channel card
Channel 1 of a 1-channel card



E13	E14	E15
Out	In	In



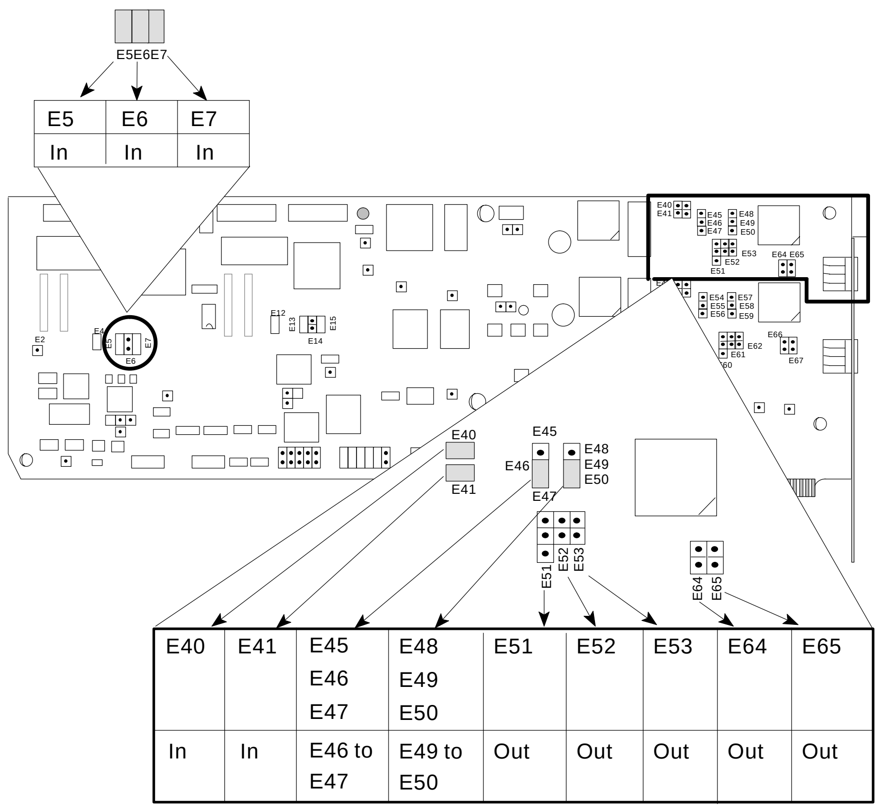
E42	E43	E54 E55 E56	E57 E58 E59	E60	E61	E62	E66	E67
Out	Out	E54 to E55	E57 to E58	Out	In	Out	In	Out

= jumper Out =

= jumper In =

FXS Interface

Channel 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

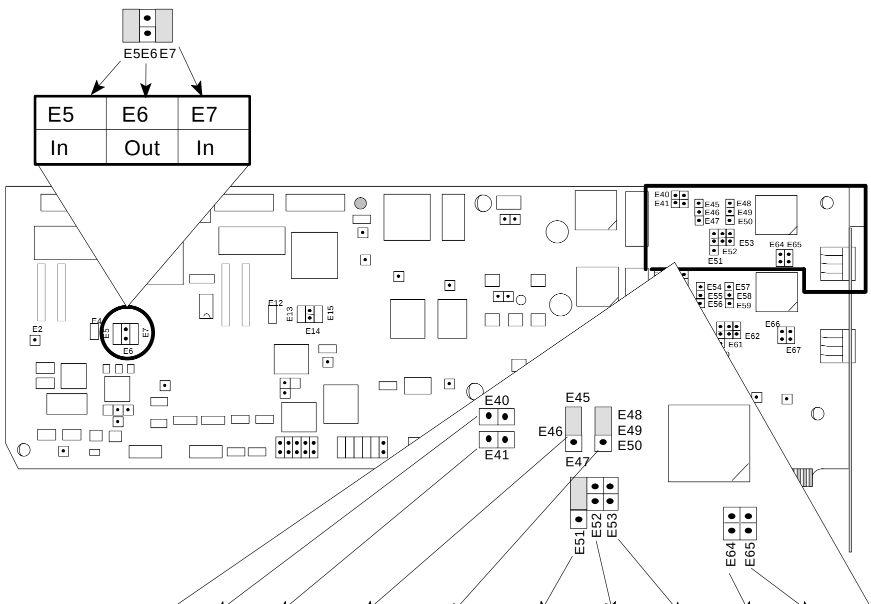
= jumper Out =

= jumper In =

E&M Type I Interface

Channel 1 of a 2-channel card only

E5	E6	E7
In	Out	In



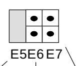
E40	E41	E45 E46 E47	E48 E49 E50	E51	E52	E53	E64	E65
Out	Out	E45 to E46	E48 to E49	1 to 2	Out	Out	Out	Out

= jumper Out =

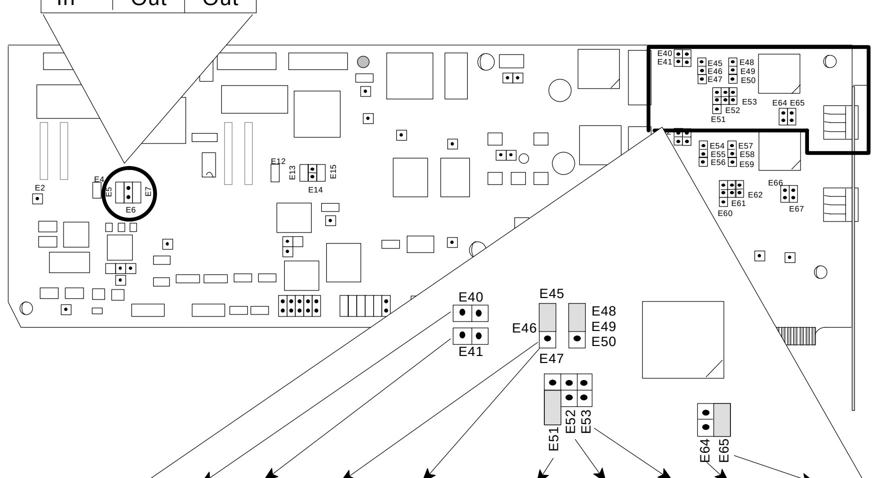
= jumper In =

E&M Type II Interface

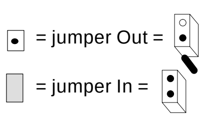
Channel 1 of a 2-channel card only



E5	E6	E7
In	Out	Out

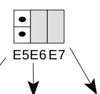


E40	E41	E45 E46 E47	E48 E49 E50	E51	E52	E53	E64	E65
Out	Out	E45 to E46	E48 to E49	2 to 3	Out	Out	Out	In

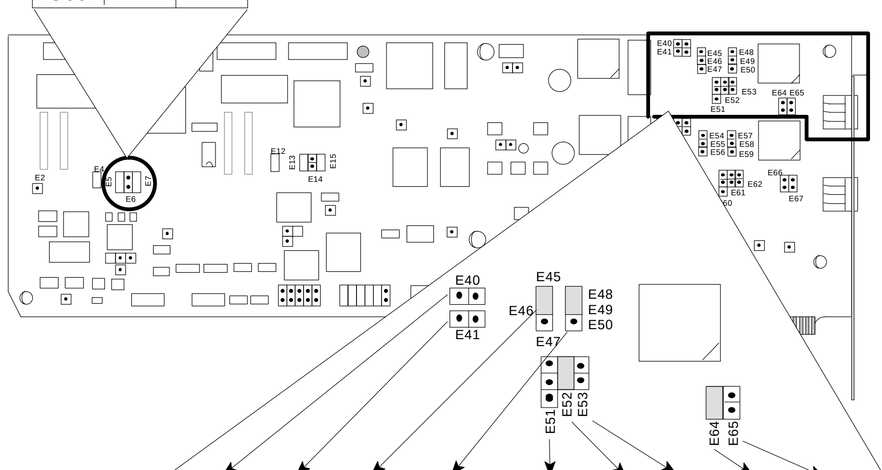


E&M Type V Interface

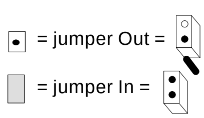
Channel 1 of a 2-channel card only



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.

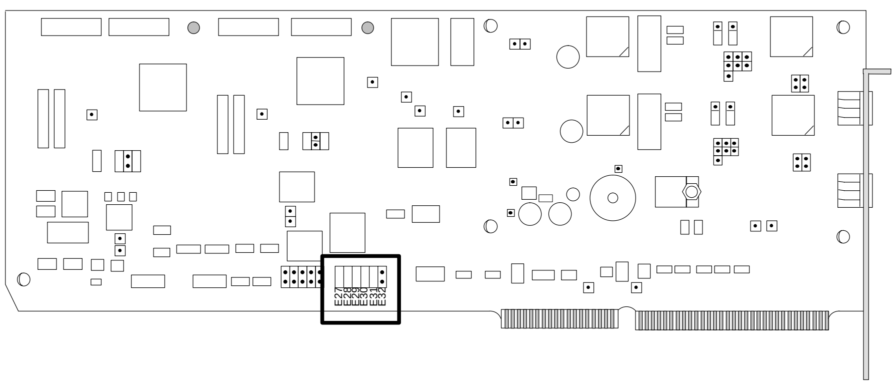
You must configure the base address for the I/O port addresses that the card requires. The I/O port addresses you can select from are:

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

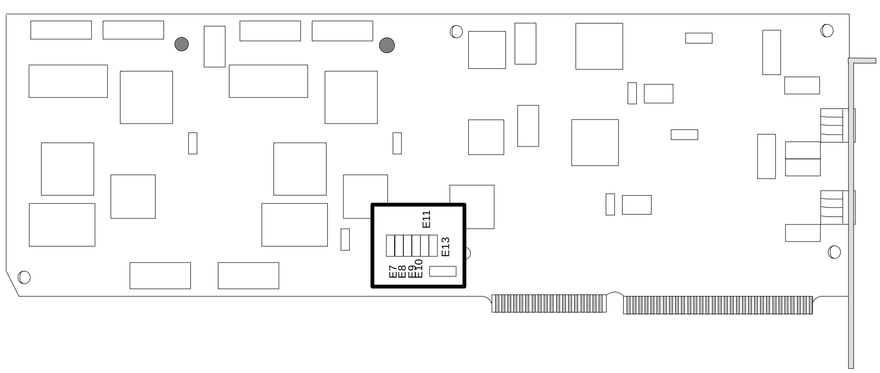
300, 310, 320, 330, 340, 350, 360, 370, 380, 390, 3A0, 3B0, 3C0, 3D0, 3E0, 3F0

Use the I/O port addresses that were assigned by Windows 95 when you performed the Add Hardware procedure. The jumper settings are shown on the next page.

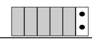
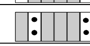
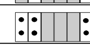
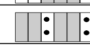
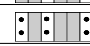
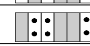
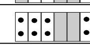
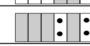
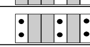
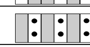
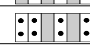
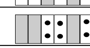
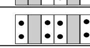
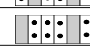
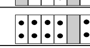
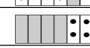
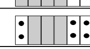
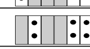
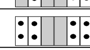
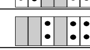
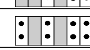
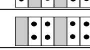
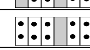
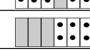
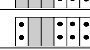
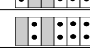
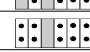
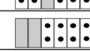
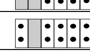
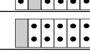
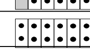
For FXS/E&M card: set the I/O port base address on jumpers E27 to E32:



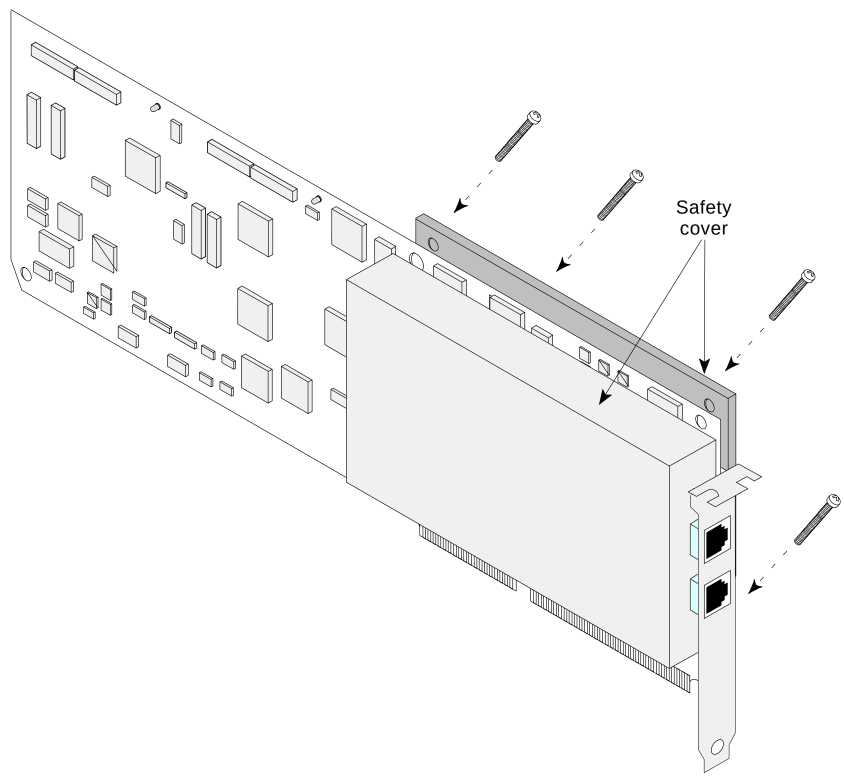
For FXO card: set the I/O port base address on jumpers E7 to E13:



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	

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



5. Install the card into the PC chassis:

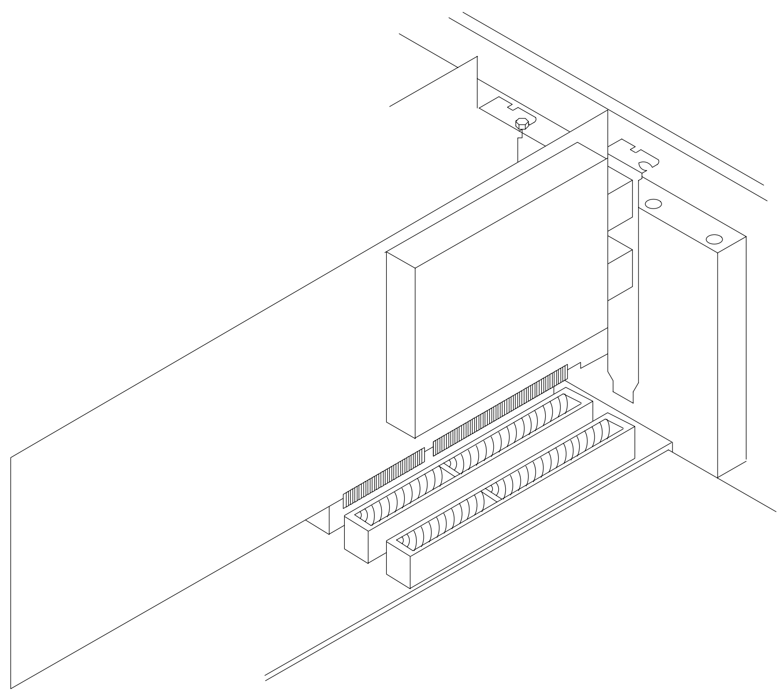
! ***WARNING***

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

! ***Caution***

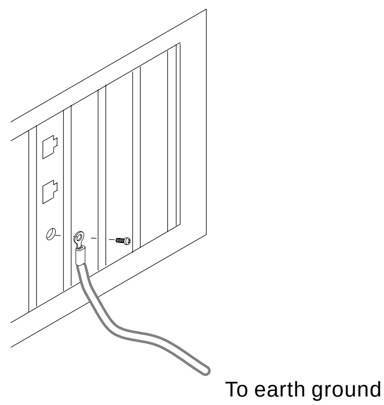
The Voice Interface Card contains static-sensitive components. Be sure to use the standard methods for handling static-sensitive components. Wear a grounding wrist strap or touch the computer chassis before handling the card.

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:



6. **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 pages 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:



7. Connect to PBX, telephone, or fax.

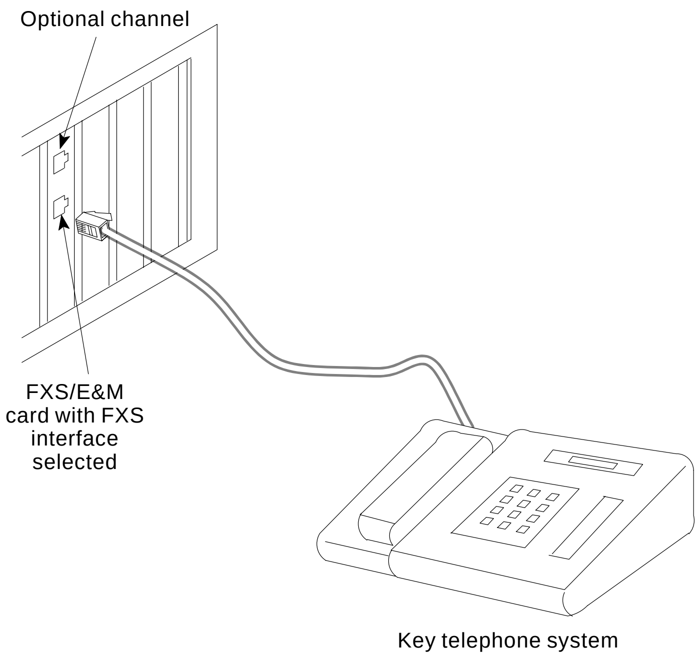
> ⚠ ***WARNING***
>
> Remove power from the computer chassis before making any connections to the analog interfaces on the Voice Interface Card.

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***

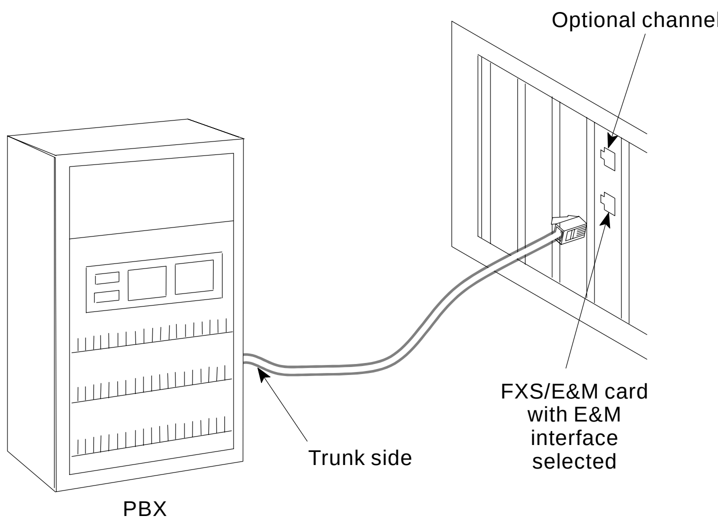
The RJ1CX connectors on the FXS/E&M card are 8-pin modular jacks. If you attempt to plug in 6-pin RJ11-type connectors, the outer pins of the RJ1CX connectors may be bent beyond the point where they can be used. The connectors will work for FXS interface, but may no longer function for E&M interface.

Here are some example analog interface connections:

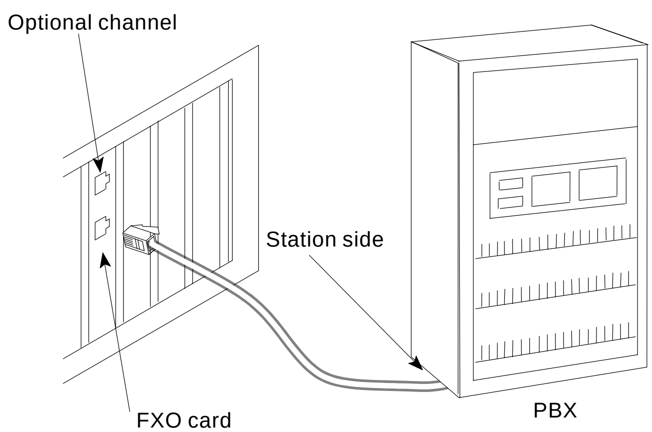


Note: The FXS interface is ***not*** intended for connection to the Public Switched Telephone Network.

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

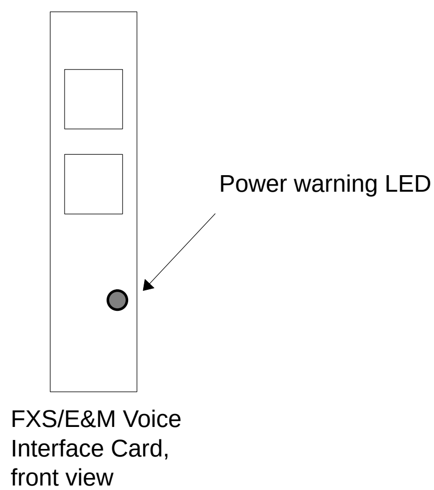
8. Power up the PC.

Remove any diskettes from the drives.

Power up the PC and verify that the machine and Voice Interface Card(s) are operating correctly. Then go on to *Configuring the System Parameters*.

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:



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:

a. Shut down the PC.

b. Check the cable(s) from the Voice Interface Card channels to your telephone equipment. The LED can come on if certain pins are shorted.

c. Powerup the PC once again.

d. If the LED stays on, shut down the PC and remove the Voice Interface Card from the PC chassis.

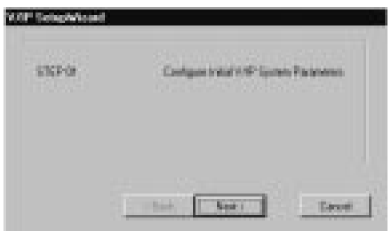
e. Verify the setting of the jumpers. An incorrect configuration of the jumpers could cause the LED to come on.

f. Reinstall the Voice Interface Card and powerup the PC once again. If the LED comes on again, contact your distributor for service.

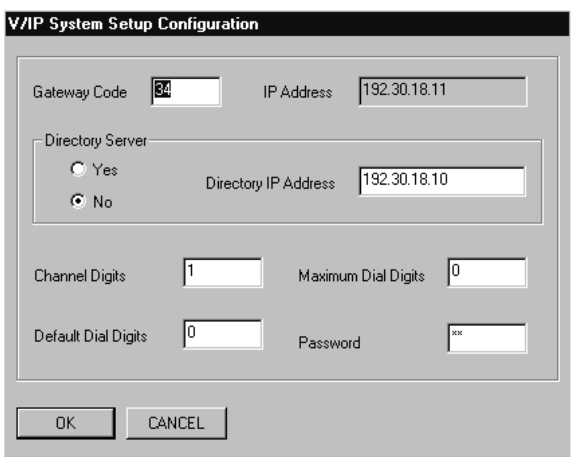
Configuring the System Parameters

1. On first time startup, the V/IP operating software will detect that there is no configuration file. This will cause the V/IP Setup Wizard to start. Click on New to perform the initial V/IP system configuration.

2. The first dialog box prompts you to configure the initial V/IP system parameters. Click on Next > .



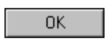
3. Enter the appropriate information in the System Setup Configuration dialog box.



Use the planning charts (located at the beginning of this section) to help you complete the entries.

The only parameter not covered by the planning charts is the password. The password accompanies all requests by Voice Interface Cards to the phone directory database server and is used to authenticate call setup and disconnect. Here are some pointers about the password:

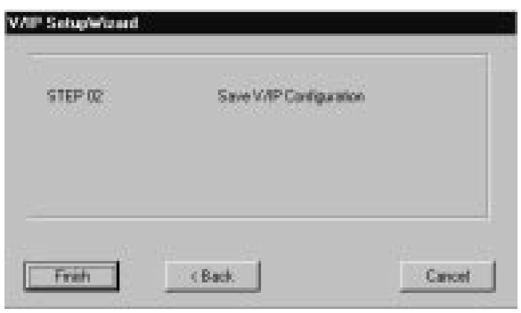
- The default is no password – the Password field is blank.
- The password is not case sensitive and can be from 1 to 31 alphanumeric characters.
- The password must be configured to the same character string in all V/IP gateway PCs in the network.
- The password is not a required entry, but its use is recommended for network security reasons.

4. Click on 

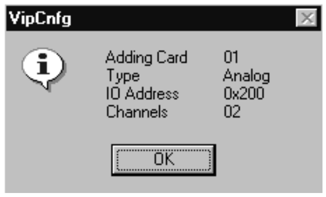
when you have completed all entries. *The Password field displays entries only as asterisks. You will have to retype the password to confirm it when you click on the OK button.*

5. The final dialog box prompts you to save the configuration. Click on 

.



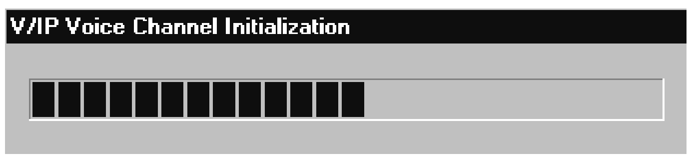
At this point, the V/IP software will locate and initialize the Voice Interface Cards within this gateway PC. You will see a message similar to the following:



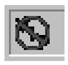
This message indicates what Voice Interface Cards the V/IP software has detected.

Click on 

. You will see the following display, showing the V/IP software initializing the Voice Interface Card channels:



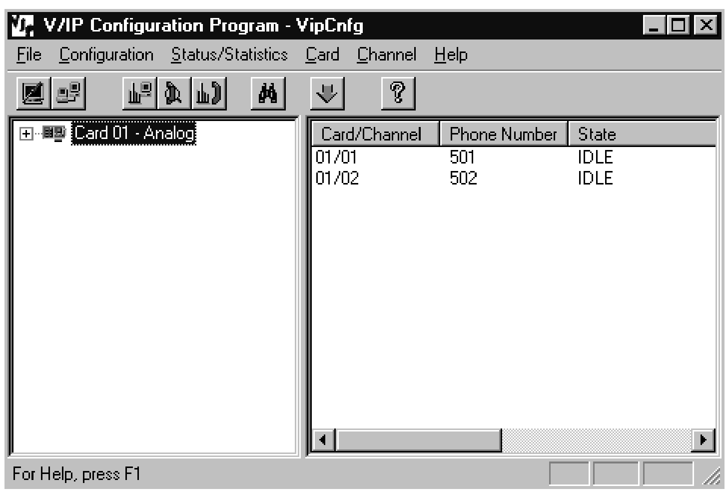
6. The V/IP software will now begin to initialize the Voice Interface Cards. Initially, in the right corner of the task bar, you will see the V/IP icon looking like this:



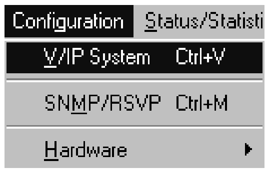
The icon will remain in this state while all Voice Interface Cards are being initialized. After this is completed, the icon changes to the following:

This means the V/IP gateway is ready to operate.

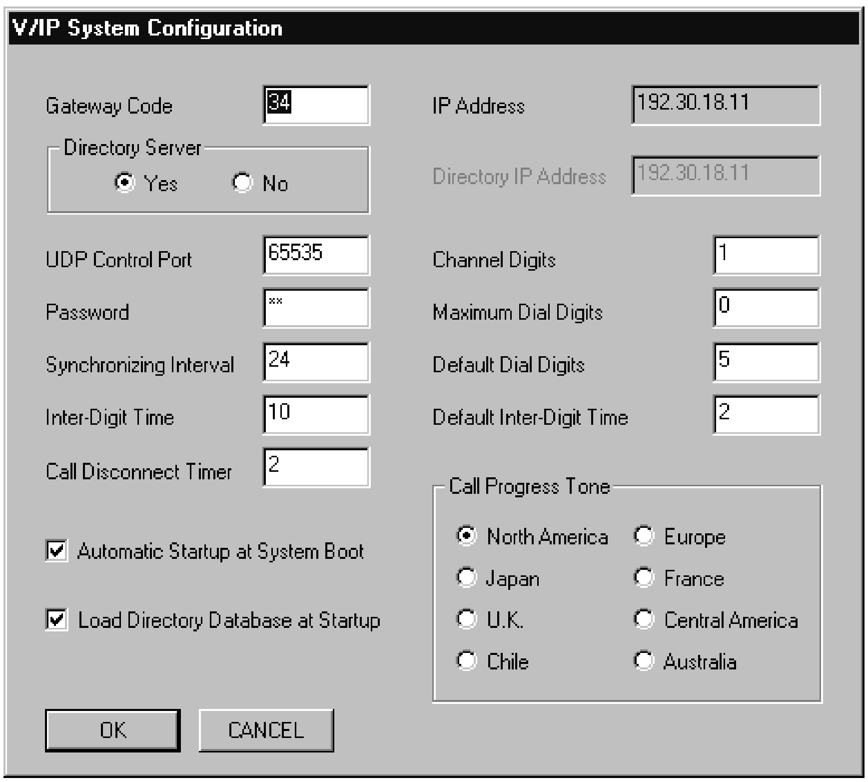
Double-click on the  icon. This will bring up the configuration program.



7. The remainder of the system parameters are configured at this point. Select Configuration ➧ V/IP System from the menu bar:



You will see the System Configuration menu:



You have already configured the most important parameters. The remainder of the parameters are self-explanatory. If you need help about any of the parameters, just select Help ➧ Help Topics from the menu bar and look up the topic *About System Parameters*.

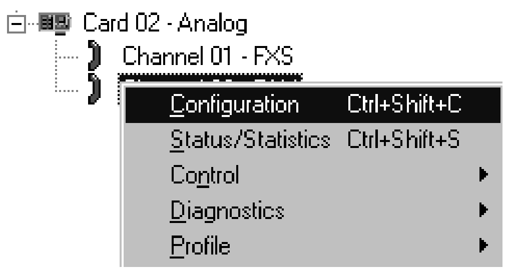
Now, continue on to *Configuring the Voice/Fax Channels*.

Configuring the Voice/Fax Channels

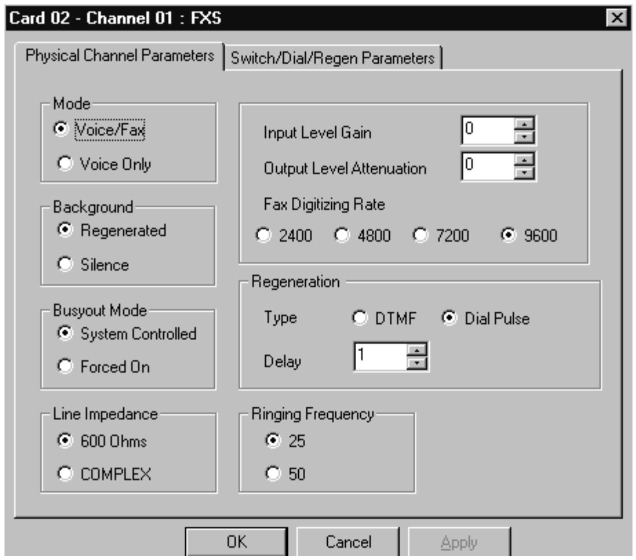
For initial operation, you may need to configure a few voice/fax channel parameters to get the channels operating correctly. In many cases, you can use the default values for the other parameters. Still, you should review the voice/fax channel parameters to make sure they are set correctly.

1. Double-click on the Card icon. The Channel icons will appear:

2. Click on the desired Channel icon and click the *right* mouse button. A popup menu will appear:



Select Configuration. You will see the channel configurations forms:

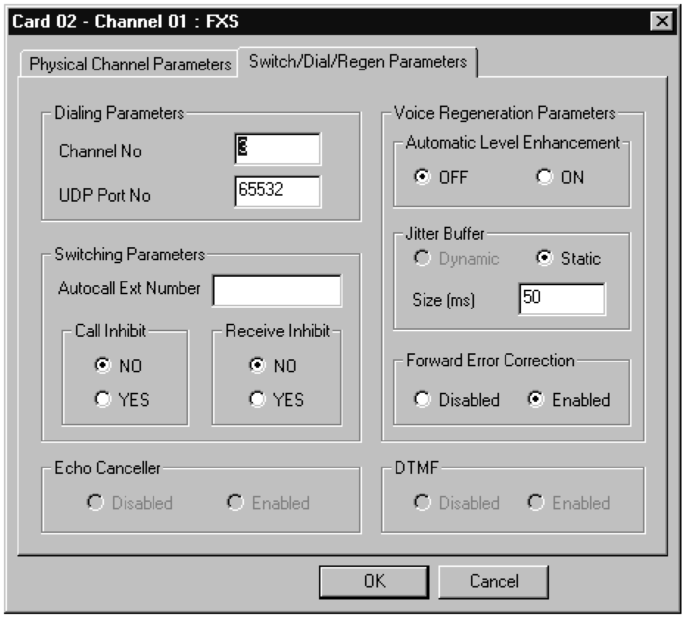


3. Select the Physical Channel Parameters tab. The parameters you might need to configure at first time installation are:

- Line Impedance (600 ohms or complex). Set to match the interfacing telephone equipment.
- Regeneration Type (DTMF or Dial Pulse). Set as required for the interfacing telephone equipment.
- Ringing Frequency (25 Hz or 50 Hz). Some countries require the 50 Hz setting.
- Analog Operation (2-wire or 4-wire, E&M interface only). Set to match the requirements of the associated PBX.

For additional information about the parameters, use the online help. Select Help ➧ Help Topics from the menu bar and look up *Physical Channel Parameters*.

4. Select the Switch/Dial/Regen Parameters tab:



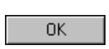
You might need to configure the Channel No parameter.

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 the V/IP gateway if all channels are to be members of one hunt group.
- 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 Channel Digits parameter setting in the System Configuration form.

For additional information about the parameters, use the online help. Select Help ➧ Help Topics from the menu bar and look up *Switching, Dialing, and Regeneration Parameters*.

5. Click on 

 when you have completed the forms for this channel. Then, continue for all the remaining channels in this PC.

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

Configuring the Routing Priority

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):

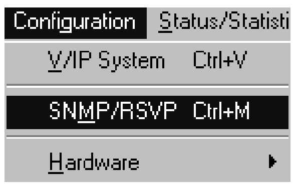
- Weighted Fair Queuing.
- Prioritize IP packets over IPX packets.
- Prioritize by UDP port numbers. The voice/fax packets sent out by the V/IP gateway are UDP packets with the source UDP port = 1EA hexadecimal (490 decimal). You should prioritize UDP packets with the source UDP port = 1EA hexadecimal.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

Configure the V/IP Gateway For RSVP

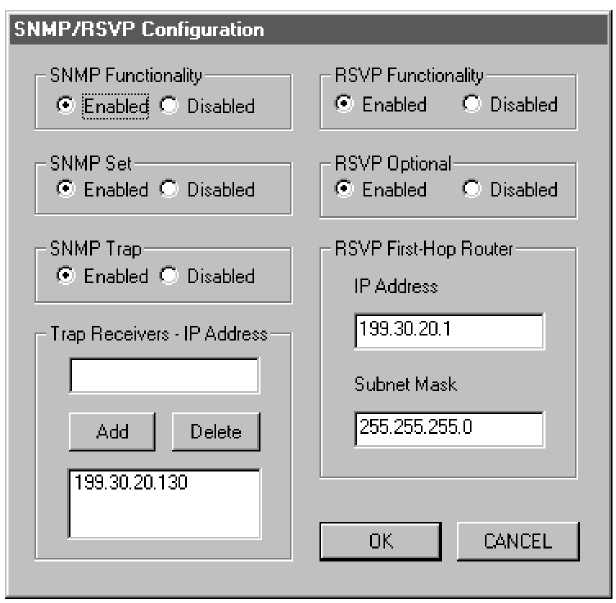
Important: To be able to use RSVP, you must have installed Microsoft Winsock 2 and Intel's PC-RSVP Service Provider prior to installing the V/IP software.

To utilize RSVP, you must enable that function on the V/IP gateway, as follows:

1. Select Configuration ➧ SNMP/RSVP from the menu bar, as follows:



A fill-in form will be displayed (if Winsock 2 was not installed, the RSVP parameters will be grayed out):



2. Configure the RSVP parameters as required. You must configure the parameters to the same values at all V/IP gateways in the network.

   For information about the parameters, use the online help. Select Help ➧ Help Topics from the menu bar and look up the topic *RSVP*.

Operation 3

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 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

To end the call, simply hang up the telephone.

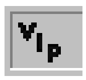
If you want more information on placing calls over the V/IP network, see the topic *About Making Calls Over the V/IP Gateway* in the V/IP Configuration Program online help.

Administration 4

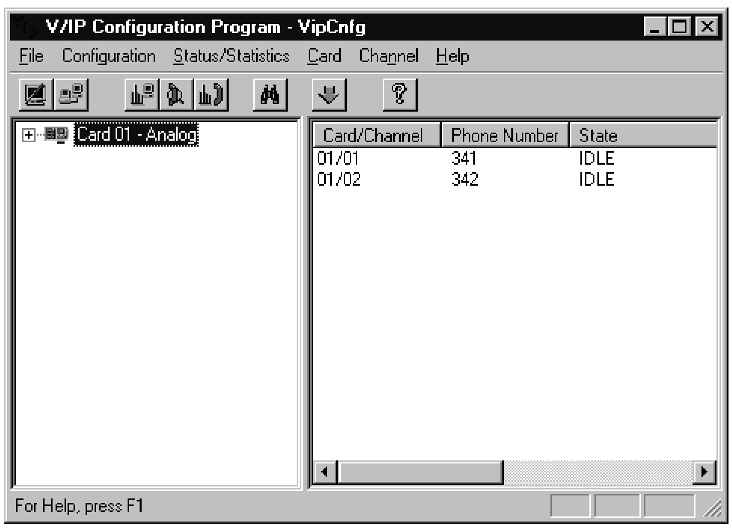
Accessing the V/IP Configuration Program

The V/IP gateway software is designed to run in the background. Regular interaction between the V/IP gateway and an operator at the computer console is not required. However, you'll need to access the V/IP Configuration Program to perform any of the procedures in this section.

To access the V/IP Configuration Program, just double-click on the V/IP icon on the task bar:

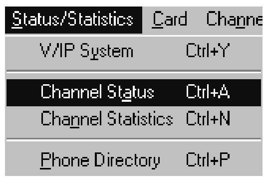


The V/IP Configuration Program will be displayed:



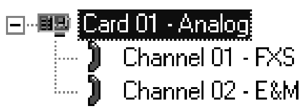
Viewing the Status of Voice/Fax Channels

You can view the status of all channels simultaneously by selecting Status/Statistics ➧ Channel Status from the menu bar:

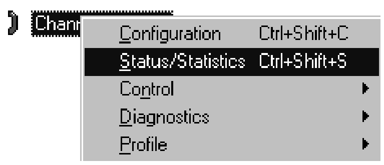


You can display the status of an individual channel in the V/IP gateway, as follows:

1. Double-click on the Card icon. The Channel icons will appear:

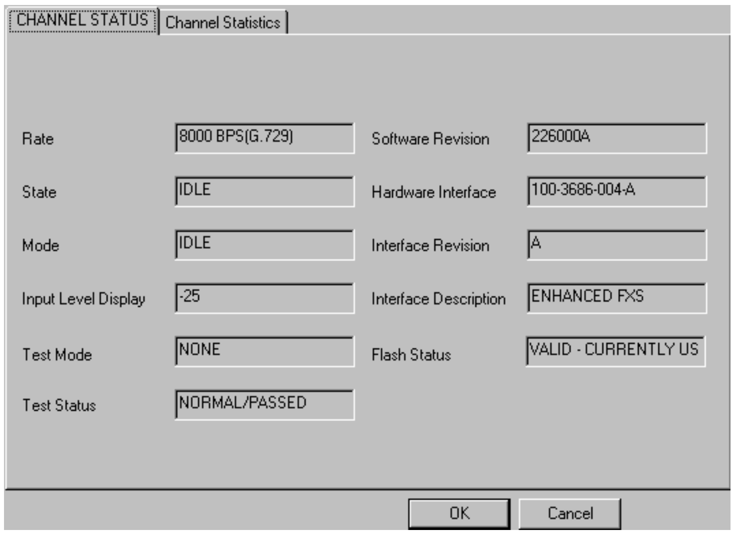


2. Click on the desired Channel icon and click the *right* mouse button. A popup menu will appear. Click on Status/Statistics:



The Channel Status/Statistics screen will be displayed.

3. Click on the Channel Status tab:



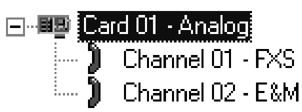
You can find technical descriptions about the displayed channel status parameters in the online help. Just select Help ➧ Help Topics from the menu bar and look up the topic *Descriptions Of Channel Status*.

Viewing the Voice/Fax Channel Statistics

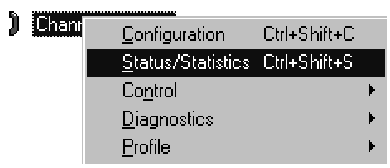
You can view the statistics for all channels, simultaneously, by selecting Status/Statistics ➧ Channel Statistics from the menu bar:

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

1. Double-click on the Card icon. The Channel icons will appear:

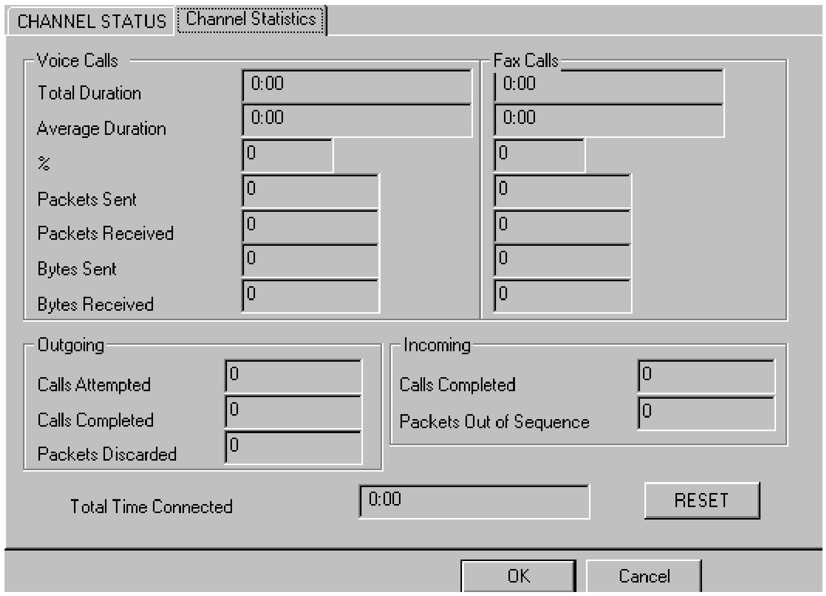


2. Click on the desired Channel icon and click the *right* mouse button. A popup menu will appear. Click on Status/Statistics:



The Channel Status/Statistics screen will be displayed.

3. Click on the Channel Statistics tab:



You can find technical descriptions about the displayed channel statistics in the online help. Just select Help ➧ Help Topics from the menu bar and look up the topic *Descriptions Of Analog Channels Statistics*.

To reset the statistics, simply click on the 

button and click on 

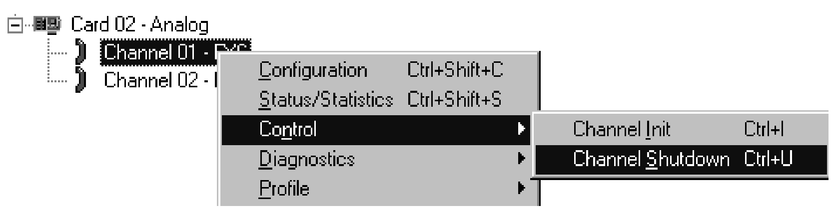
.

Changing the Configuration of a Voice/Fax Channel

You can change the configuration of a voice/fax channel only if there is no call in progress on that channel. You should also shut down the channel before making changes to its configuration.

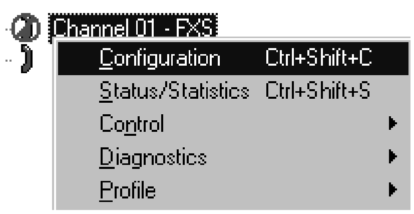
1. Double-click on the Card icon. The Channel icons will appear:

2. Click on the desired Channel icon and click the *right* mouse button. A popup menu will appear. Click on Control ➧ Channel Shutdown:

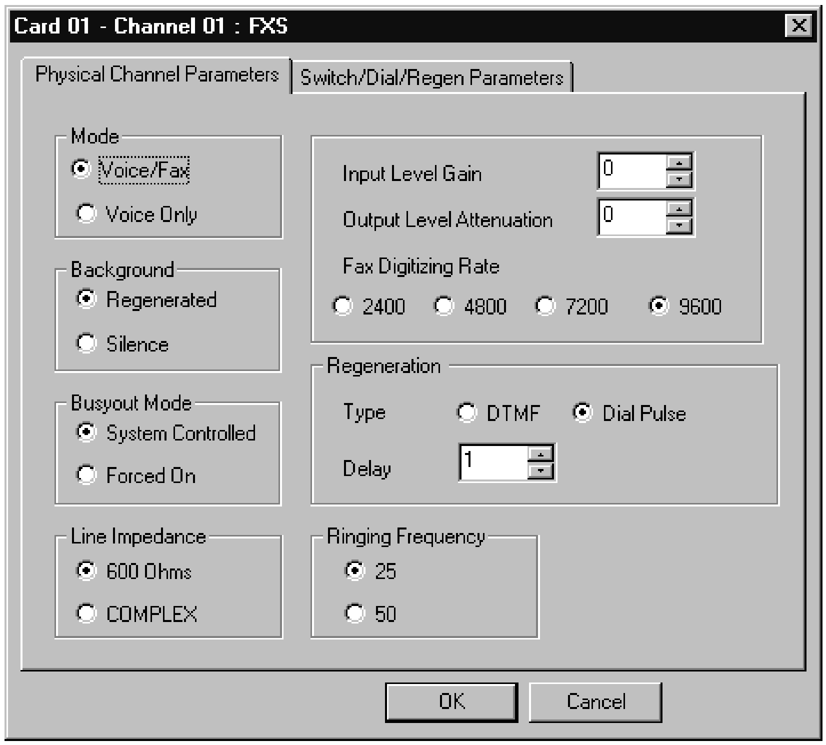


Note: If the channel is crossed out ( ), the channel has already been shut down. In this case, just continue on to step 3.

3. Click on the desired Channel icon and click the *right* mouse button. The popup menu will appear. Click on Configuration:



You will see the channel configuration forms, which consist of the Physical Channel Parameters tab and the Switch/Dial/Regen Parameters tab:



4. Make the changes you require to the channel's parameters.

   For help on the parameters shown in the Physical Channel Parameters tab, click on Help ➧ Help Topics from the menu bar and look up the topic *Physical Channel Parameters*.

   For help on the parameters shown in the Switch/Dial/Regen Parameters tab, click on Help ➧ Help Topics from the menu bar and look up the topic *About Switching, Dialing, and Regeneration Parameters*.

5. When you have completed making changes to the channel's parameters, click on 

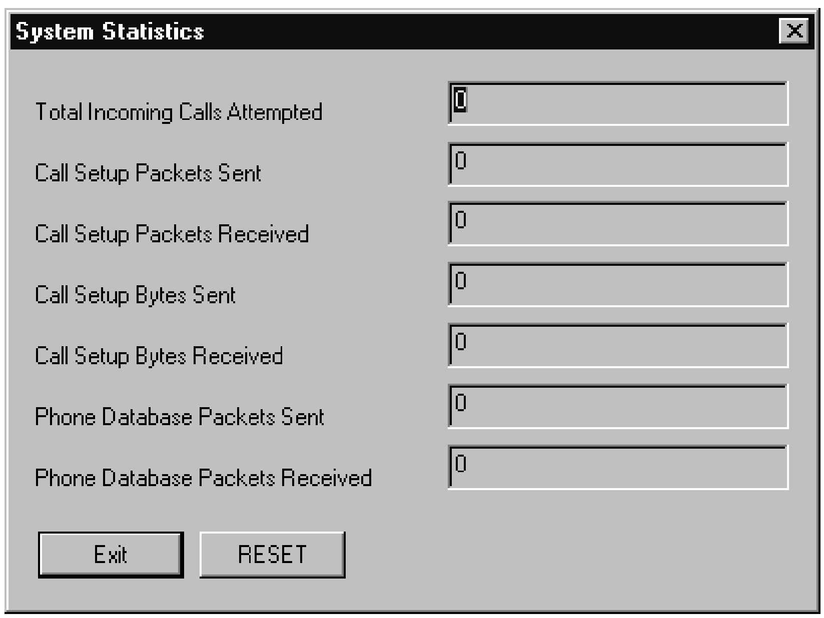
.

   The channel will automatically be reinitialized and set to the new parameters.

Viewing the System Statistics

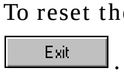
You can display the operating statistics of the V/IP gateway by selecting Status/Statistics ➧ V/IP System from the menu bar:

These statistics will be displayed like this:



You can find technical descriptions about the displayed system statistics in the online help. Just select Help ➧ Help Topics from the menu bar and look up the topic *System Statistics*.

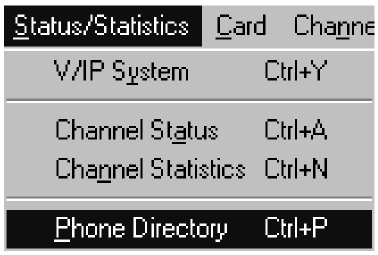
To reset the statistics, simply click on the 

button and click on 

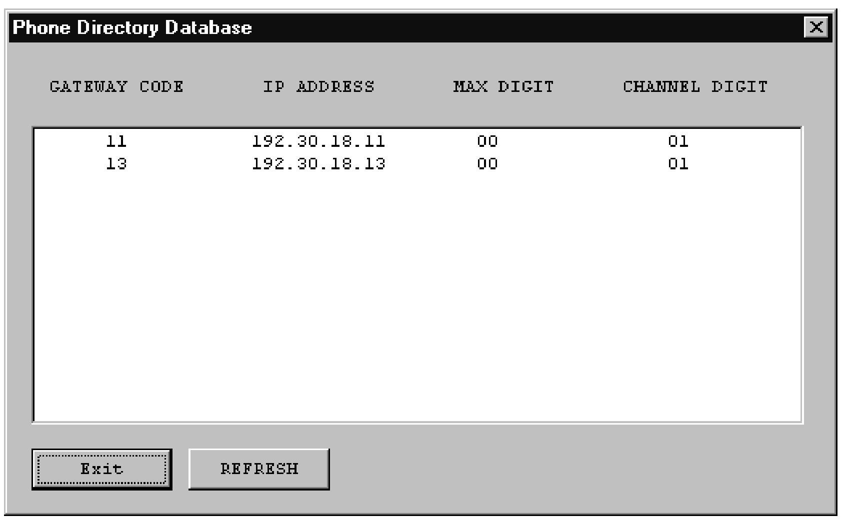
.

Viewing the Phone Directory Database

You can view the phone directory database as it exists in the local V/IP gateway by selecting Status/Statistics ➧ Phone Directory from the menu bar:



The database will be displayed like the following example:

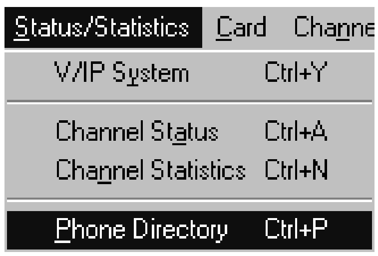


You can find technical descriptions about the entries shown in the phone directory database display in the online help. Just select Help ➧ Help Topics from the menu bar and look up the topic *About the Phone Directory Database.*

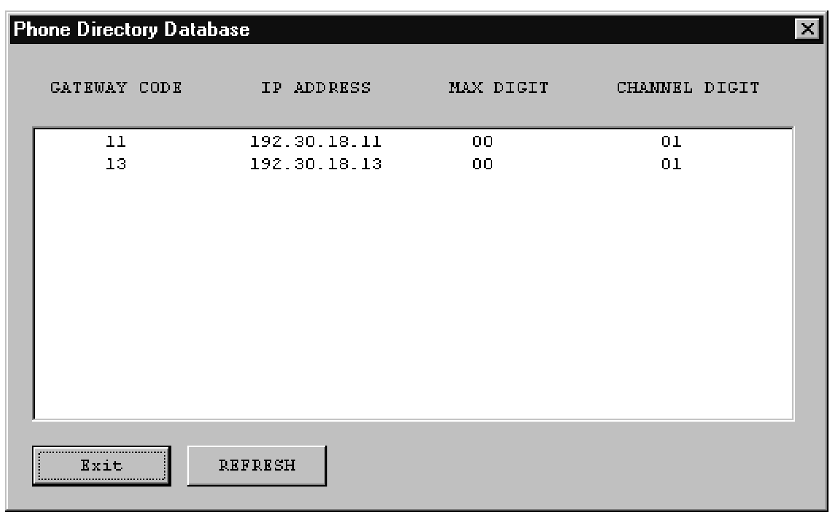
Updating the Phone Directory Database

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:

1. Select Status/Statistics ➧ Phone Directory from the menu bar:



The database will be displayed like the following example:

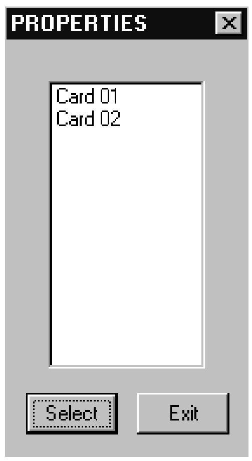


2. Click on the  button. The V/IP gateway will request a download of the database from the phone directory database server.

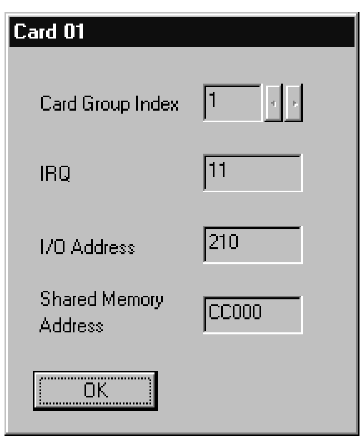
Viewing a Voice Interface Card's Hardware Parameters

1. Select Card ➧ Properties from the menu bar.

2. Select the card whose hardware parameters you want to view.



The card's hardware parameters will be displayed like the following example:



Resetting a Voice Interface Card

Resetting a card clears all calls in progress on its channels and resets all of its channels.

Click on the Card icon and click on the *right* mouse button. A popup menu will appear. Click on Control and select Reset Card:

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.

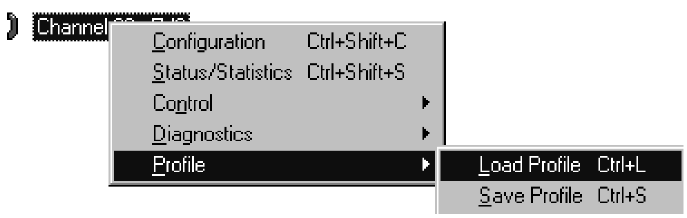
You should store profiles for all the channels in each V/IP gateway. These profiles can then be used to quickly configure the channels after a software upgrade or other major change to the overall V/IP gateway configuration.

Creating a Channel Profile

1. Make sure the channel is configured as you require. (See *Changing the Configuration of a Voice/Fax Channel* on page 4-6 if you want information about how to configure a channel.)

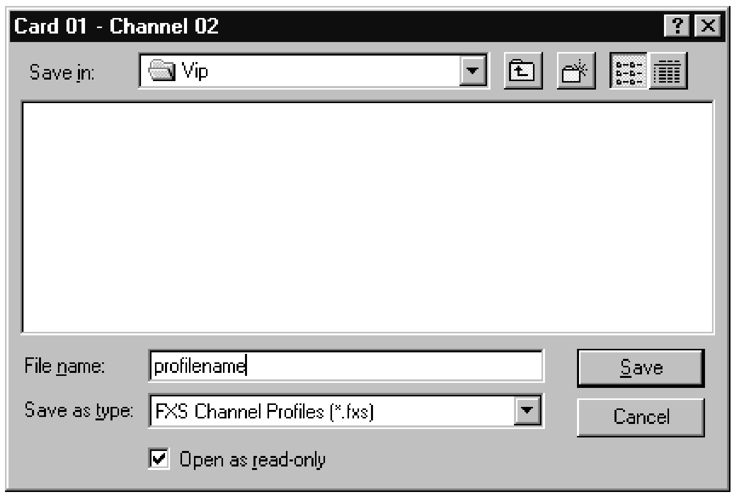
2. Double-click on the Card icon. The Channel icons will appear:

3. Click on the desired Channel icon and click the *right* mouse button. A popup menu will appear. Click on Profile ➧ Save Profile:



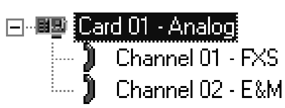
4. Enter the file name for this profile. The extension will be automatically added for you and reflects the type of channel from which this profile was saved. By default, the profile will be saved in the directory `C:\Program Files\Micom\Vip`, unless you select a different directory. Click on the 

button.

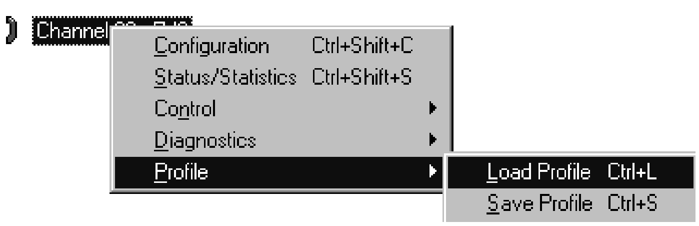


Loading a Channel Profile

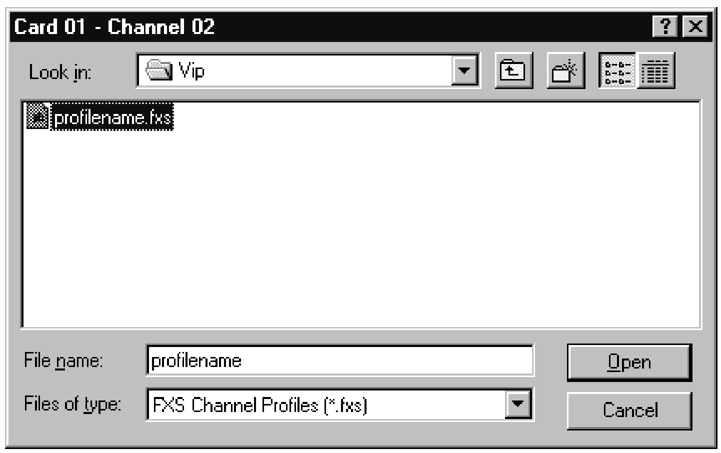
1. Double-click on the Card icon. The Channel icons will appear:



2. Click on the desired Channel icon and click the *right* mouse button. A popup menu will appear. Click on Profile ➧ Load Profile:



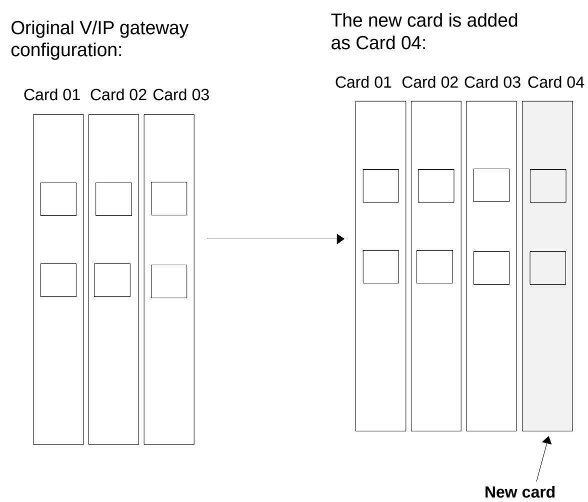
3. Select the file name of the profile to be loaded into this channel. If necessary, use the file system navigation buttons to go to the path where the profiles are saved (by default, profiles are saved in `C:\Program Files\Micom\Vip`). Click on the  button when the desired profile is selected.



The profile will be loaded into the channel and the channel will be reset. After the reset, the channel will be in an idle state and will be set to the configuration that was stored in the profile.

Adding a New Voice Interface Card

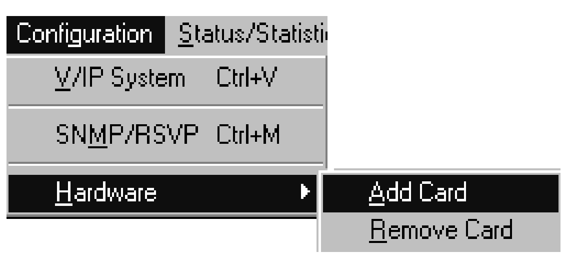
A new card is always added as the next higher card number. For example, if there are three Voice Interface Cards in the PC, the new card is installed as Card 04:



Note: You can reorganize the cards to any numbering system by using special commands to move cards up or down in the numbering sequence.

To add a new card, proceed as follows (steps 1 through 4 must done ***before*** physically installing the new card into the PC):

1. Select Configuration ➧ Hardware ➧ Add Card from the menu bar:

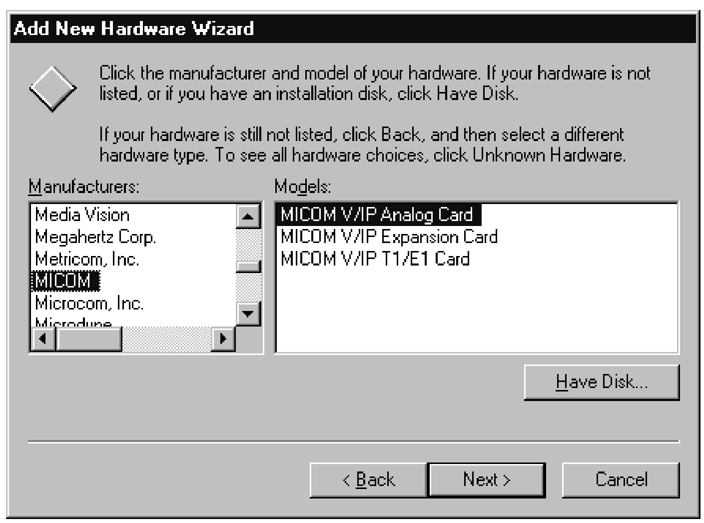


2. You will see this message:

This message reminds you that you must shut down (and power down) the computer after you've entered the information for the new card. Click on the  button.

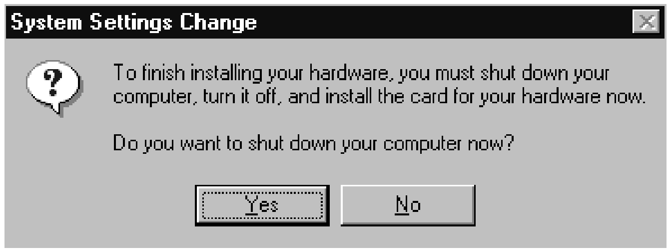
3. You will see the Windows 95 Add New Hardware Wizard. Use the wizard to setup the PC for the new card. Here are the answers to the prompts you will receive:

- When prompted "Do you want Windows to search for your new hardware," check **No**.
- When prompted "Select the type of hardware you want to install," select **Other devices**.
- When you are shown the Manufacturers and Models dialog box, select **MICOM** for Manufacturer, select **MICOM V/IP Analog Card** for Model, and click on .



- Record the resource settings that the wizard defined for the new card.

4. When you have clicked on the Finish button, you will see this prompt:

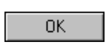


Click on 

, allow the PC to shut down, then turn off the power to the PC.

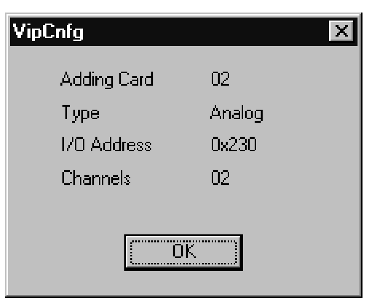
5. Install the new Voice Interface Card as detailed in the procedure starting on page 2-33 and continuing through page 2-52.

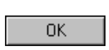
6. When the V/IP software starts up, you will see this display:

This means the V/IP software has detected the new card. Click on 

.

7. After the V/IP software has initialized the previous existing cards, it will then display a message about the new card:

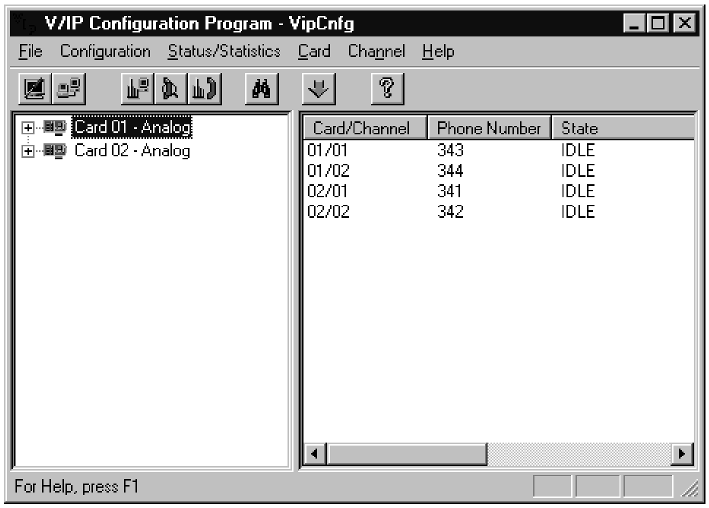


Click on 

and the card will be initialized.

Reordering the Voice Interface Cards

After adding or removing Voice Interface Cards, you may find it desirable to change the numbering order of the cards. Here is an example:



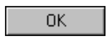
A change in Voice Interface Cards caused the channel phone numbers (extension numbers) to be in an illogical sequence according to the card numbers. Here is how to change the order so that Card 01 becomes Card 02:

1. Select Card 01.

2. *Right* click on the card and select Reorder ➧ Move Down.

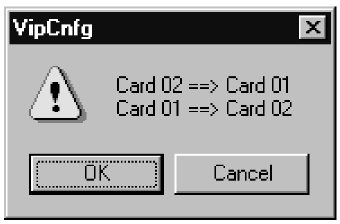
3. You will see this message:

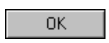
This message advises you that all channels will be taken out of service and the V/IP software will be inactivated. This state will remain until you restart the PC.

Click on 

.

4. You will see this message that the card order has been changed and all channels are out of service:



Click on 

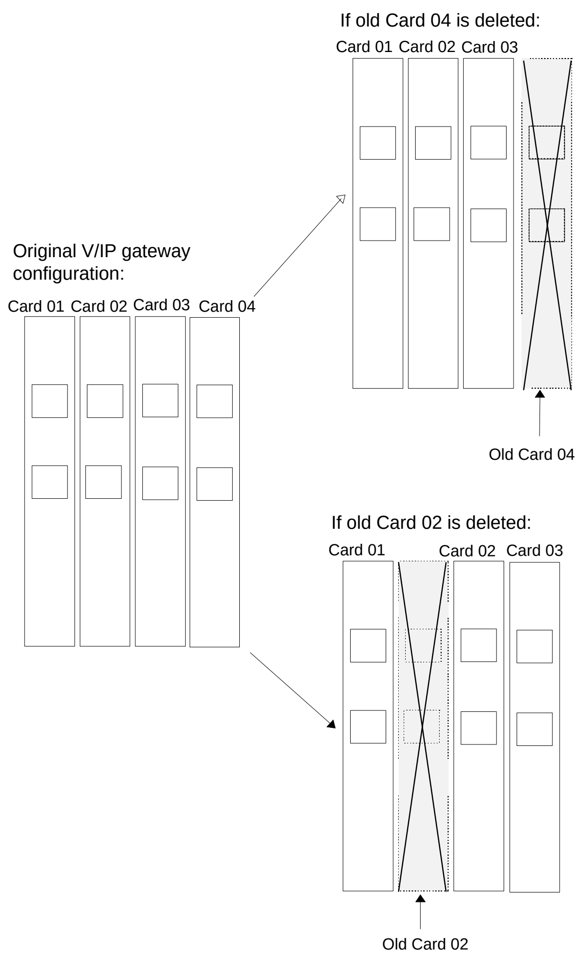
. Now, shut down the PC and restart it. When the PC comes back up, the card reorder will be in effect and all channels will be back in service.

Removing A Voice Interface Card

Removing a card affects the V/IP gateway in one of two ways:

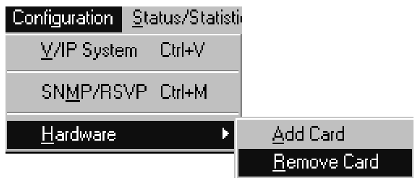
- If the card is the last card in sequence (with the highest number), the card is simply deleted from the V/IP gateway.
- Any other card, when deleted, will cause the V/IP gateway to renumber the remaining cards - the numbers will move up to take the place of the deleted card. *However, the phone numbers (extension numbers) assigned to the individual channels do* ***not*** *change.*

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

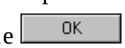


To delete a card, proceed as follows:

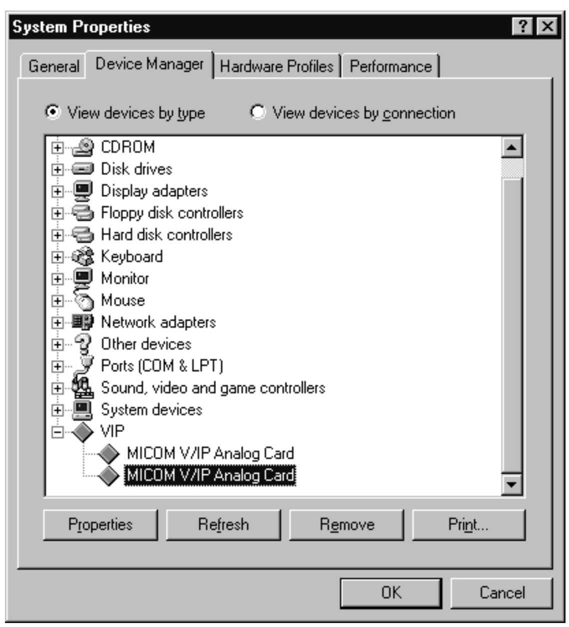
1. Select Configuration ➧ Hardware ➧ Remove Card from the menu bar:



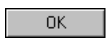
2. You will see this message:

This message reminds you that you must shut down (and power down) the computer after you've deleted the card. Click on the  button.

3. You will see the System Properties dialog box. Select Device Manager, double-click on VIP, and select the card to be deleted.

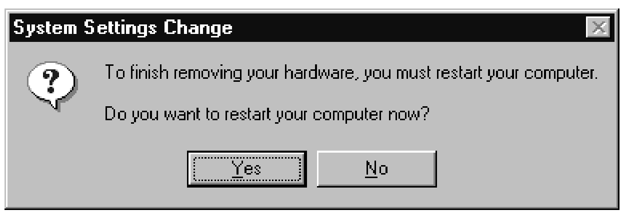


4. You will see the following confirm message:

Select 

.

5. You will see the following message:



Click on 

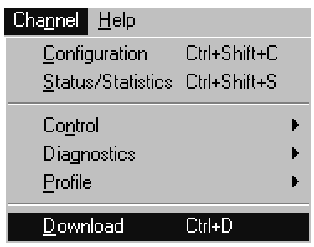
. The PC will restart and the V/IP system will make the necessary changes to logically remove the card from the system. If you want to physically remove the card from the PC, you will have to manually shut down the PC and remove the card.

Channel Code Download

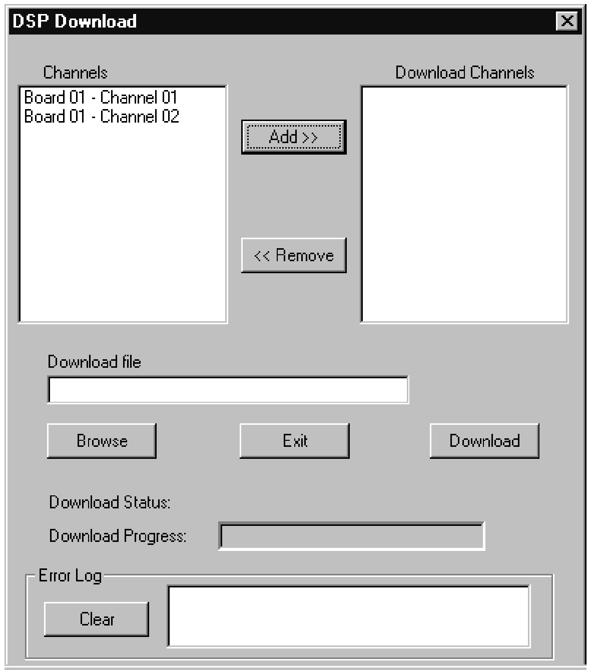
Each channel of a Voice Interface Card maintains its own copy of the operating code in firmware. The original code your Voice Interface Cards shipped with is located in the Utilities diskette, under the directory `\voice\analog`. You can command the V/IP gateway to download either the original code or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

1. Place the operating code file in the V/IP software directory (usually `C:\Program Files\Micom\Vip`).

2. Select Channel ➧ Download:



3. You will see the DSP Download dialog box:



Use the Add >> button to place channels into the Download Channels box.

4. Select the file to be downloaded to the channels. Use the Browse button to bring up a directory tree to help you find the file. Typically, the file will be something like:

```
C:\Program Files\MICOM\VIP\22600h01.voc
```

5. Click on the Download button when all is ready. The dialog box will keep you informed on the progress of the software downloads.

! *Caution*

Do not interrupt the code download while it is in progress. If the code download is interrupted, the Voice Interface Card channel(s) will end up with corrupted operating code.

Description of Files

The default path for V/IP for Win95 files is:

```
C:\Program Files\Micom\Vip
```

`Micmphon.bak`

The backup version of the Micmphon.dat phone directory database.

`Micmphon.dat`

The phone directory database as known by this V/IP gateway.

`Micmvoic.bak`

The backup version of the Micmvoic.cfg file.

`Micmvoic.cfg`

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

`readme.txt`

The release notes for the version of the V/IP software that is installed.

`Uninst.isu`

A file required for the Uninstall Wizard. This file will be needed should you need to uninstall the V/IP software at some future time.

`vipcnfg.cnt`

Contains the contents structure for the help system. It works in combination with vipcnfg.hlp to produce the V/IP help system.

`Vipcnfg.exe`

The V/IP Configuration Program.

`Vipcnfg.gid`

This file is created by the PC when accessing the V/IP help system.

`Vipcnfg.hlp`

The V/IP help system.

Troubleshooting Procedures

Windows 95 Startup Problems

If you are experiencing problems with Windows 95 starting up, you can disable the V/IP software from being loaded during the startup. When the PC boots up, press Function Key F8 when you see the statement *Starting Windows 95*. You will see a menu of selections. Select *Start Windows In Safe Mode* from the menu. This will prevent any programs in the Windows 95 Startup Group and System Tray from loading and allow you to troubleshoot system problems.

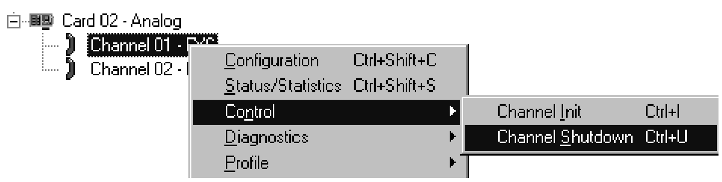
Channel Not Working

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

1. Initialize the channel:

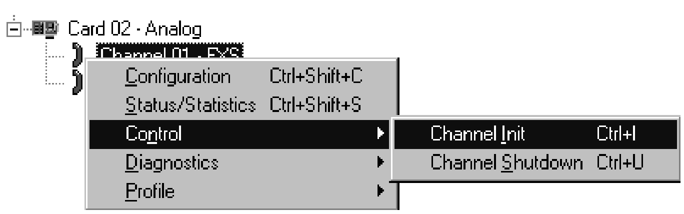
   a. Double-click on the CARD icon to display the Channel icons.

   b. Click on the Channel icon and click on the *right* mouse button. A popup menu will appear. Click on Control ➧ Channel Shutdown:



   The selected channel will be shut down.

   c. Click on the Channel icon again and click on the *right* mouse button. A popup menu will appear. Click on Control ➧ Channel Init:



   The channel will be initialized.

d. 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. Click on the Card icon and click on the *right* mouse button. A popup menu will appear. Click on Control and select Reset Card:

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.

Diagnostics

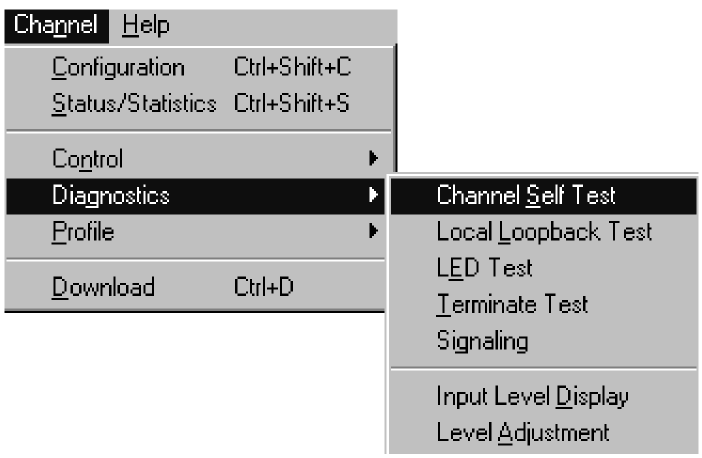
The following diagnostics functions are built into the V/IP software for testing individual channels:

- Channel Self Test
- Local Loopback Test
- Other Items on the Diagnostics Menu

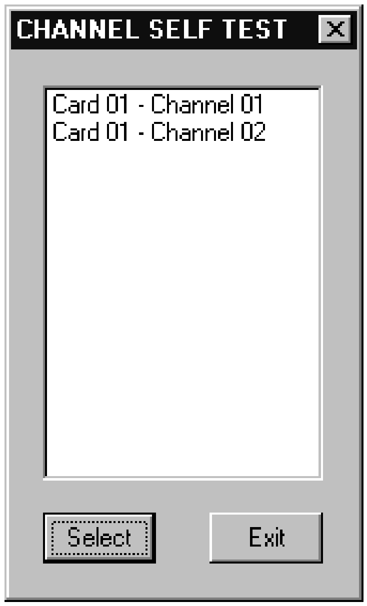
Channel Self Test

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

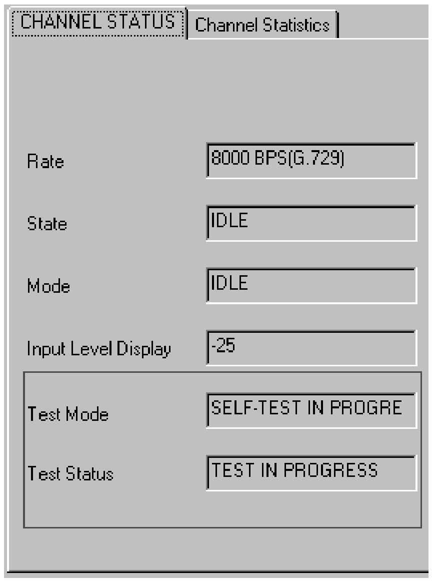
1. Select Channel ➧ Diagnostics ➧ Channel Self Test from the menu bar:



2. Select which channel to test:

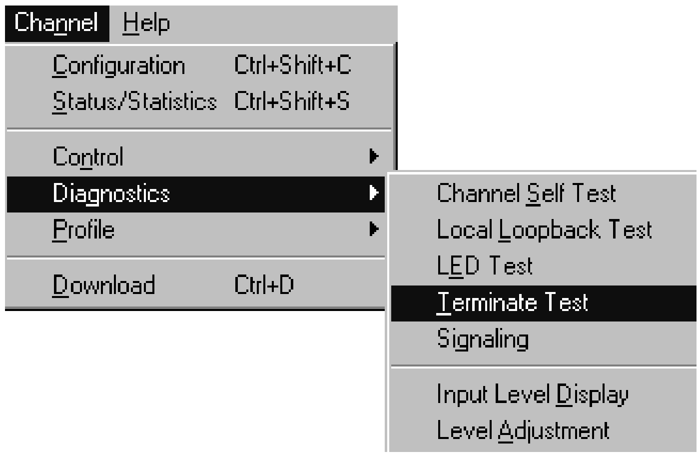


3. This is a continuous test. The status of the test can be viewed in the Channel Status display:

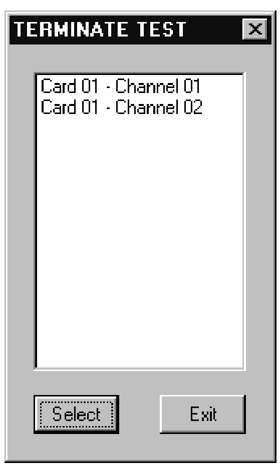


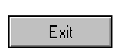
See *Viewing the Status of Voice/Fax Channels* on page 4-3 if you need the procedure for viewing the Channel Status display.

4. To stop the self test, select Channel ➧ Diagnostics ➧ Terminate Test from the menu bar.



Select the channel to terminate the self test.

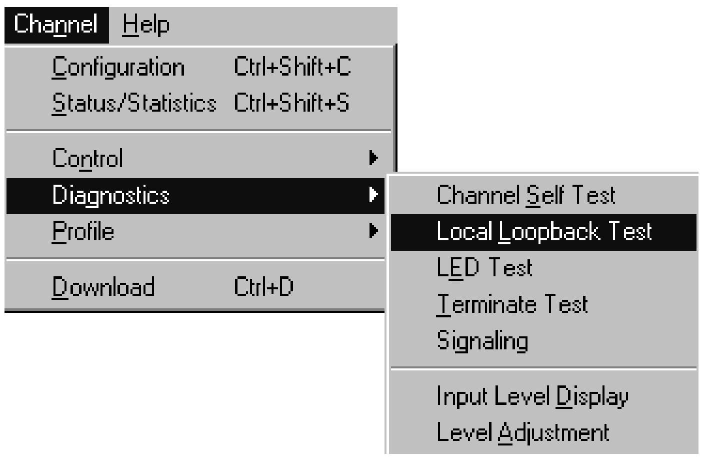


The test will be terminated. Click on  to clear out the Terminate Test display.

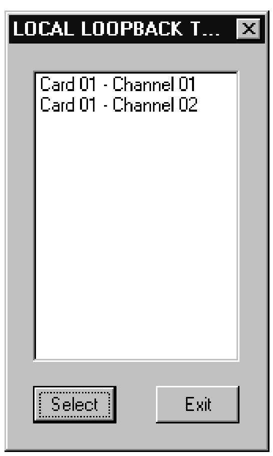
Local Loopback Test

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.

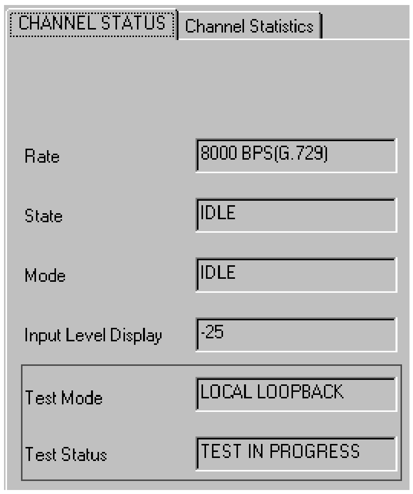
1. Select Channel ➧ Diagnostics ➧ Local Loopback Test from the menu bar:



2. Select the channel to test:



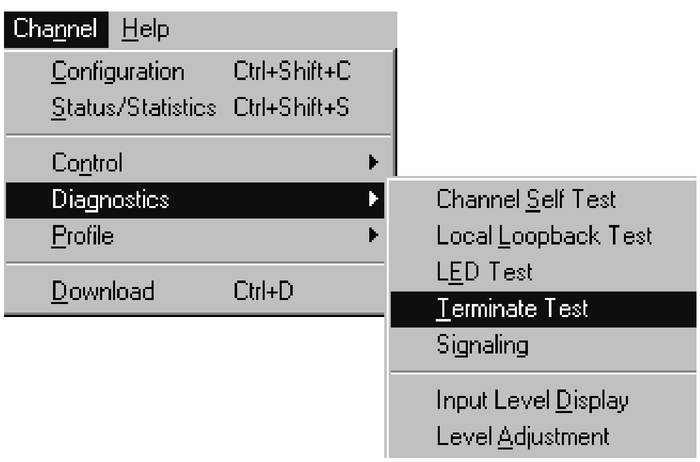
3. You can confirm that the loopback is in place from the Channel Status display:



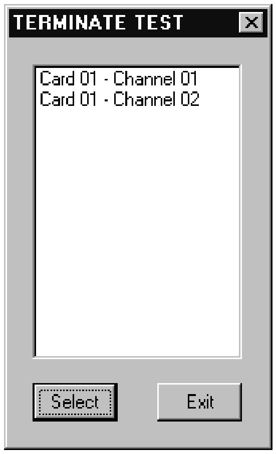
See *Viewing the Status of Voice/Fax Channels* on page 4-3 if you need the procedure for viewing the Channel Status display.

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.

4. To return the channel to normal operation, select Channel ➧ Diagnostics ➧ Terminate Test from the menu bar:



Then, select the channel that was being tested:



The loopback will be terminated. Click on 

to clear out the Terminate Test display.

Other Items on the Diagnostics Menu

You will find the following items on the Diagnostics Menu:

- LED Test. This function is reserved for future use.
- Signaling. This is a display that is used only for T1/E1 cards and has no function for analog Voice Interface Cards.
- Input Level Display. This function is used to assist in setting the channel input and output levels. The procedure for setting the levels – which uses the input level display – is in Section 5.
- Level Adjustment. This function is used to set the channel input and output levels. The level adjustment procedure is in Section 5.

Input/Output Level Adjustments 5

This section contains procedures for adjusting the input/output levels of the 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

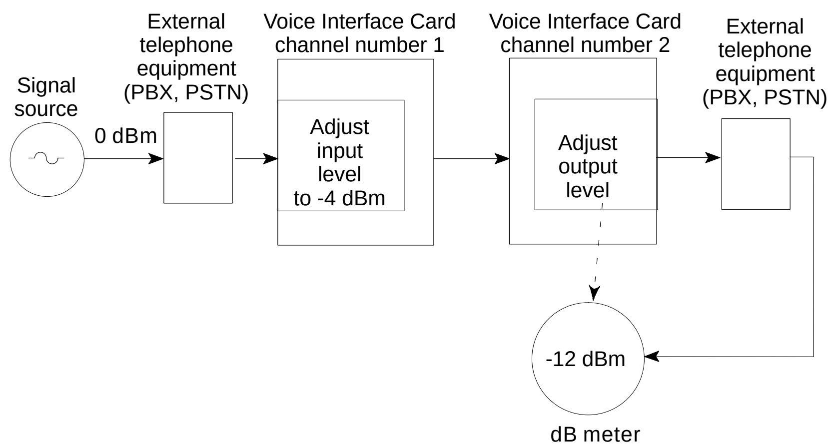
Voice Interface Card input/output levels are set at the factory for a default value of 0/0 dB. At some installations, the input/output levels need to be adjusted to compensate for external network losses and variations, to optimize the voice compression operation, and to provide a consistent communication 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

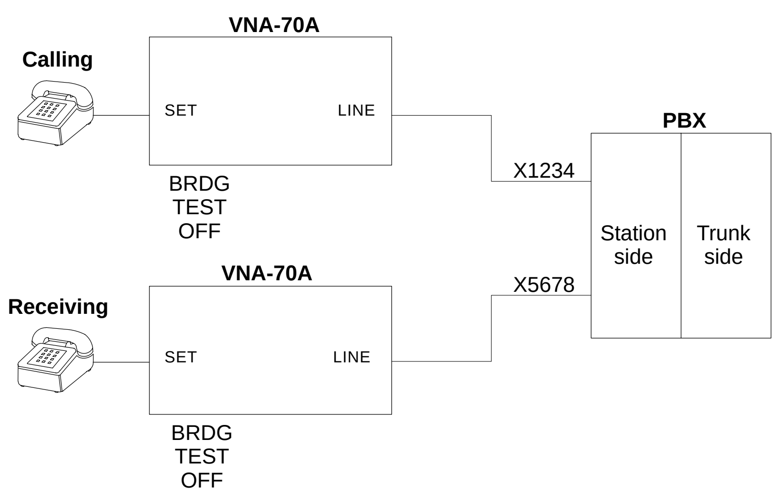
Before attempting to adjust the input output levels, configure them to the following settings:

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.

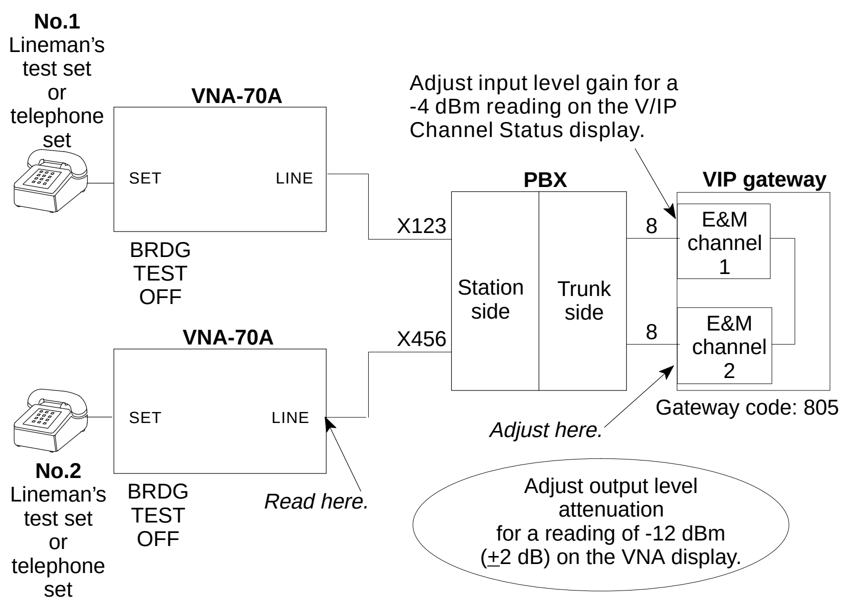
1. Use the test setup shown below. Initially, set both VNAs (Voice Network Analyzer) to BRDG, TEST and OFF. The extension numbers shown are fictitious, used here for reference only.
2. Lift the CALLING telephone set off-hook, wait for the dial tone, and dial station 5678. The RECEIVING telephone will 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

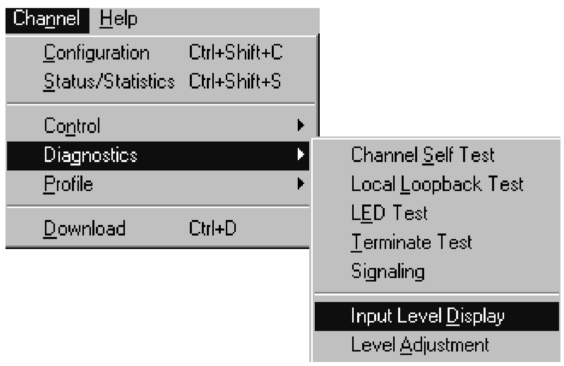
Many PBXs have selectable input/output level combinations for tie trunks, e.g., +7/-16, -16/+7 and 0/0 dB. The 0/0 dB level option should be selected. Also, PAD (2 dB) switching option, if available, should be selected.

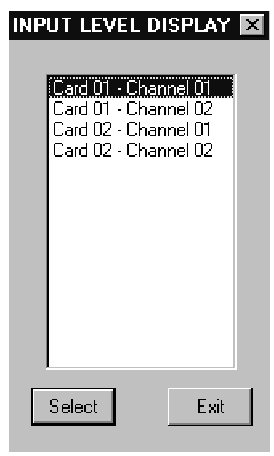
1. Use the test setup shown below. Initially, set both VNAs to BRDG, TEST and OFF. The extension numbers shown are fictitious, used here for reference only.

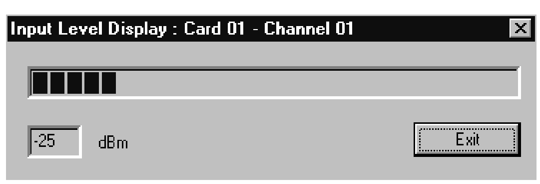


2. Select the first two E&M voice channels within the V/IP gateway to be adjusted. Set any additional voice channels to the busyout mode to insure that the test calls operate only through the channels under test.
3. Place a call from the No.1 telephone to the No.2 telephone. To do that, lift the No.1 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. Select the Input Level Display for channel No.1:







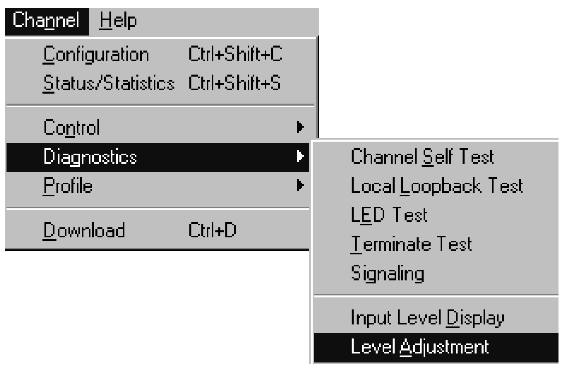
9. If the reading is more positive in the Input Level Display than -4 dBm, apply negative gain (input attenuation) in the amount that this reading is above -4 dBm.

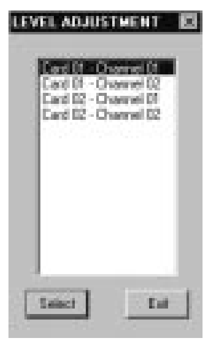
   Example: The reading is -1 dBm.

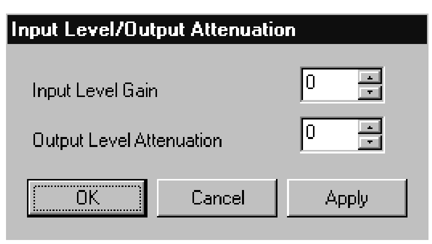
   The Input Level Gain for this channel is set to -2 dB.

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

   To adjust the input level, select the Level Adjustment menu:



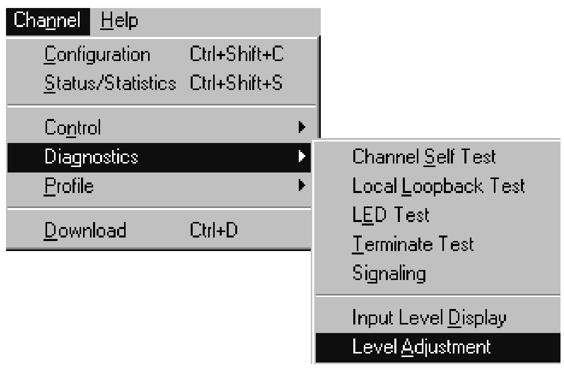


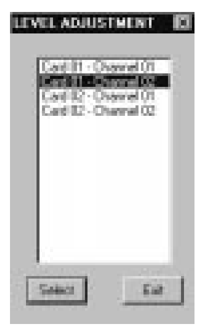


Adjust the Input Level Gain up or down until the channel input level is -4 dBm. (You will have to alternate between the Input Level Display and Level Adjustment menus.)

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 Output Level Attenuation parameter for channel No.2 (the channel that is connected to the channel just adjusted):





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. Select the Input Level Display as in step 8, but for channel number 2.
18. 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.

    To adjust the input level, enter the Level Adjustment 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 Output Level Attenuation parameter for 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).

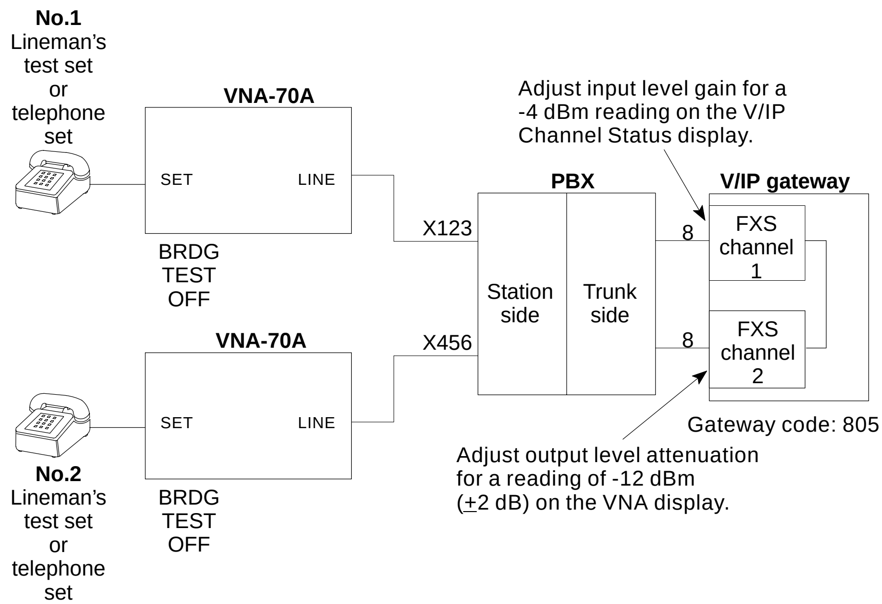
    Now that both channels are adjusted, test the voice quality. To do that, place both VNA SETUP switches to BRDG, take one telephone off-hook, dial access code 8, then dial the gateway code and channel number of the opposite channel, and then dial the extension number of the other phone. Talk to verify 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:

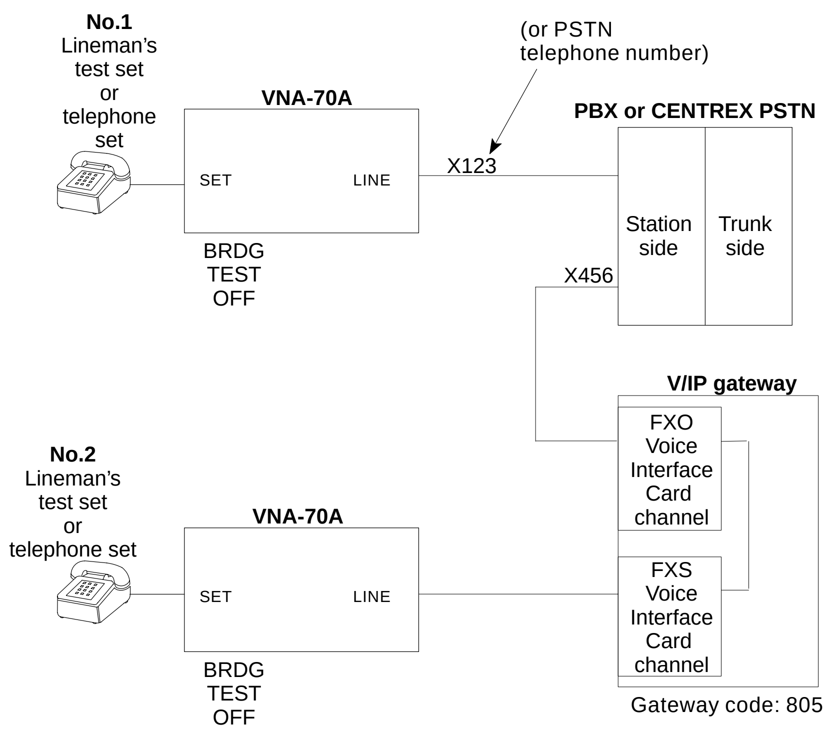


PBX Station or CENTREX PSTN Line Application Adjustments

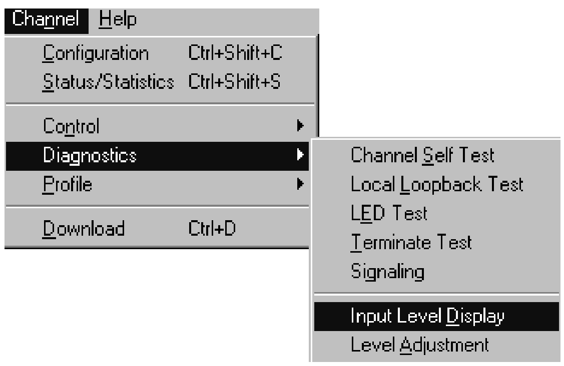
For this procedure, one interface will have to be temporarily converted for FXS operation. To do that, use a spare FXS interface Voice Interface Card, if available, as listed in the Equipment Required paragraph. If none is available, the local FXO card will have to be connected to a companion FXS card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

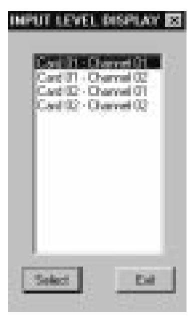
Note: If there are no single-line telephone circuits available from the PBX or CENTREX/PSTN, then the line assigned to the channel that is temporarily converted to FXS can be used for test purposes.

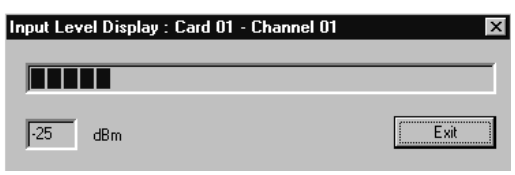
1. Use the test setup shown below. Initially, set both VNAs to BRDG, TEST, and OFF. The extension numbers shown are fictitious, used here for reference only.



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. Select the Input Level Display for channel No.1:

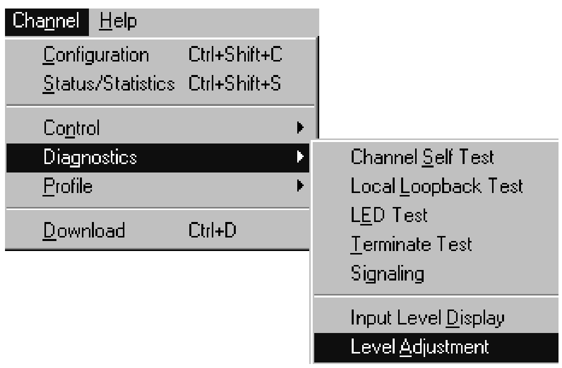


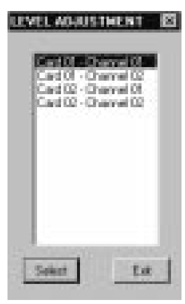


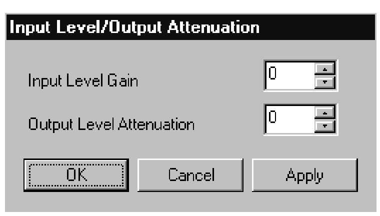


7. Observe the display on the PC screen and note the Input Level. If the reading is more positive than -4 dBm, apply attenuation (negative gain) in the amount that this reading is above -4 dBm. If the reading is more negative than -4 dBm, add gain (6 dB maximum) in the amount that the reading is below -4 dBm. Normally, station circuits require an Input Level 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 setting (-1 dB or lower). CENTREX or PSTN lines normally require an Input Level setting of +2 to +6 dB.

   To adjust the input level for the FXO channel, select the Level Adjustment menu:







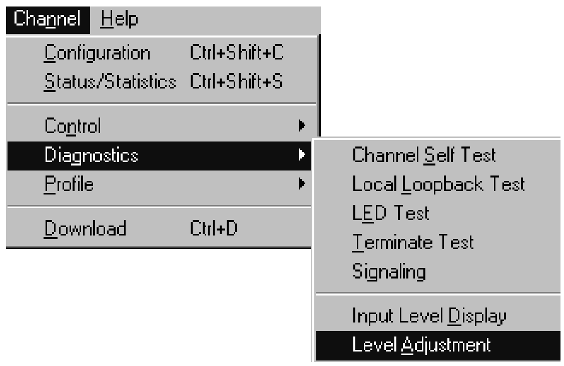
After the input level gain adjustment is completed, note the setting used to bring the input level to -4 dBm. You will need it for comparison when setting other FXO channels.

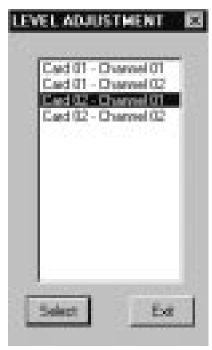
8. Read the value of the receive level on the No.2 VNA display. Note down this value for comparison when adjusting other FXO channels.

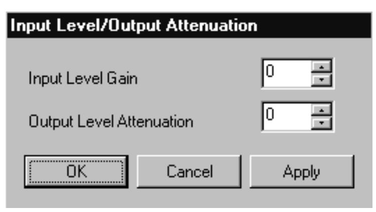
   You have now established the input level gain setting on the FXO channel. In the following steps, you will adjust the output level attenuation setting on the FXO channel.

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. Select the Input Level 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, select the Level Adjustment 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:





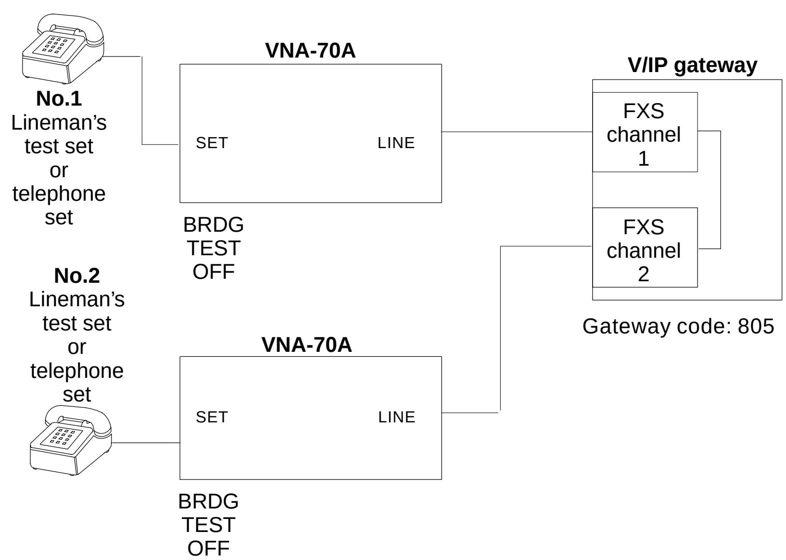


Set the output level attenuation until the receive level is approximately -12 dBm (±2 dB).

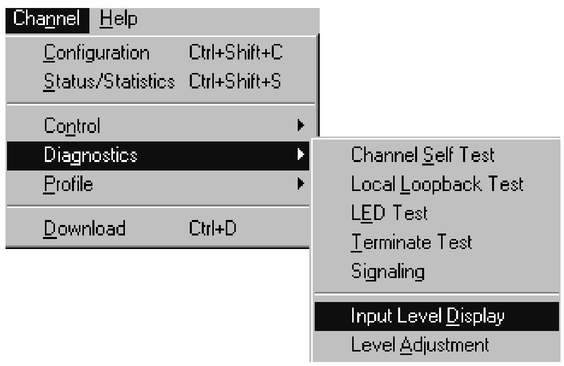
17. Now that both channels are adjusted, test the voice quality. To do that, place both VNA SETUP switches to BRDG, take the No.2 telephone off-hook, wait for the dial tone, dial the gateway code and FXO channel number, and dial extension 123. Talk to verify that there are no echoes. If there are echoes, 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.

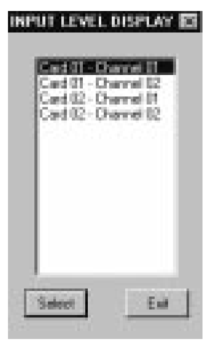
Single Line Telephone Application

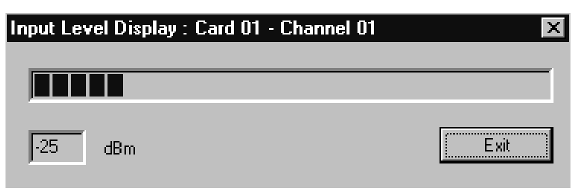
1. Use the test setup shown below. Initially, set both VNAs to BRDG, TEST, and OFF.



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. Select the Input Level Display for channel No.1:







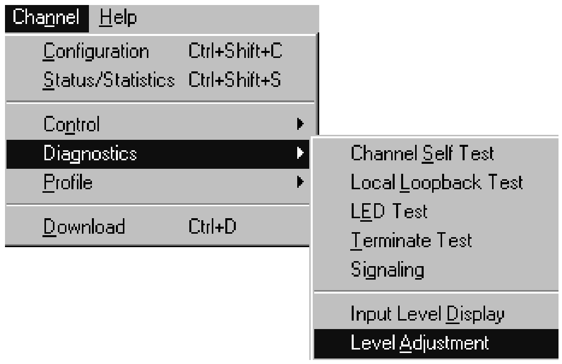
6. Observe the display on the PC screen and note the Input Level. If the reading is more positive than -4 dBm, apply attenuation (negative gain) in the amount that this reading is above -4 dBm.

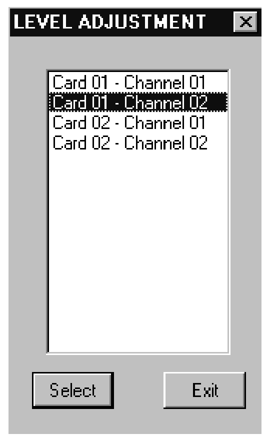
   Normally, this will require the Input Level Gain to be set between -2 and -4 dBm depending on cabling loss. (Note: if the Input Level Gain is currently set for 0 and the input level is below -10 dBm, then the FXS channel should be evaluated for a problem.)

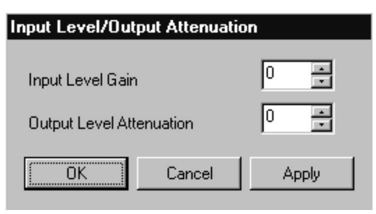
   To adjust the input level for the No.1 FXS channel, select the Level Adjustment menu as in step 7 of the PBX Station Application procedure.

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:







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. Select the Input Level Display as in step 5.
17. 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 more positive than -4 dBm.

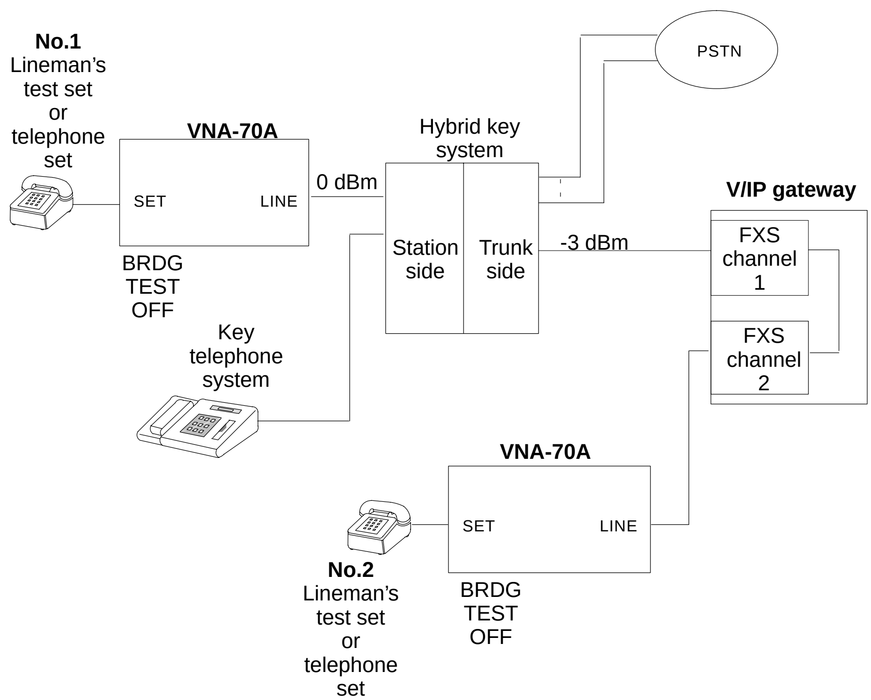
    To adjust the input level, follow the procedure of step 6 above. You will find that the setting is similar to that established in step 7.
18. Move to the voice/fax Output Level Attenuation parameter for FXS channel No.1, as in step 8 above.
19. Read the value of the receive level on the No.1 VNA display. 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.



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/-2dB with respect to 1004 Hz
Return Losses	
Echo:	$\geq$ 22 dB
Singing:	$\geq$ 18 dB
204 to 3404 Hz:	$\geq$ 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:	$\geq$ 58 dB
1004 to 3404 Hz:	$\geq$ 48 dB
Non-Linear Distortion (Multitone Signal) Second and Third Harmonics:	$\geq$ 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/-2dB 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, 2-wire:	≥ 35 dB
Echo Canceling, 2-wire:	≤ 16 milliseconds (≤ 1000 kilometers or 600 miles)
Signaling	
E-Lead Current Limit:	≤ 25 milliamperes
M-Lead Sensitivity:	48 V in series with ≤ 1400 Ω
DC Pulse Distortion:	≤ 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/-2dB 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

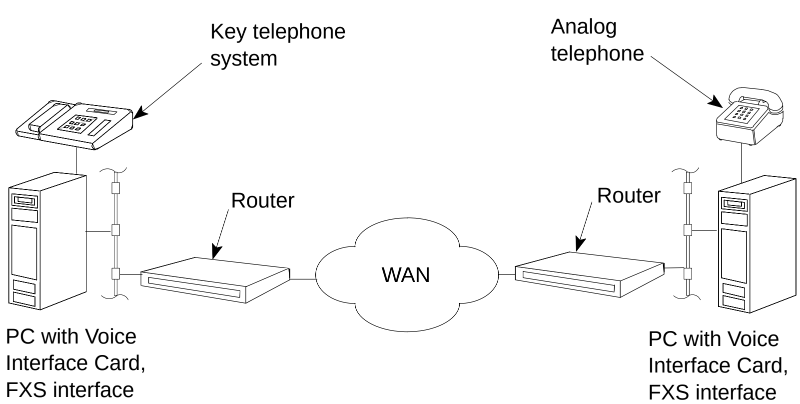
This appendix describes the FXS, E&M, and FXO interfaces.

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.

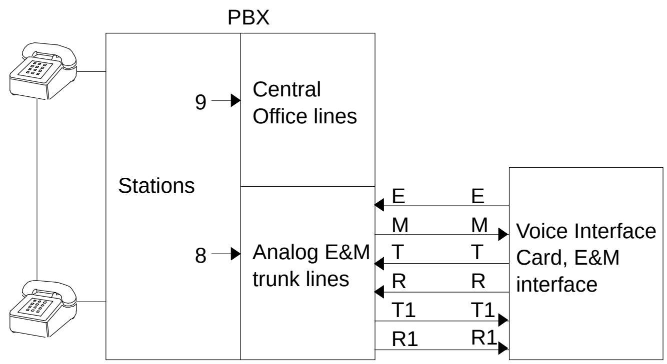
Here is a typical FXS application:



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).

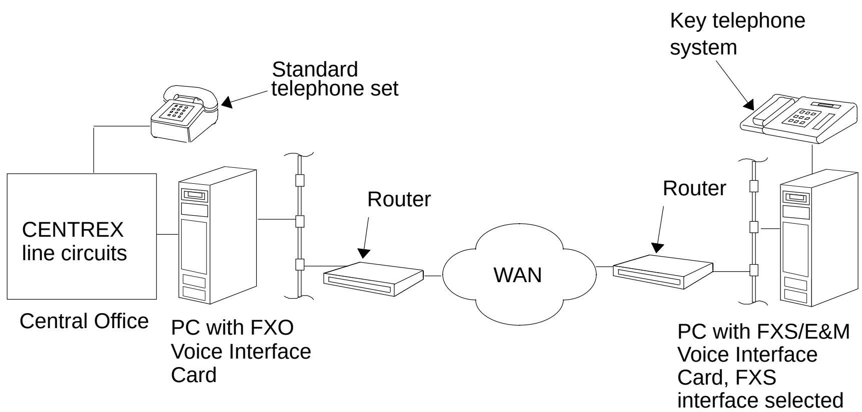
Signaling consists of an M lead and an E lead. In the E&M interface, the M lead is defined as the input, or receive, lead. The E lead is defined as the output, or transmit, lead. In some cases, an added return lead, SB, is used.

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:



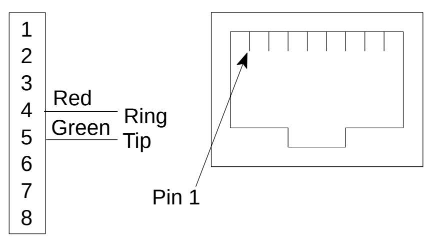
Telephone Interface Connectors

> ⚠ ***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.

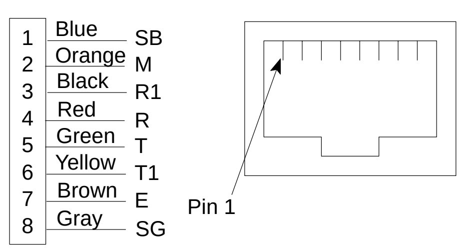
Here is the schematic diagram of the FXS connector:



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.

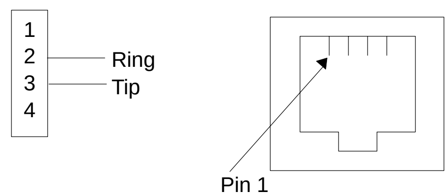
Here is the schematic diagram of the E&M connector:



FXO Interface

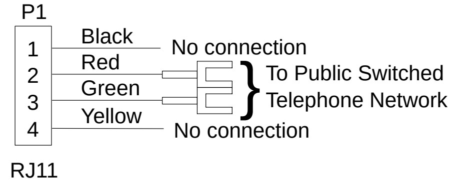
The FXO Voice Interface Card has RJ11C/W modular jacks. The card may include one or two of these connectors, depending on the model.

Here is the schematic diagram of the FXO connector:

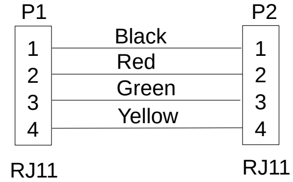


Caution

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

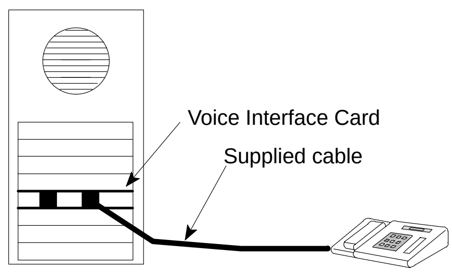


MICOM cable, part number 345-5495-014

Telephone Interface Cable Connections

FXS Interface

In most FXS interface applications, we recommend you use the cables supplied with the Voice Interface Card to connect the channels to the telephone equipment:

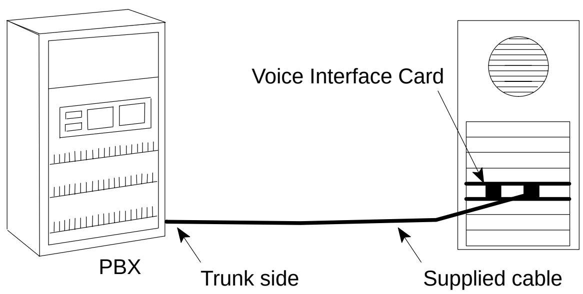


Caution

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.

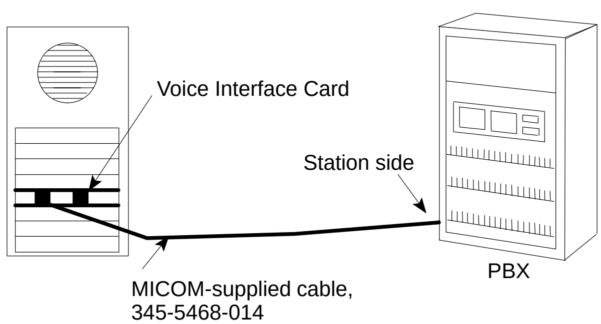
E&M Interface

For E&M interface applications, you should use the cables supplied with your Voice Interface Card. The supplied cables have 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:

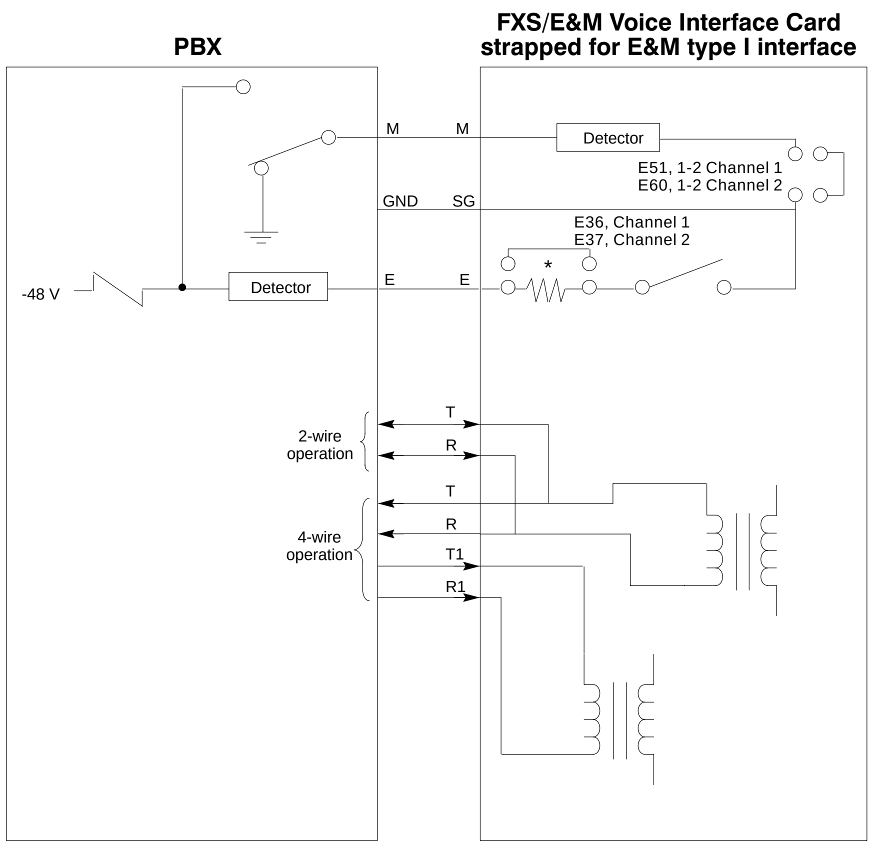


The other supplied cable has RJ11 connectors at both ends.

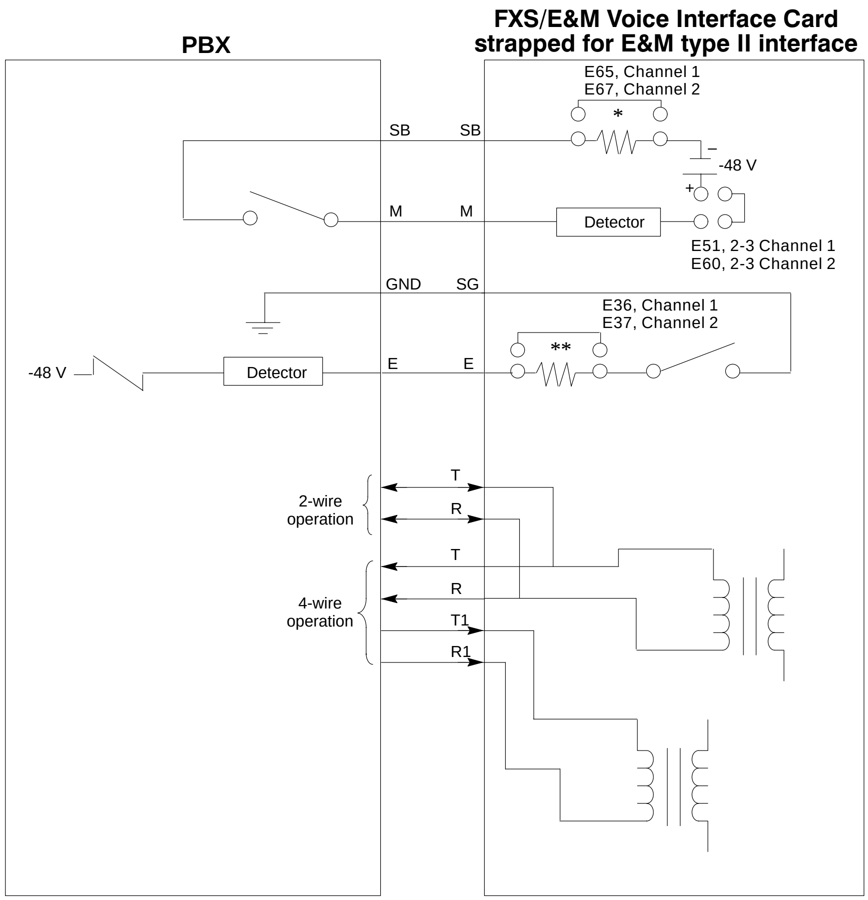
E&M Connection Diagrams

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.



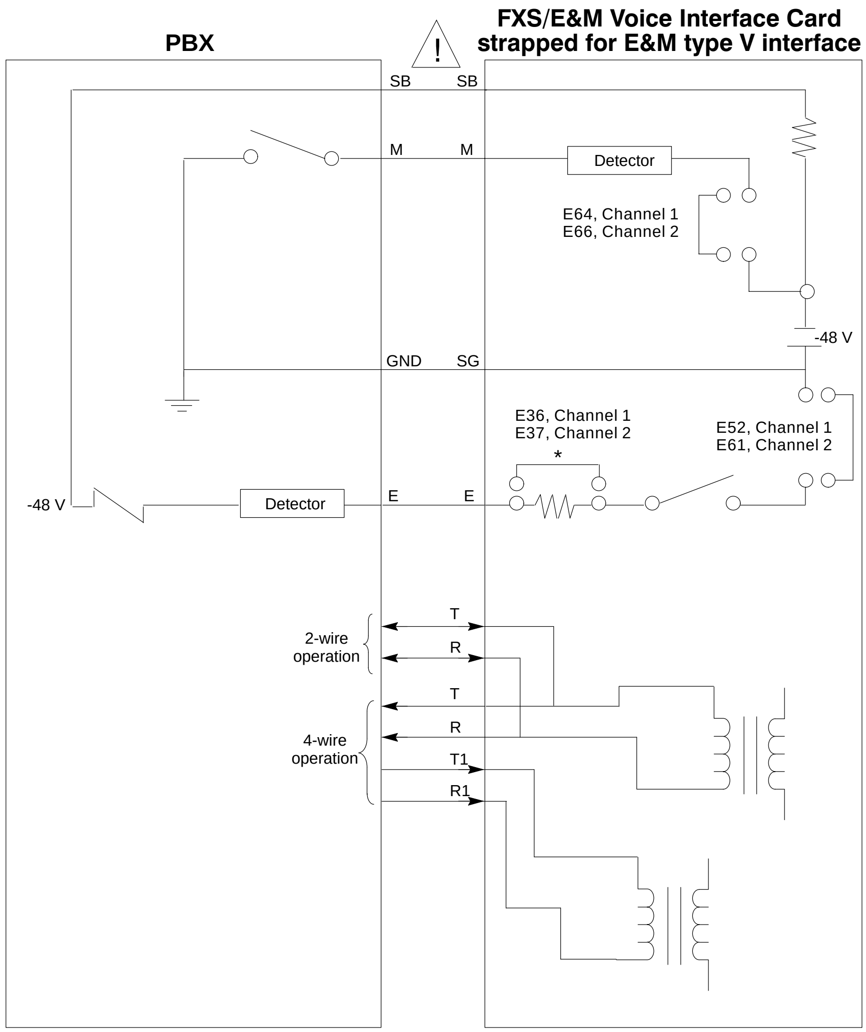
* This jumper is installed only when E lead loop resistance is too high due to long cable lengths.



* 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.

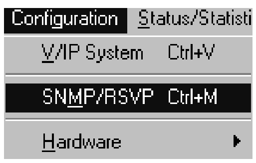
⚠ *Caution*

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

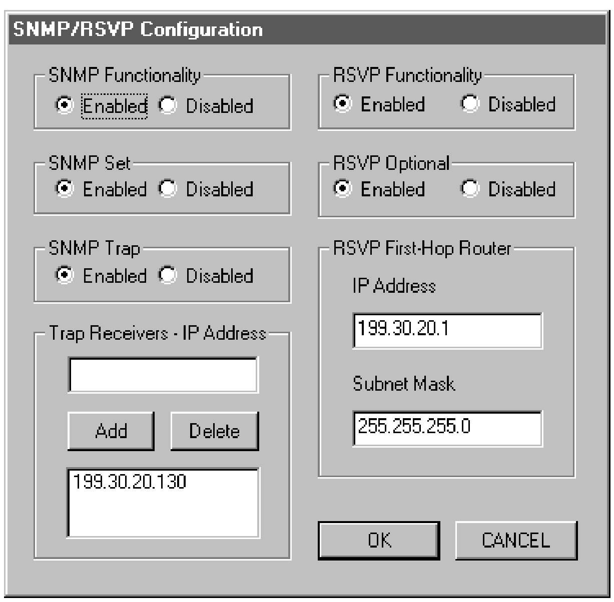
SNMP Management C

SNMP Configuration

SNMP on V/IP requires that the Microsoft SNMP Agent and the MICOM V/IP SNMP Extension Agent be installed. If not, the SNMP portion of the SNMP/RSVP Configuration form will be grayed out. To configure the V/IP gateway's SNMP operation, select Configuration ➧ SNMP/RSVP from the menu bar, as follows:



You will see the SNMP/RSVP Configuration form:



For information about SNMP parameters, use the V/IP Configuration Program online help. Select Help ➧ Help Topics from the menu bar, and look up *SNMP*.

MIB-II

The V/IP gateway supports MIB-II via the Windows 95 SNMP stack.

Enterprise MIB Extensions

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

The `micomvip.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 Extensions 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, the FCC approves this equipment, with the exception of the FXS interface, 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-2001-ISA-FXO-UK (single channel FXO card tested for BABT compliance)
  VIP-2002-ISA-FXO-UK (dual channel FXO card tested for BABT compliance)
- 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.

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.

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.

Index

W

READER'S COMMENTS

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.

Manual Name V/IP Phone/Fax IP Gateway Analog Models Microsoft Windows 95 Part No. 800-1887-11 Rev. B

	Excellent	Good	Fair	Poor
How would you rate the manual overall?	❑	❑	❑	❑
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?	❑	❑	❑	❑
Is the *index* useful?	❑	❑	❑	❑

Did you find any errors in the manual? (Please reference page, paragraph, table or figure number) ______________________________

How might we improve this manual? ______________________________

____

____

____

____

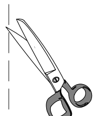
Name ______________________ Title ______________________

Company Name ______________________________

Address ______________________ Telephone (   ) ______________

____

Thank you for taking the time to fill out this form.



PCR 1077E

FOLD AS MARKED AND TAPE CLOSED BEFORE MAILING.
PLEASE DO NOT STAPLE.

Fold Here

From:________________________________

____

____

NO POSTAGE
NECESSARY
IF MAILED
IN THE
UNITED STATES

BUSINESS REPLY MAIL

FIRST CLASS PERMIT NO. 906 SIMI VALLEY, CA 93062

POSTAGE WILL BE PAID BY ADDRESSEE

MICOM Communications Corp.
ATTENTION: Manager, Technical Publications
4100 Los Angeles Avenue
Simi Valley, CA 93063–9949