# Host Media Processing Modular Line Interface Board

## TECHNICAL MANUAL

Documentation Revision 0.0: August 22, 2013


266M002

This manual applies to the PCI and the PCI Express versions of the board. Because of the differences between the bus interface, there are minor differences in the boards These differences have been noted where appropriate.

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The Host Media Processing Modular Line Interface Board Technical Manual





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800-356-9148
• 4800 Curtin Drive • McFarland, WI 53558 •
• 4145 North Service Road, Suite 200 • Burlington, Ontario L7L 6A3 •
• 266M002 •

## FCC Part 15 Requirements

WARNING: This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance with the instruction manual, may cause interference to radio communications. Operation of this equipment in a residential area is likely to cause interference in which case the user at his own expense will be required to take whatever measures may be required to correct the interference.

The authorized repair center is:

American Tel-A-System, Inc.
800-356-9148
4800 Curtin Drive
McFarland, WI 53558

There are no user serviceable components on the board. All repairs should be accomplished by returning the board to Amtelco with a description of the problem.

**WARNING:** This device contains Electrostatic Sensitive Devices. Proper care should be taken when handling this device to avoid damage from static discharges.

# Product Safety

The telephone cable(s) must remain disconnected from the telecommunications system until the card has been installed within a host which provides the necessary protection of the operator.

If it is subsequently desired to open the host equipment for any reason, the telephone cable(s) must be disconnected prior to effecting access to any internal parts which may carry telecommunications network voltages.

**Note: Only ports on FXO modules are intended to be connected directly to the PSTN network. Ports on FXS <u>should never</u> be connected directly to the PSTN network**.

## FCC Part 68 Registration

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the rear of this board is a label that contains among other information, a product identifier in the format US:AAAEQ##XXXX. If requested, this number must be provided to the telephone company.

This equipment is registered with the FCC under Part 68 as a component device for use with any generic PC Type computer or compatible. In order for FCC registration of this product to be retained, all other products used in conjunction with this product to provide your telephony function must also be FCC Part 68 registered for use with these hosts. If any of these components are not registered, then you are required to seek FCC Part 68 registration of the assembled equipment prior to connection to the telephone network. Part 68 registration specifies that you are required to maintain the approval and as such become responsible for the following:

- any component device added to your equipment, whether it bears component registration or not, will require that a Part 68 compliance evaluation is done and possibly that you have testing performed and make a modification filing to the FCC before that new component can be used;

- any modification/update made by a manufacturer to any component device within your equipment, will require that a Part 68 compliance evaluation is done and possibly that you have testing performed and make a modification filing to the FCC before the new component can be used;

- if you continue to assemble additional quantities of this compound equipment, you are required to comply with the FCC's Continuing Compliance requirements.

The network Interface Jack for this equipment is an RJ45C.

If this Amtelco HMP Modular Line Interface Board causes harm to the network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advanced notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advanced notice in order for you to make necessary modifications to maintain uninterrupted service.

In case of trouble, you may be required to disconnect the board from the telephone lines until the problem is resolved.

If trouble is experienced with this HMP Modular Line Interface Board, for warranty or repair information please contact:

American Tel-A-System, Inc.
800-356-9148
4800 Curtin Drive
McFarland, WI 53558

If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

There are no user serviceable components on the board. All repairs should be accomplished by returning the board to Amtelco with a description of the problem.

Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

If your premise has specially wired alarm equipment connected to the telephone line, ensure the installation of this HMP Modular Line Interface Board does not disable your alarm equipment. If you have questions about what will disable alarm equipment, consult your telephone company or qualified installer.

Connection to telephone company coin service is prohibited.

**Canadian Customers**
CP-01, Issue 8, Part 1
Section 14.1

**Notice:** “The industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements as prescribed in the appropriate Terminal Equipment Technical Requirements document(s). The Department does not guarantee the equipment will operate to the user’s satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs of certified equipment should be coordinated by a representative 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.

The PC chassis containing this device shall be placed in a secured location with access restricted to qualified service personnel.

## European Approvals

### CE Approval

**EN55022 EMC declaration**

This is a class B product. In a domestic environment, this product may cause radio interference in which case the user may be required to take adequate measures.

No changes or modifications to the Host Media Processing Modular Line Interface card are allowed without explicit written permission from American Tel-A-Systems, Inc., as these could void the end user's authority to operate the device.

**Notice:** The PC chassis containing this device shall be placed in a secure location with access restricted to qualified service personnel.

# Declaration of Conformity

PCI Board
Model Number: 266L007 16 Port HMP Modular Line Interface

PCIExpress Board
Model Number: 267L006 16 Port HMP Modular Line Interface

Line Interface Modules
Model Number: 266L009 4 Port FXS Module
Model Number: 266L011 4 Port FXO Module

Standards to which the conformity is declared: EN55022, EN50082-1 and EN60950-1

The undersigned declares that the equipment specified above:

- conforms to the above Standards,
- is in conformity to all the essential requirements of Directive 1999/5/EC.

Manufacturer: Amtelco

Company name: American Tel-A-Systems Inc.
DBA - Amtelco

Address: 4800 Curtin Drive
McFarland, Wisconsin 53558
USA

Paul N Henning

Signature:
Printed Name: Paul N. Henning
Position: Director of Research and Development

Date: 13 August 2013

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# 1.0 Introduction

The Host Media Processing Modular Line Interface Board is designed to provide up to sixteen analog line interfaces for use in PC based telephony systems running Asterisk or similar software. The board can be populated with a mix of FXO and FXS modules, each of which has four interfaces. FXO interfaces are used to connect to standard phone lines while FXS interfaces connect to standard telephone sets. Each interface or port provides both audio transmission and control support for basic line functions such as hook status and ring generation and detection. Each port can be programmed to conform to various national standards and practices.

There is a version of the board in both the PCI bus and the PCI Express form factor. This manual covers both versions. For the purposes of this manual, the PCI and PCI Express busses will be referred to as the PCI bus except where it is important to differentiate between them.

Unlike earlier computer telephony bus standards such as H.100, Host Media Processing relies on the processor of the PC to move the audio data between ports and to handle the details of port control. Specialized software that runs on the PC is used to accomplish this.

Asterisk, one such software platform, was devised by Digium, Inc. to allow for the easy development of custom telephone systems. It is intended for add-in boards using the PCI or PCI Express form factor. A wide variety of boards are available from a number of different vendors that will run under the Asterisk software.

## 1.1 Features and Capabilities

This section presents an overview of the features and capabilities of the Host Media Processing Modular Line Interface Board.

### 1.1.1 The FXO and FXS Interfaces

Each module contains the circuitry for four interfaces. Currently modules are available with four FXO or four FXS interfaces, though modules for other analog interfaces may be developed. The FXO and the FXS modules are designed to function as the two ends of a standard analog telephone line which is often referred to as a plain old telephone service or POTS line. The FXO interface emulates a telephone and is meant to be connected to a phone company central office or a PBX. The FXS interface is meant to be connected to a standard telephone or station set and provides the functions of a PBX or central office switch.

The interfaces provide all the necessary POTS telephony functions, that is the FXO module provides the switchhook control, ring detection and current detection, while the FXS module supplies talk battery, hook status detection and ring generation capabilities.

The module types can be mixed on a board to provide the desired number of each type of interfaces. For example, to provide the function of a small PBX, a board could be equipped with one FXO and three FXS modules to interface to four telephone lines and twelve telephone sets.

### 1.1.2 Asterisk

Asterisk is open-source software that was developed by Digium, Inc. that can be used to develop customized PBX's or other telephony applications. It runs under the Linux operating system. Boards from a number of vendors supporting a variety of telephony interfaces are available along with the necessary software drivers needed to work with

Asterisk.

Asterisk, unlike earlier telephony systems such as the H.100 bus, uses the processor of the host PC to carry out the switching and audio processing functions. This allows for the use of lower cost hardware which leads to very economical systems. Note that in Asterisk the term “channel” is synonymous with the terms “interface” or “port.”

### 1.1.3 Echo Cancellation

Some applications may require the use of echo cancellation. The Modular Line Interface Board comes equipped with the necessary circuitry to provide this function.

## 1.2 How to Use This Manual

The first five sections in this manual are organized in the order you should read and use them to get started with your HMP Modular Line Interface Board. We recommend that you begin with these three steps.

1. Read section 2.0 (Initialization) to familiarize yourself with the telephone line interfaces.

2. Follow the instructions in section 3.0 (Installation) and 4.0 (Software). These sections will allow you to get your board operating correctly within your system.

3. Read section 5.0 (Using the Modular Line Interface Board) for an overview of the features available with the Board.

The Appendices contain information on power requirements and obtaining assistance that may be helpful when installing your Modular Line Interface Board.

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# 2.0 Telephone Line Interfaces

## 2.1 The "POTS" Interface

The standard analog telephone interface, often referred to as a POTS or "plain old telephone service" line is the oldest form of telephony interface. It is the type of telephone that has commonly been supplied to residences and to small businesses where only a few telephone circuits are required. While various forms of digital services such as ISDN and VoIP have supplanted analog phone lines in many applications, they are still widely deployed both in the public telephone system and in PBX (private branch exchange) applications.

The various functions of an analog telephone circuit are sometimes summed up in the acronym BORSCHT, where the letters stand for battery, overvoltage protection, ringing, supervision, coder and decoder, hybrid, and test. While the originator of this acronym was probably just showing how clever he was, it does highlight the operation of a landline from the viewpoint of the service provider.

The most important of these function is the "talk battery." This is what actually powers the phone circuit. It's called battery because up until the last quarter of the twentieth century the phone line was driven by vast banks of batteries located in the phone company central office. These batteries supplied a nominal voltage of 48 volts, which remains the standard voltage for the public telephone network though the batteries have long since been replaced by power supplies. However, many PBX's operate at a lower voltage, most often 24 volts. The overvoltage protection in the acronym is the function of circuitry which prevents the battery voltage from getting so high that it might damage equipment.

A phone circuit consists of a pair of wires that form a loop. These wires are often referred to as tip and ring. This nomenclature is a holdover from the plugs used to complete connections on manually operated switchboards. The tip of this plug was connected to one of the wires in the circuit and the barrel of the plug or "ring" was connected to the other. By convention, the ring is referenced to ground while the tip is at a -48 volts.

In addition to the battery, the central office needs a means to alert the subscriber (the person at the phone end of the circuit) when a call is being presented. This is done by applying ringing to the circuit. This takes the form of an alternating current which has enough energy to ring a bell in the telephone set.

While ringing serves to get the attention of the person at the phone, supervision refers to signaling in the opposite direction. Normally, a telephone circuit is open, that is no current is flowing in the wires. When a switch is closed in the telephone set, current begins to flow which can be detected at the central office or PBX. This switch is often called the hook switch, a term that dates back to the old candlestick telephone sets where the receiver (the part you held to your ear) was suspended on a hook on the side of the phone. When the weight of the receiver was lifted, the hook would rise closing the circuit. When the receiver and microphone were both placed in the same handset, the hook was replaced by the button in the cradle, but the function was the same. Closing the switch is called going "off hook." Electronic devices that emulate telephone sets such as modems and analog telephony cards must still have a means of opening and closing the circuit. This may take the form of a relay, though today it is most often some sort of solid state device.

The coder/decoder function refers to the means of converting audio energy into an electric signal and vice versa. On a phone set, this is done by the microphone and receiver (a miniature loudspeaker). Electronic devices use a circuit called a CODEC.

A telephone circuit consists of a single pair of wires. This pair must carry the audio signal in both directions at the same time. However, in the phone set and switch, it is necessary to separate the incoming and outgoing signals. This is done by a circuit known as a hybrid. It should be noted that while this circuit works well enough for its purpose, it does not work perfectly, and may be a source of echo.

The final function, test, is really of concern only to phone companies which have various means of checking the integrity of the telephone circuit from the central office.

## 2.2 Types of Interfaces

The line as described above has two ends, one that lives in the central office switch or PBX, and the other which terminates in a telephone set or other terminal device. Each end has very different properties, and because of this require a different type of interface. In the terminology used by Asterisk these are called FXS and FXO interfaces respectively. where the "S" stands for station and the "O" stands for office. These terms can be a little confusing, as an FXO interface, while it is meant to connect to an FXO line, actually emulates a station set. The FXS interface is designed to provide the functions of a central office and connect to a station set.

An FXS interface, then, can supply the talk battery and ring voltage and detect current flowing in the line when the terminal device at the far end goes "off hook."

An FXO interface can detect the talk battery and the presence of ring voltage and has circuitry to go "off hook."

It is important that an FXS interface should never be connected directly to a circuit where the far end is also supplying talk battery as damage may result. Therefore an FXS interface should never be connected to an FXO line, and two FXS interfaces should not be connected together.

While connecting two FXO interfaces will not result in damage, the connection will not function either, because of the absence of the talk battery.

The modules for the Modular Line Interface Board come in two types, FXO (red) and FXS(green). Each of these modules contain four interfaces of that type. Modules can be mixed on the same board in any combination of up to four modules, that is one FXO and three FXS, two of each, or four all of the same type.

Because of PC power supply limitations, the FXS module produces a -24 V battery level.

## 2.3 Signaling

This section will describe in greater detail the nature and usage of the various signals described above.

As previously mention, the application of a ringing signal is used by the central office or PBX to indicate to a subscriber the presence of an incoming call. In North America, this ringing nominally takes the form of a 20 Hz. sine wave with a voltage of between 85 to135 volts. In Europe a frequency of 25 Hz. is commonly used, and PBXs often use 30 Hz. . While these are the nominal values, others have been used in the past and may sometimes be encountered in older central offices. One scheme that was popular for a while was used on party lines where the bell in each phone was tuned to a different frequency. This allowed the phone company to cause only one phone on the circuit to ring though they were all connected in parallel. One scheme called deciharmonic ringing used frequencies of 20 Hz., 30 Hz., 40 Hz., etc., while another scheme used frequencies of 16 1/3 Hz., 33 2/3 Hz., 50 Hz., and 66 1/3 Hz. With the demise of party lines, these schemes have mostly disappeared and frequencies other than 20 Hz. are rarely encountered in the public network.

The ringing signal is not applied continuously until a call is answered, but is interrupted by periods of silence. The pattern of ringing and silence is called the ring cadence. A number of different cadences are commonly used. In North America, the standard cadence consists of two seconds of ringing followed by four seconds of silence, which then repeats until the call is answered. PBXs often use a shortened cycle of one second of ringing and three seconds of silence. In countries outside of North America, other cadences are used, though a cadence of one second of ringing and two seconds of silence is fairly common.

In some applications, special cadences are used to indicate the nature of the incoming call. This is called "distinctive ringing" or "split ringing." An example would be .8 seconds of ringing, .4 seconds of silence, .8 seconds of ringing and 4 seconds of silence. PBXs often use split ringing to indicate whether a call is internal or external.

Ringing should never be applied to a phone circuit where the station is off-hook. At best this will result in an annoying noise and at worst may cause damage to the terminal equipment. The device supplying ringing should be designed to remove ringing as soon as an off-hook condition is detected.

An FXS module is used to generate ringing, while the FXO module can detect the ringing.

An off-hook signal by a station is used to indicate either that a call presented to the station has been answered, or that the station wishes to originate a call by dialing. Going back on-hook indicates that a call has been terminated by the station. In the middle of a call, there may be instances when the station wishes to get the attention of the switch. An example of this would be to initiate a transfer. This is normally done by the station briefly going on-hook. This is called a "hook-flash," "flash-hook" or just a "flash." The length of the flash signal will depend on the switch, but typically is between a half and a full second. European systems require a shorter flash time. Some telephones are equipped with a "flash" button that will automatically issue a timed hook-flash when pressed.

The FXO module can control the hook status, while the FXS module can detect changes in the hook status.

When originating a call, a station set will dial a set of numbers to indicate the desired destination. While originally, this was done with a rotary dial that generated a string of hook pulses, this method has largely been replaced by a system using pairs of tones. These tones are referred to as "Touchtones" or DTMF which stands for Dual-Tone Multi-Frequency. As DTMF signals are part of the audio stream, both the FXO and FXS modules can generate and detect them.

In the case where the party on the far end of a call has hung up, the central office switch or PBX may indicate this to the station, though this is not mandatory or universal on a standard loop phone circuit. Several methods are used. One consists of removing battery from the line for a period of time. This is called a "linebreak." Any interruption of battery longer than 400 msec. is considered a disconnect signal. It should be noted that not all switches generate linebreaks. Another method consists of playing dial tone, indicating that the switch is ready to accept dialing for a new call.

## 2.4 DID Circuits

The FXS and FXO modules may be used on other types of circuits in addition to the standard POTS line. The most common and useful of these is the Direct Inward Dialing or DID circuit. On this type of circuit, a number of telephone numbers are associated with a single telephone line. When one of the telephone numbers is called, the central office will route the call to the line and send a portion of the telephone number, typically the last three or four digits, to the terminal equipment to identify the number being called. This type of interface has commonly been used by telephone answering services or PBXs to reduce the number of physical telephone circuits required while still providing unique service for each number. DID numbers are typically reserved in blocks of ten or one hundred contiguous numbers, for example, 555-1200 through 555-1299.

The digits sent by the central office are called address digits. Originally, they were sent in the form of dial pulses, much like a rotary phone, but in most cases this has been replaced with DTMF signaling because it is quicker and more efficient.

The DID circuit is unusual in that the battery is supplied by the terminal equipment rather than the central switch. It is also a one-way service, in that calls can only originate at the switch interface and not the terminal interface. The switch will go off-hook to indicate that a new call is being sent to the terminal. The address digits will follow shortly, usually within a few hundred milliseconds. When the terminal wishes to answer the call, the polarity of the talk battery is reversed and remains reversed until the terminal hangs up. The switch can indicate the end of the call by going on-hook. Note that because the terminal is providing battery, an FXS interface would be used to interface to a DID circuit. An FXO interface would be used for the switch end.

Some address signaling protocols require the sending of an acknowledgment signal from the receiving side known as a “wink.” This signal, consists of a brief reversal of the battery polarity. The returning of polarity to the normal state is used as a signal to the switch that it can begin sending digits.

A DID circuit that requires a wink is called a “wink-start” circuit. One that does not is called an “immediate start” circuit. There is also a variation of the wink-start protocol that is called “delay-dial.” The difference between wink and delay has to do with the timing of the battery reversal.

The timing of DID circuits is critical, and defined in the RS-464 standard. In immediate start, the central office can begin sending digits 65 milliseconds after the initial off-hook signal. In the wink-start protocol, the wink must be between 140 and 300 milliseconds long and can not occur until 100 milliseconds after the off-hook. If a wink is not received by the switch within 10 seconds, it will typically abandon the call. The address digits may begin 70 milliseconds after the end of the wink. In delay-dial, the reversal should occur no later than 150 milliseconds after the off-hook and end when the terminal is ready to receive digits. The reversal must be at least 140 milliseconds, but may be much longer than the 300 milliseconds of a wink.

As the signaling protocols are set by the driver, they are defined by entering the information in the appropriate configuration file.

## 2.5 Cabling & Power Considerations

Connections to the board are made through the four RJ-45 connectors on the rear panel of the board. There is one connector for each of the four modules, and each connector contains four pairs of wires, one for each port on the module. The connector pinout is described in section 3.2.

Power for the telephony signaling voltages is obtained from the PC's internal power supply so that no external supply is required. However, the PC supply must have enough capacity to handle the requirements. In most applications this will not be a problem. Consult Appendix A for power requirements.

# 3.0 Installation

This section describes how to install your Host Media Processing Modular Line Interface Board into your PC.

## 3.1 PC Requirements

The Amtelco Host Media Processing Modular Line Interface Board comes in both a PCI and a PCI Express version. As the boards conform to the PCI and PCI Express standards, there are no switches to set to configure the HMP Modular Line Interface Board's memory address, I/O addresses, or interrupt. The PC's BIOS will automatically configure the board at boot time to avoid conflicts with other boards in the system.

Before attempting to install the board in your computer, you must make sure that you have the correct version (PCI or PCI Express) for the connector in your PC's backplane. Attempting to plug a board in the wrong type of slot may damage both the board and the backplane. The type of board may be determined by examining the edge connector on the bottom of the board (see the board outlines in the figures).

As the PC power supply is used to generate the telephony signal voltages, you must determine that it can supply sufficient current to power the board. In addition, it must have both +3.3 V and +5 V supplies for PCI slots and +3.3 V and +12 V supplies for PCI Express slots.

Of course, the PC must also have enough memory and have a fast enough processor to run the operating system and software platform such as LINUX and Asterisk, that the board is to be used with.

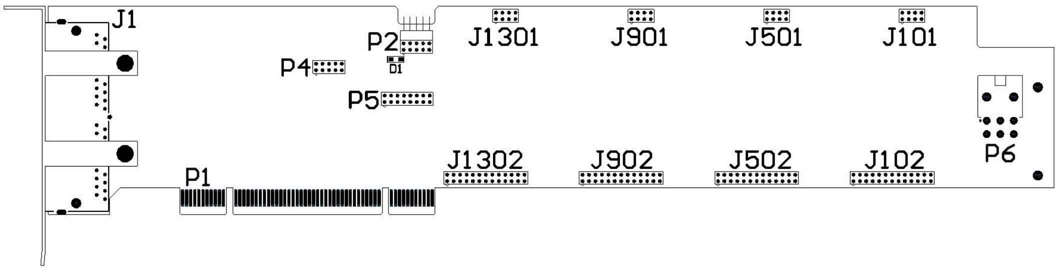


*Figure 1:* Location of Connectors and Headers for the PCI Modular Line Interface Board

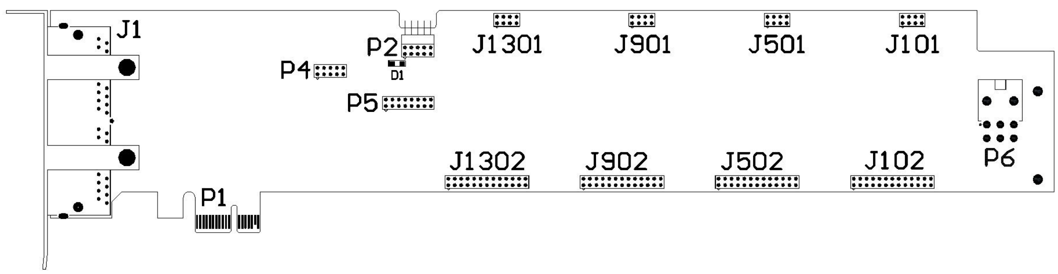


*Figure 2:* Location of Connectors and Headers for the PCI Express Modular Line Interface Board

## 3.2 Connectors: P1- P5, J1 and J101-J1302

**P1** The PCI or PCI Express connector. This plugs into the appropriate backplane connector.

**P2** Connector for synchronizing clocks on multiple boards.

**P4** Programmable logic programming connector. Do not use. For factory use only.

**P5** JTAG test connector. Do not use. For factory use only.

**P6** FXS power connector. Must have an ATX power cable connected if FXS modules are installed on the board.

**J1** Analog telephone connections. This connector is a quad RJ-45. One Connector is used for each module. There are four ports on the connector with the pinout listed below. See Figure 3.

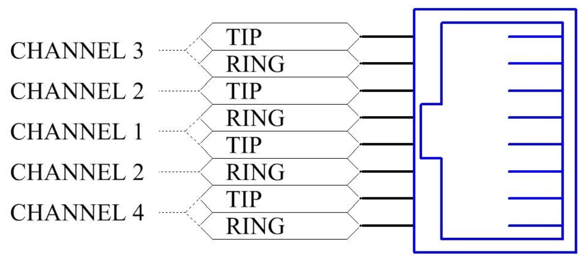


*Figure 3:* RJ-45 Pinout

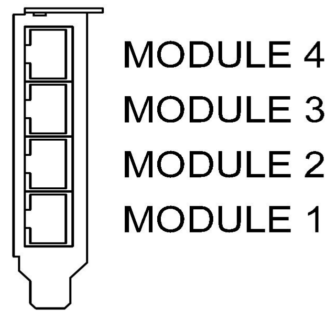


*Figure 4:* Rear Panel

**J101 & J201** Connectors for installing Module 1, ports 1-4

**J501 & J502** Connectors for installing Module 2, ports 5-8

**J901 & J902** Connectors for installing Module 3, ports 9-12

**J1301 & J1302** Connectors for installing Module 4, ports 13-16

## 3.3 Cabling

Connections to other pieces of telephony equipment are made via cables, one end of which is an RJ-45 plug that fits into the RJ-45 jacks on the rear panel of the board. The pinout of these jacks is given in *Figure 3* above. The termination and pinout of the other end will depend on the equipment to which the board is connected. Normal practice for a standard phone line is to use an RJ-11 type connector for each circuit. For such connectors an appropriate adapter cable or box will be necessary. A suitable cable assembly (266D013) may be ordered from Amtelco. One cable assembly is required for each module installed on the board (up to 4). Providing the proper cable is the responsibility of the user.

## 3.4 Installation

To install the HMP Modular Line Interface Board in your system:

1. Turn off the PC power. Remove the PC cover.

2. Do not connect the telephone cables. Verify that the board is the appropriate type for the PC backplane connector.

3. Install the FXO and FXS modules on the board as desired.

4. Insert the board into the chassis. Seat it properly in a PCI or PCI Express slot as appropriate in the PC chassis and tighten the screw in the back of the board to secure the board.

5. If one or more FXS modules are installed on the board, it is necessary to connect an ATX power cable to connector P6.

6. Reinstall the PC cover. Connect the PC to the mains supply using a socket-outlet with protective earthing connection and connect any additional protective earthing used.

7. Connect the telephone cable(s) to J1. The telephone cable terminates in an RJ-45 male connector.

If it is subsequently desired to open the host equipment chassis for any reason, the telephone cable must be detached prior to effecting access to any internal parts which may carry telecommunications network voltages.

The PC chassis containing this device shall be placed in a secure location with access restricted to qualified service personnel.

# 4.0 Software

This section describes the procedures necessary to install, configure and run the software for the Host Media Processing Modular Line Interface Board.

## 4.1 Installing the Drivers

The current drivers for the board may be downloaded from Amtelco's FTP site. The accompanying README file will give details of the procedure to install this driver. For details on accessing the FTP site, contact your Amtelco representative.

## 4.2 Configuration Files

In order for the driver to correctly set up the ports on the Modular Line Interface board, it is necessary to add information to several configuration files. Sample files are supplied with the driver software available on the FTP site as indicated in the accompanying README files.

The file **amtelco_hmp.conf** provides information which is used by the channel driver to configure the board. This file must be placed in the Asterisk configuration directory for the channel driver to function. It is used to configure ports for the interface type, proper signaling behavior, DID behavior and so on. A detailed explanation of the syntax used to set these configuration options is given in the sample file.

To include the Modular Line Interface board in the Asterisk dial plan, information will have to be added to the Asterisk configuration file **extensions.conf**. Examples of the various options for this file are given in the sample **extensions.conf.sample** file.

## 4.3 Running Asterisk

Asterisk is open source software that may be used to develop telephony applications. It may be downloaded from the site www.asterisk.org. Details of installing and running Asterisk are contained in various files in this package.

In addition, there is a variety of documentation and other information available at this site to assist the developer.

Note, that with the Modular Line Interface Board the Asterisk term "channel" is synonymous with the terms "port" and "interface."

# 5.0 Using the HMP Modular Line Interface Board

In addition to the basic capabilities of providing hook-status control and audio transmission, the Modular Line Interface Board has a number of advanced features that may be useful. This section describes how those features may be enabled and used.

## 5.1 Overview of the Advanced Features

In addition to interfacing to standards POTS lines or to connect to telephone sets, the Modular Line Interface Board may be used in a number of other applications. One of the common applications is to provide an interface to PBX's for add on systems such as voice mail or help desks. In such applications, it may be useful to pass information about the calling or called party through the use of address signaling. The Modular Line Interface Board can accommodate a number of different signaling formats and protocols.

Other uses include connecting to DID circuits from the public network or PBXs, or in ringing tie line, or security applications.

The Modular Line Interface Board also supports a number of advanced features such as echo suppression that may be useful in a variety of applications.

## 5.2 Features Provided by Asterisk

Many of the standard telephony features are provided as part of the Asterisk environment. These include the ability to detect and generate the DTMF tones used for signaling as well as the generation of the various call progress tones such as busy, reorder, and audible ringback. In addition, Asterisk provides the means of playing announcements or recording and playing back speech as in voice-mail. The user should consult the Asterisk documentation to learn how to use these features.

## 5.3 Configuring the Software for Features

Several steps are necessary when configuring the board. The Asterisk file **extensions.conf** must be modified to include the HMP Modular Line Interface board in the Asterisk dial plan. The file **amtelco_hmp.conf** must be included in the Asterisk configuration directory /etc/asterisk. This file contains information used to configure the type and operation of the board. Details on these files are given in sample files in the HMP driver package (see section 4.2).

## 5.4 Advanced Features

This section describes how to use the various features supported by the board and the associated software that go beyond the standard features provided by Asterisk.

### 5.4.1 Address Signaling Protocols

Line interfaces are sometimes used to interface to PBX's or channel banks. In these applications, a single interface may be used to direct calls to more than one destination or for more than one phone number. In these applications which are sometimes called Direct Inward Dialing or DID, one or more digits of information is sent by the originating interface to indicate the ultimate destination. These digits are referred to as "address" digits and can be sent as either DTMF or MF-R1 signals. There are several standard protocols that define the timing of the digits, i.e. the time from the off-hook signal until the digits can begin and the time between digits, and the form of the acknowledgment or ready signal sent by the destination interface.

The acknowledgment, if required, takes the form of a momentary battery reversal signal or “wink.” If no acknowledgment is required before the digits are sent, it is referred to as “immediate start”. If a “wink” is required, it is referred to as “wink start.” The “wink” is typically 200 msec. long. A variation called “delay dial” starts the wink and ends it when the digit detector is ready to accept digits.

The option **didconfig** in the file **amtelco_hmp.conf** must be set to appropriate values for DID operation.

## 5.5 Echo Cancellation

Telephony systems may under some circumstances experience the phenomena of excessive echo where speech is reflected from the far end with a noticeable delay. This is particularly true when the call involves VoIP or satellite links. To combat this echo, the HMP Modular Line Interface board is equipped with an echo cancellation module that is plugged onto the board.

This module may also provide additional capabilities such as the detection of DTMF digits on the board which may enhance system performance.

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# Appendix A: Environmental Specifications

The Host Media Processing Modular Line Interface Board meets the following environmental specifications:

TEMPERATURE EXTREMES:

Operating: 0°C (+32°F) to +50°C (+122°F).
Storage: –40°C (–40°F) to +70°C (+158°F).

AMBIENT HUMIDITY:

All boards will withstand ambient relative humidity from 0% to 95% non-condensing in both operating and storage conditions.

MECHANICAL:

All E&M boards conform to the PCI-SIG mechanical specifications for PCI or PCI Express cards.

ELECTRICAL REQUIREMENTS:

PCI HMP Modular Line Interface Board:

| | |
|---|---|
| +3.3 volts | 60 mA typical, 100 mA maximum w/o echo canceller |
| | 300 mA typical, 800 mA maximum with echo canceller |
| +5 volts | 1.0 A typical, 1.5 A maximum |

PCI Express HMP Modular Line Interface Board:

| | |
|---|---|
| +3.3 volts | 100 mA typical, 150 mA maximum w/o echo canceller |
| | 350 mA typical, 850 mA maximum with echo canceller |
| +12 volts | 500 mA typical, 750 mA maximum |

MTBF:

50,000 hours.

# Appendix B:
# Service Information

If problems should arise with your HMP Modular Line Interface Board or technical assistance be required, call Amtelco at 1-608-838-4194 ext. 168.

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