MediaPack™ User's Manual

for MP-124, MP-108, MP-104 and MP-102

VoIP SIP Gateways

Version 4.0

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US: AudioCodes Inc, 2890 Zanker Road, Suite # 200, San Jose, CA 95134.
Tel: 408-577-0488 - Fax: 408-577-0492

International: AudioCodes Ltd, 4 HaHoresh Street, PO Box 14, Yehud 56470, Israel.
Tel: +972-3-539 4000 - Fax: +972-3-539 4040

China: AudioCodes Ltd, Room 209 Tower A, Grand Pacific Building,
8A Guanghua Road, Chaoyang District, Beijing 100026, P.R.China
Tel: 86-10-6581-6622 - Fax: 86-10-6581-5599

Technical Support: support@audiocodes.com

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used, and only Industry standard terms are used throughout this manual. The $ symbol indicates hexadecimal notation.

Trademarks

Notice

This **MP-1xx/SIP** User's Manual describes the AudioCodes MediaPack Series **MP-124** 24 port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port, referred to collectively as the **MP-1xx**, supported by software version 4.0 Beta1, and enabling Users to send voice fax and data over the same IP network.

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, AudioCodes cannot guarantee accuracy of printed material after the Date Published nor can it accept responsibility for errors or omissions.

For **Technical Support** please contact:
e-mail: support@audiocodes.com
In the US, fax 408-577-0492
In other countries, fax +972-3-539-4040

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FCC Notice to Users

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Safety Notice

Installation and service of this gateway must only be performed by authorized, qualified service personnel.

Telecommunication Safety

The safety status of each port on the gateway is declared and detailed in the table below:

Ports	Safety Status
Ethernet (100 Base-T)	SELV
FXS	TNV-3

TNV-3: Circuit whose normal operating voltages exceeds the limits for an SELV circuit under normal operating conditions and on which over voltages from Telecommunication Networks are possible

SELV: Safety extra low voltage circuit.

Declaration of Conformity

We AudioCodes Ltd

Declare under our sole responsibility that the products:

MP-1xx/FXS

To which this declaration relates, is in conformity with the following standards: EN 55022 1998, EN 50024 1998, EN 60950 1992 + Amendments 1, 2, 3 & 4

As described in the European Directives: 89/336 (EMC),
73/23 (Safety),
93/68 (Safety).

Yehud, Israel, 28 Jun 2001
I. Zusmanovich
Compliance Eng.

Este producto està en conformidad con la directiva Europea 89/336, 73/23, 93/68
Dette produkt er i overensstemmelse med Europæiske Direktiver 89/336, 73/23, 93/68
Dieses Produkt ist konform mit der europäischen Richtlinie 89/336, 73/23, 93/68
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Questo prodotto è conforme con la normativa europea 89/336, 73/23, 93/68
Este producto està em conformidade com as Diretrizes Européia 89/336, 73/23, 93/68
Tuote on eurooppalaisen säännöstön mukainen 89/336, 73/23, 93/68
Denna product följer europeiska direktiv 89/336, 73/23, 93/68
Το προιόν, είναι σύμφωνο με τους κανονισμούς της Ευρωπαικής Κοινώτητας 89/336, 73/23, 93/68
Tæki þetta er í samræmi við tilskipun Evrópusambandsins 89/336, 73/23, 93/68
Dette produktet er i samhørighet med det Europeiske Direktiv 89/336, 73/23, 93/68

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MP-102, MP-104, MP-108 and MP-124 SIP Media Gateways

1 Overview

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

This section provides an overview of the features and functionality of the MP-124, MP-108, MP-104 and MP-102 telephony Media Gateways

1.1 Introduction

This document provides the User with the information about installation, configuration and operation of the **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port VoIP Media Gateways. As these units have similar functionality, except for the number of channels and some minor features, they are referred to collectively as the **MP-1xx**. It is expected that the readers are familiar with regular telephony and data networking concepts.

1.2 Gateway Description

The **MP-1xx** telephony Media Gateway provides excellent voice quality and optimized packet voice streaming over IP networks. The product enables voice, fax and data traffic to be sent over the same IP network. It is based on AudioCodes award-winning, field-proven TrunkPack design using the AudioCodes well-established DSP voice compression technology.

The **MP-1xx** incorporates up to 24 analog ports for connection, either directly to an enterprise PBX (**MP-10x**/FXO), to phones, or to fax (**MP-1xx**/FXS), supporting up to 24 simultaneous VoIP calls.

Additionally, the **MP-1xx** units are equipped with a 10/100 Base-T Ethernet port for connection to the LAN.

The **MP-1xx** Gateways are best suited for small to medium size enterprises, branch offices or for residential Media Gateway solutions.

The **MP-1xx** Gateways enable Users to make free local or international telephone/fax calls between the distributed company offices, using their existing telephones/fax. These calls are routed over the existing IP Internet or Intranet corporate data networks ensuring that voice traffic takes the minimum of space on the data network.

The **MP-1xx** Gateways are very compact devices, designed to be installed either as a desk-top unit (refer to Figure 1-2) or installed in a 19-inch rack (refer to Figure 2-1).

The **MP-1xx** supports SIP, H.323, MEGACO (H.248) and MGCP protocols, enabling the deployment of "voice over packet" solutions in environments where each enterprise or residential location is provided with a simple Media Gateway.

This provides the enterprise with a telephone connection (e.g., RJ-11), and the ability to transmit the voice and telephony signals over a packet network.

Additionally, for emergency use, the **MP-10x/FXS Gateway** provides a Life Line, connected to the unused pins on port #4 (or port #2 for **MP-102**), with a relay to an analog line, even if the **gateway** is powered off.

The layout diagram, Figure 1-5 on page 17, illustrates a typical **MP-108** and **MP-104** or **MP-102** VoIP application.

Figure 1-1: MP-124 VoIP Gateway

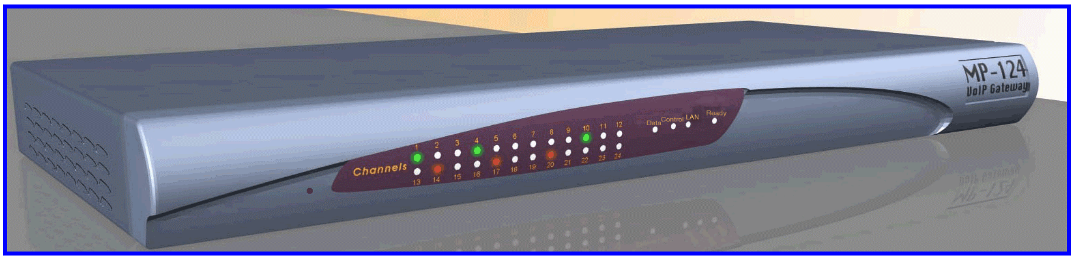


Figure 1-2: MP-108 Front View

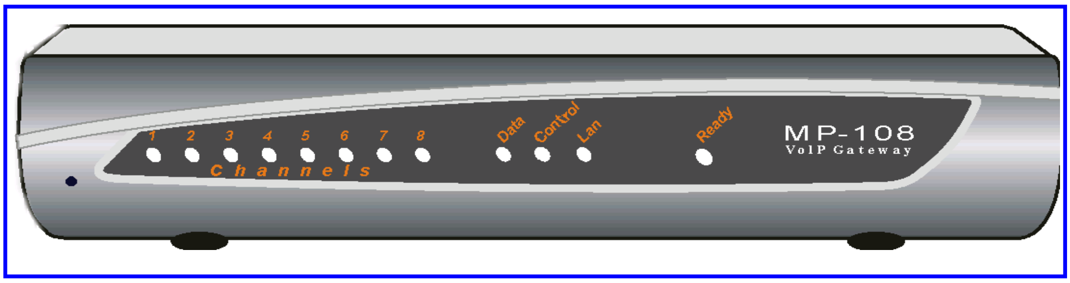


Figure 1-3: MP-104 Front View

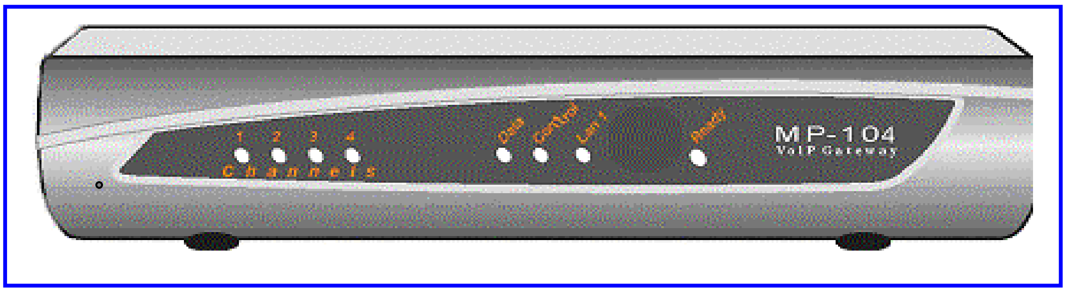


Figure 1-4: MP-102 Front View

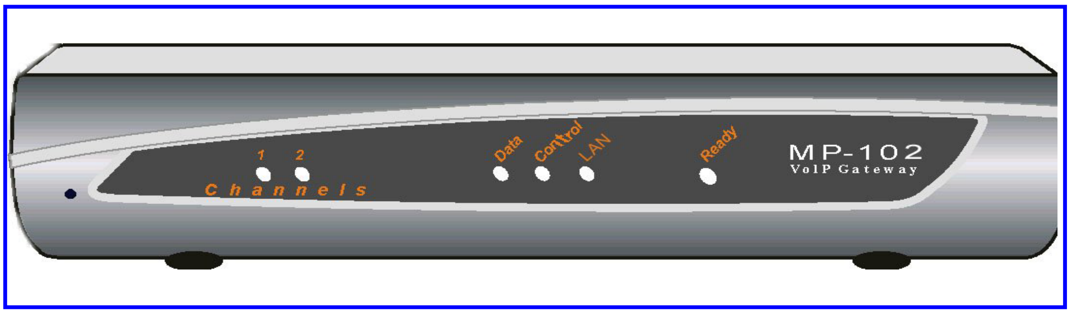
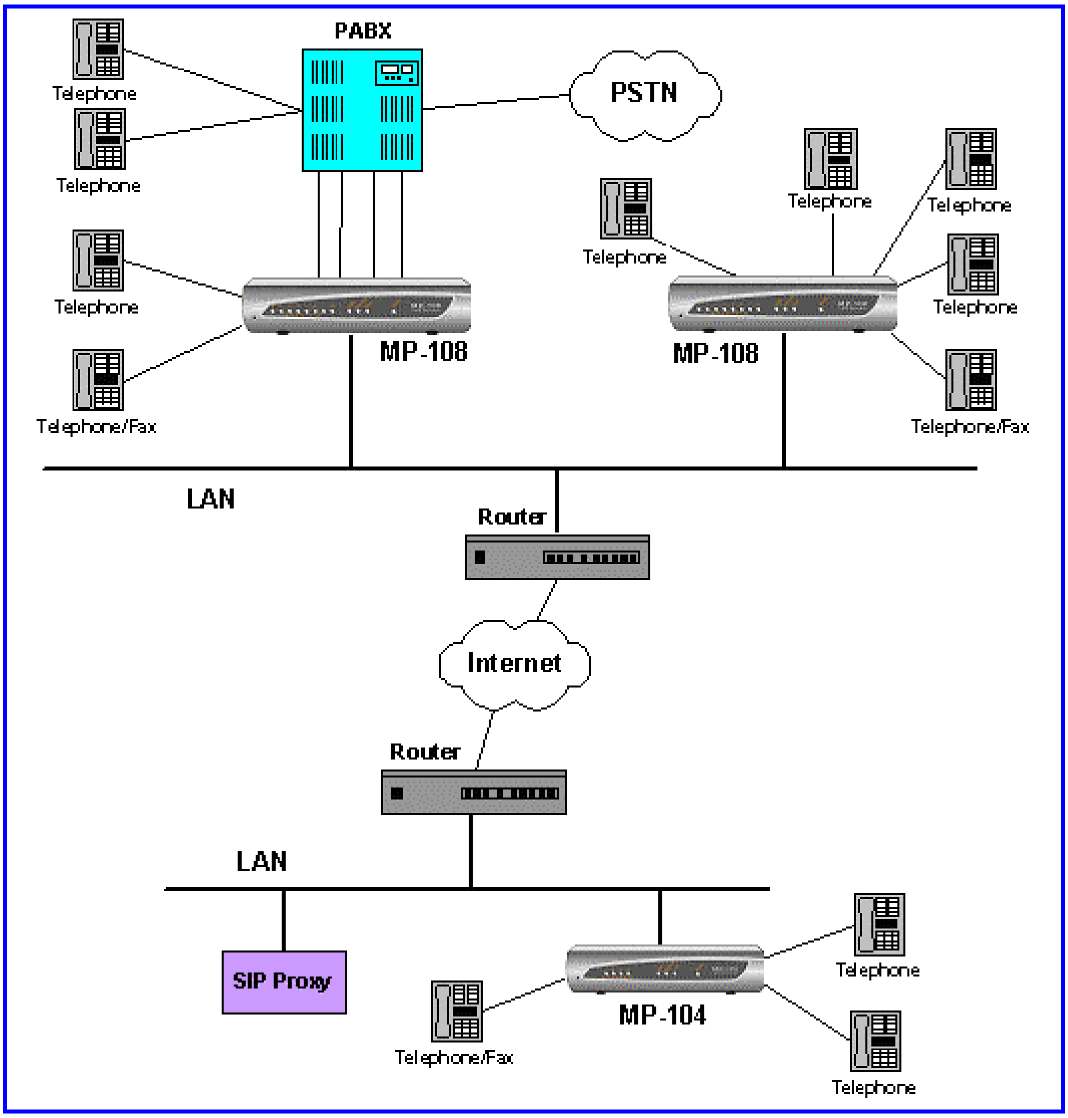


Figure 1-5: Typical MP-1xx VoIP Application



1.3 MP-1xx Key Features

- High quality Voice, Data and Fax over IP networks.
- **MP-124** supports up to 24 analog telephone loop start FXS ports as shown in Figure 1-1 on page 16.
- **MP-108** supports up to 8 analog telephone loop start FXS or FXO ports as shown in Figure 1-2 on page 16.
- **MP-104** supports up to 4 analog telephone loop start FXS or FXO ports as shown in Figure 1-3 on page 16.
- **MP-102** supports up to 2 analog telephone loop start FXS ports as shown in Figure 1-4 on page 16.
- Connected to the IP network via a 10/100 Base-T Ethernet interface.
- Coders include: G.711, G.723.1, G.726, G.727, G.729A and NetCoder at 6.4 to 8.8 kbps, selectable per channel.
- T.38 Fax with superior performance (round trip delay up to 9 sec).
- Compliant with SIP (RFC 3261), H.323, MEGACO (H.248) and MGCP.
- Life Line, connected to the unused pins on port #4 (or port #2 for **MP-102**), with a relay to an analog line, even if the **MP-10x/FXS** is powered off.
- LEDs on the front and rear panels provide information on the operating Media Gateway status and of the network interface.
- Restart button on the Front panel restarts the **MP-1xx** gateway
- **MP-10x** compact, rugged enclosure providing up to 8 analog RJ-11 ports within a compact housing of only one-half of a 19-inch rack unit, 1 U high (1.75" or 44.5 mm).
- **MP-124** 19-inch, 1 U rugged enclosure provides up to 24 analog FXS ports, using a single 50 pin Telco connector.
- Mounting option of installing two **MP-10x** Gateways in a single 19-inch rack shelf, one U high (1.75" or 44.5 mm).

2 MP-10x Hardware Installation

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

This section describes the appearance, installation procedure and functionality of the MP-10x, the LEDs on the front and rear panels, and the various connectors.

2.1 Hardware Installation Procedure

2.1.1 Unpacking

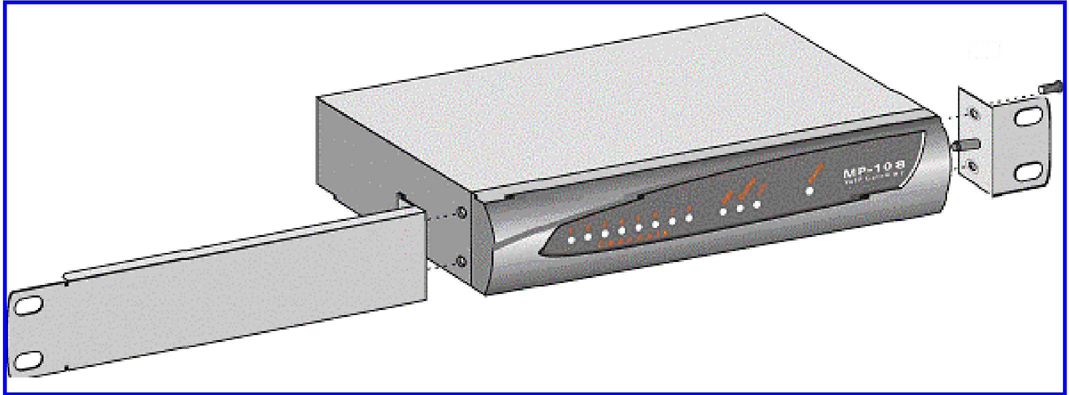
- Open the carton and remove packing materials
- Remove the **MP-10x** gateway from the carton
- Check that there is no equipment damage
- Check, retain and process any documents
- Notify AudioCodes of any damage or discrepancies
- Retain any diskettes or CDs

Safety Notice

Installation and service of this device must only be performed by authorized, qualified service personnel.

2.1.2 MP-10x Rack Mounting Installation

Figure 2-1: MP-10x Rack Mounting



The **MP-10x** is installed into a standard 19-inch rack by the addition of the 2 brackets supplied, and shown above.

➢ **To install the MP-10x, take the following steps:**

Step 1 Fasten the short bracket to the right-hand side of the **MP-10x** using the 2 screws provided, as shown in Figure 2-1, and carefully positioning the peg into a convenient ventilation hole in the side of the **MP-10x** box.

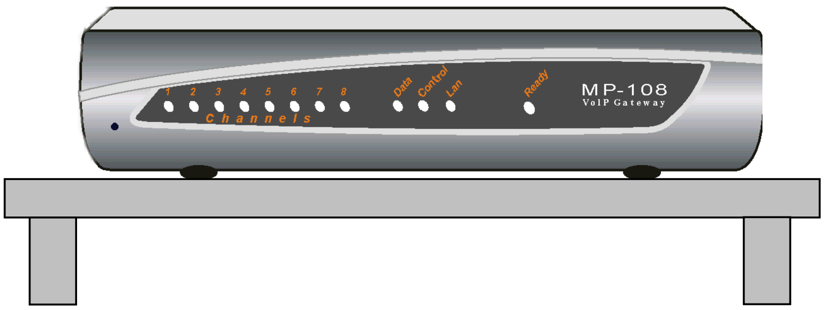
Step 2 Fasten the long bracket to the left-hand side of the **MP-10x** using the 2 screws provided as shown in Figure 2-1, and carefully positioning the peg into a convenient ventilation hole in the side of the **MP-10x** box.

Step 3 Insert the **MP-10x** into the 19-inch rack and fasten the left-hand and right-hand brackets to the vertical tracks of the 19-inch rack, using standard 19-inch rack bolts (not provided).

➢ **To connect the cables go to Step 4 (Section 2.1.4 on page 24)**

2.1.3 MP-10x Desktop Mounting Installation

Figure 2-2: MP-10x Desktop or Shelf



The **MP-10x** is installed on a desk or shelf without additional brackets as shown above.

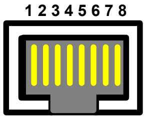
➢ **To connect the cables go to Step 4 (Section 2.1.4 on page 24)**

2.1.4 Cable Connections

The RJ-45 (Ethernet) and RJ-11 (Ports) pinouts and connectors are shown in Figure 2-3, and pins are numbered from the left with the latching finger position at the bottom.

Figure 2-3: RJ-45 LAN and RJ-11 Port Connectors and Pinouts

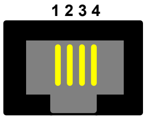
RJ-45 Connector and Pinout



1 - Tx+
2 - Tx-
3 - Rx+
6 - Rx-

4, 5, 7, 8 not connected

RJ-11 Connector and Pinout



1 - Not connected
2 - Tip
3 - Ring
4 - Not connected

➢ **To connect the cables for Desk-top or Rack-mount go to Step 4 below:**

Step 4 When using **MP-10x/FXS** gateway, insert each of the RJ-11 connectors on the 2-wire line cords of the POTS phones into the RJ-11 sockets on the rear of the gateway.

When using **MP-10x/FXO** gateway, insert each of the RJ-11 connectors on the 2-wire line cords coming from PSTN/PBX into the RJ-11 sockets on the rear of the gateway

Telephone lines and extensions of up to 7,300 m (24,000 ft) can be achieved using regular 24 AWG line cord.

Step 5 Insert the RJ-45 connector on the 10/100 Base-T cable from your LAN to the ETH RJ-45 socket (on the rear of the **MP-10x**) to provide the link to your LAN.

Step 6 Connect the **MP-10x** Gateway to the correct AC power supply, and the installation is now complete.

2.1.5 Installation of the MP-10x/FXS Life Line

Note: The **MP-124 and MP-10x/FXO** Media Gateways do **NOT** support the Life Line.

The **MP-108/FXS and MP-104/FXS** gateways provide a Life Line connection on port **#4**.

The **MP-102/FXS** gateway provides a Life Line connection on port **#2**.

This feature provides a wired phone connection to any PSTN or PBX FXS port, upon power-down conditions. When the power outage occurs, the phone that is connected to the Life Line port (see above), on pins #2 and #3, is wired to the PSTN or PBX FXS wires on pins #1 and #4 on the same connector. Therefore, the User of the **MP-10x/FXS**, can use the phone even when the **MP-10x/FXS** is not powered-on. To use this function, the User must utilize a splitter that connects pins #1 and #4 to another source of an FXS port, and pins #2 and #3 to the POTS phone.

The pinout of the Life Line RJ-11 phone connector is as follows:

1 = Life Line TIP

2 = TIP

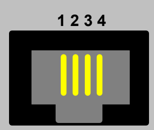
3 = RING

4 = Life Line RING

Refer to Figure 2-4 below for the RJ-11 connector pinout.

Figure 2-4: RJ-11 Connector and Life Line Pinout for MP-10x/FXS

RJ-11 Connector and Life Line Pinout

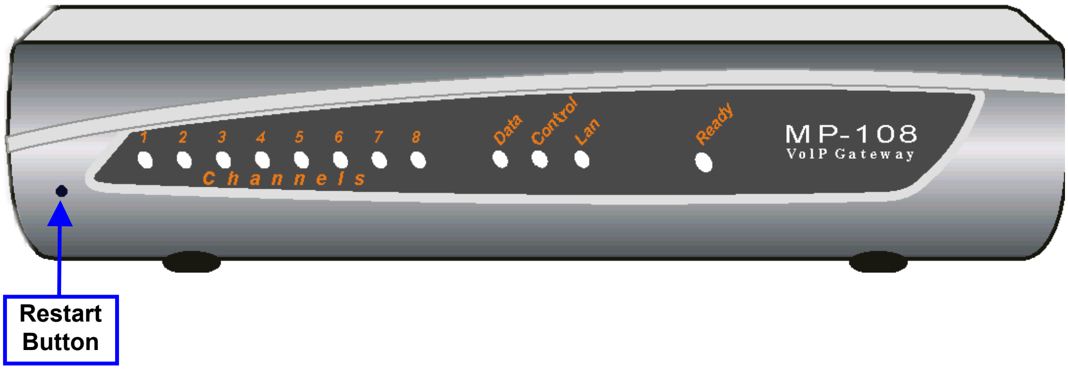


1 - Life Line Tip
2 - Tip
3 - Ring
4 - Life Line Ring

2.2 Front Panel LED Indicators

The **MP-10x** front panel LEDs indicate the Ethernet LAN status, Data (RTP) activity and state of the **MP-10x** ports.

Figure 2-5: MP-10x Front Panel LED Indicators



Functionality of the Front Panel LEDs for **MP-10x** is explained in Table 2-1.

Table 2-1: Front Panel Network LED Indicators

Label	Type	Color	State	Meaning
LAN	Ethernet Link Status	Green	ON	Valid Connection to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Data	Packet Status	Green	Blinking	Transmitting RTP Packets
		Red	Blinking	Receiving RTP Packets
		Blank	-	No traffic
Control	Control Link	Green	Blinking	Sending and receiving SIP messages.
		Red		Not supported in current release
Ready	Device Status	Green	ON	Device Powered, Self test OK
		Orange	Blinking	Software Loading/Initialization
		Red	ON	Malfunction

Table 2-2: MP-10x Channel LEDs

MP-10x with 1 to 8 Channels				
Label	**Type**	**Color**	**State**	**Meaning**
Activity	**FXS** Tel Port	Green	ON	Off-Hook/Ringing for Phone Port
	FXO Line Port	Green	ON	Line-Seize/Ringing State for Line Port

2.3 Rear Panel LED Indicators and Connectors

Figure 2-6: Rear Panel LED Indicators and Connectors

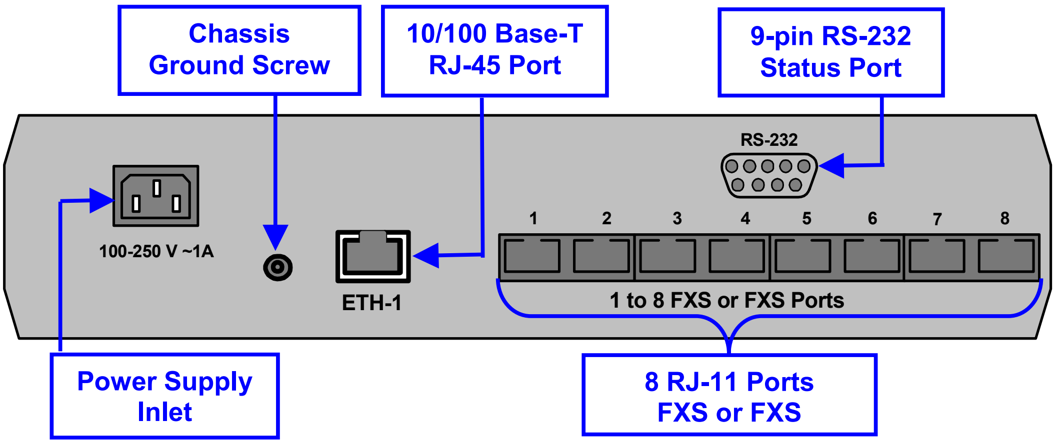


Table 2-3: Meaning of Rear Panel LED Indicators

Label	Type	Color	State	Meaning
ETH-1	Ethernet Status	**Yellow**	ON	Ethernet Port Receiving Data
		Red	ON	Collision

Table 2-4: Explanation of Rear Panel Connectors/Switches

Label	Type	Function	Comment
100-240V ~ 1A	3-pin power inlet	AC input	Connection to external power supply
1 to 8	RJ-11	8 FXS or FXO Ports	**MP-108** 2-wire Loop Start interface
1 to 4	RJ-11	4 FXS or FXO Ports	**MP-104** 2-wire Loop Start interface
1 to 2	RJ-11	2 FXS Ports	**MP-102** 2-wire Loop Start interface
ETH 1	RJ-45	10/100 Base-T Port	Shielded
RS-232	DB-9, DCE	Status Messages	Gateway connects to PC's RS-232 port with a straight cable (refer to Figure 8-1 on page 133).
⏚	Grounding screw	Chassis Ground	**MUST** be securely connected.

3 MP-124 Hardware Installation

Safety Notice

Installation and service of this device must only be performed by authorized, qualified service personnel.

This section explains the installation procedure for the MP-124 unit and describes the device's appearance and functionality, the LEDs and the various connectors.

The section has the following subsections:

3.1 Hardware Installation Procedure

3.1.1 Unpacking

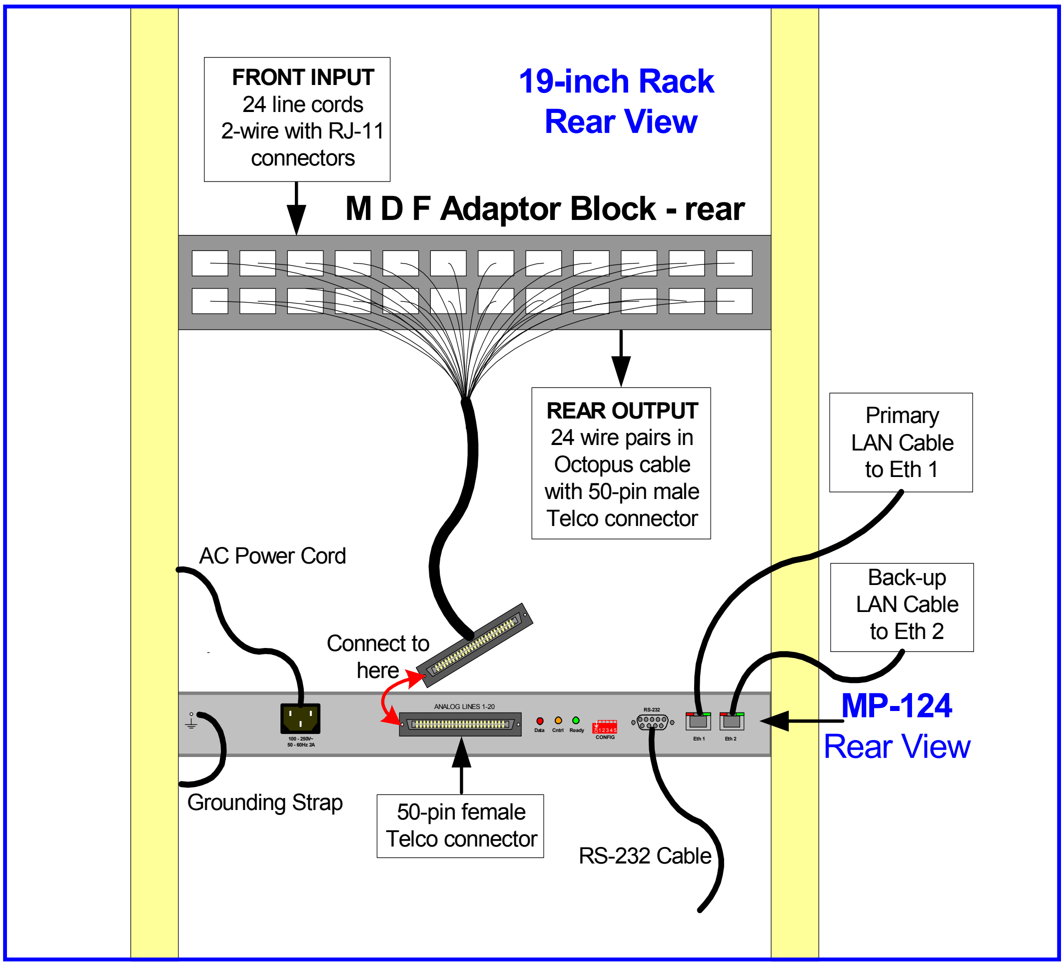
➢ To unpack the MP-124:

- Open the carton and remove packing materials
- Remove the **MP-124** from the carton
- Check that there is no equipment damage
- Check, retain and process any documents
- Notify AudioCodes of any damage or discrepancies
- Retain any diskettes or CDs

3.1.2 MDF Adaptor

To connect 24 2-wire lines into the **MP-124**, a Main Distribution Frame (MDF) Adaptor Block should be used as shown in Figure 3-1. This converts the standard RJ-11 connectorized phone line into a plain pair of wires that are terminated within a 50-pin Telco connector.

Figure 3-1: MP-124 in a 19-inch Rack with MDF Adaptor



Note:	The only equipment shown in Figure 3-1 and supplied by AudioCodes is the **MP-124** Gateway and, as an option, the MDF Adaptor.

As **input** (on the front of the 19-inch rack), the Adaptor Block takes in 24 2-wire lines with standard RJ-11 connectors.

As **output** (on the rear of the 19-inch rack), the Adaptor Block provides 24 wire pairs, which need to be terminated into a single 50-pin male Telco connector.

The 50-pin connector **must** be wired according to the pinout in Table 3-1 and Figure 3-2, shown below.

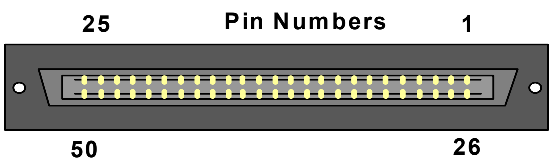
3.1.3 Cable Connections

The 50-pin Telco connector mounted on the rear of the **MP-124** is wired according to the pinout in Table 3-1 and Figure 3-2, shown below. The User's cable-mounted 50-pin Telco connector, supporting the 24 2-wire phone lines, **must** be wired identically.

Table 3-1: Pin Allocation in 50-pin Telco Connector

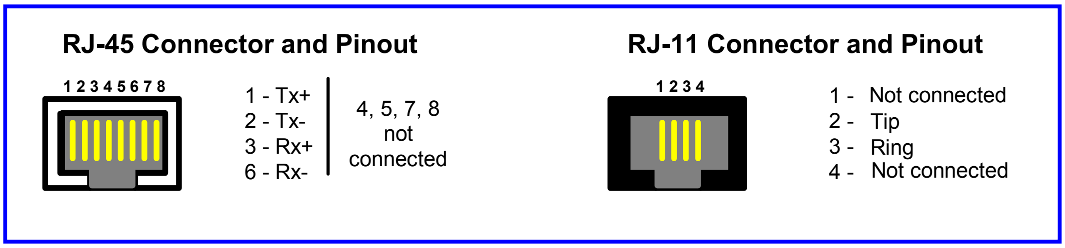
Phone Channel	Connector Pins	Phone Channel	Connector Pins
1	1/26	13	13/38
2	2/27	14	14/39
3	3/28	15	15/40
4	4/29	16	16/41
5	5/30	17	17/42
6	6/31	18	18/43
7	7/32	19	19/44
8	8/33	20	20/45
9	9/34	21	21/46
10	10/35	22	22/47
11	11/36	23	23/48
12	12/37	24	24/49

Figure 3-2: 50-pin Telco Connector

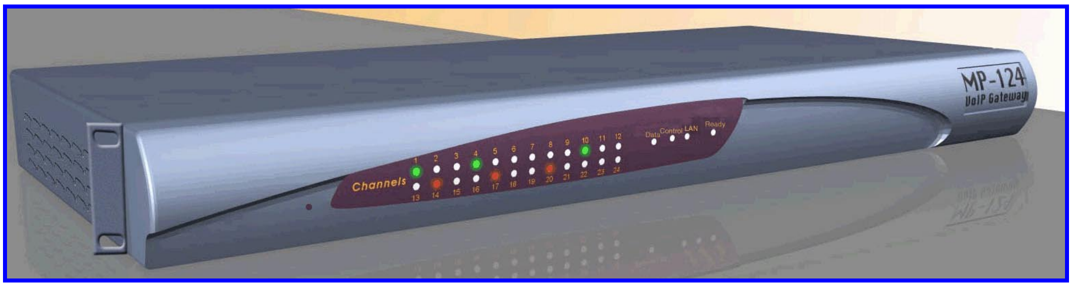


The RJ-45 (Ethernet) and RJ-11 (POTS) pinouts and connectors are shown in Figure 3-3. Pins are numbered from the left with the latching finger position at the bottom.

Figure 3-3: RJ-45 and RJ-11 Connectors and Pinouts



3.1.4 19-inch Rack Mounting



MP-124 gateway is supplied with brackets ('ears') fitted to each side of the enclosure so that the **MP-124** can be immediately installed in the 19-inch rack.

➢ To install the rack mount MP-124, take the next 9 steps:

Step 1 Insert the **MP-124** into the 19-inch rack, adjust it to the correct position and use two standard rack-screws (not supplied) to secure each of the two brackets to the rack frame.

Step 2 Insert each of the RJ-11 connectors on the 2-wire line cords of the POTS phones into the RJ-11 sockets on the front of the MDF Adaptor Block

Up to 3,000 m (10,000 feet) of 24 AWG line cord can be used to connect telephones.

Step 3 Attach to the each of the sockets on the rear of the MDF Adaptor Block one pair of wires from a 25-pair Octopus cable.

Step 4 Connect the wire-pairs at the other end of the Octopus cable to a male 50-pin Telco connector. The pinout must be that shown in Table 3-1 and Figure 3-2 on page 32.

Step 5 Insert and fasten this 50-pin connector into the female 50-pin Telco connector mounted at the rear of the **MP-124** and labeled Analog Lines 1-24.

Step 6 Insert the RJ-45 connector of the 10/100 Base-T cable into the RJ-45 connector mounted at the rear of the **MP-124** and labeled Eth 1 for connection to your LAN.

Step 7 Connect an electrically grounded strap to the chassis ground screw on the rear of the **MP-124** and fasten it securely according to the current standards.

Step 8 Connect an electric power cord of the correct rating, from a grounded supply of the correct voltage, into the power socket mounted at the rear of the **MP-124** and labeled 100 - 250 V ~ 50 – 60 Hz 2A.

Step 9 Observe the front panel LEDs to determine the functioning of the **MP-124**.

The Channel LEDs indicate that the telephones connected to the rear 50-pin connector are each in one of the following states:

ringing or in the Off Hook position	Green
normal operation	Blank
not functioning	Red

The functions of all the LEDs of the **MP-124** are shown in Table 3-2 on page 35.

3.2 Front Panel LED Indicators

Figure 3-4: Front Panel LED Indicators

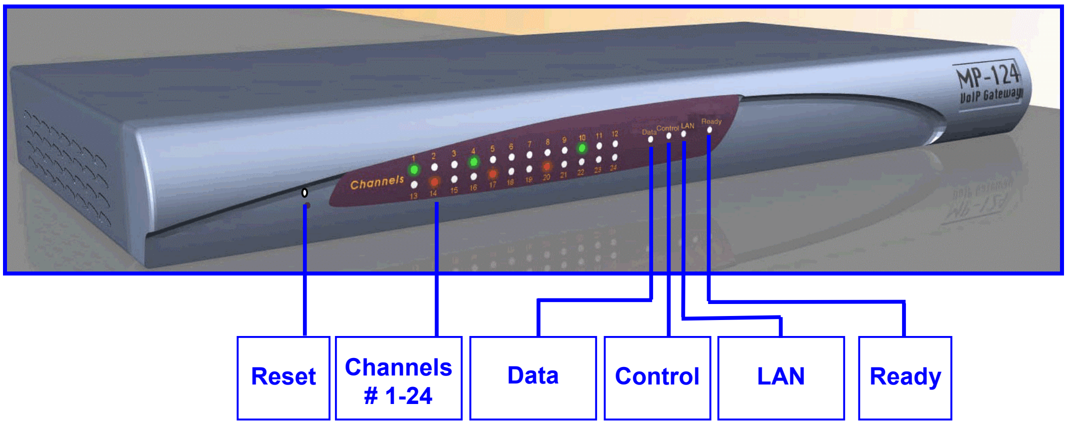


Table 3-2: Function of Front Panel LED Indicators

Label	Type	Color	State	Function
Data	Packet Status	Green	Blinking	Transmitting RTP Packets
		Red	Blinking	Receiving RTP Packets
		Blank	-	No traffic
Control	Control Link	Green	Blinking	Currently not implemented
		Red	ON	Currently not implemented
		Orange		Currently not implemented
LAN	Ethernet Status	Green	ON	Valid link to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Ready	Device Status	Green	ON	Device Powered and Self-test OK
		Orange	Blinking	Software Loading/Initialization
		Red	ON	Malfunction
Channels # 1- 24	Tel Port	Green	ON	Off-Hook/Ringing for FXS Phone Port
	Tel Status	Red	ON	Malfunction
		Blank	-	Normal

3.3 Rear Panel LED Indicators/Connectors

Figure 3-5: Rear Panel LED Indicators and Connectors

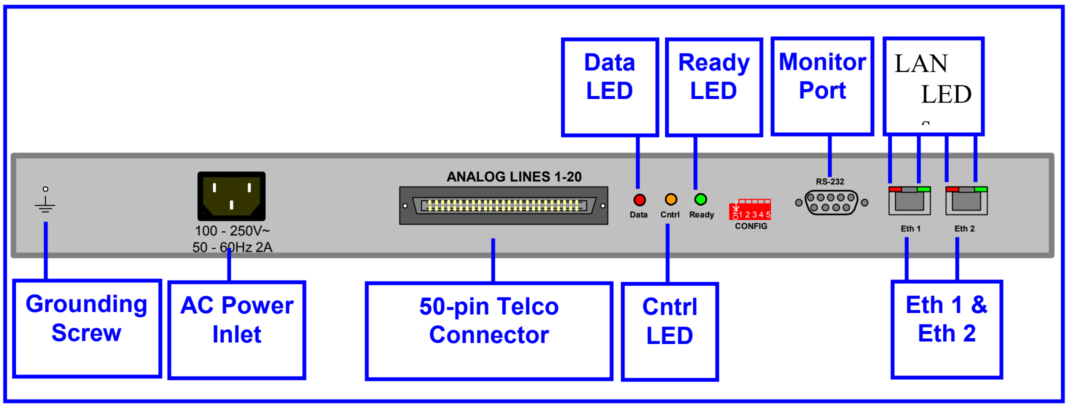


Table 3-3: Function of Rear Panel LED Indicators

Label	Type	Color	State	Function
Data	Packet Status	Green	ON	Transmitting RTP Packets
		Red	ON	Receiving RTP Packets
		Blank	-	No traffic
Cntrl	Control Link	Green	ON	Currently not implemented
		Red	ON	Currently not implemented
		Orange	ON	Currently not implemented
Ready	Device Status	Green	ON	Device Powered and Self-test OK
		Orange	ON	Software Loading/Initialization
		Red	ON	Malfunction
Eth 1	Ethernet Status	Green	ON	Valid link to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Eth 2	Ethernet Status	Green	ON	Valid link to 10/100 Base-T hub/switch
		Red	ON	Malfunction

Table 3-4: Function of Rear Panel Connectors/Switches

Label	Type	Function	Comment
100-250 V~ **50 - 60 Hz 2A**	3-pin AC	AC input	Connection to AC power cord
⏚	Grounding Screw	Chassis ground	
Analog Lines 1 to 24	50-pin Telco connector	FXS Ports	2-wire Loop Start interface
Eth 1	RJ-45	10/100 Base-T	Shielded port to Ethernet LAN. This is the default port.
Eth 2	RJ-45	10/100 Base-T	Shielded port to Ethernet LAN. The port is not in use for current SW release.
RS-232	DB-9, DCE	Status Messages	Gateway connects to PC's RS-232 port with a straight cable (refer to Figure 8-1 on page 133)..)

Note: The DIP switch located on the **MP-124** rear panel is not functional and shouldn't be used.

Reader's Notes

4 Software Installation

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

This section explains how to install the MP-1xx Software

4.1 Installation Package

The Installation Package includes the following Software and Utilities:

Software:

- *MP108_ram_fxs.cmp* - Image Software for download to **MP-10x/FXS**
- *MP108_ram_fxo.cmp* - Image Software for download to **MP-10x/FXO**
- *MP124_ram.cmp* - Image Software for download to **MP-124** gateway
- *usa_tones.dat* - Call progress tones *dat* file for download
- *usa_tones.ini* - Call progress tones *ini* file (used to create *dat* file)
- *SIPgw_FXS.ini* - *ini* example file for **MP-1xx/FXS** gateways
- *SIPgw_FXO.ini* - *ini* example file for **MP-10x/FXO** gateways
- *MP1xx_Coeff_FXS.dat* - Telephony interface configuration file for **MP-10x/FXS** gateways
- *MP10x_Coeff_FXO.dat* - Telephony interface configuration file for **MP-10x/FXO** gateways
- MIB library - Library of SNMP MIBs

Utilities:

- *ini file utility.xls* - Excel™ utility for creation of the *ini* file
- *TPDMUtil.exe* - Call progress tones file generator utility
- *Bootp_install.exe* - AudioCodes BootP & TFTP configuration utility

4.2 MP-1xx Initialization

MP-1xx Gateway comes with ready-installed software. The basic installation can be done using AudioCodes configuration utility, or from Web browser, such as from Microsoft Internet Explorer.

To change network parameters, use Web browser, AudioCodes BootP & TFTP configuration utility or use third party BootP server.

For setting the SIP parameters in the *ini* file, either edit the *ini* example file, or generate such a file using the Excel utility provided.

The *ini* file and other configuration files can be downloaded directly from the Web Browser using either HTTP protocol, or AudioCodes-provided configuration utility or any standard TFTP server. The Image software file: *ram.cmp* is only used for software upgrade.

The Call Progress tone file *usa_tones.ini* is used to define call progress tone levels and their frequency. To change the tone's parameters, first modify the file, and then using "TPDMUtil.exe" utility, convert the text *ini* file to binary *dat* file. This procedure is described in Section 6.4.

The Coeff_FXS.dat and Coeff_FXO.dat files can be used respectively to modify the **MP-1xx/FXS** and **MP-10x/FXO** telephony interface characteristics, such as DC and AC impedance, feeding current and ringing voltage. For more information, refer to Section 6.5

4.3 Quick Setup Procedure

The following procedure describes how to setup the **MP-1xx** gateway with basic parameters using standard Web Browser (such as Microsoft Explorer). It is assumed that the IP address of the gateway is known. If the IP address is unknown, use AudioCodes Configuration utility (or any standard BootP server) to set the gateway IP address and subnet mask.

Usually the **MP-1xx** gateways are shipped with following network parameters:

MP-1xx FXS gateway IP address: 10.1.10.10,

MP-10x FXO gateway IP address: 10.1.10.11

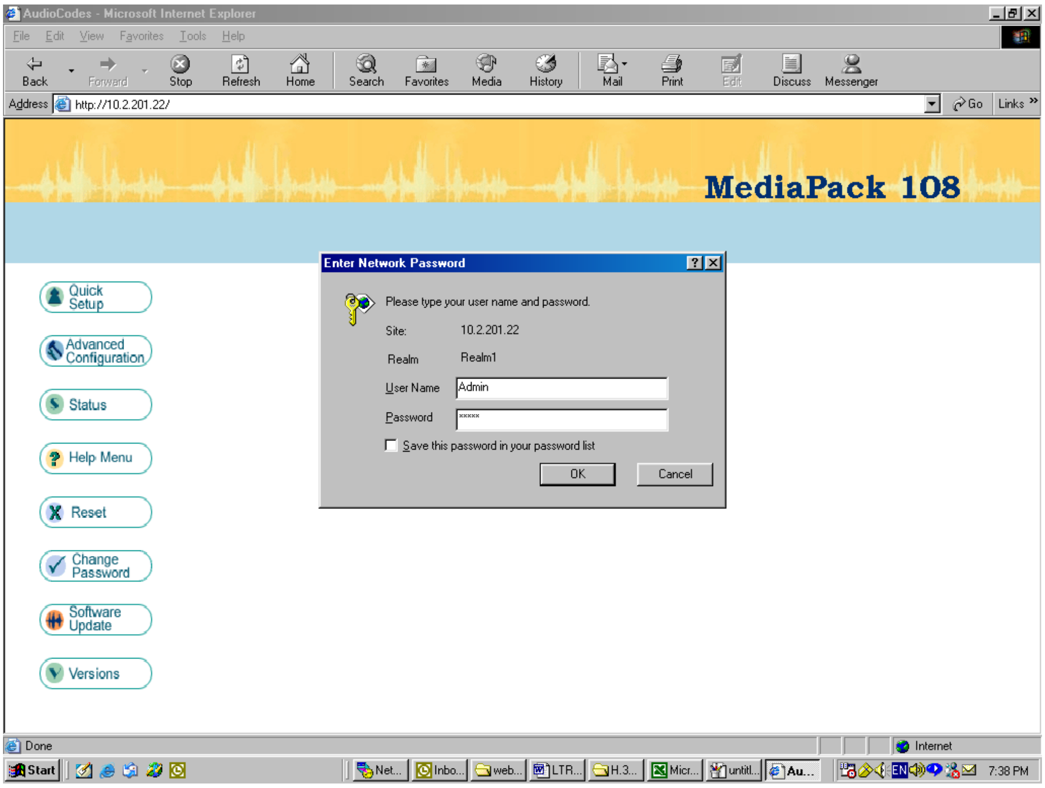
Subnet: 255.255.0.0,

Default gateway: 0.0.0.0

➢ For quick MP-1xx setup, take the next 13 steps:

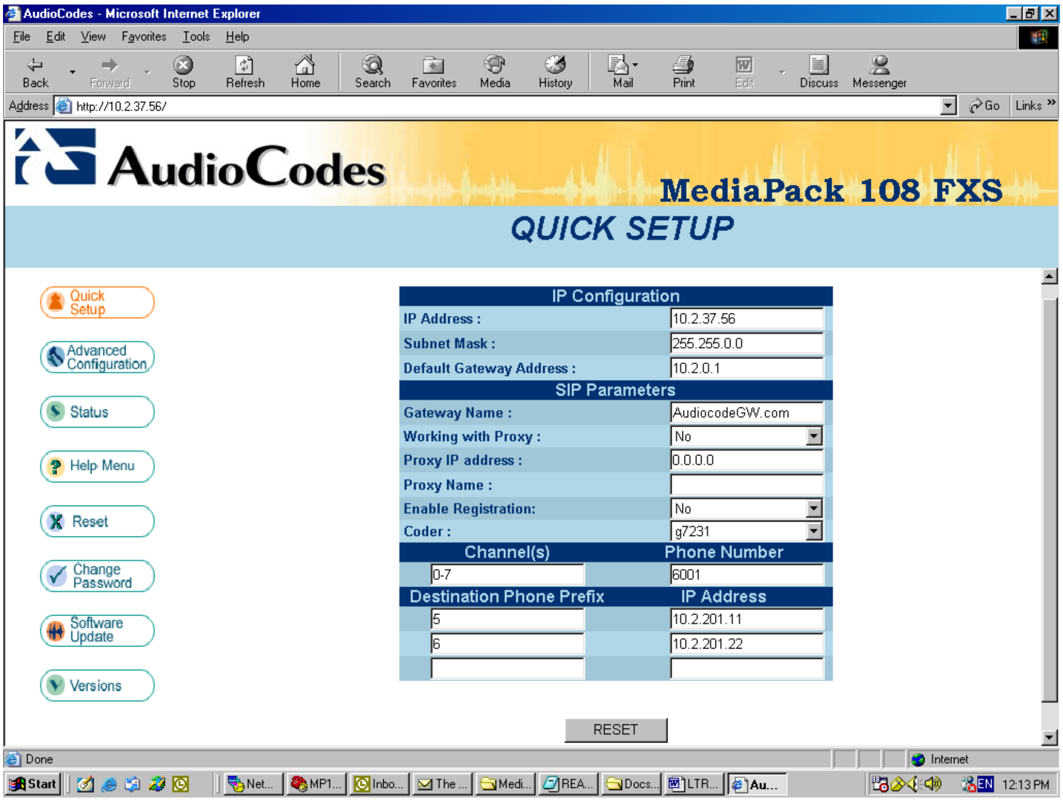
1. Power-up the gateway. After self-testing, in about 20 seconds the Ready LED on front panel turns to green. Any malfunction changes the Ready LED to red.
2. Configure your PC to have the same subnet as the gateway. Run the Web Browser and enter the gateway IP address.
3. Run Web Browser and enter gateway IP address.

Figure 4-1: Web Browser Screen



4. On **MP-1xx/** Gateway main page, enter the User Name and Password (default: Admin, Admin)
5. Change and confirm new User name and Password
6. Click “Quick Setup” button, and set the gateway new IP address, Subnet mask and Default gateway,
7. Set the basic SIP Gateway Parameters, as shown in Figure 4-2 below.

Figure 4-2: SIP Quick Setup



8. Fill in the gateway phone numbers. For example for **MP-108**, enter "0-7" in Channel field and starting Phone number, such as 6001 in adjacent Phone number field. The **MP-108** physical ports are associated with phone numbers 6001 to 6008.

9. When working without Proxy the internal routing table needs to be defined:

   Fill in the "Destination phone prefix" the prefix of called number and associated IP address. In the example above, an incoming call with called number starting with 5xxx is sent to IP address 10.2.201.11. In quick setup page, the User can define up to three routing entries. Using "Advance configuration" it is possible later to increase the size of routing table up to 20 entries.

10. Click Submit button, and then reset the gateway by clicking the Reset button from the main menu.

11. If the gateway subnet is changed, you also need to change the PC IP address and its subnet to the same subnet as the gateway.
12. Wait about 60 seconds and Refresh the Web page
13. Using “Advance Configuration” and “Status” pages you can view and modify all SIP and other gateway parameters. Refer to Section 7.2 for detailed directions of operating under Web management control.

4.4 BootP and TFTP Procedures

If the Gateway IP address is known, you can use Web Browser control for gateway configuration and provisioning. Otherwise, you can use BootP (or DHCP) and TFTP protocols for initialization and software download.

Each time the **MP-1xx** Gateway is powered-on, it performs the standard BootP procedure.

If "DHCPEnable = 1" line is included in gateway's *ini* file and if the BootP server was not found, the gateway initiates standard DHCP procedure to configure the gateway network parameters (IP address, Subnet mask and Default router address). If DHCP procedure is used, you need to find the new gateway IP address allocated by DHCP server. Usually this information can be provided by System Administrator.

If the BootP/DHCP server has not been found, the **MP-1xx** Gateway starts working from its internal flash memory.

Usually the application software already resides in the **MP-1xx** flash memory, therefore there is no need to use the BootP or TFTP procedure. Their download need only be used for changing the **MP-1xx** configuration or for a new software upgrade.

The BootP Protocol enables the network administrators to manage the configuration of the **MP-1xx** Media Gateway from a central configuration server - BootP/DHCP server.

The following RFCs (IETF Requests for Comment) describe BootP in detail: RFC 951, RFC 1542 and RFC 2132.

Downloading of the image file by the **MP-1xx** is performed using Trivial File Transfer Protocol (TFTP). TFTP protocol is described in RFC 906 and RFC 1350.

Although DHCP and BootP servers are very similar in operation, the DHCP server includes some differences that might prevent its operation with BootP clients. However, many DHCP servers, such as Windows™ NT DHCP server, are backward-compatible with BootP protocol and can be used for **MP-1xx** configuration.

Note: The BootP server is normally used to configure the **MP-1xx** initial parameters. Once this information has been provided, the BootP server is no longer needed. All parameters are stored in non-volatile memory and used when the BootP server is not accessible. The BootP server is required again if, for example, the **MP-1xx** IP address is to be changed.

Using BootP procedure, the following parameters are configured:

- Boot & *ini* File Names — Optional, refer to Note 1 below.
- Local IP Address — IP address of your **MP-1xx** Gateway.
- Gateway IP Address — If Default Gateway/Router is required, otherwise enter 0.0.0.0 address.
- Subnet Mask — Refer to Note 2 below.
- TFTP Server IP address — Refer to Note 3 below.

Note 1: Boot file name can contain one or two file names. ram.cmp file name to be used for download of application image and mp1xx.*ini* file name to be used for **MP-1xx** provisioning. Either one, two or no file names can appear in the Boot file name field. To use both file names use ";" separator (without blank spaces) between the xxx.cmp and the *yyy.ini* files (e.g., *ram.cmp;SIPgw.ini*).

Note 2: Usually TFTP and BootP servers are installed on the same Host. However, when using AudioCodes Configuration utility or Microsoft™ DHCP server, it is possible to set the IP address of TFTP server (Boot Server Host Name field), and in this case BootP and TFTP servers can run from different Hosts.

4.4.1 Configuring the TFTP Server

➢ **To configure the TFTP Server, take the next 4 steps:**

1. Set the default directory where the image file resides (C:\AudioCodes\…).
2. Copy the Image file (such as ram.cmp to the TFTP default directory on your Host PC.
3. Copy the ini file and other optional configuration files (Call Progress tones and Coeff.dat files) to the TFTP default directory on your Host PC. Ensure correct coeff.dat file is used. Two different coefficient files are provided, for **MP-1xx/FXS** and **MP-10x/FXO** gateways.
4. Set the TFTP timeout to 3 seconds and number of retransmissions to 20.

4.4.2 Using AudioCodes BootP/TFTP Configuration Utility

The AudioCodes Configuration utility provides an easy way to configure the **MP-1xx** gateway. Similar to other BootP and TFTP servers, the application can be installed on Windows™ 98 or Windows™ NT/2000. With AudioCodes' BootP/TFTP Server Configuration utility, it is possible to use the integrated TFTP server (part of the Utility) or to install TFTP server on a different host. The utility enables remote reset of the **MP-1xx** unit for triggering the initialization procedure (BootP & TFTP). For details of the procedure, refer to Appendix A in the AudioCodes "Software Utilities Manual", Catalog Number: LTRT-00602.

4.4.2.1 Configuration Utility Main Features

- BootP server supporting hundreds of entries.
- Integrated TFTP server.
- Common Log window displaying BootP and TFTP status.
- Contains all data required for provisioning of AudioCodes products.
- Provides the TFTP server address, enabling network separation of TFTP and BootP servers.
- Tools for backup and for restoring the local database.
- Templates.

- User-defined names for each entity.
- Option for changing the gateway MAC (Media Access Control) address.
- Protection against entering fault information.
- Unicast or Broadcast BootP response.

4.4.3 Configuring the Windows™ NT DHCP Server

If Microsoft™ Windows NT DHCP server is used in your organization, the server can be used in reservation mode to provide an IP address and other necessary information to the **MP-1xx** Media Gateway.

To configure the Microsoft™ Windows™ NT DHCP Server to assign IP address information to BootP clients, add a reservation for each BootP client.

For information about how to add a reservation, view the "Managing Client Reservations Help" topic in DHCP Manager.

The reservation builds an association between the media access control address (12 digits, provided in **MP-1xx** documentation) and the IP address. Windows™ NT Server provides the IP address based on the **MP-1xx** media access control (MAC) address in the BootP request frame.

To configure the Microsoft™ Windows™ NT DHCP server to provide boot file information to BootP clients, edit the BootP Table in DHCP Manager. The BootP Table is located in the Server Properties dialog box that can be accessed from the Server menu. For information about how to edit the BootP Table, view the "BootP Table" Help topic in DHCP Manager.

The following parameters must be entered:

- Local IP address — IP address of your **MP-1xx** Gateway.
- Subnet mask — Refer to Section 4.4 Note 2, for the mask limits.
- Gateway IP address — Default Gateway IP address
- Boot File name — Optional, refer to following Note.

> **Note**: Boot file name normally should not be used. This field is only used for software upgrade, refer to Section 4.5.

Refer to Utilities User Manual for detailed description of configuration of Windows NT DHCP server.

4.4.4 Other TFTP & BootP Servers

Third party TFTP and BootP servers can be used; for example Weird Solutions™ (www.weird-solutions.com).

Note: Weird Solutions™ TFTP and BootP servers must be installed on the same Host. Details can be found in the AudioCodes "Software Utilities Manual", Document #: LTRT-00702.

4.5 MP-1xx Software Upgrade

4.5.1 General Upgrade Procedure

MP-1xx includes on-board flash memory already programmed with application software. The following procedure replaces the old stored software with the new version. To run this procedure BootP and TFTP servers are required. Web Browser can be used instead of BootP server (refer to Section 7.2).

Note: The file extensions *cmp* and *ini* should be written in lower case letters.

➢ **To upgrade the integral Software take the next 5 steps:**

1. Start the TFTP and BootP servers.
2. Copy the new *ramxxx.cmp* file, *mp108.ini* file (e.g., *SIPgw.ini*) and optional configuration files to the default TFTP server directory (refer to Appendix A).
3. Set the Boot file and *ini* file names in the Web Browser Network settings page or in the BootP server: *ramxxx.cmp -fb;mp108.ini*. Other network parameters stay unchanged (IP address, subnet mask,…). If so required, it is possible to update only the *mp108.ini* parameters. For this option, set the boot file name to: *mp108.ini* (without preceding *ramxxx.cmp*). After **MP-1xx** power reset, the *ini* parameters are downloaded using the TFTP procedure and stored in the non-volatile memory.
4. Reset the **MP-1xx**. Wait about 20 seconds until the Ready LED changes to Green.
5. After accomplishing BootP and TFTP procedures, the new software is downloaded and stored in the **MP-1xx** unit's flash memory.

Note: The parameter "-fb" added to Boot file name is used to specify the burning of flash memory with new software image. To test new software version without replacing the old version, skip the "-fb" parameter. In this case, the new software is downloaded directly to RAM, and not permanently stored into flash.

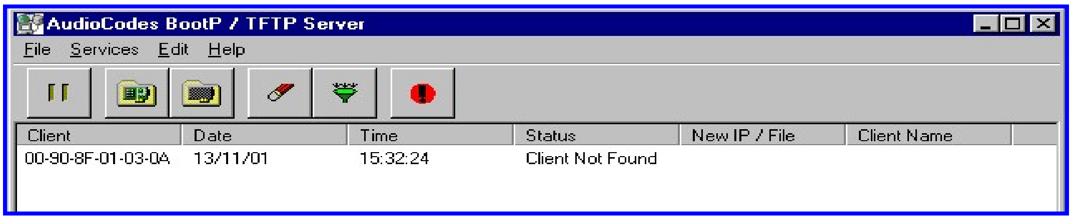
4.5.2 Upgrade Procedure Using AudioCodes Configuration Utility

The following procedure describes how to upgrade an **MP-1xx** software using the AudioCodes configuration utility.

➢ **To upgrade the Software using the AudioCodes Configuration Utility take the next 13 steps:**

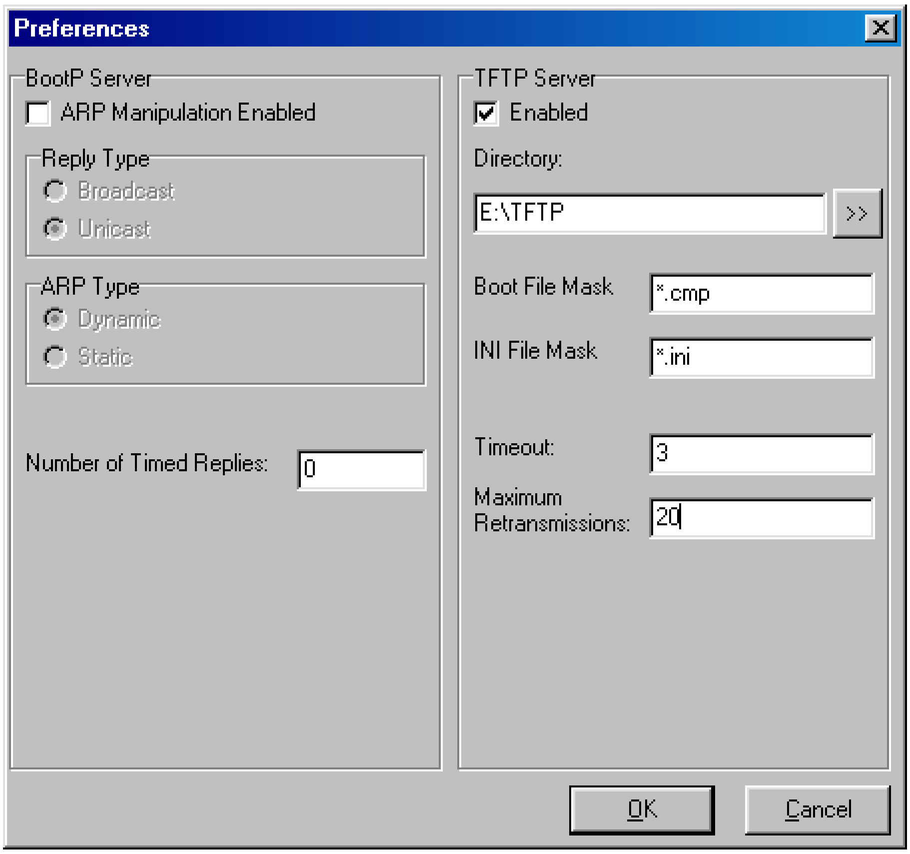
1. Install AudioCodes' Configuration utility from the AudioCodes Software CD, Catalog Number LSTC00005 (MediaPack Series).
2. Open the AudioCodes Configuration utility from Start>Programs>BootP; the AudioCodes Configuration utility main screen opens:

Figure 4-3: AudioCodes Configuration Utility Main Screen



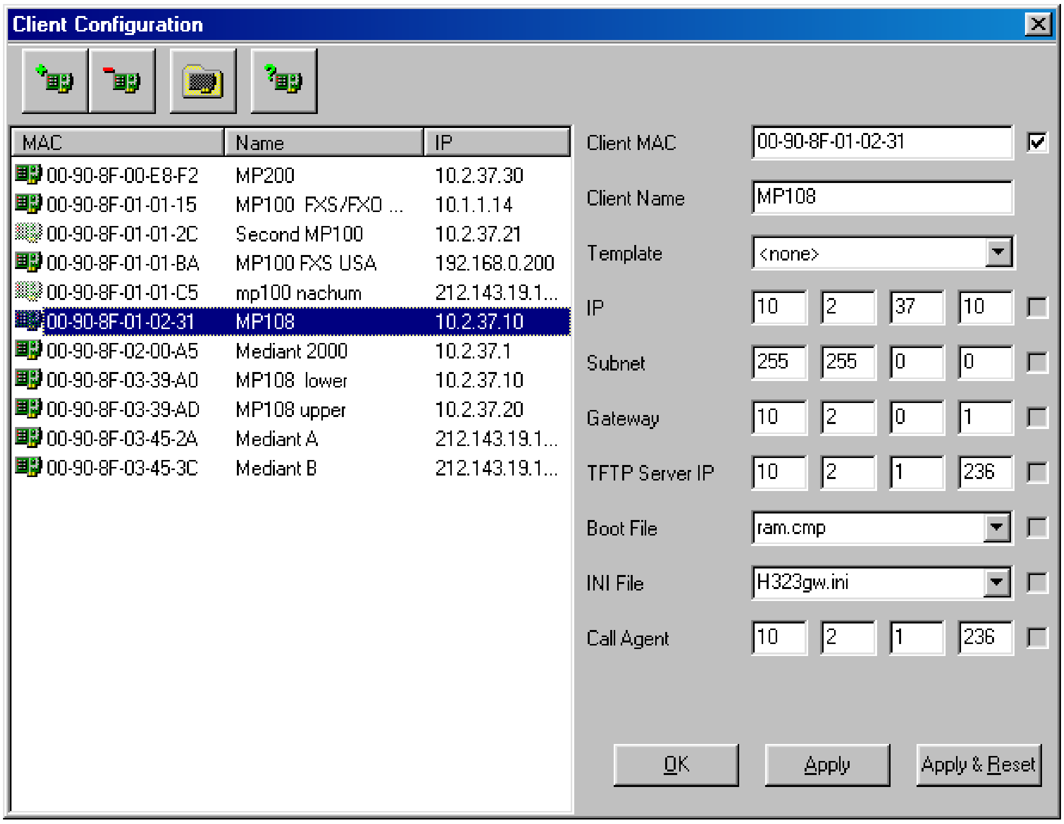
3. Click on the Edit tab to open the Edit menu.
4. Select Preferences to open Preferences window shown in Figure 4-4 on page 53.

Figure 4-4: Preferences Screen



5. In the Directory field, click on the >> button and navigate to the directory of the source *.*cmp* and *.*ini* files. All downloaded files should reside in this folder, including ram.*cmp*, mp108.*ini*, Coeff.dat and Call Progress tone.*dat* files.
6. Click OK to return to the main screen.
7. In the Services menu, choose Clients. This opens the Client Configuration screen shown in Figure 4-5 on page 54. The parameter fields displayed on the right side of the screen constitute the **MP-1xx** software profile configuration. The parameter fields are all blank in the case of a Client Not Found.

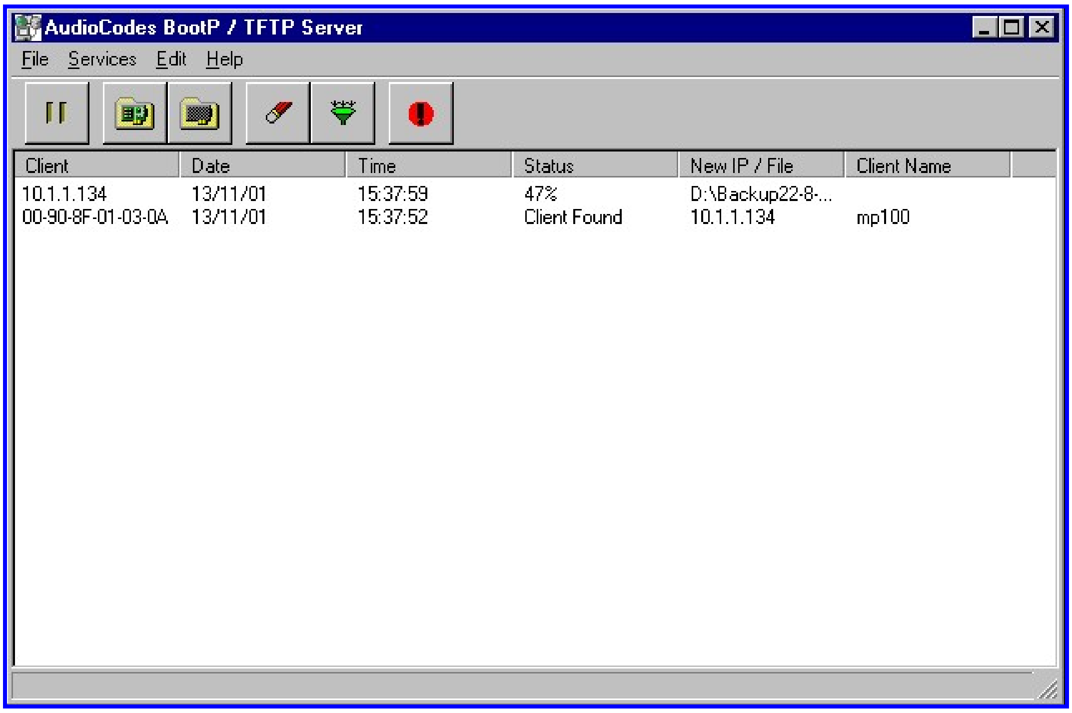
Figure 4-5: Client Configuration



8. Enter the Client MAC address and Client Name.
9. Enter the IP address (such as 10.2.37.1).
10. Enter the Subnet (such as 255.255.0.0); set the Subnet to a valid value in accordance with the IP address.
11. Enter the IP address of the default Gateway; it can be any address within the subnet.
12. Select the required Boot and *ini* Files
13. To permanently store the new image file in the **MP-1xx** flash memory, add –fb suffix to Boot file name, such as “ram.cmp –fb”. After entering the file names, click “Apply & Reset” button.

The following status messages is displayed in the AudioCodes BootP/TFTP Server main screen:

Figure 4-6: AudioCodes Configuration Utility – TFTP download



Reader's Notes

5 Profiling & Operation

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

This section describes the MP-1xx SIP Profile, Operation and Configuration of its various supported functions.

5.1 SIP Profile

5.1.1 Supported SIP Features

The **MP-1xx** complies with RFC 2543 bis IETF standard.

MP-1xx main SIP features are:

- Works with Proxy or without Proxy, using an internal routing table.
- Proxy or Registrar Registration, such as:

```
REGISTER sip:proxyname SIP/2.0
VIA: SIP/2.0/UDP 212.179.22.229;branch=z9hG4bRaC7AU234
From: <sip:101@sipgatewayname>;tag=1c29347
To: <sip:101@sipgatewayname>
Call-ID: 10453@212.179.22.229
Seq: 1 REGISTER
Expires: 3600
Contact: sip:101@212.179.22.229
Content-Length: 0
```

  Where the "proxyname" and "sipgatewayname" strings are defined in **MP-10x** *ini* file (or configured from Web). The REGISTER message is sent per each **MP-10x** FXS port.

  The MP-10x FXO gateway registrates just once, using the "username" parameter in left side of sip URL, such as sip: "username@sipgatewayname". The "username" parameter can be defined in INI file of from Web browser.

- Proxy and Registrar Authentication (handling 401 and 407 responses) is supporting both Basic or Digest methods. "User Name" used for FXS channel authentication is equal to the channel phone number. Single password is used for all gateway endpoints.
- SIP-URL: sip:"phone number"@IP address (such as 122@10.1.2.4, where "122" is the phone number of the source or destination phone number) or sip:"phone_number"@"domain name", such as 122@myproxy.com
- Reliable UDP transport, with retransmissions
- Supported codecs: G.711 A-law, G.711 µ-law, G.723 (6.3 kbps) , G.729A, NetCoders.
- Can negotiate codec from a list of given codecs
- SIP Responses:
  - Informational Responses: 100, 180, 181,182, 183

- Successful responses: 200OK
- Failure Responses: 4xx

For detailed information, refer to the latest **MP-1xx**/SIP Release Notes, Catalog Number: LTRT-00656.

5.2 Using SIP Gateway Features

Details of how to use and configure the SIP Gateway Parameters are shown below in Table 5-1.

Table 5-1: Using SIP Gateway Features (continues on pages 61 to 63)

Feature	Parameter	Sheet of Excel Utility	Value
SIPGatewayName	AudioCodes gateway name. If specified this name is used in right side of SIP URL in "FROM" header, otherwise the gateway IP address is used.	SIPgw	
Use SIP Proxy	IsProxyUsed	SIPgw	1
	ProxyIp Proxy IP address	SIPgw	IP
ProxyName	Proxy Domain Name used in SIP Request-URI. If this parameter is not specified, the Proxy IP address will be used instead in SIP URL.	SIPgw	
No Proxy, Use routing table	IsProxyUsed	SIPgw	0
	Define routing table using: a. PREFIX = <prefix, IP address> list b. IP = *,<IP address>	Phone or/& Prefix Tables	Phone numbers, Prefixes and IPs
	When Proxy isn't used, it is necessary to define IP routing table, to enable **MP-1xx** gateway to find destination IP address, according to received dial number. The routing table is defined in *ini* file, using PREFIX or per phone number definitions. The gateway first searches for Phone table to find a destination IP address, than it looks for Prefix parameter, and later for "Prefix = *,<IP address>" definition. "Prefix = *, <IP address>" defines destination IP address for any other phone number.		
Set numbers to end points	Channel2Phone= <channel>,<phone> or ChannelList = port, number, phone	EndPoints	Phone numbers
	The easiest way to define endpoint phone number is to use ChannelList parameter. For example, to define 101 to 107 numbers for an **MP-108**, use a single line: ChannelList = 0,8,101. The first parameter (0) indicates the first endpoint number.		

Table 5-1: Using SIP Gateway Features (continues on pages 61 to 63)

Feature	Parameter	Sheet of Excel Utility	Value
	The second parameter (8) indicates the number of endpoints. The third parameter (101) indicates the first endpoint phone number. Up to ten such ChannelList definitions can appear in the same *ini* file. One or more phone numbers in the ChannelList can be modified by using Channel2Phone definition, following the ChannelList parameter in the *ini* file.		
Dial plan	MaxDigits Maximal number of digits in dialed number.	General	Max digits
	TimeBetweenDigits Timeout between dialed digits, used to terminate dialing. Usually it is set to 4 seconds.	General	Time in seconds
Choose Coder	CoderName	General	Preferred coder name
Several Coders	CoderName	General	List of coders
	In this mode, several codecs are sent in SDP message. On receiving the remote response (200 OK) with its SDP, a process of matching coders is done between the local set of coders (from the *ini* file) and the remote set. The local coders are the preferred ones, and if the first local coder is included in the remote SDP response, then it is selected, otherwise next local coder is tested for match.		
Automatic dialing	IsDialNeeded	General	0
	TargetOfChannel	Automatic Dialing	Phone numbers to dial
	This is used to perform automatic dialing once OFF HOOK is detected in FXS gateway or ringing is detected on FXO port. There is no need to dial in this mode. For each channel, define destination phone number, using “TargetOfChannel<channel> = phone number” definition.		
One Stage	IsTwoStageDial	SIPgw	0

Table 5-1: Using SIP Gateway Features (continues on pages 61 to 63)

Feature	Parameter	Sheet of Excel Utility	Value
Dialing, **IP → FXO calls**	IsUseFreeChannel	General	1
	MP-10x/FXO seizes the next available FXO line, and dials the destination phone number received in INVITE message. Use the 'IsWaitForDialTone' parameter to specify whether the dialing comes after detection of dial tone, or immediately after seizing the line. The FXO gateway releases the call if busy or fast busy (reorder) tone is detected on the FXO port.	General	
Two Stage Dialing, **IP → FXO calls**	IsTwoStageDial IsUseFreeChannel	SIPgw General	1 1
	For 'Two Stage Dialing' the **MP-10x**/FXO seizes the next available PSTN/PBX line, without performing any dial, the remote user is connected over IP to PSTN/PBX, and all further signaling (dialing and call progress tones) is done directly with the PBX without gateway intervention. Usually the phone number received in INVITE message is not used, however if 'IsUseFreeChannel = 0', the phone number received in INVITE, is used for seizing specific FXO line that has same number.		
	The FXO gateway releases the call if busy or fast busy (reorder) tone is detected on the FXO port.		

5.3 Getting Started SIP Gateway Example

In this section, two **MP-108** FXS gateways are configured to be used as SIP Gateways. The end-point numbers are 101, 102, … 108 for the first gateway and 201 to 208 for the second gateway. After finishing the configuration, the User can perform telephone calls between telephones connected to a single **MP-108** unit, or between both **MP-108** gateways. SIP Proxy is not used in this example, and call routing is performed using an internal phone to an IP table.

➢ To configure the call take the following 4 steps:

Step 1: Setup

Check connections and tools setup (TFTP, BootP, HyperTerminal).

Step 2: Build *ini* file

The *ini* file is a text file containing a list of parameters for the **MP-108**. The file can be written manually or generated by the Excel utility provided. To use the Excel utility, first install the Microsoft Office 2000™ Excel application.

In this example, the EXCEL utility is used:

- Invoke the EXCEL utility
- On the “Endpoints“ sheet, define local phone numbers for each **MP-108** gateway. For the first **MP-108** gateway, define local phone numbers: 101 to 108. For the second **MP-108** gateway, define local phone numbers: 201 to 208.
- On the “Phones Prefix routing Table” sheet, define routing IP addresses for each dialed number. (This is required when Proxy is not used.)
- On the “SIPgw” sheet, define that SIP Proxy is not used.
- Click on the “Generate SIP *ini* File” button in the “General” sheet.
- Check that the “SIPgw.ini” file was generated in the folder “C:/SIPgw/”.

5.3.1 Example of *ini* file

Figure 5-1: Example of *ini* File for the First MP-108 Gateway

```
MGControlProtocolType = 8

MaxDigits = 3
CoderName = g711Alaw64k

IsProxyUsed = 0

; Phone of each end point
ChannelList = 0,8,101

; Logger information
EnableSyslog = 0
LoggerFormat = 0

; IP to Phones routing table

Prefix = 10,10.2.37.10
Prefix = 20,10.2.37.20
```

Step 3: Download Configuration

Download the *ini* file using TFTP and BootP procedures and check (viewing the RS-232 terminal) that there are no errors.

Step 4: Try!

Pick up the phone connected to port #1 of the first **MP-108** and dial 102, to the phone connected to port #2 of the same gateway. Check for progress tones in the calling end-point and for ringing in the called end-point. Answer in the called end-point and check for voice quality.

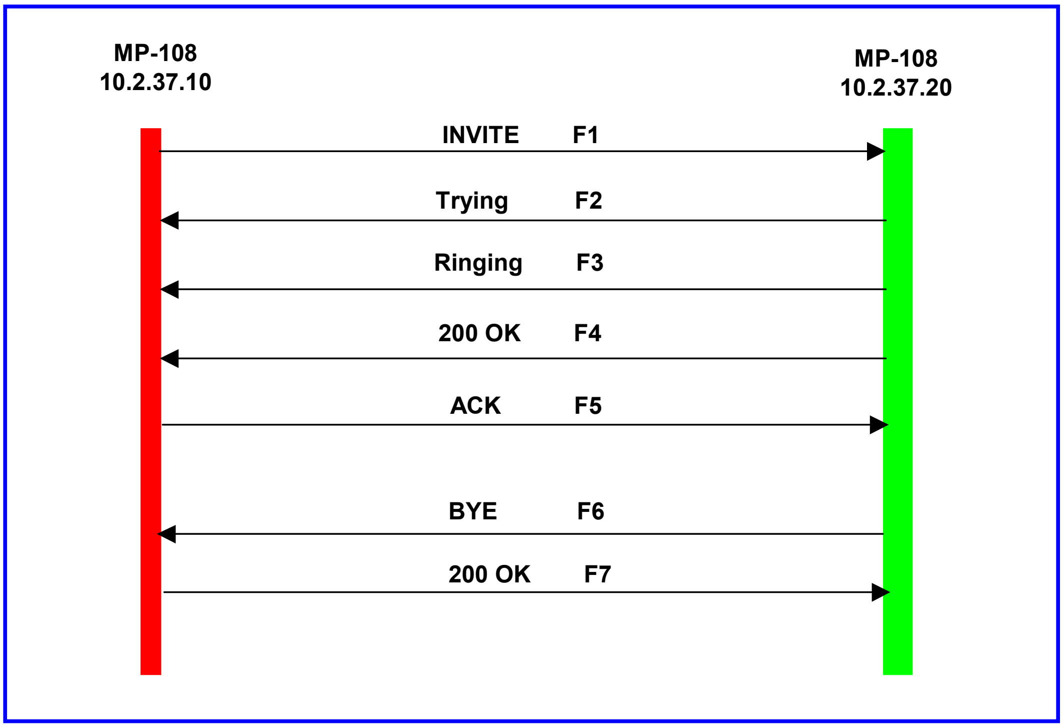
Dial 201 from the phone connected to port #1 of the first **MP-108** gateway; the phone connected to port #1 of the second **MP-108** will ring. Answer the call and check for voice quality.

5.3.2 SIP Call Flow

The following Call Flow describes SIP messages exchanged between two **MP-108** gateways during simple call.

Phone "101" dials "201", sending INVITE message to Gateway 10.2.37.20

Figure 5-2: SIP Call Flow



F1 10.2.37.10 ==> 10.2.37.20 INVITE

INVITE sip:201@10.2.37.20 SIP/2.0
Via: SIP/2.0/UDP 10.2.37.10
From: <sip:101@10.2.37.10>;tag=1c87419
To: <sip:201@10.2.37.20>
Call-ID: 87419@10.2.37.10
CSeq: 1 INVITE
Accept-Language: en

Session-Expires: 1000

Contact: 101 <sip:101@10.2.37.10>
Content-Type: application/sdp
Content-Length: 131

v=0
o=MP100 1234 5678 IN IP4 10.2.37.10
s=phone-call
c=IN IP4 10.2.37.10
t=0 0
m=audio 6000 RTP/AVP 8
a=rtpmap:8 pcma/8000

F2 10.2.37.20 ==> 10.2.37.10 100 TRYING

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.2.37.10
From: <sip:101@10.2.37.10>;tag=1c87419
To: <sip:201@10.2.37.20>;tag=3363
Call-ID: 87419@10.2.37.10
CSeq: 1 INVITE
Content-Length: 0

F3 10.2.37.20 ==> 10.2.37.10 180 RINGING

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.2.37.10
From: <sip:101@10.2.37.10>;tag=1c87419
To: <sip:201@10.2.37.20>;tag=3363
Call-ID: 87419@10.2.37.10
CSeq: 1 INVITE
Content-Length: 0

Phone "201" answers the call, and sends "200 OK" message to Gateway 10.2.37.10

F4 10.2.37.20 ==> 10.2.37.10 200 OK

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.2.37.10
From: <sip:101@10.2.37.10>;tag=1c87419
To: <sip:201@10.2.37.20>;tag=3363
Call-ID: 87419@10.2.37.10
CSeq: 1 INVITE
Session-Expires: 1000
Allow: REGISTER, OPTIONS, INVITE, ACK, CANCEL, BYE
Contact: 201 <sip:201@10.2.37.20>
Content-Type: application/sdp
Content-Length: 131

v=0
o=MP100 1234 5678 IN IP4 10.2.37.20
s=phone-call
c=IN IP4 10.2.37.20
t=0 0
m=audio 6000 RTP/AVP 8
a=rtpmap 8 pcma/8000

F5 10.2.37.10 ==> 10.2.37.20 ACK

ACK sip:201@10.2.37.20 SIP/2.0
Via: SIP/2.0/UDP 10.2.37.10
From: <sip:101@10.2.37.10>;tag=1c87419
To: <sip:201@10.2.37.20>;tag=3363
Call-ID: 87419@10.2.37.10
CSeq: 1 ACK
Content-Length: 0

Phone "201" goes onhook, gateway 10.2.37.20 sends "BYE" to Gateway 10.2.37.10

F6 10.2.37.20 ==> 10.2.37.10 BYE

```
BYE sip:101@10.2.37.10 SIP/2.0
Via: SIP/2.0/UDP 10.2.37.20
From: <sip:201@10.2.37.20>;tag=3363
To: <sip:101@10.2.37.10>;tag=1c87419
Call-ID: 87419@10.2.37.10
CSeq: 101 BYE
Content-Length: 0
```

F7 10.2.37.10 ==> 10.2.37.20 200 OK

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.2.37.20
From: <sip:201@10.2.37.20>;tag=3363
To: <sip:101@10.2.37.10>;tag=1c87419
Call-ID: 87419@10.2.37.10
CSeq: 101 BYE
Content-Length: 0
```

5.4 SIP Authentication Example

MP-108 gateway supports basic and digest authentication types, according to the SIP standard. A proxy server might require authentication before forwarding an INVITE message. A registrar server may also require authentication for client registration. A proxy replies to an unauthenticated INVITE with a 407 Proxy Authorization Required response, containing a Proxy-Authenticate header with the form of the challenge. After sending an ACK for the 407, the User Agent can then resend the INVITE with a Proxy-Authorization header containing the credentials.

User Agent, redirect or registrar servers typically use 401 Unauthorized response to challenge authentication containing a WWW-Authenticate header, and expect the re-INVITE to contain an Authorization header.

The following example describes the Digest Authentication procedure including computation of User Agent credentials.

The REGISTER request is send to registrar server for registration, as follows:

REGISTER sip:10.2.2.222 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c17940
To: <sip: 122@10.1.1.200>
Call-ID: 634293194@10.1.1.200
CSeq: 1 REGISTER
Contact: sip:122@10.1.1.200:
Expires:3600
On receiving this request the registrar returns 401 Unauthorized response.

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.2.1.200
From: <sip:122@10.2.2.222 >;tag=1c17940
To: <sip:122@10.2.2.222 >
Call-ID: 634293194@10.1.1.200
Cseq: 1 REGISTER
Date: Mon, 30 Jul 2001 15:33:54 GMT
Server: Columbia-SIP-Server/1.17
Content-Length: 0
WWW-Authenticate: Digest realm="audiocodes.com",
nonce="11432d6bce58ddf02e3b5e1c77c010d2",
stale=FALSE,
algorithm=MD5

According to the sub-header present in the WWW-Authenticate header the correct REGISTER request is formed.

Since the algorithm used is MD5, take:
The username is equal to the endpoint phone number: 122
The realm return by the proxy: audiocodes.com
The password from the *ini* file: AudioCodes.
The equation to be evaluated: (according to RFC this part is called A1).
"122:audiocodes.com:AudioCodes".
The MD5 algorithm is run on this equation and stored for future usage.
The result is: "a8f17d4b41ab8dab6c95d3c14e34a9e1"

Next we need to evaluate the par called A2. We take:
The method type “REGISTER”
Using SIP protocol “sip”
Proxy IP from *ini* file “10.2.2.222”

The equation to be evaluated:
“REGISTER:sip:10.2.2.222”.

The MD5 algorithm is run on this equation and stored for future usage.
The result is:”a9a031cfddcb10d91c8e7b4926086f7e”

The final stage:
The A1 result
The nonce from the proxy response: “11432d6bce58ddf02e3b5e1c77c010d2”
The A2 result

The equation to be evaluated:
“A1:11432d6bce58ddf02e3b5e1c77c010d2:A2”.

The MD5 algorithm is run on this equation. The outcome of the calculation is the response needed by the GW to be able top register with the Proxy.
The response is: “b9c45d0234a5abf5ddf5c704029b38cf”

At this time a new REGISTER request is issued with the response:

REGISTER sip:10.2.2.222 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c23940
To: <sip: 122@10.1.1.200>
Call-ID: 654982194@10.1.1.200
CSeq: 1 REGISTER
Contact: sip:122@10.1.1.200:
Expires:3600
Authorization: Digest, username: 122,
realm="audiocodes.com”,

nonce="11432d6bce58ddf02e3b5e1c77c010d2",

uri="10.2.2.222",

response="b9c45d0234a5abf5ddf5c704029b38cf"

On receiving this request, if accepted by the Proxy, the proxy will return a 200OK response closing the REGISTER transaction.

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.1.1.200

From: <sip: 122@10.1.1.200>;tag=1c23940

To: <sip: 122@10.1.1.200>

Call-ID: 654982194@10.1.1.200

Cseq: 1 REGISTER

Date: Thu, 26 Jul 2001 09:34:42 GMT

Server: Columbia-SIP-Server/1.17

Content-Length: 0

Contact: <sip:122@10.1.1.200>; expires="Thu, 26 Jul 2001 10:34:42 GMT"; action=proxy; q=1.00

Contact: <122@10.1.1.200:>; expires="Tue, 19 Jan 2038 03:14:07 GMT"; action=proxy; q=0.00

Expires: Thu, 26 Jul 2001 10:34:42 GMT

5.5 Remote Extension with FXO & FXS Gateways Example

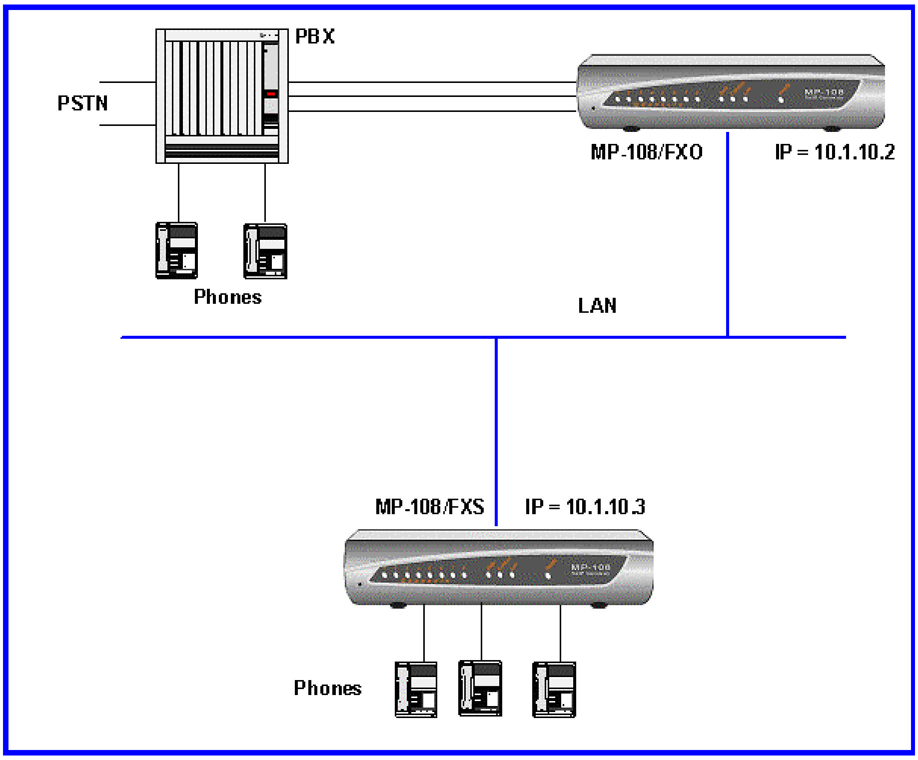
This application explains how to demonstrate remote extension via IP, using **MP-108/FXO** and **MP-108/FXS** gateways. In this configuration, PBX incoming calls are routed to "Remote Extension" via the **MP-108/FXO** and **MP-108/FXS** gateways.

➢ Requirements

- One **MP-108/FXO** gateway
- One **MP-108/FXS** gateway
- Analog phones (POTS)
- PBX – one or more PBX loop start lines
- LAN.

Connect the **MP-108/FXO** ports directly to PBX lines as shown in the diagram below:

Figure 5-3: MP-108/FXS & MP-108/FXO Layout



5.5.1.1 Dialing from Remote Extension (Phone connected to MP-108/FXS)

➢ **To configure the call take the next 6 steps:**

1. Take the handset off, to hear the dial tone coming from PBX, as if the phone was connected directly to PBX.
2. **MP-108/FXS** and **MP-108/FXO** establish a voice path connection from the phone to the PBX immediately after the phone handset was raised.
3. Dial the destination number (the DTMF digits are sent, over IP, directly to PBX).
4. All tones heard are generated from PBX (such as ringback, busy or fast busy tones).
5. There is one-to-one mapping between **MP-108/FXS** ports and PBX lines.
6. The call is disconnected when the phone connected to **MP-108/FXS** goes on-hook.

5.5.1.2 Dialing from other PBX line, or from PSTN

➢ **To configure the call take the next 5 steps:**

1. Dial the PBX subscriber number the same way as if the user's phone was connected directly to PBX.
2. Immediately as PBX rings into **MP-108**/FXO, the ring signal is "send" to phone connected to **MP-108/FXS**.
3. Once the phone's handset, connected to **MP-108/FXS**, is raised, the **MP-108/FXO** seizes the PBX line and the voice path is established between the phone and the PBX line.
4. There is a one to one mapping between PBX lines and **MP-108/FXS** ports. Each PBX line is routed to the same phone (connected to **MP-108/FXS**).
5. The call is disconnected when phone connected to **MP-108/FXS** goes on-hook.

5.5.1.3 MP-108/FXS Configuration (using the FXS *ini* file)

➢ **To Configure the MP-108/FXS *ini* file take these 4 steps:**

1. Configure in FXS ini file the endpoint numbers from 100 to 107.
2. Configure TargetOfChannel table to include phone numbers of **MP-108/FXO** gateway: such as TargetOfChannel0 = 200. (When phone connected to port #0 goes off-hook, the FXS gateway automatically dials "200" number).
3. Configure IP to phone table, to IP address and numbers of the FXO gateway: such as Prefix=20,10.1.10.2 (where 10.1.10.2 is an IP address of **MP-108/FXO**).
4. Set 'IsDialNeeded = 0' to activate automatic dialing, when the handset goes off-hook.

```
IsDialNeeded = 0
;------------------------------------
; Phone of each end point
;------------------------------------
ChannelList = 0,8,100

;------------------------------------
; Automatic dialed numbers
;------------------------------------
TargetOfChannel0 = 200
TargetOfChannel1 = 201
TargetOfChannel2 = 202
TargetOfChannel3 = 203
TargetOfChannel4 = 204
TargetOfChannel5 = 205
TargetOfChannel6 = 206
TargetOfChannel7 = 207

; Phones to IP routing table
;------------------------------------
Prefix = 20,10.1.10.2
```

5.5.1.4 MP-108/FXO configuration (using the FXO *ini* file)

➢ **To Configure the MP-108/FXO *ini* file take these 4 steps:**

1. Configure in FXO ini file the endpoint numbers from 200 to 207.
2. Configure TargetOfChannel table to include phone numbers of the **MP-108/FXS** gateway: such as TargetOfChannel0= 100 (when ringing signal is detected at port #0 of FXO gateway, the FXO gateway automatically dials "100" number).
3. Configure IP to phone table, to IP address and numbers of the FXS gateway: such as Prefix=10, 10.1.10.3 (where 10.1.10.3 is an IP address of **MP-108/FXS**).
4. Set 'IsDialNeeded = 0' to activate automatic dialing when ringing is detected at FXO port.

IsDialNeeded = 0

;-----------------------------------

; Phone of each end point

;-----------------------------------

ChannelList = 0,8,200

;-----------------------------------

; **Automatic dialed numbers**

;-----------------------------------

TargetOfChannel0 = 100

TargetOfChannel1 = 101

TargetOfChannel2 = 102

TargetOfChannel3 = 103

TargetOfChannel4 = 104

TargetOfChannel5 = 105

TargetOfChannel6 = 106

TargetOfChannel7 = 107

;-----------------------------------

; **Phones to IP routing table**

;-----------------------------------

Prefix = 10,10.1.10.3

Reader's Notes

6 Provisioning

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

This section provides details of MP-1xx provisioning.

6.1 Provisioning for SIP Operation

Initial configuration of the **MP-1xx** is provided through Web browser control or by loading of *mp108.ini* configuration file. The configuration *ini* file can be downloaded from Web browser using HTTP or TFTP protocols or by using AudioCodes configuration utility. The *ini* file name is provided in the field 'Boot File Name' of the BootP server.

To create an *ini* file, it is recommended to use the Excel™ utility provided. To use the Excel™ utility, first install the Microsoft™ Office 2000™ Excel™ application.

The *ini* file contains the following information:

- Basic and Logging Parameters shown in Table 6-1 on page 82
- Channel Parameters shown in Table 6-2 on page 84.
- SIP parameters shown in Table 6-3 on page 87).
- Names for optional Call Progress Tone file. For detailed information, refer to Section 6.4.
- Name for optional Telephony Interface (Coeff.dat) Configuration file. For **MP-1xx/FXS** and for **MP-10x/FXO** two different files should be used. Refer to Section 6.5 for more details.

Note: The names of Call Progress and Coeff.dat files in *ini* file must be enclosed in quotation marks ('…').

All *ini* file data is downloaded at startup and stored in non-volatile memory. The provisioning procedure should be used again only to modify **MP-1xx** parameters; otherwise, BootP and TFTP is not needed again.

The Default Channel Parameters are applied to all **MP-1xx** channels.

Users do not have to specify all parameters, as each unspecified parameter is set to its default value. Using the *ini* file resets all unspecified parameters to their default values.

The Channel Parameters define the DTMF/MF, Fax and Modem transfer modes. Refer to Appendix F for a detailed description of these modes.

6.1.1 Basic, Logging and Web Parameters

Table 6-1: Basic and Logging Parameters (continues on pages 82 to 83)

ini File Field Name	Valid Range and Description
MGControlProtocolType	8 = for SIP gateway
DSPVersionTemplateNumber	0 = Firmware DSP version supports PCM/ADPCM, G723 and G729 Coders (default) 1 = Firmware DSP version supports PCM/ADPCM, and NetCoder coders
EthernetPhyConfiguration	0 = 10 Base-T half-duplex 1 = 10 Base-T full-duplex 2 = 100 Base-T half-duplex 3 = 100 Base-T full-duplex 4 = auto-negotiate (Default) Auto-negotiate falls back to half-duplex mode (HD) when the opposite port is not in auto-negotiate, but the speed (10 Base-T, 100 Base -T) in this mode is always configured correctly.
DNSPriServerIP	IP address of primary DNS server
DNSSecServerIP	IP address of secondary DNS server
DHCPEnable	0 – Disable (default) 1 – Enable After the gateway is powered up it will try first to communicate with BootP server . If BootP server is not responding and " DHCPEnable =1" the gateway will send DHCP request to configure its IP address and other network parameters from enterprise DHCP server.
BootPRetries	1 = Single BootP request. 2 = 2 BootP retries - (3 seconds). 3 = 3 BootP retries - (default, 6 seconds) 4 = 10 BootP retries - (30 seconds). 5 = 20 BootP retries - (60 seconds). 6 = 40 BootP retries - (120 seconds). 7 = 100 BootP retries - (300 seconds). 15 = BootP retries forever. Number of BootP retries, and then DHCP retries (if DHCPEnable = 1) at gateway startup.
Note that BootPRetries parameter becomes active after the **MP-1xx** is reset and *ini* file is loaded. To change the parameters, first modify the *ini* file, and then reset the gateway.	
EnableDiagnostics	0 = No diagnostics (default) 1 = Perform diagnostics
WatchDogStatus	0 = Disable gateway's watch dog 1 = Enable gateway's watch dog (default)
SysLogServerIP	IP address in dotted format notation, e.g., '192.10.1.255'

Table 6-1: Basic and Logging Parameters (continues on pages 82 to 83)

ini File Field Name	Valid Range and Description
EnableSyslog	0 = Disable SysLog (default) 1 = Enable SysLog If SysLog is disabled all Logs and error messages are sent to RS-232 serial port if "DisableRS232 = 0"
DisableRS232	0 – RS-232 serial port is enabled (default) 1 – RS-232 serial port is disabled To enable sending of all log and error messages to the RS-232 serial port, define: "EnableSyslog = 0" and "DisableRS232 = 0"
LoggerFormat	0 = name + msg 1 = time + msg 2 = name + time + msg 3 = SysLog prefix + msg (default)
DisableWebTask	0 = Enable Web management (default) 1 = Disable Web management
ResetWebPassword	Allows resetting to default of Web password to: Username: "Admin" Password: "Admin"
Disable WebConfig	0 = Enable changing parameters from Web (default) 1 = Operate Web server in "read only" mode
SNMPManagerIP	IP address of SNMP Manager. The SNMP manager is used for receiving SNMP Traps. For example: SNMPManagerIP = 10.2.1.10
DisableSNMP	0 = SNMP is enabled (default) 1 = SNMP is disabled

6.1.2 Channel Parameters

Table 6-2: Channel Parameters (continues on pages 84 to 86)

ini File Field Name	Valid Range and Description
DJBufMinDelay	0 to 150 msec (default = 70) Dynamic Jitter Buffer Minimum Delay. (Described in the AudioCodes "VoPLib Reference Library User Manual", Catalog Number LTRT-00644, Section 2.3.18)
DJBufOptFactor	0 to 12 (default = 7) Dynamic jitter buffer frame error/delay optimization. This is described in the AudioCodes "VoPLib Reference Library User Manual", Catalog Number LTRT-00644 Section 2.3.18
BaseUDPPort	Range 6000 -64000 (default 6000) Lower boundary of UDP ports to be used by the gateway for RTP, RTCP and T.38 channels. The upper boundary is the BaseUDPPort + 10*(number of gateway's channels). For details refer to the Appendix A
ECHybridLoss	0 = 6 dB (default) 1 = 9 dB 2 = 0 dB 3 = 3 dB Sets the four wire to two wire worst case Hybrid loss, the ratio between the signal level sent to the hybrid and the echo level returning from the hybrid.
FaxModemBypassM	1, 2 (default = 1) Number of 20 msec payloads to be used for generating one RTP fax/modem bypass packet.
FaxModemRelayVolume	-18 to -3, corresponding to -18 dBm to -3 dBm in 1 dB steps. (Default = -12 dBm) Fax gain control.
FaxRelayECMEnable	0 = Disable using ECM mode during Fax Relay 1 = Enable using ECM mode during Fax Relay. (default)
FaxRelayEnhanced RedundancyDepth	0 to 4 (default =0) Number of repetitions applied to control packets when using T.38 standard.
FaxRelayRedundancyDepth	0 to 2 (default =0) Number of repetitions to be applied to each fax relay payload when transmitting to network (applicable only when T38ProtectionMode = 0).

Table 6-2: Channel Parameters (continues on pages 84 to 86)

ini File Field Name	Valid Range and Description
FaxRelayMaxRate	Limits the maximum rate at which fax messages are transmitted. 0 = 2.4 kbps 1 = 4.8 kbps 2 = 7.2 kbps 3 = 9.6 kbps 4 = 12.0 kbps 5 = 14.4 kbps, (default).
FaxTransportMode	Sets the Fax transport 0 = disable 1 = relay, (default) 2 = bypass.
UseT38orFRF11	0 = Use proprietary FRF.11 syntax to send/receive fax relay. 1 = Use T.38 protocol to send/receive fax relay, (default).
V21ModemTransportType	0 = Transparent, (default) 2 = ModemBypass.
V22ModemTransportType	0 = Transparent 2 = ModemBypass, (default).
V23ModemTransportType	0 = Transparent 2 = ModemBypass, (default).
V32ModemTransportType (For V.32 & V.32bis modems)	0 = Transparent 2 = ModemBypass, (default).
V34ModemTransportType (For V.34 & V.90 modems)	0 = Transparent 2 = ModemBypass, (default).
FaxModemBypassCoderType	Coder to be used while performing fax/modem bypass. Refer to acTCoders enumeration. Usually, high bit rate coders such as G.711 and G.726/G.727 should be used.
	0 = G711 A-law =0, (default)
	1 = G711 µ-law=1,
	4 = G726_32
	11 = G727_32.
T38ProtectionMode	0 = Use redundancy packets for protecting T.38 fax relay stream, (default) 1 = Use Forward Error Correction (FEC) algorithm to protect T.38 fax relay stream (isn't implemented)
DTMFVolume	-31 to 0, corresponding to -31 dBm to 0 dBm in 1 dB steps (default = -11 dBm) DTMF gain control.

Table 6-2: Channel Parameters (continues on pages 84 to 86)

ini File Field Name	Valid Range and Description
DTMFTransportType	0 = erase digit from voice stream, do not relayed to remote. 1 = erase digit from voice stream, relay to remote. (Default) 2 = digits remains in voice stream. 3 = erase digit from voice stream, relay to remote according to RFC 2833 standard
MFTransportType	0 = erase MFs from voice transport, not relayed to remote. 1 = erase MFs from voice transport, relay to remote. (= default) 2 = MFs are not erased from voice, not relayed to remote.
InputGain	-31 to 31 corresponding to -31 dB to +31 dB in 1 dB steps. (Default = 1 dB). PCM input gain.
RTPRedundancyDepth	0 = Disable redundancy packets generation (default) 1 = Enable generation of RFC 2198 redundancy packets.
VoiceVolume	-31 to 31, corresponding to -31 dB to +31 dB in 1 dB steps. (Default = 1 dB). Voice gain control
M	Number of codec payloads (5,10,20 or 30msec, depending on selected codec) to be used for generating one RTP packet. M = n payloads (n = 1, 2 or 3); M = 1 (default)
SCE	0 = silence compression disabled (default) 1 = silence compression enabled
ECE	0 = Echo Canceler disabled 1 = Echo Canceler Enabled (default)
IPPrecedence	0 to 7 (default 0) Sets the value of the IP precedence field in the IP header for all RTP packets
IPTOS	0 to 15 (default 0) Sets the value of the IP Type Of Service field in the IP header for all RTP packets
DTMFDigitLength	Time in msec for generating DTMF to PSTN side Default = 100 msec
DTMFInterDigitInterval	Time in msec between generated DTMFs to PSTN side Default = 100 msec
TestMode	0 = CoderLoopback, encoder-decoder loopback inside DSP. 1 = PCMLoopback, loopback the incoming PCM to the outgoing PCM. 2 = ToneInjection, generates a 1000 Hz tone to outgoing PCM. 3 = NoLoopback, (default).

6.1.3 SIP Parameters

Table 6-3: SIP Parameters (continues on pages 87 to 90)

ini File Field Name	Valid Range and Description
GatewayVersion	Version of Gateway, for example "GatewayVersion = 4.0 GA".
SIPGatewayName	**MP-1xx** gateway Domain Name, if specified the name is used in right side of SIP URL. If not specified, the gateway IP address is used instead (default)
IsProxyUsed	0 = no Proxy used [internal phones table used] (default) 1 = Proxy is used
ProxyIp	IP address of Proxy Server Used if IsProxyUsed = 1
ProxyName	Proxy Domain Name. If specified, the name is used as Request-URI in REGISTER, INVITE and other SIP messages. If the proxy name isn't specified, the Proxy IP address is used instead.
UserName	User name used for Registration and for BASIC/DIGEST authentication process with Proxy. This parameter is applicable only for **MP-10x** FXO gateway. For **MP-1xx** FXS the channel phone number will be used instead of the UserName (eight usernames for **MP-108**).
Password	Password used for BASIC/DIGEST authentication process with Proxy. Single password is used for all gateway ports.
Cnonce	String used by the server and client to provide mutual authentication. (Free format i.e. "Cnonce = 0a4f113b")
IsRegisterNeeded	0 = Gateway will not register to Proxy (default) 1 = Gateway will register to Proxy at power up The gateway will register up to eight times (for **MP-108** FXS) with its channel's phone numbers.
IsSpecialDigits	0 = "#" digit will terminate DTMF dialing (Default) 1 = "#" digit will not terminate dialing
SipT1Rtx	Timer T1 value for retransmission in msec. SipT1Rtx = 500
SipT2Rtx	Timer T2 value for retransmission in msec. SipT2Rtx = 4000
CoderName	Coder name that is used. g711Ulaw64k – G.711 μ-law, 20 msec g711Alaw64k – G.711 A-law, 20 msec g711Ulaw64k,5 – G.711 μ-law, 5 msec g711Alaw64k,5 – G.711 A-law, 5 msec g711Ulaw64k,10 – G.711 μ-law, 10 msec g711Alaw64k,10 – G.711 A-law, 10 msec g7231 – G.723 6.3 kbps (default) g726 – G.726 ADPCM 16 kbps g729 – G.729A

Table 6-3: SIP Parameters (continues on pages 87 to 90)

ini File Field Name	Valid Range and Description
	NetCoder6_4 – NetCoder 6.4 kbps NetCoder7_2 – NetCoder 7.2 kbps NetCoder8 – NetCoder 8.0 kbps NetCoder8_8 – NetCoder 8.8 kbps This parameter can appear several times. If several coders are used, IsMSAlgorithmOn should be set to 1 for Normal connect procedure, otherwise, only the first coder is used.
ChannelList	List of phone numbers for **MP-1xx** channels a, b, c a = first channel b = number of channels starting from “a” c = the phone number of the first channel example: ChannelList = 0,8,101 Defines phone numbers 101 to 108 for up to 8 **MP-108** channels. The *ini* file can include up to ten “ChannelList = “ entries The “ChannelList = “ can be used instead or in addition to Channel2Phone parameter.
Channel2Phone	Phone number of channel. Its format: Channel2Phone = “<channel>, <number>” <channel> is 0...23. Example: “Channel2Phone = 0, 1002” Appears once for each channel: 8 times for **MP-108**, or 4 times for **MP-104** and twice for **MP-102**. For 8-port and 24-port gateways it is suggested to use “ChannelList = “ parameter, where in a single line, all gateway’s phone numbers can be defined. The “Channel2Phone” can be used instead or in addition to “ChannelList = “ parameter.
Prefix	Mapping phone number to IP address, using phone number prefix Example: Prefix = 20,10.2.10.2 Any dialed number that starts with “20” is routed to IP address “10.2.10.2”. Needed when Gatekeeper is not used. Can appear up to 20 times. Maximal prefix size is 7 digits
IsDialNeeded	0 = no dial needed (automatic dialing) 1 = dial needed (default) If 0 = TargetOfChannel parameters define the automatic dialed number. This parameter is applicable for both FXS and FXO gateways. If "DialisNeeded =1" the FXO gateway will seize the line (after detecting the ringing signal), play a dial tone, collect DTMF digits and send INVITE to IP destination.
TargetOfChannel###	Automatic dialed phone number. The automatic dialed number, used if OFF HOOK detected in FXS channel, or ringing signal is detected in FXO channel.

Table 6-3: SIP Parameters (continues on pages 87 to 90)

ini File Field Name	Valid Range and Description
	Applicable when IsDialNeeded = 0. Its format: "TargetOfChannel<channel> = <number>". Example: "TargetOfChannel1 = 123" The parameter, if used, should be defined per gateway FXS or FXO port (channel)
IsUseFreeChannel	0 = Select the FXO channel according to destination phone number received in INVITE message, (default) 1 = Select the next available FXO channel Used for IP→ **MP-1xx**/FXO calls The next available FXO channel is selected, out of the gateway channels defined in 'ChannelList' and Channel2Phone' parameters. When using one stage dialing, ('IsTwoStageDial =0'), 'IsUseFreeChannel' should be equal to '1'. For one stage dialing the **MP-1xx**/FXO selects the next free channel, and dials into the FXO line the destination phone number received in INVITE message.
IsTwoStageDial	0 = One stage dialing 1 = Two Stage Dialing (default) Used for IP → **MP-10x**/FXO calls For 'Two Stage Dialing the **MP-10x**/FXO seizes the PSTN/PBX line, without performing any dial, the remote User is connected over IP to PSTN/PBX, and all further signaling (dialing and call progress tones) is done directly with the PBX without gateway intervention. For 'One Stage Dialing' **MP-10x**/FXO seizes the next available channel ('IsUseFreeChannel' should be '1'), and dials the destination phone number received in INVITE message. Use the 'IsWaitForDialTone' parameter to specify whether the dialing should come after detection of dial tone, or immediately after seizing of the line.
IsWaitForDialTone	0 = don't wait for dial tone 1 = Wait for dial tone (default) Used for **MP-1xx**/FXO, for 'One Stage Dialing'. If IsWaitForDialTone = 0, **MP-10x**/FXO dials phone number immediately after seizing the PSTN/PBX line, without 'listening' to dial tone. If IsWaitForDialTone = 1, **MP-10x**/FXO dials phone number only after it detects a dial tone (it can take 3-5 sec to detect a dial tone). The correct dial tone parameters should be configured in call progress tone file.
MaxDigits	2 to 19 (default 4). Maximum number of digits that can be dialed. Dialing ends when maximum number of digits dialed or timeout between digits expired (TimeBetweenDigits parameter), or '#' is dialed.

Table 6-3: SIP Parameters (continues on pages 87 to 90)

ini File Field Name	Valid Range and Description
TimeBetweenDigits	0 to 5 (default 4) Inter-digit timeout in seconds, used to terminate dialed numbers.
CallerDisplayInfo#	Caller DisplayInfo table is used to send Caller Identification information per FXS gateway port to remote IP terminal. This parameter can appear up to eight times (for **MP-108**), with up to 18 characters per string. For **MP-124** This parameter can appear up to 24 times with up to 10 characters per string. For example: CallerDisplayInfo0 = AudioCodes 1001 CallerDisplayInfo1 = AudioCodes 1002 ; ; CallerDisplayInfo8 = AudioCodes 1008
EnableCallerID	0 = Don't send CallerID signal to Gateway's FXS port (default) 1 = Calling number and Display text is sent to gateway FXS port, between first and second rings, to be displayed on phone's caller ID display for incoming call. In FXO gateway if "EnableCallerID=1", the Caller ID signal will be detected and send to IP in SIP INVITE message (as "Display" element).
EnableReversalPolarity	0 = The line polarity is not changed on answer (default) 1 = The line polarity is changed on call answer and then changed back on call release. Applicable for **MP-1xx** FXS gateways
TimeForReorderTone	Duration of played reorder tone in seconds (default 5 sec). Applicable for FXO port. The tone is played before releasing the FXO line.
TimeForDialTone	Duration of played dial tone (default 16 sec), The dial tone is played at FXS gateway port, after phone is picked up, or after the FXO gateway seizes the line in respond to ringing. During play of the dial tone, gateway waits for DTMF digits. Applicable for both FXS and FXO gateways when "Automatic dialing" feature is disabled, "IsDialNeeded = 0"

6.1.4 Loading Configuration Files

Users can use the *ini* file in order to specify Call Progress Tone table files and Line Characteristics control file to be downloaded to the **MP-1xx** during the configuration phase, either directly from the Web Browser or by using TFTP procedure. It is also possible to define whether the downloaded files are stored in non-volatile memory so the TFTP process is not required every time the gateway boots up.

The following *ini* file fields are related to this operation:

"**CallProgressTonesFilename**" – The name (and path) of the file containing the call progress tones definition. Refer to Section 6.4.4 for additional information on how to create and download this file.

"FXSCoefFileName" – The name (and path) of the file providing the FXS line characteristic parameters.

"FXOCoefFileName" – The name (and path) of the file providing the FXO line characteristic parameters.

"BurnCallProgressTonesFile" – Stores the call progress tones configuration file in non-volatile memory, if set to 1.

"BurnCoefFile" – Stores the line characteristics file in non-volatile memory, if set to 1.

6.2 The *ini* File Structure

The *ini* file can contain any number of parameters. The parameters are divided into groups by their functionality. The general form of the *ini* file is shown below.

Figure 6-1: *ini* File Structure

```
[Sub Section Name]

Parameter_Name = Parameter_Value
Parameter_Name = Parameter_Value
.
..

; REMARK

 [Sub Section Name]
      .
```

6.2.1 The *ini* File Structure Rules

- Lines beginning with a semi-colon ';' (as the first character) are ignored.
- Carriage Return must be the final character of each line.
- Number of spaces before and after "=" is not relevant.
- If there is a syntax error in the parameter name, the value is ignored.
- Syntax errors in the parameter value field can cause unexpected errors (because parameters may be set to the wrong values).
- Sub-section names are optional.
- The File name String parameters, should be placed between two inverted commas ('…'). For example CallProgressTonesFileName = 'cpusa.dat'
- The parameter field is NOT case sensitive.
- Parameter values should be entered only in decimal format, except for the Call Agent IP address.
- The *ini* file should be ended with one or more carriage returns.
- "[Files] " line should precede the CallProgressTonesFileName, and these lines, if used, should be placed at the end of *ini* file; (refer to the *ini* file

example below in Figure 6-2).

6.2.2 The *ini* File Example

An example of an *ini* file for an SIP gateway is shown in Figure 6-2.

Figure 6-2: SIP *ini* File Example

```
MGControlProtocolType = 8

[Channel Params]
DJBufferMinDelay = 75
RTPRedundancyDepth = 1

IsProxyUsed = 1
ProxyIp = 192.168.122.179
MaxDigits = 3
CoderName = g7231

; Phone of each end point
Channel2Phone = 0, 101
Channel2Phone = 1, 102
Channel2Phone = 2, 103
Channel2Phone = 3, 104

EnableSyslog = 0
LoggerFormat = 0

[Files]

CallProgressTonesFilename = 'CPUSA.dat'
BurnCallProgressTonesFile = 1
FXSCOEFFILENAME = 'coeff.dat'
BurnCoefFile = 1
```

Note 1: Using Windows Properties Display, verify that the MS-DOS name of the *ini* file is in fact *mp108.ini*, and NOT by mistake *mp108.ini.ini,* or mp108~.ini.

Note 2: To restore **MP-1xx** default configuration parameters, use the *mp1xx.ini* file without any valid parameters or with semicolon (;) character preceding all lines in the file.

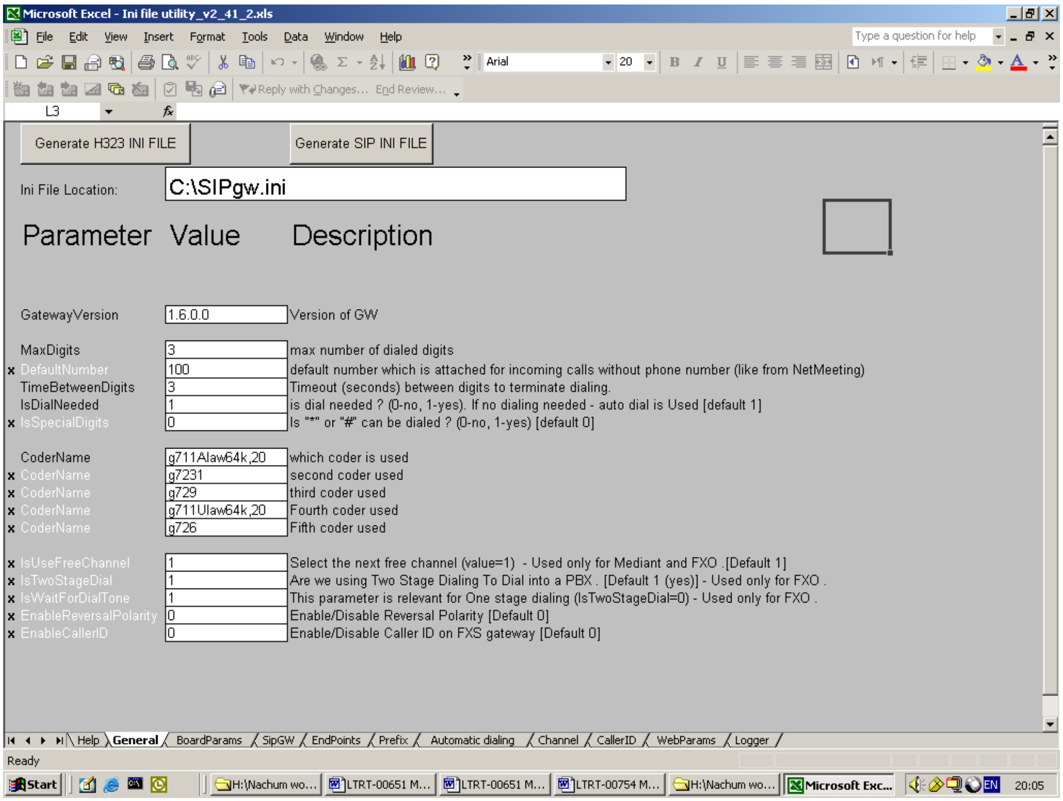
6.3 Excel Utility for *ini* File Generation

The Excel™ Utility enables easy generation of **MP-1xx** and other MediaPack series Gateway *ini* files. To use the Excel utility, first install the Microsoft™ Office 2000 Excel™ application.

Currently the utility can be used to generate *ini* files only for H.323 and SIP gateways.

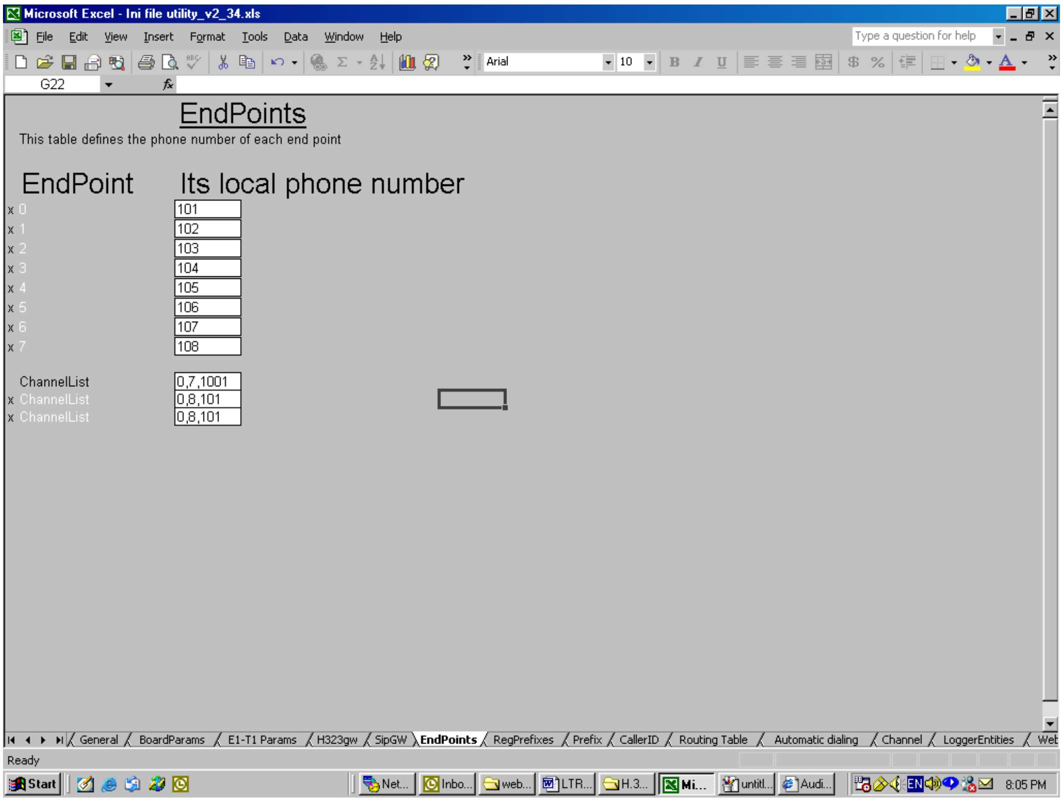
6.3.1 General Data Sheet

Figure 6-3: General Data Sheet



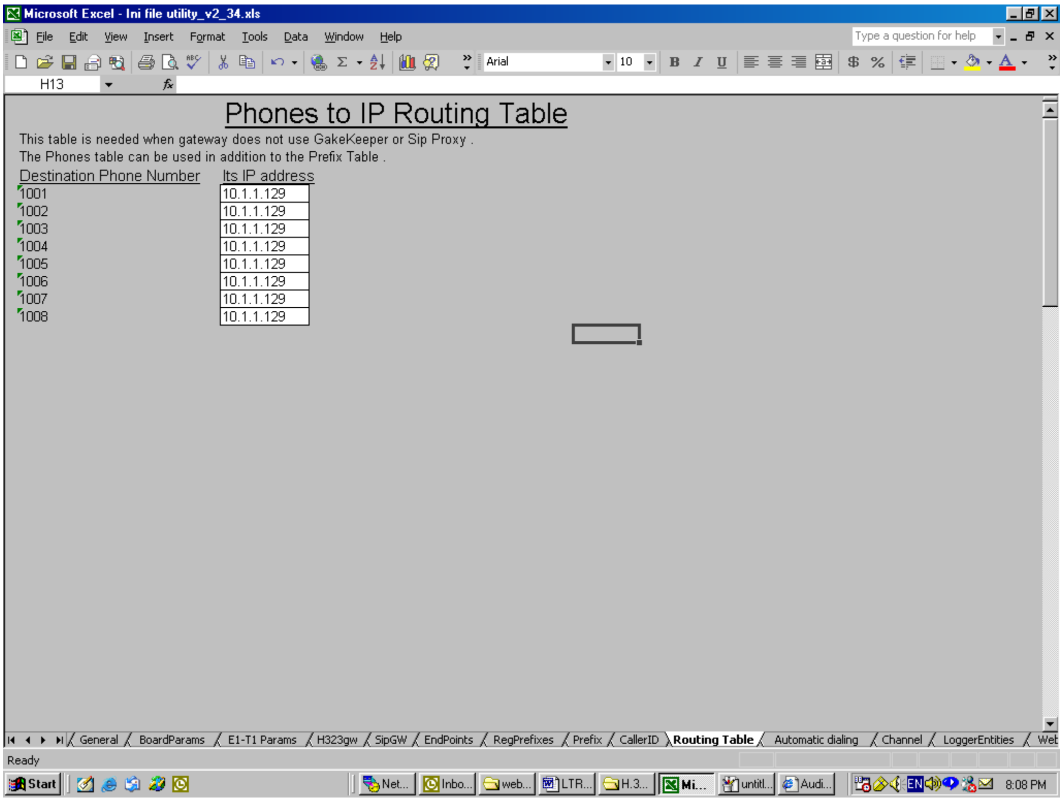
6.3.2 End Points Page

Figure 6-4: End Points Page



6.3.3 Phones to IP Routing Table

Figure 6-5: Phones to IP Routing Table



6.4 Using Call Progress Tones and Ringing

The Call Progress Tones Configuration File contains the definitions of the call progress tones and characteristics of Ringing signal to be detected/generated by the **MP-1xx**. Users can use either **MP-1xx,** one of the configuration files supplied by AudioCodes, or construct their own file.

The Call Progress Tones Configuration File used by the **MP-1xx** is a binary file (with the extension *dat*). Users can construct their own configuration file by starting from *tone.ini* file format, then modifying the file, and finally converting it into binary format using the "Download conversion utility" supplied with the **MP-1xx** package.

Please select "Convert dBm values" checkbox in the Conversion Utility.

To download the Call Progress Tones File to the **MP-1xx**, a correct definition should be used in the *mp108.ini* file. Refer to Section 6.4.4 for the description of the procedure on how to generate and download the Call Progress Tones file.

6.4.1 Format of the Call Progress *ini* File

The Call Progress Tones section of the *ini* file format starts from the following string:

- **[NUMBER OF CALL PROGRESS TONES]** – containing only the following key: "Number of Call Progress Tones" defining the number of call progress tones to be defined in the file.
- **[CALL PROGRESS TONE #X]** – containing the Xth tone definition (starting from 1 and not exceeding the number of call progress tones defined in the first section) using the following keys:
    - **Tone Type –** Call Progress tone type

      1 Dial Tone

      2 Ringback Tone

      3 Busy Tone

      4 Congestion Tone

      5 Special Information Tone

      6 Warning Tone

      7 Reorder Tone

      8 Confirmation Tone

      9 Call Waiting Tone

- **Low Freq [Hz] –** Frequency in hertz of the lower tone component in case of dual frequency tone, or the frequency of the tone in case of single tone.
- **High Freq [Hz] –** Frequency in hertz of the higher tone component in case of dual frequency tone, or zero (0) in case of single tone.
- **Low Freq Level [-dBm] –** Generation level 0 dBm to –31 dBm in [dBm].
- **High Freq Level –** Generation level. 0 to –31 dBm. The value should be set to '32' in the case of a single tone.
- **First Signal On Time [10 msec] – "**Signal On" period (in 10 msec units) for the first cadence on-off cycle.
- **First Signal Off Time [10 msec] – "**Signal Off" period (in 10 msec units) for the first cadence on-off cycle.
- **Second Signal On Time [10 msec] – "**Signal On" period (in 10 msec units) for the second cadence on-off cycle.
- **Second Signal Off Time [10 msec] – "**Signal Off" period (in 10 msec units) for the second cadence on-off cycle.

Using this configuration file, the User can create up to 16 different call progress tones using up to 15 different frequencies (in the range of 300 Hz to 2000 Hz). Each one of the call progress tones is specified by the following two parameters: the tone frequency (either single or dual frequencies are supported) and the tone cadence. This is specified by 2 sets of ON/OFF periods, but Users can discard the use of the first On/Off cycle by setting the relevant parameters to zero. When the tone is made up of a single frequency, the second frequency field should be set to zero.

For a continuous tone (such as dial tone), only the "First Signal On time" should be specified. In this case, the parameter specifies the detection period. For example if it equals 300, then the tone is detected after 3 seconds (300 x 10 msec).

Note 1: When defining several continuous tones, the "First Signal On Time" parameter should have the same value for all tones.

Note 2: The tones frequency should differ by at least 40 Hz from one tone to other defined tones.

6.4.2 Default Template for Call Progress Tones

The **MP-1xx** is initialized with the default Call Progress Tones configuration template shown in Table 6-4. If you need to change one of the tones, edit the default call progress.txt file.

For example: to change the dial tone to 440 Hz only, replace the #Dial tone section in Table 6-4 with the following text:

#Dial tone

[CALL PROGRESS TONE #1]

Tone Type=1

Low Freq [Hz]=440

High Freq [Hz]=0

Low Freq Level [-dBm]=10 (-10 dBm)

High Freq Level [-dBm]=32 (use 32 only if a single tone is required)

First Signal On Time [10msec]=300; the dial tone is detected after 3 sec

First Signal Off Time [10msec]=0

Second Signal On Time [10msec]=0

Second Signal Off Time [10msec]=0

Users can specify several tones of the same type using Tone Type definition. These additional tones are used only for tone detection. Generation of specific tone is according to the first definition of the specific tone. For example, the User can define an additional dial tone by appending the second dial tone definition lines to the tone *ini* file. The **MP-1xx** reports dial tone detection if either one of the two tones has been detected.

Table 6-4: Call Progress Tones Template (continues on pages 99 to 102)

[NUMBER OF CALL PROGRESS TONES]
Number of Call Progress Tones=9
#Dial tone
[CALL PROGRESS TONE #0]
Tone Type=1
Low Freq [Hz]=350
High Freq [Hz]=440
Low Freq Level [-dBm]=13
High Freq Level [-dBm]=13
First Signal On Time [10msec]=300

Table 6-4: Call Progress Tones Template (continues on pages 99 to 102)

First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
#Dial tone
[CALL PROGRESS TONE #1]
Tone Type=1
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=10
High Freq Level [-dBm]=32
First Signal On Time [10msec]=300
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
#Ringback
[CALL PROGRESS TONE #2]
Tone Type=2
Low Freq [Hz]=440
High Freq [Hz]=480
Low Freq Level [-dBm]=19
High Freq Level [-dBm]=19
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=200
Second Signal Off Time [10msec]=400
#Ringback
[CALL PROGRESS TONE #3]
Tone Type=2
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=16
High Freq Level [-dBm]=32

Table 6-4: Call Progress Tones Template (continues on pages 99 to 102)

First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=100
Second Signal Off Time [10msec]=300
#Busy
[CALL PROGRESS TONE #4]
Tone Type=3
Low Freq [Hz]=480
High Freq [Hz]=620
Low Freq Level [-dBm]=24
High Freq Level [-dBm]=24
First Signal On Time [10msec]=50
First Signal Off Time [10msec]=50
Second Signal On Time [10msec]=50
Second Signal Off Time [10msec]=50
#Busy
[CALL PROGRESS TONE #5]
Tone Type=3
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=20
High Freq Level [-dBm]=32
First Signal On Time [10msec]=50
First Signal Off Time [10msec]=50
Second Signal On Time [10msec]=50
Second Signal Off Time [10msec]=50
#Reorder tone
[CALL PROGRESS TONE #6]
Tone Type=7
Low Freq [Hz]=480
High Freq [Hz]=620
Low Freq Level [-dBm]=24

Table 6-4: Call Progress Tones Template (continues on pages 99 to 102)

High Freq Level [-dBm]=24
First Signal On Time [10msec]=25
First Signal Off Time [10msec]=25
Second Signal On Time [10msec]=25
Second Signal Off Time [10msec]=25
#Confirmation tone
[CALL PROGRESS TONE #7]
Tone Type=8
Low Freq [Hz]=350
High Freq [Hz]=440
Low Freq Level [-dBm]=20
High Freq Level [-dBm]=20
First Signal On Time [10msec]=10
First Signal Off Time [10msec]=10
Second Signal On Time [10msec]=10
Second Signal Off Time [10msec]=10
#Call Waiting Tone
[CALL PROGRESS TONE #8]
Tone Type=9
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=20)
High Freq Level [-dBm]=32
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=30
Second Signal Off Time [10msec]=900

6.4.3 Format of the Ringing Definition

The ringing pattern configures the ringing tone frequency and up to 4 ringing cadences. It is applicable for the **MP-1xx/FXS** gateways. Only single ringing pattern can be defined, if not a default ringing pattern applies. The ringing frequency can be configured in the range from 10 Hz up to 200 Hz with a 5 Hz resolution. Each ringing cadence period can be defined as single ringing burst. Refer to the examples below.

The distinctive ringing section of the *ini* file format contains the following strings:

- [NUMBER OF DISTINCTIVE RINGING PATTERNS]
  - Number of Ringing patterns = 1
  - [Ringing Pattern #0]
  - Ring Type =0

  - **Freq [Hz] –** Frequency in hertz of the ringing tone.
  - **First Ring On Time [10 msec] – "**Ring On" period (in 10 msec units) for the first cadence on-off cycle.
  - **First Ring Off Time [10 msec] – "**Ring Off" period (in 10 msec units) for the first cadence on-off cycle.
  - **Second Ring On Time [10 msec] – "**Ring On" period (in 10 msec units) for the second cadence on-off cycle.
  - **Second Ring Off Time [10 msec] – "**Ring Off" period (in 10 msec units) for the second cadence on-off cycle.
  - **Third Ring On Time [10 msec] – "**Ring On" period (in 10 msec units) for the third cadence on-off cycle.
  - **Third Ring Off Time [10 msec] – "**Ring Off" period (in 10 msec units) for the third cadence on-off cycle.
  - **Fourth Ring On Time [10 msec] – "**Ring Off" period (in 10 msec units) for the forth cadence on-off cycle.
  - **Fourth Ring Off Time [10 msec] – "**Ring Off" period (in 10 msec units) for the forth cadence on-off cycle.
  - **Burst –** configures the ringing signal to be a single ringing burst comprised of all specified above cadences. The "Burst" string is defined per each ringing cadence and it must appear between "First/Second/Third/Forth" string and the "Ring On/Off Time".

6.4.3.1 Examples of Various Ringing Signals

#Regular North American Ringing Pattern: 20 Hz, 2 sec On, 4 sec Off

[NUMBER OF DISTINCTIVE RINGING PATTERNS]

Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=20
First Ring On Time [10msec]=200
First Ring Off Time [10msec]=400

#GR-506-CORE Ringing Pattern 3: 20 Hz ringing comprised of three cadences
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=20
First Ring On Time [10msec]=40
First Ring Off Time [10msec]=20
Second Ring On Time [10msec]=40
Second Ring Off Time [10msec]=20
Third Ring On Time [10msec]=80
Third Ring Off Time [10msec]=400

#EN 300 001 Ring – Finland: informative ringing nr. 3: three ringing bursts followed by repeated ringing of 1 sec on and 3 sec off.
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=25
First Burst Ring On Time [10msec]=30
First Burst Ring Off Time [10msec]=30
Second Burst Ring On Time [10msec]=30
Second Burst Ring Off Time [10msec]=30
Third Burst Ring On Time [10msec]=30
Third Burst Ring Off Time [10msec]=30
Fourth Ring On Time [10msec]=100
Fourth Ring Off Time [10msec]=400

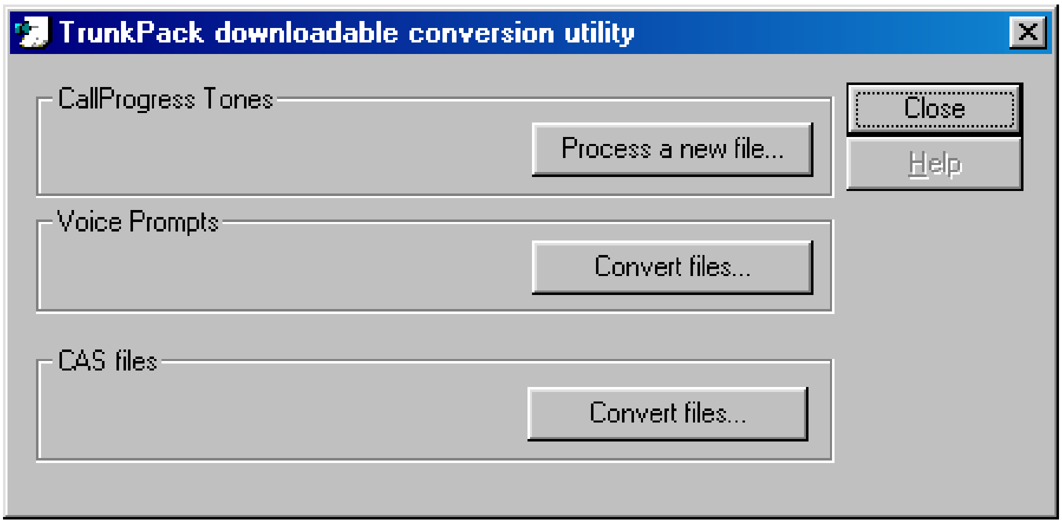
6.4.4 Call Progress Tone and Ringing Generation and Download Procedure

Follow the directions below for generation and download of the Call Progress Tone file.

➢ **To run the procedure take the following 10 steps:**

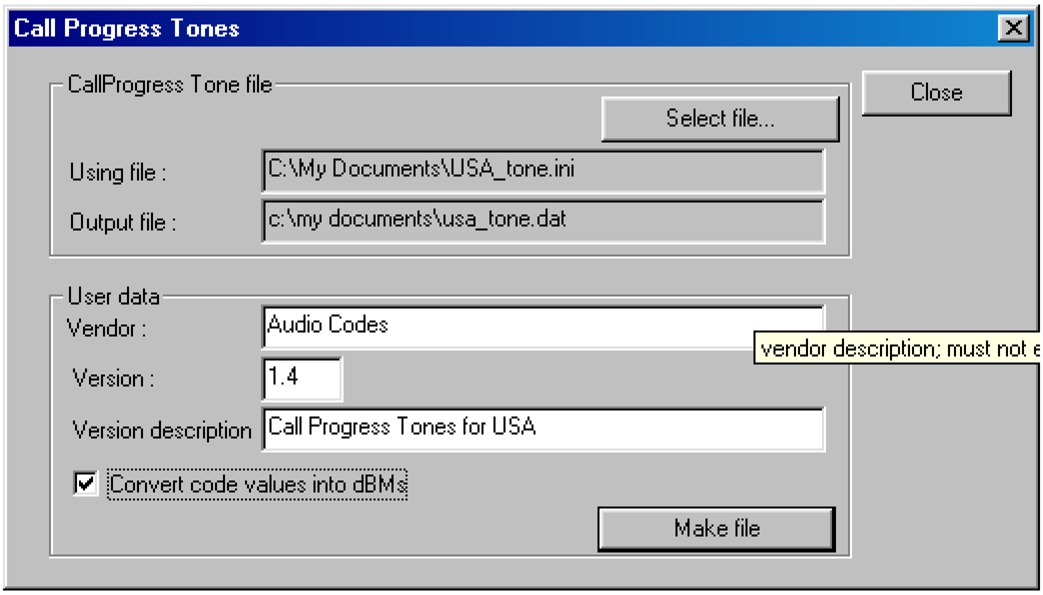
1. Prepare the ***tone.ini*** file including call progress tones and ringing parameters.
2. Use the "Download conversion utility" to generate binary ***tone.dat*** file

Figure 6-6: Download Selection Screen



3. Click "Process a new file.
4. Select input file such as ***usa_tone.ini*** and fill the Vendor and Version fields.

Figure 6-7: File Selection Screen



5. Select the 'Convert Code into dBm' checkbox
6. Click "Make File" button and then close the application.
7. Edit the ***mp-1xx8.ini*** file and add the following two lines:

   CallProgressTonesFilename = 'usa_tone.dat'
   BurnCallProgressTonesFile = 1

8. Save the "***usa_tone.dat***" and "***mp108.ini***" files in TFTP folder
9. Set the Boot file name in the BootP server: ***mp108.ini***.
10. Activate the BootP and TFTP servers and reset the **MP-1xx** gateway (refer to Section 6.1, describing **MP-1xx** provisioning).

6.5 The coeff.dat Configuration File

The purpose of the coeff.dat configuration file is to provide best feed and transmission quality adaptation for different phone line types. Two different *coeff.dat* files are needed for **MP-1xx/FXS** and for **MP-10x/FXO** gateways. The file consists of a set of parameters for the signal processor of the loop interface devices. This parameter set provides control of the following AC and DC interface parameters:

- DC (battery) feed characteristics
- AC impedance matching
- Transmit gain
- Receive gain
- Hybrid balance
- Frequency response in transmit and receive direction
- Hook thresholds
- Ringing generation and detection parameters

This means, for example, that changing impedance matching or hybrid balance requires no hardware modifications, so that a single device is able to meet requirements for different markets. The digital nature of the filters and gain stages also ensures high reliability, no drifts (over temperature or time) and simple variations between different line types.

The *coeff.dat* configuration file is produced by AudioCodes for each market after comprehensive performance analysis and testing, and can be modified on request. The current file supports US line type of 600 ohm AC impedance and 40 V RMS ringing voltage for REN = 2.

In future software releases, it will be expanded to consist of different sets of line parameters, which can be selected in the *ini* file, for each port.

To support different types of countries and markets, it is necessary to support loading of a new *Coefficients.ini* file. This file consist of AC and DC line parameters for the peripheral devices. This file is loaded into the **MP-1xx** using the TFTP, in the same way as for the *tones.dat* file.

Reader's Notes

7 SNMP and Web Management

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

This section describes MP-1xx SNMP and Web Management and its various supported functions.

7.1 SNMP Management

7.1.1 SNMP Overview

SNMP (Simple Network Management Protocol) is a standard network-based client/server-based control protocol to manage devices in the Network. The client program (called the Network Manager) makes connections to a server program, called the SNMP Agent. The SNMP Agent, embedded on a remote network device, serves information to the Network Manager regarding the device's status. The database used by the Agent to retrieve information, is referred to as the SNMP Management Information Base (MIB), and is a standard set of statistical and control values. Apart from the standard MIBs documented in IETF's RFCs, SNMP additionally allows the usage of private MIBs, containing non-standard information set.

Directives, issued by the network manager client to an SNMP Agent, consist of the identifiers of SNMP variables (referred to as MIB object identifiers or MIB variables) along with instructions to either get the value for the identifier, or set the identifier to a new value.

The definitions of MIB variables supported by a particular agent are incorporated in descriptor files, written in Abstract Syntax Notation (ASN.1) format, made available to network management client programs so that they can become aware of MIB variables and their usage.

The **MP-1xx** contains an embedded SNMP Agent supporting both general network MIBs (such as the IP MIB), VoP-specific MIBs (such as RTP, MGCP, etc.) and a proprietary MIB (known also as AudioCodes MIB) enabling a deeper probe into the inter-working of the gateway. All the supported MIBs files are supplied as part of the release.

7.1.2 SNMP Message Standard

Four types of SNMP messages are defined:

- "**Get**" Request that returns the value of a named object.
- "**Get-Next**" Request that returns the next name (and value) of the "next" object supported by a network device given a valid SNMP name.
- "**Set**" Request that sets a named object to a specific value.
- "**Trap**" Message generated asynchronously by network devices. It notifies the network manager of a problem apart from polling of the device.

Each of the following message types fulfills a particular requirement of network managers:

- **Get Request**: Specific values can be fetched via the "get" request to determine the performance and state of the device. Typically, many different values and parameters can be determined via SNMP without the overhead associated with logging into the device, or establishing a TCP connection with the device.
- **Get Next Request**: Enables the SNMP standard network managers to "walk" through all SNMP values of a device (via the "get-next" request) to determine all names and values that the device supports. This is accomplished by beginning with the first SNMP object to be fetched, fetching the next name with a "get-next", and repeating this operation until an error is encountered (indicating that all MIB object names have been "walked").
- **Set Request**: The SNMP standard provides a method of effecting an action associated with a device (via the "set" request) to accomplish activities such as disabling interfaces, disconnecting Users, clearing registers, etc. This provides a way of configuring and controlling network devices via SNMP.
- **Trap Message**: The SNMP standard furnishes a mechanism by which devices can "reach out" to a network manager on their own (via the “trap" message) to notify the manager of a problem with the device. This typically requires each device on the network to be configured to issue SNMP traps to one or more network devices that are awaiting these traps. The Trap messages are send to SNMP Manager. The IP address of SNMP Manager is defined in the *ini* file or via Web Browser (in Network Settings)

7.1.3 SNMP MIB Objects

The SNMP MIB is arranged in a tree-structured fashion, similar in many ways to a disk directory structure of files. The top level SNMP branch begins with the ISO "internet" directory, which contains four main branches:

- The "**mgmt**" SNMP branch contains the standard SNMP objects usually supported (at least in part) by all network devices.
- The “**private**" SNMP branch contains those "extended" SNMP objects defined by network equipment vendors.
- The "**experimental**" and "**directory**" SNMP branches, also defined within the "internet" root directory, are usually devoid of any meaningful data or objects.

The "tree" structure described above is an integral part of the SNMP standard, however the most pertinent parts of the tree are the "leaf" objects of the tree that provide actual management data regarding the device. Generally, SNMP leaf objects can be partitioned into two similar but slightly different types that reflect the organization of the tree structure:

- **Discrete MIB Objects**: Contain one precise piece of management data. These objects are often distinguished from "Table" items (below) by adding a ".0" (dot-zero) extension to their names. The operator must merely know the name of the object and no other information.
- **Table MIB Objects**: Contain multiple pieces of management data. These objects are distinguished from "Discrete" items (above) by requiring a "." (dot) extension to their names that uniquely distinguishes the particular value being referenced. The "." (dot) extension is the "instance" number of an SNMP object. In the case of "Discrete" objects, this instance number is zero. In the case of "Table" objects, this instance number is the index into the SNMP table. SNMP tables are special types of SNMP objects, which allow parallel arrays of information to be supported. Tables are distinguished from scalar objects, in that tables can grow without bounds. For example, SNMP defines the "ifDescr" object (as a standard SNMP object) that indicates the text description of each interface supported by a particular device. Since network devices can be configured with more than one interface, this object could only be represented as an array.

By convention, SNMP objects are always grouped in an "Entry" directory, within an object with a "Table" suffix. (The "ifDescr" object described above resides in the "ifEntry" directory contained in the "ifTable" directory).

7.1.4 SNMP Extensibility Feature

One of the principal components of any respectable SNMP manager is a “MIB Compiler" which allows new MIB objects to be added to the management system. When a MIB is compiled into an SNMP manager, the manager is made "aware" of new objects that are supported by agents on the network. The concept is similar to adding a new schema to a database.

Typically, when a MIB is compiled into the system, the manager creates new folders or directories that correspond to the objects. These folders or directories can typically be viewed with a "MIB Browser", which is a traditional SNMP management tool incorporated into virtually all network management systems.

The act of compiling the MIB allows the manager to know about the special objects supported by the agent and access these objects as part of the standard object set.

7.1.5 MP-1xx Gateway Supported MIBs

The **MP-1xx** gateway contains an embedded SNMP Agent supporting the following MIBs:

- **The Standard MIB (MIB-II)** - The various SNMP values in the standard MIB are defined in RFC 1213. The standard MIB includes various objects to measure and monitor IP activity, TCP activity, UDP activity, IP routes, TCP connections, interfaces, and general system description.
- **RTP MIB** - The RTP MIB is supported per the RFC 2959. It contains objects relevant to the RTP streams generated and terminated by the gateway and to the RTCP information related to these streams.
- **AcBoard MIB** - This proprietary MIB contains objects related both to the configuration of the gateway and channels as well as to run-time information. Through this MIB, the User can set up the gateway configuration parameters, reset the gateway, and monitor the gateway's operational robustness and quality of service during run-time.

7.2 Web Management

7.2.1 Overview

The **MP-1xx gateway** contains an Embedded Web Server to be used both for gateway configuration, including downloading of configuration files, and for run-time monitoring. The Web Server can be accessed from any standard Web browser, such as Microsoft™ Internet Explorer, Netscape™ Navigator, etc. Specifically, Users can employ this facility to set up the gateway configuration parameters needed to configure the gateway. Users also have the option to reset the gateway to apply the new set of parameters.

Access to the Embedded Web Server is controlled by protection and security mechanisms described below.

Note: The **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have identical functionality (the **MP-102** supports FXS only), except for the number of channels, and are referred to collectively in this manual as the **MP-1xx**.

7.2.2 Password Control

The Embedded Web Server is protected by a unique username-password combination. The first time a browser request (click on one of the buttons in the Home Page) is made, the User is requested to provide its username-password so that the User can obtain access. Subsequent requests are negotiated by the browser on behalf of the User, so that the User doesn't have to re-enter the username-password for each request, but the request is still authenticated.

An additional level of protection is obtained by a restriction that no more than three IP addresses can access the Embedded Web Server concurrently. With this approach, a fourth User is told that the Server is busy, even if the correct username-password was provided.

7.2.2.1 The Embedded Web Server Username-Password

The default username-password for all gateways is:

- Username = "Admin"
- Password = "Admin "

Change the Web password using the "Configuration Menu > Change Password" selection and then follow the pop-up window directives. The password can be a maximum of 7 characters. The new password is active only after restarting the gateway using the reset button of the Embedded Web Server. Otherwise, the "old" password is still active.

The User can reset the Web password (to the default values) using an *ini* file parameter called “RESETWEBPASSWORD”. The Web password is automatically the default password.

7.2.3 Web Configuration

The Embedded Web Server can be configured using *ini* file parameters.

7.2.3.1 Read-only Mode

The Embedded Web Server can be initialized to “read-only mode” by setting the “DISABLEWEBCONFIG = 1” *ini* file parameter (the default state is read-write mode). In this mode, all the Web pages are presented in read-only mode. By selecting this mode, the User disables the capability to modify the configuration data. In addition, the User does NOT have access to the “Change Password” page or to the reset page. When the gateway is controlled through PCI, the Embedded Web server is always in read-only mode.

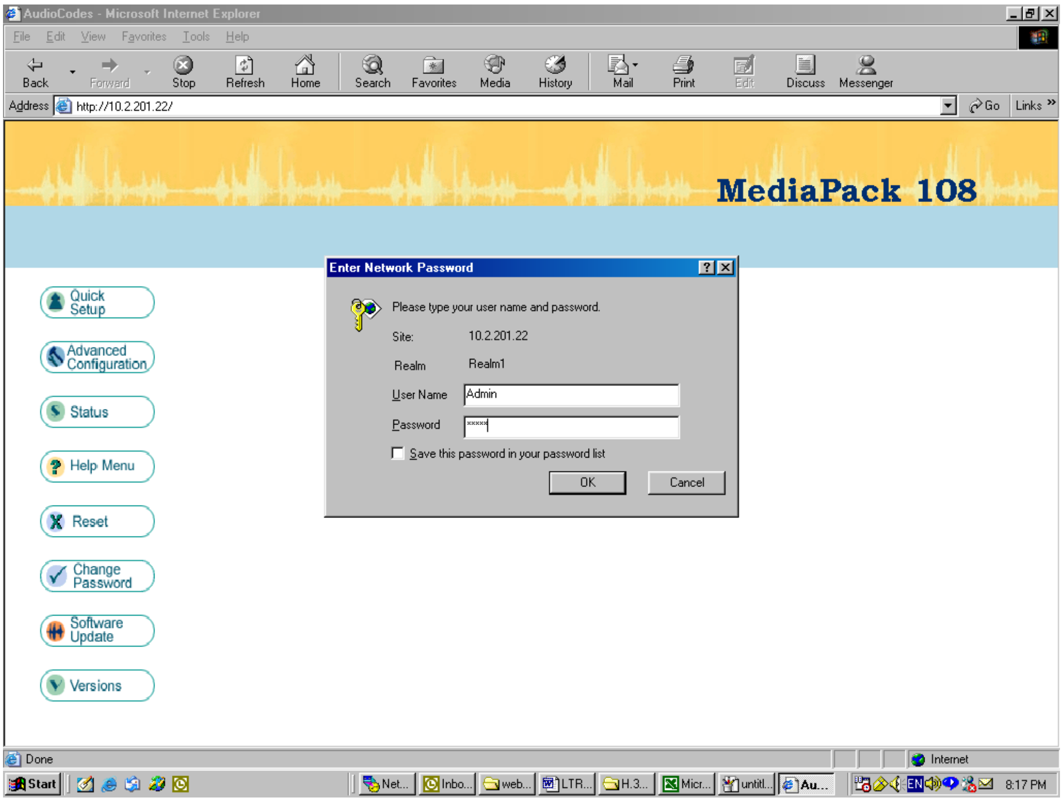
7.2.3.2 Disable/Enable Embedded Web Server

To deny access to the gateway through HTTP protocol, the User has the capability to disable the embedded Web server task. To disable the Web task, use an *ini* file parameter called “DISABLEWEBTASK = 1”. The default is to Web task enabled. When the gateway is controlled through PCI, the Embedded Web server is always activated. The User cannot disable the task in PCI mode.

7.2.4 Using the Embedded Web Server

This section explains how to use the Embedded Web Server. After the initial IP address is set to the gateway, it is possible to connect with the integral Web-based configuration application. To access this Web application, invoke any standard Web-browsing application such as Microsoft™ Internet Explorer, Netscape™ Navigator, etc., and specify the IP address of the gateway in the address field; the Embedded Web Server screen appears, as shown in Figure 7-1. After entering for the first time, the default User name and the Password (Admin, Admin), the user is requested to enter the new User Name and Password.

Figure 7-1: Embedded Web Server – Home Page

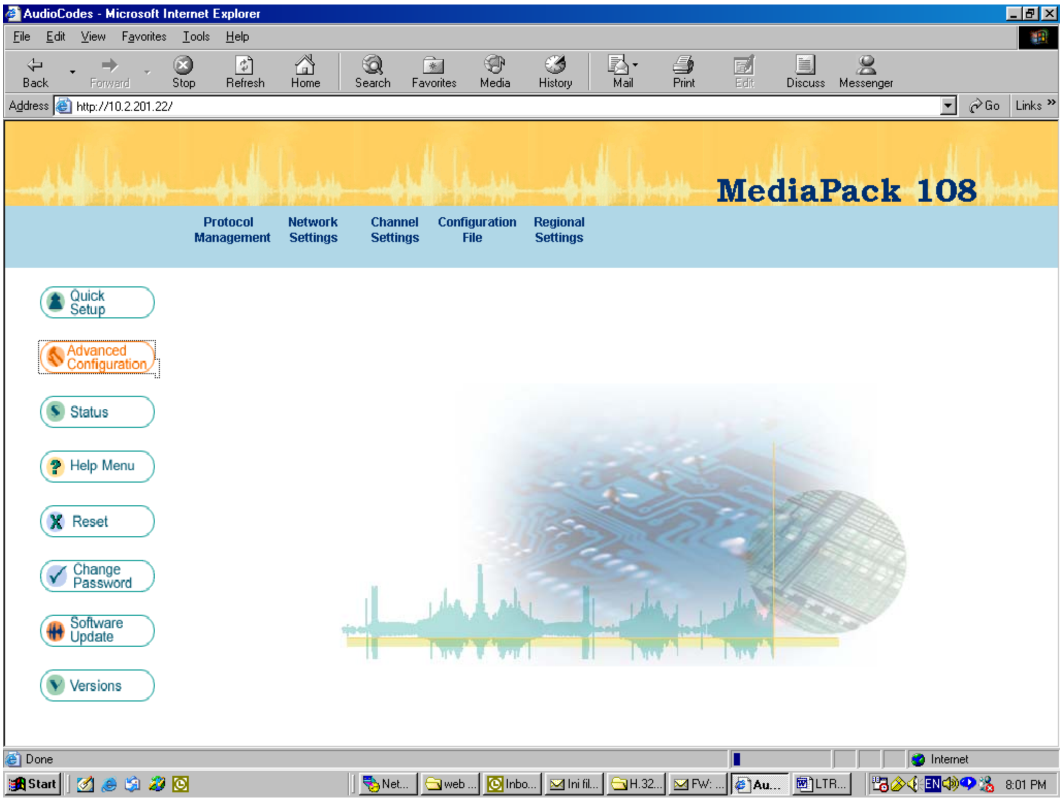


7.2.4.1 Set Up Gateway Configuration Parameters

To configure the gateway parameters you can use either the “Quick Setup” menu or go directly to “Advance Configuration” menu. Quick setup provides a basic set of gateway configuration settings. An example of the “Quick Setup” configuration is described in Section 4.3 on page 43.

Clicking the “Advance Configuration” button leads to the following screen.

Figure 7-2: Embedded Web-Server - Gateway Parameters

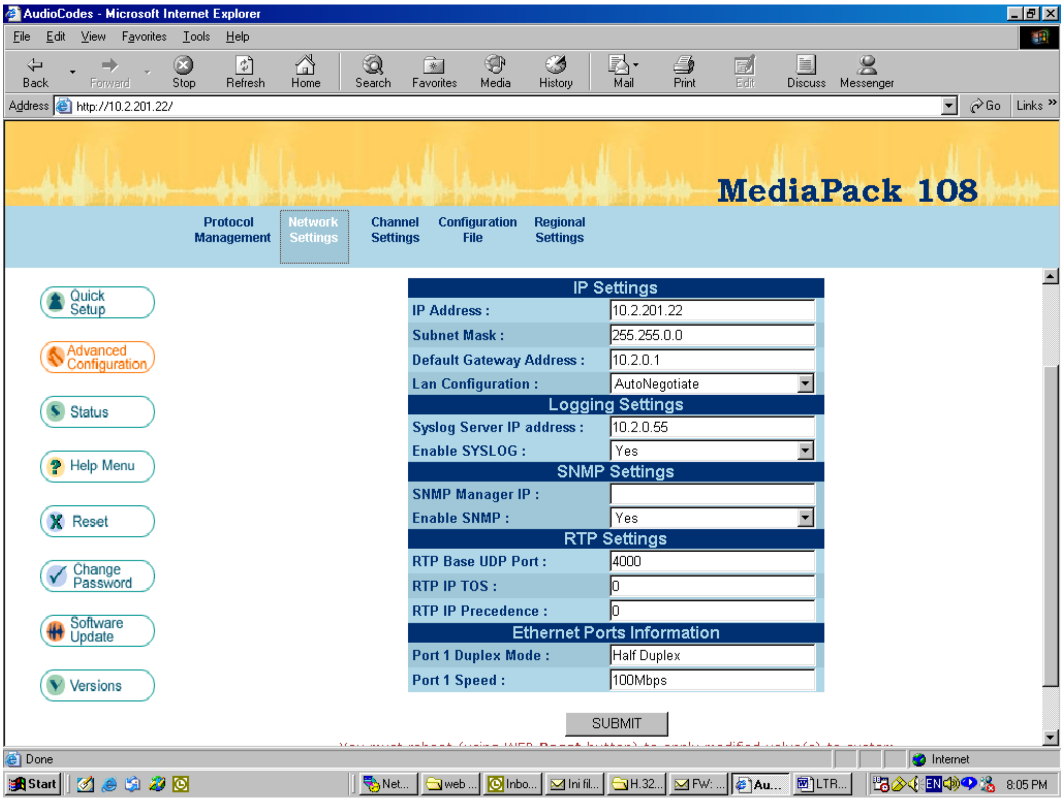


Selecting each of the sub menus shows the active configuration of each section and the values of the relevant parameters.

7.2.4.2 Set up Gateway Network Parameters

To change the gateway network settings, select the "Network Settings" tab shown below.

Figure 7-3: Web Server – Network Settings



From network settings page the User can define:

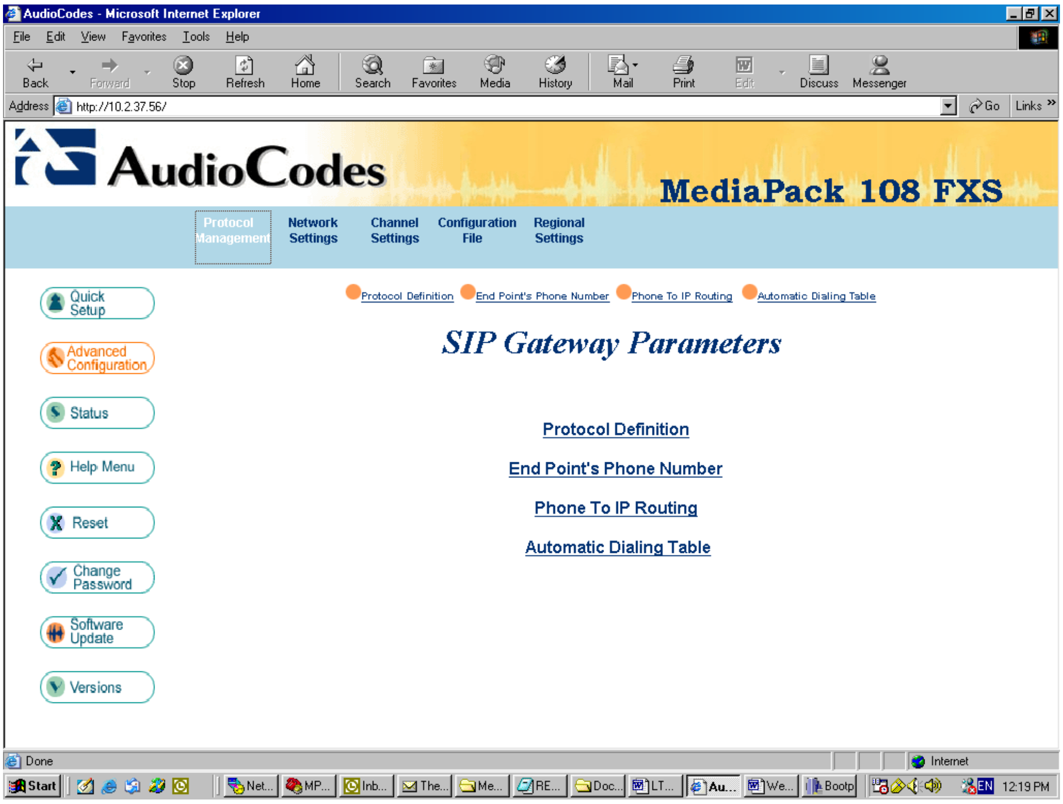
- IP settings including the gateway IP address and subnet mask.
- Logging settings, such as IP address of SysLog Server. If the SysLog Server is disabled, the logging data is sent to the gateway's serial RS-232 port.
- SNMP settings.
- RTP settings, including RTP base port and IP TOS and Precedence QOS parameters.
- Ethernet status

7.2.5 Configuration of MP-108 SIP Parameters

To configure **MP-108** SIP parameters, select “Protocol Management” tab from “Advance Configuration” page.

From SIP Gateway Parameters screen you can view and configure the SIP Gateway parameters, to set gateway endpoint’s phone numbers and Phone to IP routing table (which is needed if Proxy is not used)

Figure 7-4: SIP Gateway Parameters



7.2.5.1 SIP Protocol Definition

From this screen you can view and define SIP Gateway parameters, VoIP coder(s), Proxy IP address and more. After changing the parameters press the "Submit" button and then reset the gateway using the "Reset" button.

Figure 7-5: SIP Protocol Definition Page

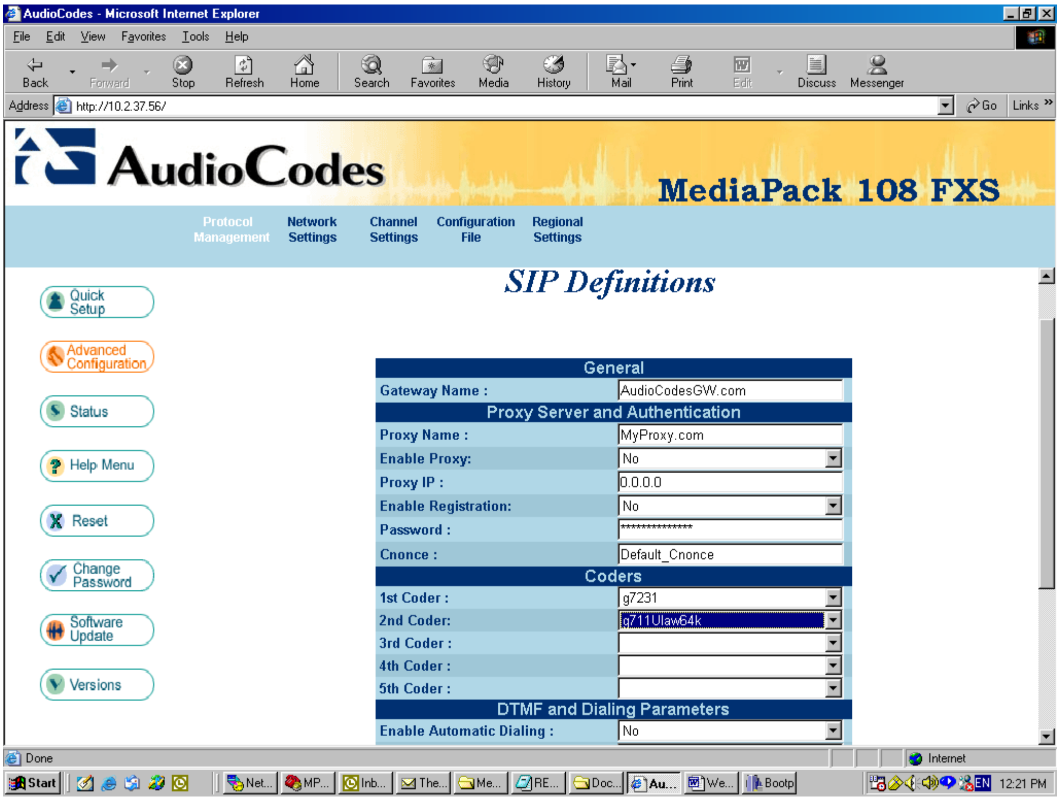


Figure 7-6: SIP Parameters

General	
Gateway Name :	AudioCodesGW.com
Proxy Server and Authentication	
Proxy Name :	MyProxy.com
Enable Proxy:	No
Proxy IP :	0.0.0.0
Enable Registration:	No
Password :	**************
Cnonce :	Default_Cnonce
Coders	
1st Coder :	g7231
2nd Coder:	g711Ulaw64k
3rd Coder :	
4th Coder :	
5th Coder :	
DTMF and Dialing Parameters	
Enable Automatic Dialing :	No
*Max Digits In Phone Num:	6
*Digits Timeout[sec] :	4
*Use '#' digit for dialing termination :	No

FXO Gateway Parameters (Applicable only for **MP-1xx/FXO** gateway) shown below in Figure 7-7.

Figure 7-7: FXO Gateway Parameters

FXO Gateway Parameters	
*Select Next Available Channel :	No
*Dialing Mode :	Two Stage
*Waiting For Dial Tone :	Yes
*Reorder Tone Duration[sec] :	5
*Dial Tone Duration[sec] :	16

7.2.5.2 Endpoint's Phone Numbers

From this screen, you can view and define SIP Gateway phone numbers.

Figure 7-8: Endpoint's Phone Numbers

Channel(s)	Phone Number
0-7	101

"**Endpoint's Phone Number" Table** is used to allocate phone numbers to gateway ports, and to enable/disable gateway ports. The table defines phone numbers for gateway endpoints. The endpoints that aren't defined are disabled. In Channel(s) field a range of endpoints can be entered, such as "0-7" for **MP-108**. For a single endpoint, a single number can be entered in the channel field.

After changing these phone numbers, press the "Submit" button and then reset the gateway using the "Reset" button.

7.2.5.3 Phone to IP Routing Table

The “**Phone to IP Routing” Table** is needed if the gateway operates without a Proxy. It contains up to 20 rows. Each row associates a called phone number prefix with destination IP address. The Phone to IP Routing Table is shown in Figure 7-9 below.

Figure 7-9: Phone to IP Routing Table

Destination Phone Prefix	IP Address
25	10.2.201.11
3	10.2.32.150
4	10.2.32.150
5	10.2.32.154
11	10.2.32.155

In the example above, all incoming calls with dialed numbers starting with 25 will be routed to IP address 10.2.201.11.

This table can be changed on-the-fly, without resetting the gateway.

7.2.5.4 Automatic Dialing Table

"Automatic Dialing Table" defines destination numbers for phone ➔ IP calls. It can be used if "Automatic Dialing" feature is enabled. The table is applicable for FXS and FXO analog gateways, for outgoing, phone ➔ IP calls. The table contains pre configured phone numbers per gateway port. The number is automatically dialed if phone is picked up. The Automatic Dialing Table is shown in Figure below, using the **MP-108** as the example.

Figure 7-10: Automatic Dialing Table

GW Port	Destination Phone Number
Port 1	1002
Port 2	1003
Port 3	1004
Port 4	1101
Port 5	1102
Port 6	1105
Port 7	1234
Port 8	2222

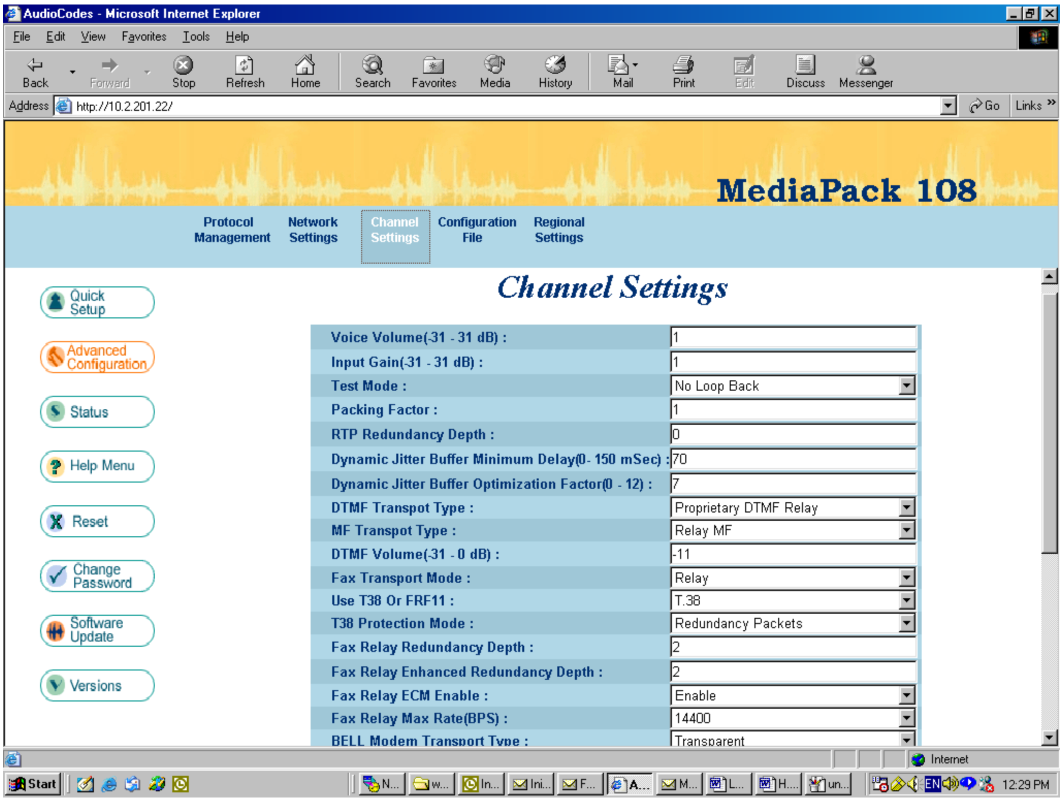
7.2.5.5 Caller ID Display Table

The table contains Caller ID display information per **MP-1xx/FXS** gateway port. This information is sent in INVITE message to remote party, for Phone ➔ IP calls. Remote party can use this display information for caller identification. The caller ID string can contain up to 18 characters.

7.2.5.6 Channel Settings Menu

Selecting the “Channel Settings” tab, enables to view and to modify the gateway channel parameters, such as Input & Output voice gain, Jitter buffer characteristics, Modem, Fax and DTMF transport modes etc. These parameters apply to all gateway ports.

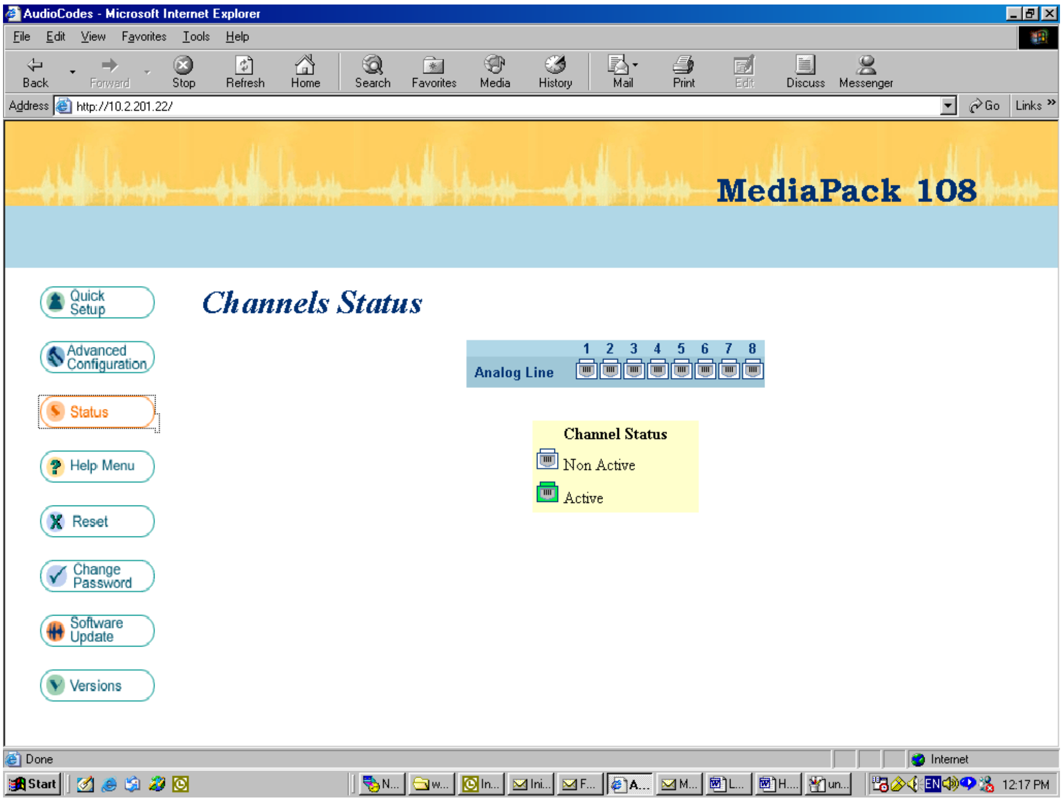
Figure 7-11: Channel Settings



7.2.5.7 Channel Status Menu

Selecting the "**Status"** button on left side of the page provides real time monitoring of the current channel status, as shown in the example in Figure 7-12.

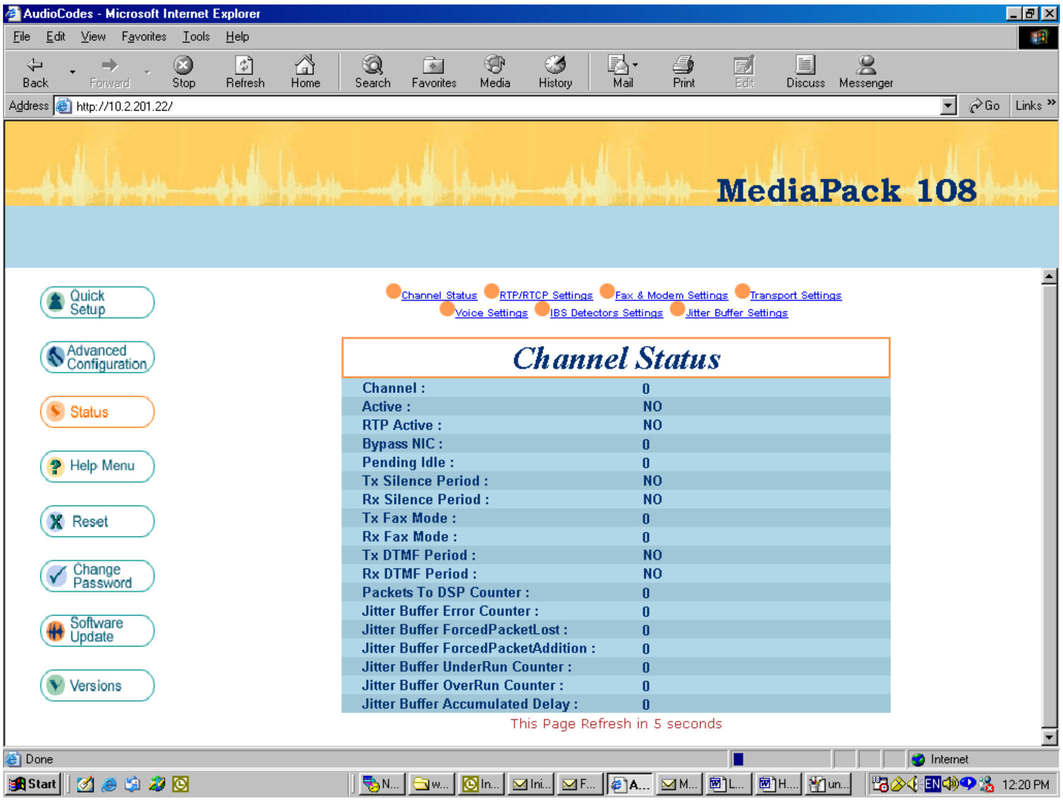
Figure 7-12: Web-Server – Channel Status (1)



Active Channels are colored green.

Selecting a channel shows, for example, the following information of the selected channel (refer to Figure 7-13 on page 128).

Figure 7-13: Web-Server - Channel Status (2)



Selecting each of the sub menus shows the active channel configuration of each section and the values of its relevant parameters, as shown in the above example.

8 Diagnostics

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

This section provides details of the features and functionality of the MP-1xx diagnostics and troubleshooting.

8.1 Diagnostics Overview

AudioCodes provides several diagnostic tools to enable the User to identify an error condition and to provide a solution or work around when working with **MP-1xx** gateway.

- LED Indication of channel activity status, data, control and LAN status.
- **MP-1xx** Self-Testing on hardware initialization.
- RS-232 terminal Notification Messages
- SysLog Event Notification Messages.
- Solutions to Common Problems.

They are described in the following pages.

8.2 MP-1xx Gateway Alarms & SNMP Traps

8.2.1 LED Visual Indicator Status and Alarms

Table 8-1: Indicator LEDs on the MP-1xx Front Panel

Label	Type	Color	State	Meaning
LAN	Ethernet Link Status	Green	ON	Valid Connection to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Data	Packet Status	Green	Blinking	Transmitting RTP Packets
		Red	Blinking	Receiving RTP Packets
		Blank	-	No traffic
Control	Control Link	Green	Blinking	Sending and receiving SIP messages.
		Red		Not supported in current release
Ready	Device Status	Green	ON	Device Powered, Self test OK
		Orange	Blinking	Software Loading/Initialization
		Red	ON	Malfunction

Table 8-2: MP-1xx Channel LEDs

MP-1xx with 1 to 8 Channels				
Label	**Type**	**Color**	**State**	**Meaning**
Activity	**FXS** Tel Port	Green	ON	Off-Hook/Ringing for Phone Port
	FXO Line Port	Green	ON	Line-Seize/Ringing State for Line Port

8.3 MP-1xx Self-Testing

The **MP-1xx** features two self-testing modes: rapid and detailed.

Rapid self-test mode is invoked each time the Media Gateway completes the initialization process. This is a short test phase in which the only error detected and reported is failure in initializing hardware components. All Status and Error reports in this self-test phase are reported through Network Interface ports, as well as indicated by the LED Status Indicators.

Detailed self-test mode is invoked when initialization of the Media Gateway is completed and if the configuration parameter EnableDiagnostics is set to 1 (this parameter can be configured through the *ini* file mechanism). In this mode, the Media Gateway tests all the hardware components (memory, DSP, etc.), outputs the status of the test results, and ends the test. To continue operational running, reset the Media Gateway again but this time configure the EnableDiagnostics parameter to 0.

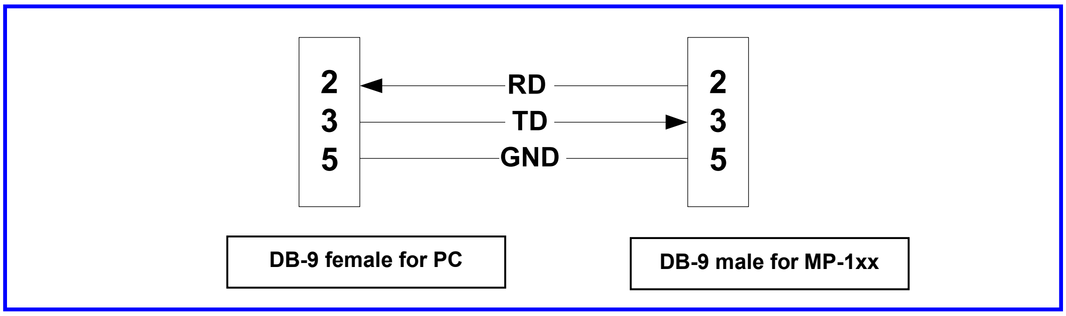
8.4 RS-232 Terminal

The **MP-1xx** status and error messages can be viewed via a terminal connected to the RS-232 management port.

➢ To connect MP-1xx to a HyperTerminal, take this step:

With a standard RS-232 straight cable (not a cross-over cable) with DB-9 connectors, connect the **MP-1xx** RS-232 port (it is marked "RS232") to either COM1 or COM2 RS-232 communication port on the PC. The connector pinout and gender are shown below in Figure 8-1.

Figure 8-1: RS-232 Cable Wiring



➢ To configure the HyperTerminal, take these 5 steps:

1. On a PC running a Windows™ operating system, open Start>Programs>Accessories>Communications>HyperTerminal; the Connection Description dialog opens.
2. Enter a name for the new connection in the Name field and click OK; the Connect To dialog opens.
3. In the Connect To dialog, enter COM1 or COM2, depending on the physical connection you performed when connecting the **MP-1xx** to the PC with the RS-232 cable; the COM1/2 Properties dialog opens.

4. In the COM1/2 Properties dialog, enter the following settings for the serial communication port:

Baud Rate:	115,200 bps
Data bits:	8
Parity:	None
Stop bits:	1
Flow control:	Hardware

5. Click OK; the HyperTerminal main screen opens.

After applying power or reset the following information is printed on the terminal screen shown in Figure 8-2.

Figure 8-2: Status and Error Messages

```
MAC address = 00-90-8F-01-00-9E
Local IP address = 10.1.37.6
Subnet mask = 255.255.0.0
Default gateway IP address = 10.1.1.5
TFTP server IP address = 10.1.1.167
Boot file name = ram35136.cmp
INI file name = mp108.ini
Call agent IP address = 10.1.1.18
Log server IP address = 0.0.0.0
Full/Half Duplex state = HALF DUPLEX
Flash Software Burning state = OFF
Serial Debug Mode = OFF
Lan Debug Mode = OFF

BootLoad Version 1.75
Starting TFTP download ... Done.
MP108 Version 3.80.00
```

8.5 SysLog Support

8.5.1 Overview

SysLog protocol is an event notification protocol that allows a machine to send event notification messages across IP networks to event message collectors - also known as SysLog servers. The SysLog protocol is defined in RFC 3164 IETF standard.

Since each process, application and operating system was written somewhat independently, there is little uniformity to SysLog messages. For this reason, no assumption is made on the contents of the messages other than the minimum requirements of its priority.

SysLog uses UDP as its underlying transport layer mechanism. The UDP port that has been assigned to SysLog is 514.

The SysLog message is transmitted as an ASCII message. The message starts with a leading "<" ('less-than' character), followed by a number, which is followed by a ">" ('greater-than' character). This is optionally followed by a single ASCII space.

The number described above is known as the Priority and represents both the Facility and Severity as described below. The Priority number consists of one, two, or three decimal integers.

Example:

<37> Oct 11 16:00:15 mymachine su: 'su root' failed for lonvick on /dev/pts/8

8.5.2 SysLog Operation

8.5.2.1 Sending the SysLog Messages

AudioCodes' SysLog client, embedded in the firmware of the **MP-1xx**, sends error reports/events generated by the **MP-1xx** unit application to a SysLog server, using IP/UDP protocol. AudioCodes does NOT provide a SysLog server as several are provided as shareware that can be downloaded from the Internet.

Examples of SysLog Servers downloadable from the Internet:

1. Kiwi Enterprises: http://www.kiwi-enterprises.com/software_downloads.htm
2. The US CMS Server: http://uscms.fnal.gov/hanlon/uscms_server/
3. TriAction Software: http://www.triaction.nl/Products/SyslogDaemon.asp
4. Netal SL4NT 2.1 Syslog Daemon: *http://www.netal.com*

A typical SysLog server application enables filtering of the messages according to priority, IP sender address, time, date, …

8.5.2.2 Setting the SysLog Server IP Address

A SysLogServerIP Address parameter is supplied via Web browser or from an *ini* file in order to determine the address of the SysLog server.

8.5.2.3 Controlling the Activation of the SysLog Client

Activation of the SysLog client is controlled by an EnableSyslog *ini* file parameter. Setting it to 1 enables the SysLog protocol log.

8.5.2.4 The *ini* File Example for SysLog

Figure 8-3: The *ini* File Example for SysLog

```
[Syslog]
SyslogServerIP=10.2.0.136
EnableSyslog =1
```

8.6 Solutions to Possible Problems

8.6.1 General

If there is a problem, check the following resources:

- Web Browser status and channel parameter pages.
- Log messages of **MP-1xx** in HyperTerminal screen.
- BootP & TFTP log messages (for startup problems).
- Log messages in SysLog server.

8.6.2 Possible Common Problems

Possible common problems are described in Table 8-3.

Table 8-3: Possible Common Problems (continues on pages 137 to 138)

Problem	Possible Cause	What to do
No communication	Software does not function in **MP-1xx**	Try to "ping" to **MP-1xx**. If ping fails, check for network problems/definitions and try to reset the **MP-1xx**
	Network problem	Check cables.
	Network definitions	Check if default gateway can reach IP of box.
		Check if box got the correct IP (it can be seen in the HyperTerminal screen).
		Check the validity of IP address, subnet and default gateway. If default gateway is not used, enter 0.0.0.0
	BootP didn't reply to box	Check if BootP server replied to **MP-1xx** at restart; it is seen in the BootP server log.
		Try to restart BootP server.
		Check the MAC address of the box in BootP server.
***ini* file was not loaded**	TFTP server down	Check if TFTP server working.
	TFTP server didn't get the request	Check this in its log.
	MP-1xx didn't request the file from your TFTP	Look in HyperTerminal for the TFTP server IP address that the **MP-1xx** is trying to use.
	TFTP server bug	Try to restart TFTP server.

Table 8-3: Possible Common Problems (continues on pages 137 to 138)

Problem	Possible Cause	What to do
	BootP sent to MP wrong TFTP server address	Check in HyperTerminal screen the address of used TFTP.
	Ini file does not exists in default directory of TFTP	Check default directory of TFTP server and check that ini file exists there.
	Wrong ini file name	Verify in windows explorer that file extensions are displayed and the ini file isn't by mistake "XXX.ini.ini". Verify that extension *ini* is in lowercase letters.
	TFTP have too short timeout	Verify that: Timeout = 2 sec, # of retransmission = 10
Wrong *ini* file loaded	.ini file is not in the correct position	Old ini file was probably loaded. Check which ini file was loaded. This can be done using HyperTerminal screen. The Gateway displays contents of ini file before it began.
	.ini file corrupted	check ini file syntax
BootP reply from wrong BootP server	Other BootP servers contain MAC address of gateway	Check that only your BootP server contains **MP-1xx** MAC address.

9 Specifications

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

This section describes the Specifications of the MP-1xx Gateway

9.1 MP-1xx Specifications

Table 9-1: MP-1xx Functional Specifications (continues on pages 141 to 143)

MP-1xx FXS Functionality	
FXS Capabilities	Short or Long Haul up to 3,000 m (10,000 ft) using 24 AWG line cord.
	Includes lightning and high voltage protection for outdoor operation.
	Caller ID generation: Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation.
	Programmable Line Characteristics: Battery feed, line current, hook thresholds, AC impedance matching, hybrid balance, Tx & Rx frequency response, Tx & Rx Gains.
	Programmable ringing signal. Up to three cadences and frequency 10 to 200 Hz.
	Drive up to 4 phones per port (total 32 phones) simultaneously in Off-hook and Ring states. **MP-124** REN = 2 **MP-10x** REN = 5
	Over-temperature protection for abnormal situations as shorted lines.
	Loop-backs for testing and maintenance.
MP-10x FXO Functionality	
FXO Capabilities (does not apply to **MP-102**)	Short or Long Haul up to 7,000 m (24,000 ft) using 24 AWG line cord.
	Includes lightning and high voltage protection for outdoor operation.
	Programmable Line Characteristics: AC impedance matching, hybrid balance, Tx & Rx frequency response, Tx & Rx Gains, ring detection threshold, DC characteristics.
	Caller ID detection: Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation.
Voice & Tone Characteristics	
Voice Compression	G.711 PCM at 64 kbps µ-law/A-law G.723.1 MP-MLQ at 6.3 kbps G.726/G.727 at 16 to 40 kbps ADPCM and E-ADPCM G.729A CS-ACELP at 8 kbps NetCoder at 6.4 to 8.8 kbps, 800-bit increments (proprietary coder)
Silence Suppression	G.723.1 Annex A G.729 Annex B PCM and ADPCM - Proprietary Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) NetCoder
Echo Canceler	G.168, 25 msec

Table 9-1: MP-1xx Functional Specifications (continues on pages 141 to 143)

Gain Control	Programmable
DTMF Transport	Mute, transfer in RTP payload or relay in compliance with RFC 2833
DTMF Detection and Generation	Dynamic range 0 to -25 dBm, compliant with TIA 464B and Bellcore TR-NWT-000506.
Call Progress Tone Detection and Generation	15 tones: single tone or dual tones, programmable frequency & amplitude; 16 frequencies in the range 300 to 2000 Hz, 1 or 2 cadences per tone, up to 2 sets of ON/OFF periods.
Output Gain Control	-31 dB to +31 dB in steps of 1 dB
Input Gain Control	-31 dB to +31 dB in steps of 1 dB
Fax/Modem Relay	
Fax Relay	Group 3 fax relay up to 14.4 kbps with auto fallback T.38 compliant, real time fax relay Tolerant network delay (up to 9 sec round trip)
Modem Relay	Up to 14.4 kbps V.32bis (optional)
Modem Transparency	Auto switch to PCM or ADPCM on V.34 or V.90 modem detection
Protocols	
Control Protocols	SIP (rfc3261), H.323 ITU,, MEGACO (H.248) and MGCP
Communication Protocols	RTP/RTCP packetization. IP stack (UDP, TCP, RTP). Remote Software download (TFTP & BootP support).
Line Signaling Protocols	Loop start, FXS and FXO
Interfaces	
FXS Telephony Interface	2, 4, 8 or 24 Analog FXS phone or fax ports, loop start
FXO Telephony Interface	4 or 8 Analog FXO PSTN/PBX loop start ports
Network Interface	RJ-45 shielded connector, 10/100 Base-T.
RS-232 Interface	RS-232 Terminal Interface for maintenance, diagnostic reports and code tracing. DB-9 connector on rear panel
Life Line (MP-10x/FXS)	Life Line, connected to the unused pins on port #4 (port #2 for **MP-102**/FXS), with a relay to an analog line, even if the **MP-10x/FXS** is powered off (refer to Section 2.1.5 for details). Does NOT function with **MP-124** and **MP-10x**/FXO gateways.
Connectors & Switches	
Rear Panel	
24 Analog Lines (MP-124)	50-pin Telco shielded connector
8 Analog Lines (MP-108)	8 RJ-11 connectors

Table 9-1: MP-1xx Functional Specifications (continues on pages 141 to 143)

4 Analog Lines (MP-104)	4 RJ-11 connectors
2 Analog Lines (MP-102)	2 RJ-11 connectors
Ethernet	10/100 Base-T, RJ-45 shielded connector
RS-232	Console port - DB-9
Front Panel	
Reset	Resets the **MP-1xx**
Physical	
MP-10x Enclosure Dimensions	Width: 221 mm 8.7 in Height: 44.5 mm 1.75 in Depth: 240 mm 9.5 in Weight: 1.24 kg 2.5 lb
MP-124 Enclosure Dimensions	1U, 19-inch Rack Width: 445 mm 17.5 in Height: 44.5 mm 1.75 in Depth: 269 mm 10.6 in Weight: 2.24 kg 4.9 lb
Environmental	Operational: 0° to 45° C 32° to 113° F Storage: -10° to 70° C 14° to 158° F Humidity: 10 to 90% non-condensing
Installation	Desk-top, shelf, or 19-inch rack mount with side brackets.
Electrical	Universal 90-260 VAC, 1A, 47-63 Hz
Type Approvals	
Telecommunication	FCC part 68 & CE CTR21
Safety and EMC	UL 1950, FCC part 15 Class B CE Mark (EN 60950, EN 55022, EN 55024)
Management	
Configuration	Gateway configuration using Web browser, *ini* files or local RS-232 console
Management and Maintenance	SNMP
	SysLog, per RFC 3164
	Local RS-232 terminal
	Web Management

Reader's Notes

Appendices A to G

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO gateways.

This section contains the following Appendices:

> **Note**: The following 4 Appendices are all located in the AudioCodes "Software Utilities Manual", Catalog Number: LTRT-00602.

Appendix A - AudioCodes BootP/TFTP Configuration Utility

The AudioCodes **BootP/TFTP Configuration Utility** enables easy configuration and provisioning of AudioCodes products. It contains BootP and TFTP servers with specific adaptations to AudioCodes' requirements. For details of the configuration routine and descriptive example screens refer to Appendix A, "AudioCodes BootP/TFTP Configuration Utility" in the AudioCodes "Software Utilities Manual", Catalog Number: LTRT-00602.

Appendix B - Windows™ NT DHCP Server Configuration

> **Note**: For correct operation with BootP clients, install Windows™ NT4 service pack 4 after enabling the DHCP server service on the NT server.

For details of the **DHCP Server Configuration** routine and descriptive example screens refer to Appendix B in the AudioCodes "Software Utilities Manual", Catalog Number: LTRT-00602.

Appendix C - Weird Solutions BootP Server Configuration

The **BootP Server** 95 can be downloaded from www.weird-solutions.com; it can be installed on Windows™ 95/98 or Windows™ NT. For details of the configuration routine and descriptive example screens refer to Appendix C in the AudioCodes "Software Utilities Manual", Catalog Number: LTRT-00602.

> **Note**: The BootP and TFTP servers must be located on the same host.

Appendix D - TFTP Server Configuration and Installation

The **TFTP Server** ("TFTP Turbo 98") can be downloaded from www.weird-solutions.com; it can be installed on Windows™ 95/98 or Windows™ NT. For details of the configuration routine and example screens refer to Appendix D in the AudioCodes "Software Utilities Manual", Catalog Number: LTRT-00602.

Appendix E - Default RTP/RTCP/T.38 Ports

The following table shows the **MP-1xx** Default RTP/RTCP/T.38 Port Allocation for SIP protocol.

Table E-1: MP-1xx Default RTP/RTCP/T.38 Port Allocation

Channel Number	RTP Port	RTCP Port	T.38 Port
1	6000	6001	6002
2	6010	6011	6012
3	6020	6021	6022
4	6030	6031	6032
5	6040	6041	6042
6	6050	6051	6052
7	6060	6061	6062
8	6070	6071	6072
n	**6000+10(n-1)**	**6001+10(n-1)**	**6002+10(n-1)**

Appendix F - RTP/RTCP Payload Types

RTP Payload Types are defined in RFC 1889/1890. AudioCodes has added new payload types to enable advanced use of other coder types. These types are reportedly not used by other applications.

Note: Refer to the AudioCodes "MP-Series Release Notes", Catalog Number: LTRT-00316, for the supported coders.

F.1 Packet Types Defined in RFC 1890

Table F-2: Packet Types Defined in RFC 1890

Payload Type	Description	Basic Packet Rate [msec]
0	G.711 µ-Law	20
2	G.726-32	20
4	G.723 (6.3/5.3 kbps)	30
8	G.711 A-Law	20
18	G.729	20
200	RTCP Sender Report	Randomly, approximately every 5 sec (when packets are sent by channel)
201	RTCP Receiver Report	Randomly, approximately every 5 sec (when channel is only receiving)
202	RTCP SDES packet	
203	RTCP BYE packet	
204	RTCP APP packet	

F.2 AudioCodes Defined Payload Types

Table F-3: AudioCodes Defined Payload Types

Payload Type	Description	Basic Packet Rate [msec]
35	G.726 16 kbps	20
36	G.726 24 kbps	20
38	G.726 40 kbps	20
39	G.727 16 kbps	20
40	G.727 24-16 kbps	20
41	G.727 24 kbps	20
42	G.727 32-16 kbps	20
43	G.727 32-24 kbps	20
44	G.727-32 kbps	20
45	G.727 40-16 kbps	20
46	G.727 40-24 kbps	20
47	G.727 40-32 kbps	20
51	NetCoder 6.4 kbps	20
52	NetCoder 7.2 kbps	20
53	NetCoder 8.0 kbps	20
54	NetCoder 8.8 kbps	20
55	NetCoder 9.6 kbps	20
56	Transparent PCM	20
100	DTMF relay	20
101	Fax Relay	Different packet rates
102	Fax Bypass	20
103	Modem Bypass	20
104	RFC2198 (Redundancy)	Same as channel's voice coder.

Appendix G- DTMF, Fax and Modem Transport Modes

Users can configure parameters to control the transport method of:

- DTMF/MF digits.
- Fax
- Modem

G.1 DTMF/MF Relay Settings

Users can control the way DTMF/MF digits are transported to the remote endpoint, using the DTMFTransport/MFTransport configuration parameters. The following three modes are supported:

DTMF/MFTransportType= 0 (MuteDTMF/MF). In this mode, DTMF/MF digits are erased from the audio stream and are **not** relayed to the remote side. Instead silence is sent in the RTP stream.

DTMF/MFTransportType= 1 (RelayDTMF/MF). In this mode, DTMF/MF digits are erased from the audio stream and **are** relayed to the remote side using a proprietary RTP syntax.

DTMF/MFTransportType= 2 (TransparentDTMF/MF). In this mode, DTMF/MF digits are **left** in the audio stream and the DTMF/MF relay is disabled.

DTMF/MFTransportType= 3 (RelayDTMF/MF). In this mode, DTMF/MF digits are erased from the audio stream and **are** relayed to the remote side according to RFC 2833 standard.

G.2 Fax/Modem Settings

Users can choose to use for fax, and for each modem type (V.21/V.22/V.23/Bell/V.32/V.34), one of the following transport methods:

Fax relay mode (demodulation / remodulation, not applicable for Modem),
Bypass (using a high bit rate coder to pass the signal), or
Transparent (passing the signal in the current voice coder).

When any of the relay modes are enabled, distinction between fax and modem is not immediately possible at the beginning of a session. The channel is therefore in “Answer Tone” mode until a decision is made The packets sent to the network at this stage are fax relay packets (The packets can be either T.38-complaint, or FRF.11-based proprietary syntax, selected by setting the channel’s configuration parameter UseT38orFRF11.)

G.2.1 Configuring Fax Relay Mode

When FaxTransportType= 1 (relay mode), then when fax is detected the channel automatically switches from the current voice coder to answer tone mode, and then to fax relay mode. The UseT38orFRF11 configuration parameter defines either T.38-compliant network packets or proprietary FRF.11-based packets (the last mode should be used mostly for backward-compatibility with older software versions).

When fax transmission ends, the reverse is carried out, and fax relay switches to voice. This mode switch occurs automatically, both at the local and remote endpoints.

Users can limit the fax rate using the FaxRelayMaxRate parameter and can enable/disable ECM fax mode using the FaxRelayECMEnable parameter.

When using T.38 mode, the User can define a redundancy feature to improve Fax transmission over congested IP network. This feature is activated by "FaxRelayRedundancyDepth" and "EnhancedFaxRelayRedundancyDepth" parameters. Although this is a proprietary redundancy scheme, it should not create problems when working with other T.38 decoders.

When using FRF.11 mode, only "FaxRelayRedundancyDepth" parameter can be used.

Note: T.38 mode currently supports only the T.38 UDP syntax.

G.2.2 Configuring Fax/Modem ByPass Mode

When VxxTransportType=2 (FaxModemBypass, Vxx can be either V32/V22/V21/Bell/V34/Fax), then when fax/modem is detected, the channel automatically switches from the current voice coder to a high bit-rate coder, as defined by the User, with the FaxModemBypassCoderType configuration parameter.

If fax relay is enabled for one of the modes, then the Answer Tone mode packets are relayed as fax relay packets.

During the bypass period, the coder is used the packing factor (by which a number of basic coder frames are combined together in the outgoing WAN packet) set by the User in the FaxModemBypassM configuration parameter. The network packets to be generated and received during the bypass period are regular voice RTP packets (per the selected bypass coder) but with a different RTP Payload type.

When fax/modem transmission ends, the reverse is carried out, and bypass coder is switched to regular voice coder.

> **Note**: When Fax relay is enabled, V21TransportType must be set to disable (Transparent) mode.

G.2.3 Supporting V.34 Faxes

V.34 fax machine support is available only in bypass mode (fax relay is not supported) when the channel is configured in one of the configurations described below:

FaxTransportMode = 2 (Bypass)

V34ModemTransportType = 2 (Modem bypass)

In this configuration both T.30 and V.34 faxes will work in Bypass mode

Or

FaxTransportMode = 1 (Relay)

V34ModemTransportType = 2 (Modem bypass)

In this configuration T.30 faxes use relay mode (T.38) while V.34 Fax use Bypass mode.

In order to use V.34 fax in Relay mode (T.38), you must configure:

FaxTransportMode = 1 (Relay)

V34ModemTransportType = 0 (Transparent)

V32ModemTransportType = 0

V23ModemTransportType = 0

V22ModemTransportType = 0

V21ModemTransportType = 0

This configuration forces the V.34 fax machine to work in T.30 mode.

Reader's Notes

US: AudioCodes Inc, 2890 Zanker Road, Suite # 200, San Jose, CA 95134.
Tel: 408-577-0488 - Fax: 408-577-0492

International: AudioCodes Ltd, 4 HaHoresh Street, PO Box 14, Yehud 56470, Israel.
Tel: +972-3-539 4000 - Fax: +972-3-539 4040

China: AudioCodes Ltd, Room 209 Tower A, Grand Pacific Building,
8A Guanghua Road, Chaoyang District, Beijing 100026, P.R.China
Tel: 86-10-6581-6622 - Fax: 86-10-6581-5599

Technical Support: support@audiocodes.com

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