AudioCodes™ CPE & Access Analog Gateways

SIP

MediaPack™ MP-124 & MP-11x

User's Manual Version 5.0

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Notices

Notice

This document describes the AudioCodes MediaPack series Voice over IP (VoIP) gateways.

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. Updates to this document and other documents can be viewed by registered Technical Support customers at www.audiocodes.com under Support / Product Documentation.

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Pursuant to the WEEE EU Directive, electronic and electrical waste must not be disposed of with unsorted waste. Please contact your local recycling authority for disposal of this product.

Customer Support

Customer technical support and service are provided by AudioCodes' Distributors, Partners, and Resellers from whom the product was purchased. For Customer support for products purchased directly from AudioCodes, contact support@audiocodes.com.

Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used. Only industry-standard terms are used throughout this manual. Hexadecimal notation is indicated by 0x preceding the number.

Related Documentation

Document #	Manual Name
LTRT-656xx (e.g., LTRT-65601)	MP-11x & MP-124 SIP Release Notes
LTRT-598xx	MP-11x & MP-124 MGCP-H.323-SIP Fast Track Guide
LTRT-665xx	CPE Configuration Guide for IP Voice Mail

Note 1: MediaPack refers to the MP-124, MP-118, MP-114, and MP-112 VoIP gateways.

Note 2: MP-11x refers to the MP-118, MP-114, and MP-112 VoIP gateways.

Note: Where 'network' appears in this manual, it means Local Area Network (LAN), Wide Area Network (WAN), etc. accessed via the gateway's Ethernet interface.

Note: **FXO** (Foreign Exchange Office) is the interface replacing the analog telephone and connects to a Public Switched Telephone Network (PSTN) line from the Central Office (CO) or to a Private Branch Exchange (PBX). The FXO is designed to **receive** line voltage and ringing current, supplied from the CO or the PBX (just like an analog telephone). An FXO VoIP gateway interfaces between the CO/PBX line and the Internet.

FXS (Foreign Exchange Station) is the interface replacing the Exchange (i.e., the CO or the PBX) and connects to analog telephones, dial-up modems, and fax machines. The FXS is designed to **supply** line voltage and ringing current to these telephone devices. An FXS VoIP gateway interfaces between the analog telephone devices and the Internet.

Warning: Ensure that you connect FXS ports to analog telephone or to PBX-trunk lines only and FXO ports to CO / PBX lines only.

Warning: The MediaPack is supplied as a sealed unit and must only be serviced by qualified service personnel.

Warning: Disconnect the MediaPack from the mains and from the Telephone Network Voltage (TNV) before servicing.

1 Overview

This document provides you with the information on installation, configuration and operation of the VoIP analog gateways listed in the table below:

Table 1-1: Supported Product Configurations

Product Name	FXS	FXO	Combined FXS / FXO	Number of Channels
MP-124	✓	×	×	24
MP-118	✓	✓	4 + 4	8
MP-114	✓	✓	2 + 2	4
MP-112*	✓	×	×	2

* The MP-112 differs from the MP-114 and MP-118 in that its configuration excludes the RS-232 connector, Lifeline option, and outdoor protection.

As these units have similar functionality (with the exception of their number of channels and some minor features), they are collectively referred to throughout this manual as the *MediaPack*.

1.1 Gateway Description

The MediaPack series analog VoIP gateways are cost-effective, cutting edge technology products. These stand-alone analog VoIP gateways provide superior voice technology for connecting legacy telephones, fax machines and PBX systems with IP-based telephony networks, as well as for integration with new IP-based PBX architecture. These products are designed and tested to be fully interoperable with leading softswitches and SIP servers.

The MediaPack gateways incorporate up to 24 analog ports for connection, directly to an enterprise PBX (FXO), or / and to phones, fax machines and modems (FXS), supporting up to 24 simultaneous VoIP calls.

Additionally, the MediaPack units are equipped with a 10/100 Base-TX Ethernet port for connection to the network.

The MediaPack gateways are best suited for small to medium size enterprises, branch offices or for residential media gateway solutions.

The MediaPack 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 network ensuring that voice traffic uses minimum bandwidth.

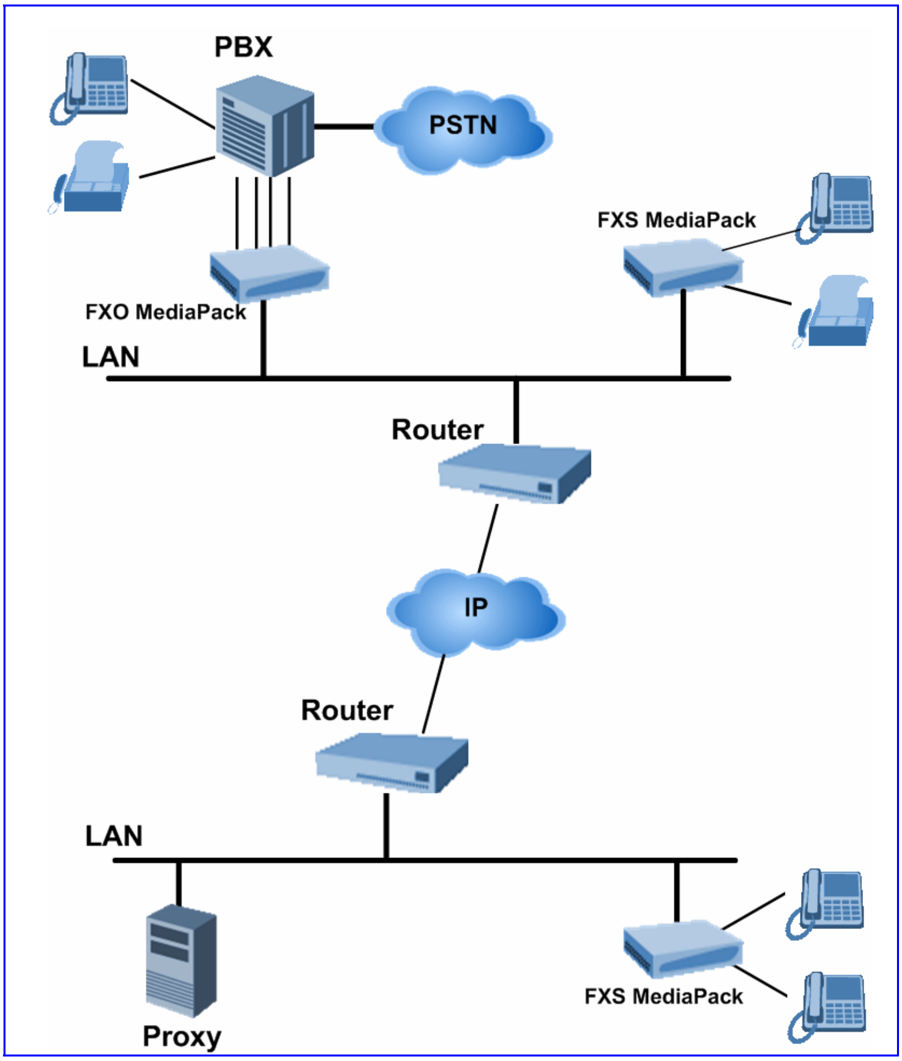
The MediaPack gateways are very compact devices that can be installed as a desk-top unit, on the wall or in a 19-inch rack.

The MediaPack gateways support SIP (Session Initiation Protocol) protocol, enabling the deployment of ‘voice over IP’ 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 capability to transmit the voice and telephony signals over a packet network.

The layout diagram (Figure 1-1), illustrates a typical MediaPack VoIP application.

Figure 1-1: Typical MediaPack VoIP Application



1.2 SIP Overview

SIP (Session Initialization Protocol) is an application-layer control (signaling) protocol used on the MediaPack for creating, modifying, and terminating sessions with one or more participants. These sessions can include Internet telephone calls, media announcements and conferences.

SIP invitations are used to create sessions and carry session descriptions that enable participants to agree on a set of compatible media types. SIP uses elements called Proxy servers to help route requests to the user's current location, authenticate and authorize users for services, implement provider call-routing policies and provide features to users.

SIP also provides a registration function that enables users to upload their current locations for use by Proxy servers. SIP, on the MediaPack, complies with the IETF (Internet Engineering Task Force) RFC 3261 (refer to http://www.ietf.org).

1.3 MediaPack Features

This section provides a high-level overview of some of the many MediaPack supported features.

1.3.1 General Features

- Superior, high quality Voice, Data and fax over IP networks.
- Toll quality voice compression.
- Enhanced capabilities including MWI, long haul, metering, CID and out door protection.
- Proven integration with leading PBXs, IP-PBXs, Softswitches and SIP servers.
- Spans a range of 2 to 24 FXS/FXO analog ports.
- Selectable G.711 or multiple Low Bit Rate (LBR) coders per channel.
- T.38 fax with superior performance (handling a round-trip delay of up to nine seconds).
- Echo Canceler, Jitter Buffer, Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) support.
- Comprehensive support for supplementary services.
- Web Management for easy configuration and installation.
- EMS for comprehensive management operations (FCAPS).
- Simple Network Management Protocol (SNMP) and Syslog support.
- SMDI support for Voice Mail applications.
- ThroughPacket™ proprietary feature that aggregates payloads from several channels into a single IP packet to reduce bandwidth overhead.
- Supports load balancing with proxy.
- T.38 fax fallback to PCM (or NSE).
- Can be integrated into a Multiple IPs and a VLAN-aware environment.
- Capable of automatically updating its firmware version and configuration.
- Secured Web access (HTTPS) and Telnet access using SSL / TLS.
- IPSec and IKE protocols are used in conjunction to provide security for control (e.g., SIP) and management (e.g., SNMP and Web) protocols.
- Secured RTP (SRTP) according to RFC 3711, used to encrypt RTP and RTCP transport.

1.3.2 MP-11x Hardware Features

- Combined FXS/FXO gateways (4 FXS and 4 FXO ports on the MP-118; 2 FXS and 2 FXO ports on the MP-114).
- MP-11x compact, rugged enclosure only one-half of a 19-inch rack unit, 1 U high.
- Lifeline - provides a wired phone connection to PSTN line when there is no power, or the network fails (combined FXS/FXO gateways provide a Lifeline connection available on all FXS ports).
- LEDs on the front panel that provide information on the operating status of the media gateway and the network interface.
- Reset button on the rear panel that restarts the MP-11x gateway, and is also used to restore the MP-11x parameters to their factory default values.

1.3.3 MP-124 Hardware Features

- MP-124 19-inch, 1U rugged enclosure provides up to 24 analog FXS ports, using a single 50 pin Telco connector.
- LEDs on the front and rear panels that provide information on the operating status of the media gateway and the network interface.
- Reset button on the front panel that restarts the MP-124 gateway, and is also used to restore the MP-124 parameters to their factory default values.

1.3.4 SIP Features

The MediaPack SIP gateway complies with the IETF RFC 3261 standard.

- Reliable User Datagram Protocol (UDP) transport, with retransmissions.
- Transmission Control Protocol (TCP) Transport layer.
- SIPS using TLS.
- T.38 real time Fax (using SIP).

  Note: If the remote side includes the fax maximum rate parameter in the SDP body of the INVITE message, the gateway returns the same rate in the response SDP.
- Works with Proxy or without Proxy, using an internal routing table.
- Fallback to internal routing table if Proxy is not responding.
- Supports up to four Proxy servers. If the primary Proxy fails, the gateway automatically switches to a redundant Proxy.
- Supports domain name resolving using DNS NAPTR and SRV records for Proxy, Registrar and domain names that appear in the Contact and Record-Route headers.
- Supports Load Balancing over Proxy servers using Round Robin or Random Weights.
- Proxy or Registrar Registration, such as:

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

  - The "servername" string is defined according to the following rules:
  - The "servername" is equal to "RegistrarName" if configured. The "RegistrarName" can be any string.
  - Otherwise, the "servername" is equal to "RegistrarIP" (either FQDN or numerical IP address), if configured.
  - Otherwise the "servername" is equal to "ProxyName" if configured. The "ProxyName" can be any string.
  - Otherwise the "servername" is equal to "ProxyIP" (either FQDN or numerical IP address).

  The parameter GWRegistrationName can be any string. If the parameter is not defined, the parameter UserName is used instead.

  The 'sipgatewayname' parameter (defined in the *ini* file or set from the Web browser), can be any string. Some Proxy servers require that the 'sipgatewayname' (in REGISTER messages) is set equal to the Registrar / Proxy IP address or to the Registrar / Proxy domain name.

The REGISTER message is sent to the Registrar's IP address (if configured) or to the Proxy's IP address. The message is sent per gateway or per gateway endpoint according to the AuthenticationMode parameter. Usually the FXS gateways are registered per gateway port, while FXO gateways send a single registration message, where Username is used instead of phone number in From/To headers. The registration request is resent according to the parameter 'RegistrartionTimeDivider'. For example, if 'RegistrationTimeDivider = 70' (%) and Registration Expires time = 3600, the gateway resends its registration request after 3600 x 70% = 2520 sec. The default value of 'RegistrartionTimeDivider' is 50%.

- Proxy and Registrar Authentication (handling 401 and 407 responses) using Basic or Digest methods. Accepted challenges are kept for future requests to reduce the network traffic.
- Single gateway Registration or multiple Registration of all gateway endpoints.
- Supported methods: INVITE, CANCEL, BYE, ACK, REGISTER, OPTIONS, INFO, REFER, UPDATE, NOTIFY, PRACK, SUBSCRIBE and PUBLISH.
- Modifying connection parameters for an already established call (re-INVITE).
- Working with Redirect server and handling 3xx responses.
- Early media (supporting 183 Session Progress).
- PRACK reliable provisional responses (RFC 3262).
- Call Hold and Transfer Supplementary services using REFER, Refer-To, Referred-By, Replaces and NOTIFY messages.
- Supports RFC 3711, Secured RTP and Key Exchange according to <draft-ietf-mmusic-sdescriptions-12>.
- Supports RFC 3489, Simple Traversal of UDP Through NATs (STUN).
- Supports RFC 3327, Adding 'Path' to Supported header.
- Supports RFC 3581, Symmetric Response Routing.
- Supports RFC 3605, RTCP Attribute in SDP.
- Supports RFC 3326, Reason header.
- Supports RFC 4028, Session Timers in SIP.
- Supports network asserted identity and privacy (RFC 3325 and RFC 3323).
- Supports RFC 3911, The SIP Join Header.
- Supports RFC 3903, SIP Extension for Event State Publication.
- Supports RFC 3953, The Early Disposition Type for SIP.
- Supports RFC 3966, The tel URI for Telephone Numbers.
- Supports RFC 4244, An Extension to SIP for Request History Information.
- Supports Tel URI (Uniform Resource Identifier) according to RFC 2806 bis.
- Supports ITU V.152 - Procedures for supporting Voice-Band Data over IP Networks.
- Remote party ID <draft-ietf-sip-privacy-04.txt>.
- Supports obtaining Proxy Domain Name(s) from DHCP (Dynamic Host Control Protocol) according to RFC 3361.
- Supports handling forking proxy multiple responses.
- RFC 2833 Relay for DTMF Digits, including payload type negotiation.
- DTMF out-of-band transfer using:
    - INFO method <draft-choudhuri-sip-info-digit-00.txt>
    - INFO method, compatible with Cisco gateways
    - NOTIFY method <draft-mahy-sipping-signaled-digits-01.txt>

- SIP URL: sip:"phone number"@IP address (such as 1225556@10.1.2.4, where "122556" is the phone number of the source or destination) or sip:"phone_number"@"domain name", such as 122556@myproxy.com. Note that the SIP URI host name can be configured differently per called number.
- Supports RFC 4040, RTP payload format for a 64 kbit/s transparent data.
- Can negotiate coder from a list of given coders.
- Supports negotiation of dynamic payload types.
- Supports multiple ptime values per coder.
- Supports RFC 3389, RTP Payload for Comfort Noise.
- Supports RFC 3824, Using E.164 numbers with SIP (ENUM).
- Supports reception and DNS resolution of FQDNs received in SDP.
- Supports <draft-ietf-sip-gruu-09>, Obtaining and Using Globally Routable User Agent (UA) URIs (GRUU) in SIP
- Responds to OPTIONS messages both outside a SIP dialog and in mid-call. Generates SIP OPTIONS messages as Proxy keep-alive mechanism.
- Publishes the total number of free Tel channels in a 200 OK response to an OPTIONS requests.
- Implementation of MWI IETF <draft-ietf-sipping-mwi-04.txt>, including SUBSCRIBE (to the MWI server). The MediaPack FXS gateways can accept an MWI NOTIFY message that indicates waiting messages or indicates that the MWI is cleared.
- Supports 3-Way Conference using an external media server..

For more updated information on the gateway's supported features, refer to the latest MP-11x & MP-124 SIP Release Notes.

2 MediaPack Physical Description

This section provides detailed information on the hardware, the location and functionality of the LEDs, buttons and connectors on the front and rear panels of the MP-11x (refer to Section 2.1 below) and MP-124 (refer to Section 2.2 on page 27) gateways.

For detailed information on installing the MediaPack, refer to Chapter 0 on page 29.

2.1 MP-11x Physical Description

2.1.1 MP-11x Front Panel

Figure 2-1 illustrates the front layout of the MP-118 (almost identical on MP-114 and MP-112). Table 2-1 lists and describes the front panel LEDs on the MP-11x.

Tip:	MP-11x (FXS/FXO) gateways feature similar front panel LEDs; they only differ in the number of **Channels Status** LEDs, which correspond to the number of channels.

Figure 2-1: MP-118 Front Panel Connectors

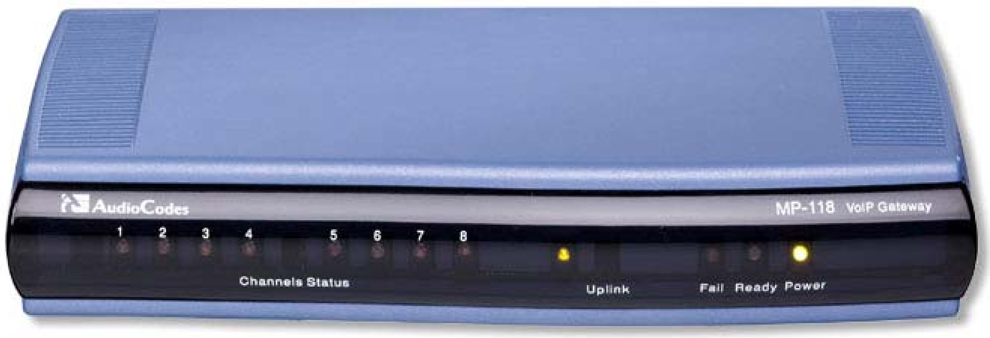


Table 2-1: Definition of MP-11x Front Panel LED Indicators (continues on pages 26 to 26)

LED	Type	Color	State	Definition
Channels Status	Telephone Interface	Green	Blinking	The phone is ringing (incoming call, before answering).
			Fast Blinking	Line malfunction
			Off	Normal onhook position
			On	Offhook
				Ringing
Uplink	Ethernet Link Status	Green	On	Valid 10/100 Base-TX Ethernet connection
			Off	No uplink
Fail	Failure Indication	Red	On	Failure (fatal error). Or system initialization.
			Off	Normal working condition
Ready	Device Status	Green	On	Device powered, self-test OK
			Off	Software loading or System failure
Power	Power Supply Status	Green	On	Power iscurrently being supplied to the device
			Off	Either there's a failure / disruption in the AC power supply or power is currently not being supplied to the device through the AC power supply entry.

2.1.2 MP-11x Rear Panel

Figure 2-2 illustrates the rear layout of the MP-118 (almost identical on MP-114 and MP-112). Table 2-2 lists and describes the rear panel connectors and button on the MP-11x.

Figure 2-2: MP-118 Rear Panel Connectors

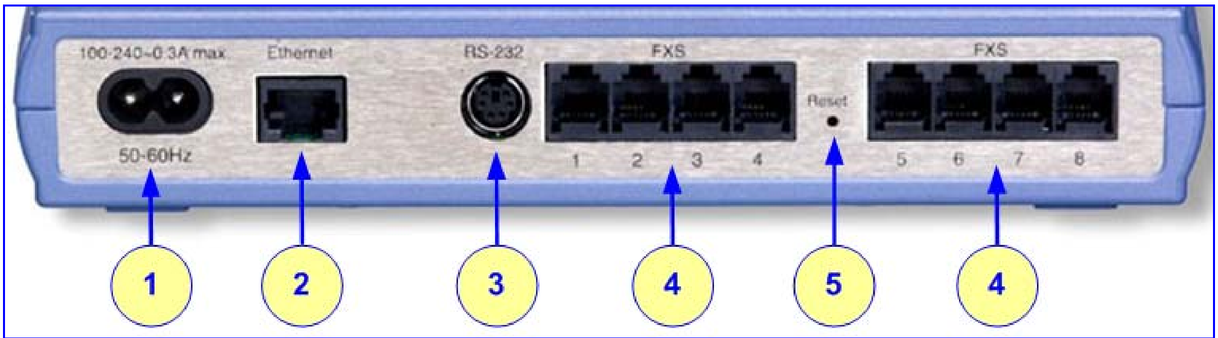


Table 2-2: MP-11x Rear Panel Component Descriptions

Item #	Label	Component Description
1	**100-240~0.3A max.**	AC power supply socket
2	**Ethernet**	10/100 Base-TX Uplink port
3	**RS-232**	RS-232 status port (requires a DB-9 to PS/2 adaptor)
4	**FXS or FXO**	2, 4 or 8 FXS / FXO ports
5	**Reset**	Reset button

2.2 MP-124 Physical Description

2.2.1 MP-124 Front Panel

Figure 2-3 illustrates the front layout of the MP-124. Table 2-3 describes the Reset button located on the front panel. Table 2-4 lists and describes the front panel LEDs.

Figure 2-3: MP-124 Front Panel

Table 2-3: Front Panel Buttons on the MP-124

Type	Function	Comment
Reset button	Resets the MP-124	Press the reset button with a paper clip or any other similar pointed object, until the gateway is reset.
	Restores the MP-124 parameters to their factory default values	Refer to Section 10.1 on page 261.

Table 2-4: Indicator LEDs on the MP-124 Front Panel

Label	Type	Color	State	Function
Ready	Device Status	Green	On	Device Powered, self-test OK
		Orange	Blinking	Software Loading/Initialization
		Red	On	Malfunction
LAN	Ethernet Link Status	Green	On	Valid 10/100 Base-TX Ethernet connection
		Red	On	Malfunction
Control	Control Link	Green	Blinking	Sending and receiving SIP messages
		Blank	--	No traffic
Data	Packet Status	Green	Blinking	Transmitting RTP (Real-Time Transport Protocol) Packets
		Red	Blinking	Receiving RTP Packets
		Blank	--	No traffic
Channels	Telephone Interface	Green	On	Offhook / Ringing for FXS Phone Port
				FXO Line-Seize/Ringing State for Line Port
		Green	Blinking	There's an incoming call, before answering
		Red	On	Line Malfunction
		Blank	--	Normal

2.2.2 MP-124 Rear Panel

The figure below illustrates the rear panel of the MP-124. For descriptions of the rear panel components, refer to Table 2-5. For the functionality of rear panel Ethernet LEDs, refer to Table 2-6.

Figure 2-4: MP-124 (FXS) Rear Panel Connectors

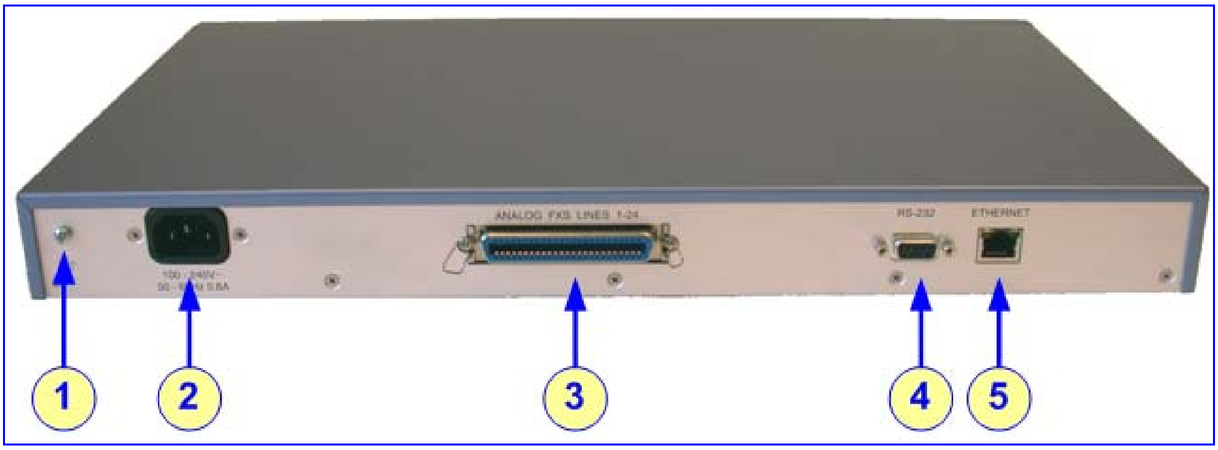


Table 2-5: MP-124 Rear Panel Component Descriptions

Item #	Label	Component Description
1	⏚	Protective earthing screw (mandatory for all installations). Accepts a 6-32 UNC screw.
2	**100-250 V~ 50 - 60 Hz 2A**	AC power supply socket.
3	**ANALOG FXS LINES 1 –24**	50-pin Telco for 1 to 24 analog lines.
4	**RS-232**	9 pin RS-232 status port.
5	**ETHERNET**	10/100 Base-TX Ethernet connection.

The Ethernet LEDs are located within the RJ-45 socket. The table below describes these LEDs.

Table 2-6: Ethernet LEDs on the MP-124 Rear Panel

Label	Type	Color	State	Function
ETHERNET	Ethernet Status	Green	On	Valid 10/100 Base-TX Ethernet connection
		Red	On	Malfunction

3 Installing the MediaPack

This section provides information on the installation procedure for the MP-11x (refer to Section 3.1 below) and the MP-124 (refer to Section 3.2 on page 36). For information on how to start using the gateway, refer to Chapter 4 on page 41.

Caution Electrical Shock

The equipment must only be installed or serviced by qualified service personnel

3.1 Installing the MP-11x

➢ To install the MP-11x, take these 4 steps:

1. Unpack the MP-11x (refer to Section 3.1.1).
2. Check the package contents (refer to Section 3.1.2).
3. Mount the MP-11x (refer to Section 3.1.4 on page 30).
4. Cable the MP-11x (refer to Section 3.1.5 on page 38).

After connecting the MP-11x to the power source, the **Ready** and **Power** LEDs on the front panel turn to green (after a self-testing period of about two minutes). Any malfunction in the startup procedure changes the **Fail** LED to red and the **Ready** LED is turned off (refer to Table 2-1 on page 26 for details on the MP-11x LEDs).

When you have completed the above relevant sections you are then ready to start configuring the gateway (Chapter 5 on page 49).

3.1.1 Unpacking

➢ To unpack the MP-11x, take these 6 steps:

1. Open the carton and remove the packing materials.
2. Remove the MP-11x gateway from the carton.
3. Check that there is no equipment damage.
4. Check, retain and process any documents.
5. Notify AudioCodes or your local supplier of any damage or discrepancies.
6. Retain any diskettes or CDs.

3.1.2 Package Contents

Ensure that in addition to the MP-11x, the package contains:

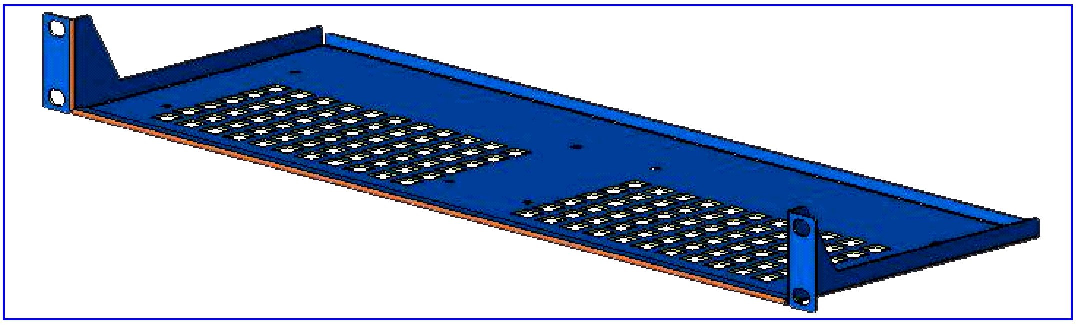
- AC power cable.
- Small plastic bag containing four anti-slide bumpers for desktop installation.
- A CD with software and documentation may be included.
- The MediaPack Fast Track Installation Guide.

3.1.3 19-inch Rack Installation Package (Optional)

An additional option is available for installing the MP-11x in a 19-inch rack. The 19-inch rack installation package contains a single shelf (shown in Figure 3-1) and eight shelf-to-device screws.

Note: The 19-inch rack shelf is not supplied in the standard package kit, but can be ordered separately: Bulk Pack package (MCMK00015) containing 10 rack mounting shelves for MP-11x. For ordering and pricing, please contact your AudioCodes' distributor.

Figure 3-1: 19-inch Rack Shelf



3.1.4 Mounting the MP-11x

The MP-11x can be mounted on a desktop (refer to Section 3.1.4.1 below), on a wall (refer to Section 3.1.4.2) or installed in a standard 19-inch rack (refer to Section 3.1.4.3).

Figure 3-2 below describes the MP-11x base.

Figure 3-2: View of the MP-11x Base

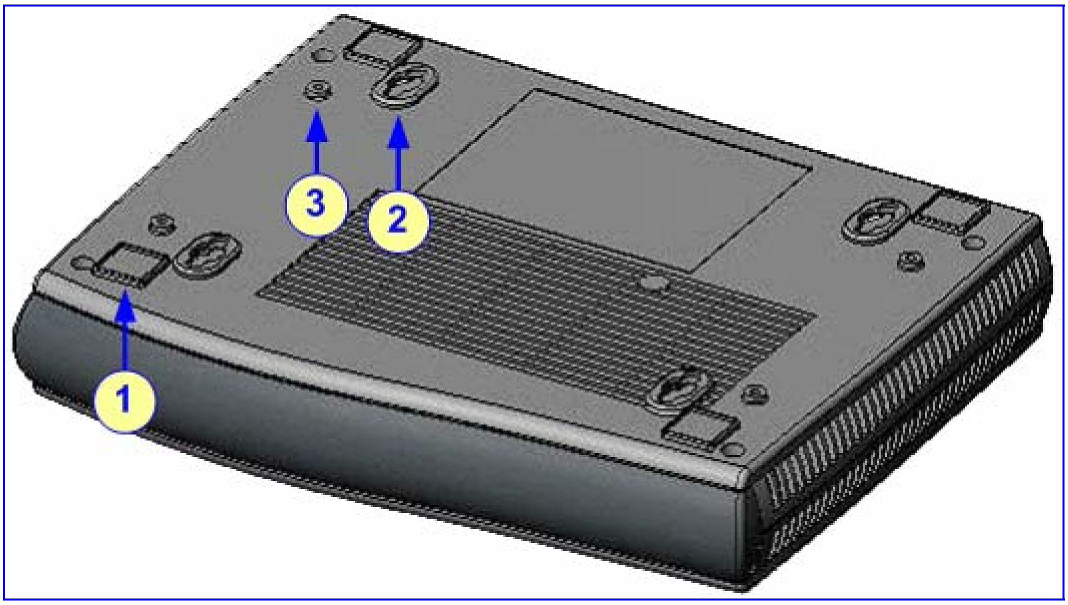


Table 3-1: View of the MP-11x Base

Item #	Functionality
1	Square slot used to attach anti-slide bumpers (for desktop mounting)
2	Oval notch used to attach the MP-11x to a wall
3	Screw opening used to attach the MP-11x to a 19-inch shelf rack

3.1.4.1 Mounting the MP-11x on a Desktop

Attach the four (supplied) anti-slide bumpers to the base of the MP-11x (refer to item #1 in Figure 3-2) and place it on the desktop in the position you require.

3.1.4.2 Mounting the MP-11x on a Wall

➢ **To mount the MP-11x on a wall, take these 4 steps:**

1. Drill four holes according to the following dimensions:
   - Side-to-side distance 140 mm.
   - Front-to-back distance 101.4 mm.
2. Insert a wall anchor of the appropriate size into each hole.
3. Fasten a DIN 96 3.5X20 wood screw (not supplied) into each of the wall anchors.
4. Position the four oval notches located on the base of the MP-11x (refer to item #2 in Figure 3-2) over the four screws and hang the MP-11x on them.

3.1.4.3 Installing the MP-11x in a 19-inch Rack

The MP-11x can be installed in a standard 19-inch rack by placing it on an AudioCodes' 19-inch rack-mounting shelf that is pre-installed in the rack. The shelf can hold up to two MP-11x gateways. This shelf can be ordered separately from AudioCodes.

> **Note:** The 19-inch rack shelf is not supplied in the standard package kit, but can be ordered separately (Bulk Pack package MCMK00015 with 10 rack mounting shelves for MP-11x). For ordering and pricing, please contact your AudioCodes' distributor.

Figure 3-3: MP-11x Rack Mount

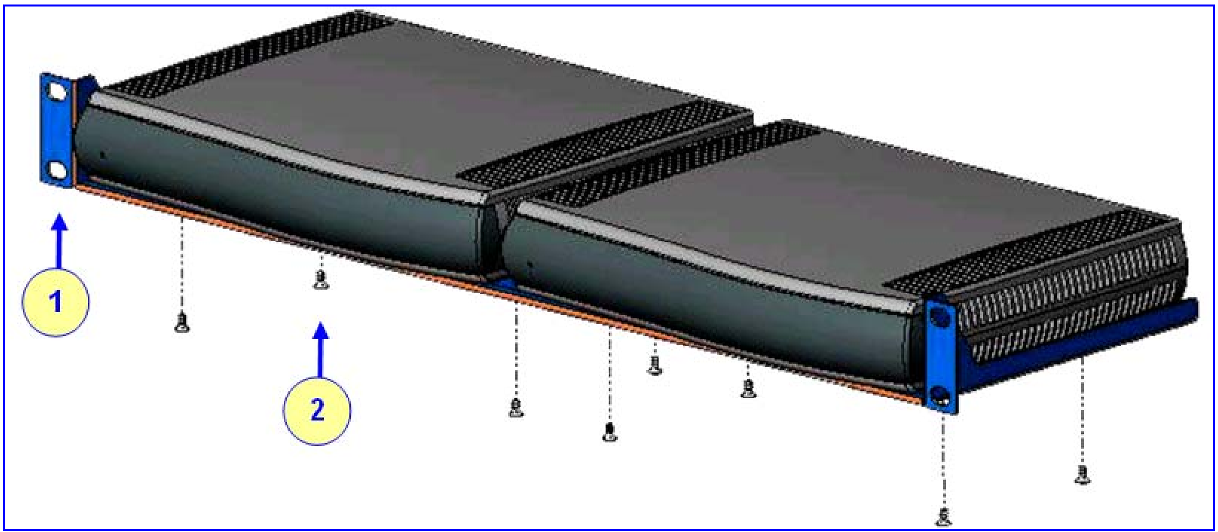


Table 3-2: MP-11x Rack Mount

Item #	Functionality
1	Standard rack holes used to attach the shelf to the rack
2	Eight shelf-to-device screws

➢ To install the MP-11x in a 19-inch rack, take these 3 steps:

1. Use the shelf-to-device screws (supplied) to attach one or two MP-11x devices to the shelf.
2. Position the shelf in the rack and line up its side holes with the rack frame holes.
3. Use four standard rack screws (not supplied) to attach the shelf to the rack.

3.1.5 Cabling the MP-11x

Cable your MP-11x according to each section of Table 3-3. For detailed information on the MP-11x rear panel connectors, refer to Table 2-2 on page 26.

Table 3-3: MP-11x Cables and Cabling Procedure

Cable	Cabling Procedure	
RJ-45 Ethernet cable	Connect the Ethernet connection on the MP-11x directly to the network using a crossover RJ-45 Ethernet cable. For connector pinouts refer to Figure 3-4 below. Note that when assigning an IP address to the MP-11x using HTTP (under Step 1 in Section 4.2.1), you may be required to disconnect this cable and re-cable it differently.	
RJ-11 two-wire telephone cords	Connect the RJ-11 FXS connectors to fax machines, modems, or phones.	Ensure that FXS and FXO ports are connected to the correct devices, otherwise damage can occur. The RJ-11 connector pinouts is described in Figure 3-5).
	Connect the RJ-11 FXO connectors to telephone exchange analog lines or PBX extensions.	
Lifeline	For detailed information on setting up the Lifeline, refer to the procedure in Section 3.1.5.2 on page 34.	
RS-232 serial cable	For detailed information on connecting the MP-11x RS-232 port to your PC, refer to Section 3.1.5.1.	
AC Power cable	Connect the MP-11x power socket to the mains.	

Warning: To reduce the risk of fire, use only No. 26 AWG or larger telecommunication line cords.

Warning: Units must be connected by service personnel to a socket-outlet with a protective earthing connection.

Figure 3-4: RJ-45 Ethernet Connector Pinouts

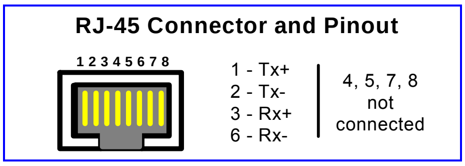
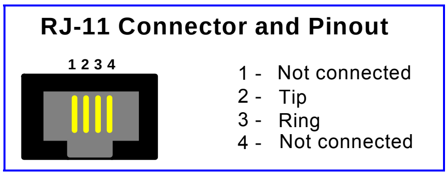


Figure 3-5: RJ-11 Phone Connector Pinouts



3.1.5.1 Connecting the MP-11x RS-232 Port to Your PC

Using a standard RS-232 straight cable (not a cross-over cable) with DB-9 connectors, connect the MP-11x RS-232 port (using a DB-9 to PS/2 adaptor) to either COM1 or COM2 RS-232 communication port on your PC. The pinouts of the PS/2 connector is shown below in Figure 3-6.

A PS/2 to DB-9 adaptor is not included with the MP-11x package. For the PS/2 to DB-9 pinouts, refer to Figure 3-7 below.

For information on establishing a serial communications link with the MP-11x, refer to Section 10.2 on page 262.

Figure 3-6: PS/2 Connector Pinouts

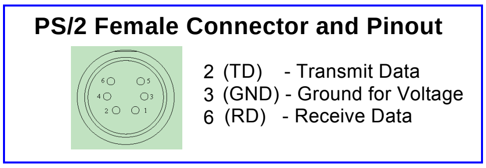
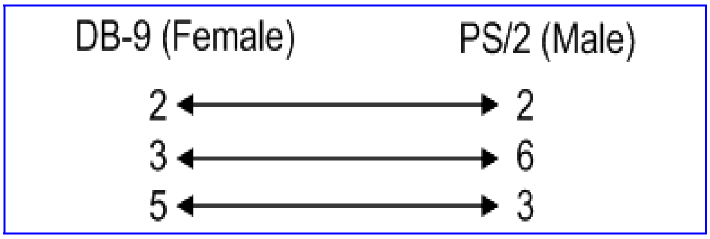


Figure 3-7: PS/2 to DB-9 Adaptor Pinouts



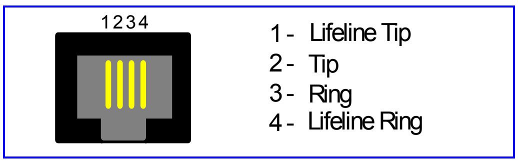
3.1.5.2 Cabling the MP-11x/FXS Lifeline

The Lifeline provides a wired analog POTS phone connection to any PSTN or PBX FXS port when there is no power, or when the network connection fails. Users can therefore use the Lifeline phone even when the MP-11x is not powered on or not connected to the network.

On FXS gateways, a single Lifeline, connected to port #1 via a splitter (not supplied), is available. On combined FXS/FXO gateways a splitter isn't required; all FXS ports are automatically connected to FXO ports (FXS port 0 to FXO port 4 and so forth). On FXO gateways a Lifeline isn't available.

The Lifeline's splitter connects pins #1 and #4 to another source of an FXS port, and pins #2 and #3 to the POTS phone. Refer to the Lifeline splitter pinouts in Figure 3-8.

Figure 3-8: Lifeline Splitter Pinouts and RJ-11 Connector



➢ **To cable the MP-11x/FXS Lifeline, take these 3 steps:**

1. Connect the Lifeline splitter to port #1 on the MP-11x (the Lifeline splitter is a special order option).
2. Connect the Lifeline phone to Port A on the Lifeline splitter.
3. Connect an analog PSTN line to Port B on the Lifeline splitter.

➢ To cable the combined MP-11x FXS/FXO Lifeline, take these 2 steps:

1. Connect a fax machine, modem, or phone to each of the FXS ports.
2. Connect an analog PSTN line to each of the FXO ports.

Note: The use of the Lifeline on network failure can be disabled using the 'LifeLineType' *ini* file parameter (described in Table 5-55 on page 182).

3.2 Installing the MP-124

➢ To install the MP-124, take these 4 steps:

1. Unpack the MP-124 (refer to Section 3.2.1 below).
2. Check the package contents (refer to Section 3.2.2 below).
3. Mount the MP-124 (refer to Section 3.2.3 on page 36).
4. Cable the MP-124 (refer to Section 3.2.4 on page 38).

After connecting the MP-124 to the power source, the Ready and LAN LEDs on the front panel turn to green (after a self-testing period of about 1 minute). Any malfunction changes the Ready LED to red.

When you have completed the above relevant sections you are then ready to start configuring the gateway (Chapter 4 on page 41).

3.2.1 Unpacking

➢ To unpack the MP-124, take these 6 steps:

1. Open the carton and remove packing materials.
2. Remove the MP-124 gateway from the carton.
3. Check that there is no equipment damage.
4. Check, retain and process any documents.
5. Notify AudioCodes or your local supplier of any damage or discrepancies.
6. Retain any diskettes or CDs.

3.2.2 Package Contents

Ensure that in addition to the MP-124, the package contains:

- AC power cable.
- 2 short equal-length brackets and bracket-to-device screws for the 19-inch rack installation.
- A CD with software and documentation may be included.
- The MediaPack Fast Track Installation Guide.

3.2.3 Mounting the MP-124

The MP-124 can be mounted on a desktop or installed in a standard 19-inch rack. Refer to Section 3.2.4 on page 38 for cabling the MP-124.

3.2.3.1 Mounting the MP-124 on a Desktop

No brackets are required. Simply place the MP-124 on the desktop in the position you require.

Figure 3-9: Desktop or Shelf Mounting

Rack Mount Safety Instructions (UL)

When installing the chassis in a rack, be sure to implement the following Safety instructions recommended by Underwriters Laboratories:

- **Elevated Operating Ambient** - If installed in a closed or multi-unit rack assembly, the operating ambient temperature of the rack environment may be greater than room ambient. Therefore, consideration should be given to installing the equipment in an environment compatible with the maximum ambient temperature (Tma) specified by the manufacturer.
- **Reduced Air Flow** - Installation of the equipment in a rack should be such that the amount of air flow required for safe operation on the equipment is not compromised.
- **Mechanical Loading** - Mounting of the equipment in the rack should be such that a hazardous condition is not **achieved** due to uneven mechanical loading.
- **Circuit Overloading** - Consideration should be given to the connection of the equipment to the supply circuit **and** the effect that overloading of the circuits might have on overcurrent protection and supply wiring. Appropriate consideration of equipment nameplate ratings should be used when addressing this concern.
- **Reliable Earthing** - Reliable earthing of rack-mounted equipment should be maintained. Particular attention should be given to supply connections other than direct connections to the branch circuit (e.g., use of power strips.)

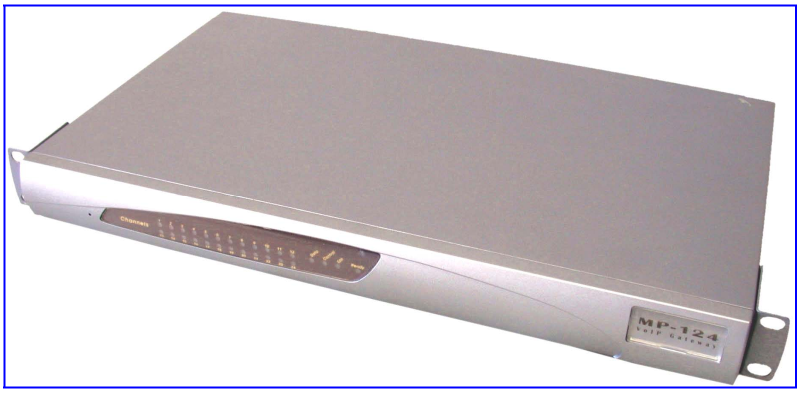
3.2.3.2 Installing the MP-124 in a 19-inch Rack

The MP-124 is installed into a standard 19-inch rack by the addition of two short (equal-length) supplied brackets. The MP-124 with brackets for rack installation is shown in Figure 3-10.

➢ To install the MP-124 in a 19-inch rack, take these 7 steps:

1. Remove the two screws on one side of the device nearest the front panel.
2. Insert the peg on one of the brackets into the third air vent down on the column of air vents nearest the front panel.
3. Swivel the bracket until the holes in the bracket line up with the two empty screw holes on the device.
4. Use the supplied screws to attach the bracket to the side of the device.
5. Repeat steps 1 to 4 to attach the second bracket to the other side of the device.
6. Position the device in the rack and line up the bracket holes with the rack frame holes.
7. Use four standard rack screws (not supplied) to attach the device to the rack.

Figure 3-10: MP-124 with Brackets for Rack Installation



3.2.4 Cabling the MP-124

Cable your MP-124 according to each section of Table 3-4. For detailed information on the MP-124 rear panel connectors, refer to Section 2.2.2 on page 28.

Table 3-4: MP-124 Cables and Cabling Procedure

Cable	Cabling Procedure
Protective earthing strap	Connect an earthed strap to the chassis protective earthing screw (6-32 UNC screw) and fasten it securely according to the safety standards.
RJ-45 Ethernet cable	Connect the Ethernet connection on the MP-124 directly to the network using a crossover RJ-45 Ethernet cable. For connector pinouts refer to Figure 3-11 below. Note that when assigning an IP address to the MP-124 using HTTP (under Step 1 in Section 4.2.1), you may be required to disconnect this cable and re-cable it differently.
50-pin Telco cable (MP-124 devices only). **An Octopus cable is not included with the MP-124 package.**	Refer to the MP-124 Safety Notice below. ▪ Wire the 50-pin Telco connectors according to the pinouts in Figure 3-12 on page 38, and Figure 3-13 on page 39. ▪ Attach each pair of wires from a 25-pair Octopus cable to its corresponding socket on the MDF Adaptor Block's rear. ▪ Connect the wire-pairs at the other end of the cable to a male 50-pin Telco connector. ▪ Insert and fasten this connector to the female 50-pin Telco connector on the MP-124 rear panel (labeled Analog FXS Lines 1-24). ▪ Connect the telephone lines from the Adaptor Block to a fax machine, modem, or telephones by inserting each RJ-11 connector on the 2-wire line cords of the POTS phones into the RJ-11 sockets on the front of an MDF Adaptor Block as shown in Figure 3-13 on page 39.
RS-232 serial cable	For detailed information on connecting the MP-124 RS-232 port to your PC, refer to Section 3.2.4.1 on page 40.
AC Power cable	Connect the MP-124 power socket to the mains.

MP-124 Safety Notice

To protect against electrical shock and fire, use a 26 AWG min wire to connect analog FXS lines to the 50-pin Telco connector.

Figure 3-11: RJ-45 Ethernet Connector Pinouts

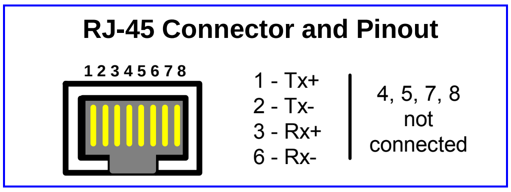


Figure 3-12: 50-pin Telco Connector (MP-124/FXS only)

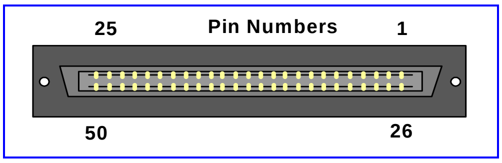


Figure 3-13: MP-124 in a 19-inch Rack with MDF Adaptor

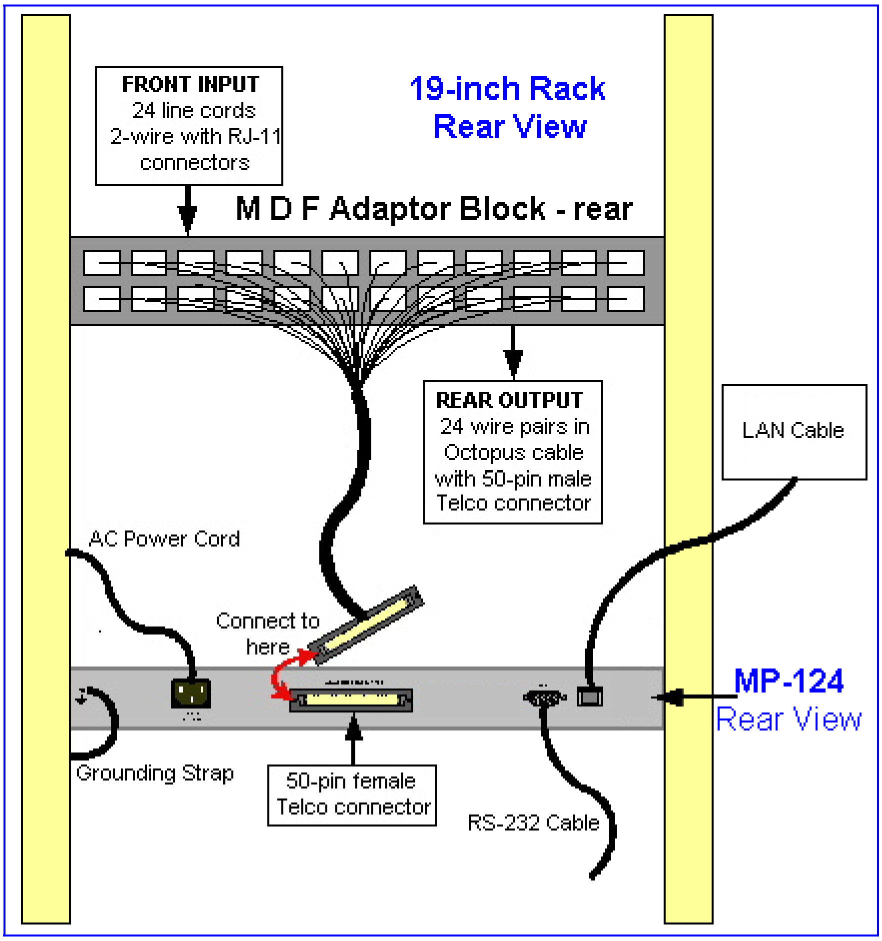


Table 3-5: Pin Allocation in the 50-pin Telco Connector

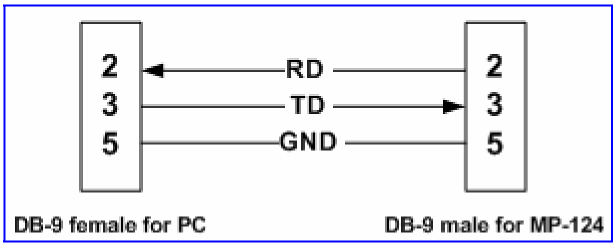
Phone Channel	Connector Pins	Phone Channel	Connector Pins
1	1/26	13	13/38
2	2/27	14	14/39
3	3/28	15	15/40
4	4/29	16	16/41
5	5/30	17	17/42
6	6/31	18	18/43
7	7/32	19	19/44
8	8/33	20	20/45
9	9/34	21	21/46
10	10/35	22	22/47
11	11/36	23	23/48
12	12/37	24	24/49

3.2.4.1 Connecting the MP-124 RS-232 Port to Your PC

Using a standard RS-232 straight cable (not a cross-over cable) with DB-9 connectors, connect the MP-124 RS-232 port to either COM1 or COM2 RS-232 communication port on your PC. The required connector pinouts and gender are shown below in Figure 3-14.

For information on establishing a serial communications link with the MP-124, refer to Section 10.2 on page 262.

Figure 3-14: MP-124 RS-232 Cable Wiring



4 Getting Started

The MediaPack is supplied with default networking parameters (show in Table 4-1 below) and with an application software already resident in its flash memory (with factory default parameters).

Before you begin configuring the gateway, change its default IP address to correspond with your network environment (refer to Section 4.2) and learn about the configuration methods available on the MediaPack (refer to Section 4.1 below).

For information on quickly setting up the MediaPack with basic parameters using a standard Web browser, refer to Section 4.2.3 on page 43.

Table 4-1: MediaPack Default Networking Parameters

FXS or FXO	Default Value
FXS	10.1.10.10
FXO	10.1.10.11
FXS / FXO	10.1.10.10
MediaPack default subnet mask is 255.255.0.0, default gateway IP address is 0.0.0.0	

4.1 Configuration Concepts

Users can utilize the MediaPack in a wide variety of applications, enabled by its parameters and configuration files (e.g., Call Progress Tones (CPT)). The parameters can be configured and configuration files can be loaded using:

- A standard Web Browser (described and explained in Chapter 5 on page 49).
- A configuration file referred to as the *ini* file. For information on how to use the *ini* file, refer to Chapter 6 on page 209.
- An SNMP browser software (refer to Chapter 14 on page 305).
- AudioCodes' Element Management System (EMS) (refer to Section 14.10 on page 322 and to AudioCodes' EMS User's Manual or EMS Product Description).

To upgrade the MediaPack (load new software or configuration files onto the gateway) use the Software Upgrade wizard, available through the Web Interface (refer to Section 5.9.1 on page 197), or alternatively use the BootP/TFTP configuration utility (refer to Section 7.3.1 on page 213).

For information on the configuration files, refer to Chapter 6 on page 209.

4.2 Assigning the MediaPack IP Address

To assign an IP address to the MediaPack use one of the following methods:

- HTTP using a Web browser (refer to Section 4.2.1 below).
- BootP (refer to Section 4.2.2 on page 42).
- Voice Menu using a standard touch-tone telephone connected to one of the FXS analog ports (refer to Section 4.2.3 on page 43). This method doesn't apply to FXO gateways.
- The embedded Command Line Interface (CLI) accessed via Telnet or RS-232 (refer to Section 4.2.4 on page 44).
- DHCP (refer to Section 7.2 on page 212).
- Use the 'Reset' button at any time to restore the MediaPack networking parameters to their factory default values (refer to Section 10.1 on page 261).

4.2.1 Assigning an IP Address Using HTTP

➢ **To assign an IP address using HTTP, take these 8 steps:**

1. Disconnect the MediaPack from the network and reconnect it to your PC using one of the following two methods:
   - Use a standard Ethernet cable to connect the network interface on your PC to a port on a network hub / switch. Use a second standard Ethernet cable to connect the MediaPack to another port on the same network hub / switch.
   - Use an Ethernet cross-over cable to directly connect the network interface on your PC to the MediaPack.
2. Change your PC's IP address and subnet mask to correspond with the MediaPack factory default IP address and subnet mask, shown in Table 4-1. For details on changing the IP address and subnet mask of your PC, refer to Windows™ Online Help (Start > Help).
3. Access the MediaPack Embedded Web Server (refer to Section 5.3 on page 51).
4. In the 'Quick Setup' screen (shown in Figure 4-1), set the MediaPack 'IP Address', 'Subnet Mask' and 'Default Gateway IP Address' fields under 'IP Configuration' *to correspond with your network IP settings.* If your network doesn't feature a default gateway, enter a dummy value in the 'Default Gateway IP Address' field.
5. Click the **Reset** button, and then at the prompt, click **OK**; the MediaPack applies the changes and restarts.

Tip:	Record and retain the IP address and subnet mask you assign the MediaPack. Do the same when defining new username or password. If the Embedded Web Server is unavailable (for example, if you've lost your username and password), use the BootP/TFTP (Trivial File Transfer Protocol) configuration utility to access the device, 'reflash' the load and reset the password (refer to Appendix C on page 349 for detailed information on using a BootP/TFTP configuration utility to access the device).

6. Disconnect your PC from the MediaPack or from the hub / switch (depending on the connection method you used in Step 1).
7. Reconnect the MediaPack and your PC (if necessary) to the LAN.
8. Restore your PC's IP address & subnet mask to what they originally were. If necessary, restart your PC and re-access the MediaPack via the Embedded Web Server with its new assigned IP address.

4.2.2 Assigning an IP Address Using BootP

Note:	BootP procedure can also be performed using any standard compatible BootP server.

Tip:	You can also use BootP to load the auxiliary files to the MediaPack (refer to Section 5.9.2.1 on page 203).

➢ To assign an IP address using BootP, take these 3 steps:

1. Open the BootP application (supplied with the MediaPack software package).
2. Add client configuration for the MediaPack, refer to Section C.11.1 on page 355.
3. Use the reset button to *physically* reset the gateway causing it to use BootP; the MediaPack changes its network parameters to the values provided by the BootP.

4.2.3 Assigning an IP Address Using the Voice Menu Guidance

Initial configuration of the gateway can be performed using a standard touch-tone telephone connected to one of the FXS analog ports. The voice menu can also be used to query and modify basic configuration parameters.

➢ To assign an IP address using the voice menu guidance, take these 7 steps:

1. Connect a telephone to one of the FXS ports. Lift the handset and dial ***12345 (three stars followed by the digits 1, 2, 3, 4, 5).
2. Wait for the 'configuration menu' voice prompt to be played.
3. To change the IP address, press 1 followed by the pound key (#).
   - The current IP address of the gateway is played. Press # to change it.
   - Dial the new IP address; use the star (*) key instead of dots ("."), e.g., 192*168*0*4 and press # to finish.
   - Review the new IP address, and press 1 to save it.
4. To change the subnet mask, press 2 followed by the # key.
   - The current subnet mask of the gateway is played. Press # to change it.
   - Dial the new subnet mask; e.g. 255*255*0*0 and press # to finish.
   - Review the new subnet mask, and press 1 to save it.
5. To change the default gateway IP address, press 3 followed by the # key.
   - The gateway's current default gateway address is played. To change, press #.
   - Dial the new default gateway address; e.g., 192*168*0*1 and press # to finish.
   - Review the new default gateway address, and press 1 to save it.
6. Hang up the handset. Access the gateway's Embedded Web Server with the new IP address you assigned (refer to Section 5.3 on page 51).
7. Complete the gateway's configuration and save it to non-volatile memory (refer to Section 5.10.2 on page 205).

The following configuration parameters can be queried or modified via the voice menu:

Table 4-2: Configuration Parameters Available via the Voice Menu
(continues on pages 43 to 44)

Item Number at Menu Prompt	Description
1	IP address
2	Subnet mask
3	Default gateway IP address
4	Primary DNS server IP address
7	DHCP enable / disable
11	MGCP call agent IP address (N/A)

Table 4-2: Configuration Parameters Available via the Voice Menu (continues on pages 43 to 44)

Item Number at Menu Prompt	Description
12	MGCP call agent port number (N/A)
99	Voice menu password (initially 12345). **Note:** The voice menu password can also be changed using the parameter 'VoiceMenuPassword' (refer to Table 5-50 on page 174).

4.2.4 Assigning an IP Address Using the CLI

First access the CLI using a standard Telnet application or using a serial communication software (e.g., HyperTerminal™) connected to the MediaPack RS-232 port (refer to Section 4.2.4.1 below). Then assign the MediaPack an IP address (refer to Section 4.2.4.2 below).

4.2.4.1 Access the CLI

➢ **To access the CLI via the Embedded Telnet Server, take these 3 steps:**

1. Enable the Embedded Telnet Server:
   a. Access the MediaPack Embedded Web Server (refer to Section 5.3 on page 51).
   b. Set the parameter 'Embedded Telnet Server' (under **Advanced Configuration** > **Network Settings** > **Application Settings**) to 'Enable (Unsecured)' or 'Enable Secured (SSL)'.
   c. Click the **Maintenance** button on the main menu bar; the 'Maintenance Actions' screen is displayed.
   d. From the 'Burn to FLASH' drop-down list, select 'Yes'.
   e. Click the **Reset** button; the MediaPack is shut down and re-activated. A message about the waiting period is displayed. The screen is refreshed.
2. Use a standard Telnet application to connect to the MediaPack Embedded Telnet Server. Note that if the Telnet server is set to SSL mode, a special Telnet client is required on your PC to connect to the Telnet interface over a secured connection.
3. Login using the username ('Admin') and password ('Admin').

➢ **To access the CLI via the RS-232 port, take these 2 steps:**

1. Connect the RS-232 port to your PC (For the MP-124, refer to Section 3.2.4.1 on page 40. For the MP-11x, refer to Section 3.1.5.1 on page 34).
2. Use serial communication software (e.g., HyperTerminal™) to connect to the MediaPack.
   Set your serial communication software to the following communications port settings:
   - Baud Rate: 115,200 bps (MP-124), 9,600 bps (MP-11x)
   - Data bits: 8
   - Parity: None
   - Stop bits: 1
   - Flow control: None

The CLI prompt becomes available.

4.2.4.2 Assign an IP Address

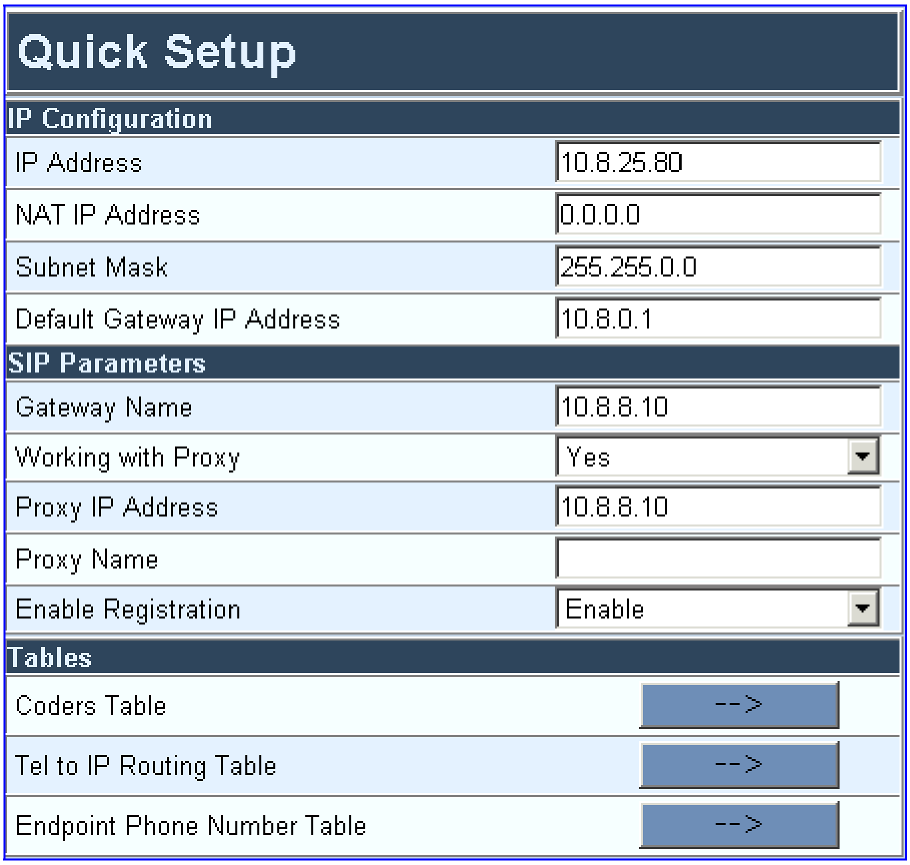
➢ To assign an IP address via the CLI, take these 4 steps:

1. At the prompt type ‘conf’ and press enter; the configuration folder is accessed.
2. To check the current network parameters, at the prompt, type **`GCP IP`**, and then press Enter; the current network settings are displayed.
3. Change the network settings by typing:
   `SCP IP` `[ip_address] [subnet_mask] [default_gateway]`
   (e.g., ‘SCP IP 10.13.77.7 255.255.0.0 10.13.0.1’); the new settings take effect on-the-fly. Connectivity is active at the new IP address.
   Note: This command requires you to enter all three network parameters (each separated by a space).
4. To save the configuration, at the prompt, type **`SAR`**, and then press Enter; the MediaPack restarts with the new network settings.

4.3 Configure the MediaPack *Basic* Parameters

To configure the MediaPack *basic* parameters use the Embedded Web Server's 'Quick Setup' screen (shown in Figure 4-1 below). Refer to Section 5.3 on page 51 for information on accessing the 'Quick Setup' screen.

Figure 4-1: Quick Setup Screen



➢ To configure basic SIP parameters, take these 9 steps:

1. If the MediaPack is connected to a router with Network Address Translation (NAT) enabled, perform the following procedure. If it isn't, leave the 'NAT IP Address' field undefined.
   - Determine the 'public' IP address assigned to the router (by using, for instance, router Web management). If the public IP address is static, enter this in the 'NAT IP Address' field.
   - Enable the DMZ (Demilitarized Zone) configuration on the router for the LAN port where the MediaPack gateway is connected. This enables unknown packets to be routed to the DMZ port.
2. Under 'SIP Parameters', enter the MediaPack domain name in the field 'Gateway Name'. If the field is not specified, the MediaPack IP address is used instead (default).
3. When working with a Proxy server, set 'Working with Proxy' field to 'Yes' and enter the IP address of the primary Proxy server in the field 'Proxy IP Address'. When no Proxy is used, the internal routing table is used to route the calls.
4. Enter the Proxy Name in the field 'Proxy Name'. If Proxy name is used, it replaces the Proxy IP address in all SIP messages. This means that messages are still sent to the physical Proxy IP address but the SIP URI contains the Proxy name instead.

5. Configure ‘Enable Registration’ to ‘Yes’ or ‘No’:
   - ‘No’ = the MediaPack does not register to a Proxy server/Registrar (default).
   - ‘Yes’ = the MediaPack registers to a Proxy server/Registrar at power up and every ‘Registration Time’ seconds; The MediaPack sends a REGISTER request according to the ‘Authentication Mode’ parameter. For detailed information on the parameters ‘Registration Time’ and ‘Authentication Mode’, refer to Table 5-4 on page 66.
6. To configure the Coders Table, click the arrow button next to ‘Coders Table’. For information on how to configure the Coders Table, refer to Section 5.5.1.3 on page 72.
7. To configure the Tel to IP Routing Table, click the arrow button next to ‘Tel to IP Routing Table’. For information on how to configure the Tel to IP Routing Table, refer to Section 5.5.5.2 on page 100.
8. To configure the Endpoint Phone Number Table, click the arrow button next to ‘Endpoint Phone Number’. For information on how to configure the Endpoint Phone Number Table, refer to Section 5.5.7 on page 115.
9. Click the **Reset** button, and then at the prompt, click **OK**; the MediaPack applies the changes and restarts.

You are now ready to start using the VoIP gateway. To prevent unauthorized access to the MediaPack, it is recommended that you change the username and password that are used to access the Web Interface. Refer to Section 5.6.5.1 on page 168 for details on how to change the username and password.

Tip: Once the gateway is configured correctly back up your settings by making a copy of the VoIP gateway configuration (*ini* file) and store it in a directory on your PC. This saved file can be used to restore configuration settings at a future time. For information on backing up and restoring the gateway’s configuration, refer to Section 5.6.3 on page 165.

Reader's Notes

5 Web Management

The Embedded Web Server is used both for gateway configuration, including loading of configuration files, and for run-time monitoring. The Embedded Web Server can be accessed from a standard Web browser, such as Microsoft™ Internet Explorer, Netscape™ Navigator, etc. Specifically, users can employ this facility to set up the gateway configuration parameters. Users also have the option to remotely reset the gateway and to permanently apply the new set of parameters.

5.1 Computer Requirements

To use the Embedded Web Server, the following is required:

- A computer capable of running your Web browser.
- A network connection to the VoIP gateway.
- One of the following compatible Web browsers:
  - Microsoft™ Internet Explorer™ (version 6.0 and higher).
  - Netscape™ Navigator™ (version 7.2 and higher).

Note: The browser must be Java-script enabled. If java-script is disabled, access to the Embedded Web Server is denied.

5.2 Protection and Security Mechanisms

Access to the Embedded Web Server is controlled by the following protection and security mechanisms:

- User accounts (refer to Section 5.2.1 below).
- Read-only mode (refer to Section 5.2.2 on page 51).
- Disabling access (refer to Section 5.2.3 on page 51).
- Secured HTTP connection (HTTPS) (refer to Section 12.2.2 on page 288).
- Limiting access to a predefined list of IP addresses (refer to Section 5.6.5.2 on page 170).
- Managed access using a RADIUS server (refer to Section 12.3 on page 293).

5.2.1 User Accounts

To prevent unauthorized access to the Embedded Web Server, two user accounts are available, a primary and secondary. Each account is composed of three attributes: username, password and access level. The username and password enable access to the Embedded Web Server itself; the access level determines the extent of the access (i.e., availability of screens and read / write privileges). Note that additional accounts can be defined using a RADIUS server (refer to Section 12.3 on page 293).

Table 5-1 lists the available access levels and their privileges.

Table 5-1: Available Access Levels and their Privileges

Access Level	Numeric Representation*	Privileges
Security Administrator	200	Read / write privileges for all screens
Administrator	100	Read-only privilege for security-related screens and read / write privileges for the others
User Monitor	50	No access to security-related and file-loading screens and read-only access to the others
No Access	0	No access to any screen

* The numeric representation of the access level is used only to define accounts in a RADIUS server (the access level ranges from 1 to 255).

The access level mechanism operation is as follows (for both Web and RADIUS accounts):

Each Web screen features two (hard-coded) minimum access levels, read and write. The read access level determines whether the screen can be viewed. The write access level determines whether the information in the screen can be modified.

When a user tries to access a specific Web screen, the user's access level is compared with the access levels of the screen:

- If the access level of the user is less than the screen's read access level, the screen cannot be viewed.
- If the access level of the user is equal to or greater than the screen's read access level but less than the write access level, the screen is read only.
- If the access level of the user is equal to or greater than the screen's write access level, the screen can be modified.

The default attributes for the two accounts are shown in Table 5-2 below:

Table 5-2: Default Attributes for the Accounts

Account / Attribute	Username (Case-Sensitive)	Password (Case-Sensitive)	Access Level
Primary Account	Admin	Admin	Security Administrator*
Secondary Account	User	User	User Monitor

* The access level of the primary account cannot be changed; all other account-attributes can be modified.

The first time a browser request is made, users are requested to provide their account's username and password to obtain access. If the Embedded Web Server is left idle for more than five minutes, the session expires and the user is required to re-enter username and password.

Tip: To access the Embedded Web Server with a different account, click the **Log Off** button and then re-access with a new username and password.

For details on changing the account attributes, refer to Section 5.6.5.1 on page 168. Note that the password and username can be a maximum of 19 case-sensitive characters.

To reset the username and password of both accounts to their defaults, set the *ini* file parameter 'ResetWebPassword' to 1.

5.2.2 Limiting the Embedded Web Server to Read-Only Mode

Users can limit access to the Embedded Web Server to read-only mode by changing the *ini* file parameter 'DisableWebConfig' to 1. In this mode all Web screens, regardless of the access level used, are read-only and cannot be modified. In addition, the following screens cannot be accessed: 'Quick Setup', 'Web User Accounts', 'Reset', 'Save Configuration' and all of the file-loading screens.

Notes:

- Read-only policy can also be applied to selected users by setting the access level of the secondary account to 'User Monitor' (DisableWebConfig = 0) and distributing the primary and secondary accounts to users according to the organization's security policy.
- When DisableWebConfig is set to 1, read-only privileges are applied to all accounts regardless of their access level.

5.2.3 Disabling the Embedded Web Server

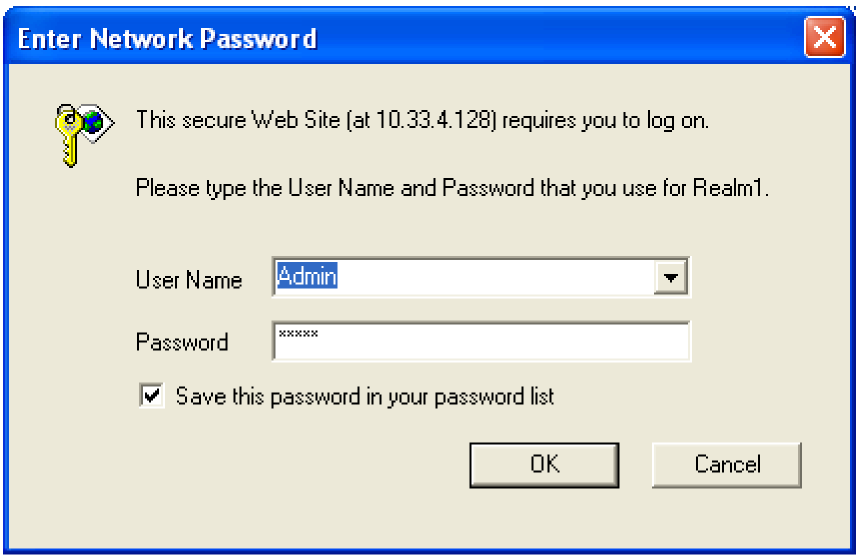
Access to the Embedded Web Server can be disabled by using the *ini* file parameter 'DisableWebTask = 1'. The default is access enabled.

5.3 Accessing the Embedded Web Server

➢ **To access the Embedded Web Server, take these 4 steps:**

1. Open a standard Web-browsing application such as Microsoft™ Internet Explorer™ or Netscape™ Navigator™.
2. In the Uniform Resource Locator (URL) field, specify the IP address of the MediaPack (e.g., http://10.1.10.10); the Embedded Web Server's 'Enter Network Password' screen appears, shown in Figure 5-1.

Figure 5-1: Embedded Web Server Login Screen



3. In the 'User Name' and 'Password' fields, enter the username (default: 'Admin') and password (default: 'Admin'). Note that the username and password are case-sensitive.
4. Click the **OK** button; the 'Quick Setup' screen is accessed (shown in Figure 4-1).

5.3.1 Using Internet Explorer to Access the Embedded Web Server

Internet explorer's security settings may block access to the gateway's Web browser if they're configured incorrectly. In this case, the following message is displayed:

Unauthorized
Correct authorization is required for this area. Either your browser does not perform authorization or your authorization has failed. RomPager server.

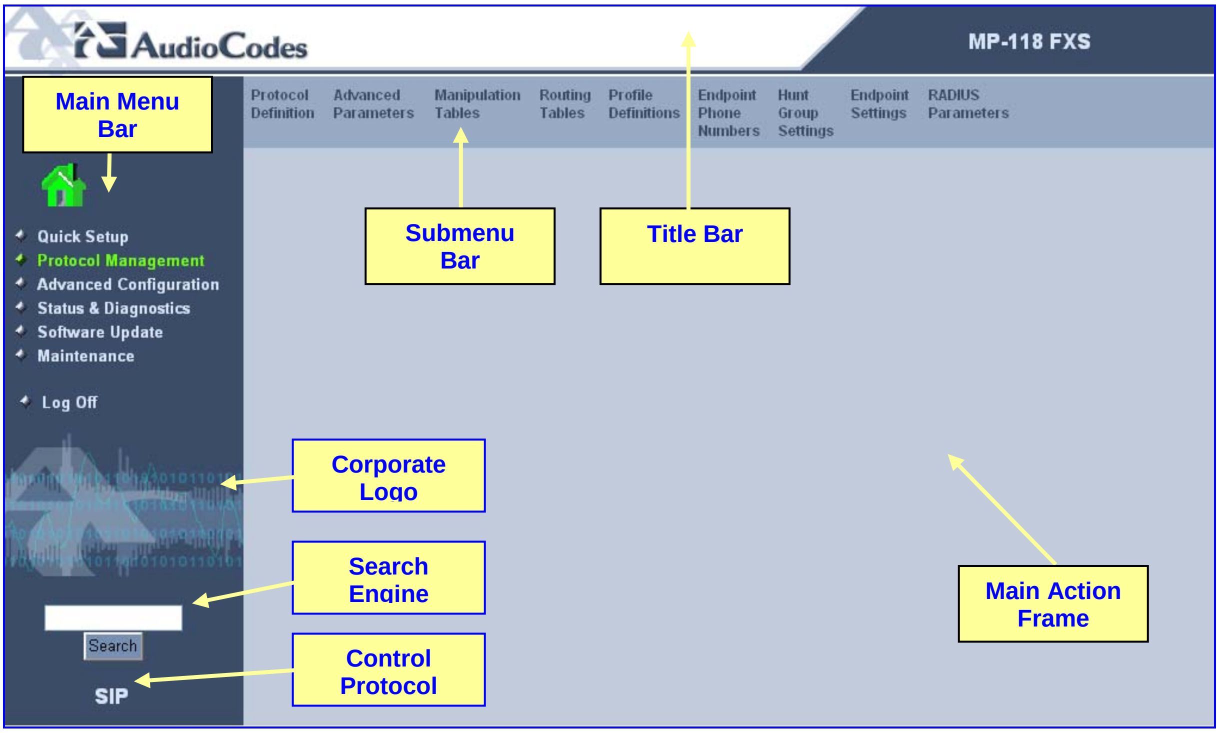
- **To troubleshoot blocked access to Internet Explorer™, take these 3 steps**

1. Delete all cookies from the Temporary Internet files. If this does not clear up the problem, the security settings may need to be altered (continue with Step 2).
2. In Internet Explorer > Tools > Internet Options:
   - Select the **Security** tab, select Custom Level. Scroll down until the Logon options are displayed and change the setting to: **Prompt for username and password**.
   - Select the **Advanced** tab, scroll down until the HTTP 1.1 Settings are displayed and verify that the **Use HTTP 1.1** option is checked.
3. Restart the browser.

5.4 Getting Acquainted with the Web Interface

Figure 5-2 shows the general layout of the Web Interface screen.

Figure 5-2: MediaPack Web Interface (e.g., MP-118 FXS)



The Web Interface screen features the following components:

- **Title bar:** contains three configurable elements: corporate logo, a background image and the product's name. For information on how to modify these elements, refer to Section 10.5 on page 267.
- **Product name:** the gateway model name.
- **Main menu bar:** always appears on the left of every screen to quickly access parameters, submenus, submenu options, functions and operations.
- **Submenu bar:** appears on the top of screens and contains submenu options.
- **Main action frame:** the main area of the screen in which information is viewed and configured.
- **Home icon:** when clicked it opens the 'Trunk & Channel Status' screen (refer to Section 5.8 on page 195).
- **Corporate logo:** AudioCodes' corporate logo. For information on how to remove this logo, refer to Section 10.5 on page 267.
- **Search Engine:** for searching *ini* file parameters configurable by the Embedded Web Server (refer to Section 5.4.4 on page 54).
- **Control Protocol:** the MediaPack control protocol (e.g., SIP).

5.4.1 Main Menu Bar

The main menu bar of the Web Interface is divided into the following menus:

- **Quick Setup:** Use this menu to configure the gateway's basic settings; for the full list of configurable parameters go directly to 'Protocol Management' and 'Advanced Configuration' menus. An example of the Quick Setup configuration is described in Section 4.2.3 on page 43.
- **Protocol Management:** Use this menu to configure the gateway's control protocol parameters and tables (refer to Section 5.5 on page 56).
- **Advanced Configuration:** Use this menu to set the gateway's advanced configuration parameters (for advanced users only) (refer to Section 5.6 on page 137).
- **Status & Diagnostics:** Use this menu to view and monitor the gateway's channels, Syslog messages, hardware / software product information, and to assess the gateway's statistics and IP connectivity information (refer to Section 5.7 on page 187).
- **Software Update:** Use this menu when you want to load new software or configuration files onto the gateway (refer to Section 5.9 on page 197).
- **Maintenance:** Use this menu to remotely lock/unlock the device (refer to Section 5.10 on page 204), save configuration changes to the non-volatile flash memory (refer to Section 5.10.2 on page 205), and remotely reset the gateway (refer to Section 5.10.3 on page 206).

When positioning your curser over a parameter name (or a table) for more than 1 second, a short description of this parameter is displayed. Note that parameters preceded by an exclamation mark (!) are *not* changeable on-the-fly and require that the device be reset.

5.4.2 Saving Changes

To save changes to the volatile memory (RAM) click the **Submit** button (changes to parameters with on-the-fly capabilities are immediately available, other parameter are updated only after a gateway reset). Parameters that are only saved to the volatile memory revert to their previous settings after hardware reset. When performing a software reset (i.e., via Web or SNMP) you can choose to save the changes to the non-volatile memory. To save changes so they are available after a power fail, you must save the changes to the non-volatile memory (flash).

To save the changes to flash, refer to Section 5.10.2 on page 205.

5.4.3 Entering Phone Numbers in Various Tables

Phone numbers entered into various tables on the gateway, such as the Tel to IP routing table, must be entered without any formatting characters. For example, if you wish to enter the phone number 555-1212, it must be entered as 5551212 without the hyphen (-). If the hyphen is entered, the entry does not work. The hyphen character is used in number entry only, as part of a range definition. For example, the entry [20-29] means 'all numbers in the range 20 to 29'.

5.4.4 Searching for Configuration Parameters

The Embedded Web Server provides a search engine that allows you to search any *ini* file parameter that is configurable by the Web server. The **Search** button, located near the bottom of the Main menu bar (refer to Figure 5-2 on page 52) is used to perform parameter searches.

You can search for a specific parameter (e.g., "EnableIPSec") or a sub-string of that parameter (e.g., "sec"). If you search for a sub-string, the Embedded Web Server lists all parameters that contain the searched sub-string in their parameter names.

➤ **To search for an *ini* file parameter configurable by the Web server, take these 3 steps:**

1. In the 'Search Engine' field, enter the *ini* parameter name or sub-string of the parameter name.
2. Click **Search**. The 'Searched Result' screen appears, listing all searched parameter results.

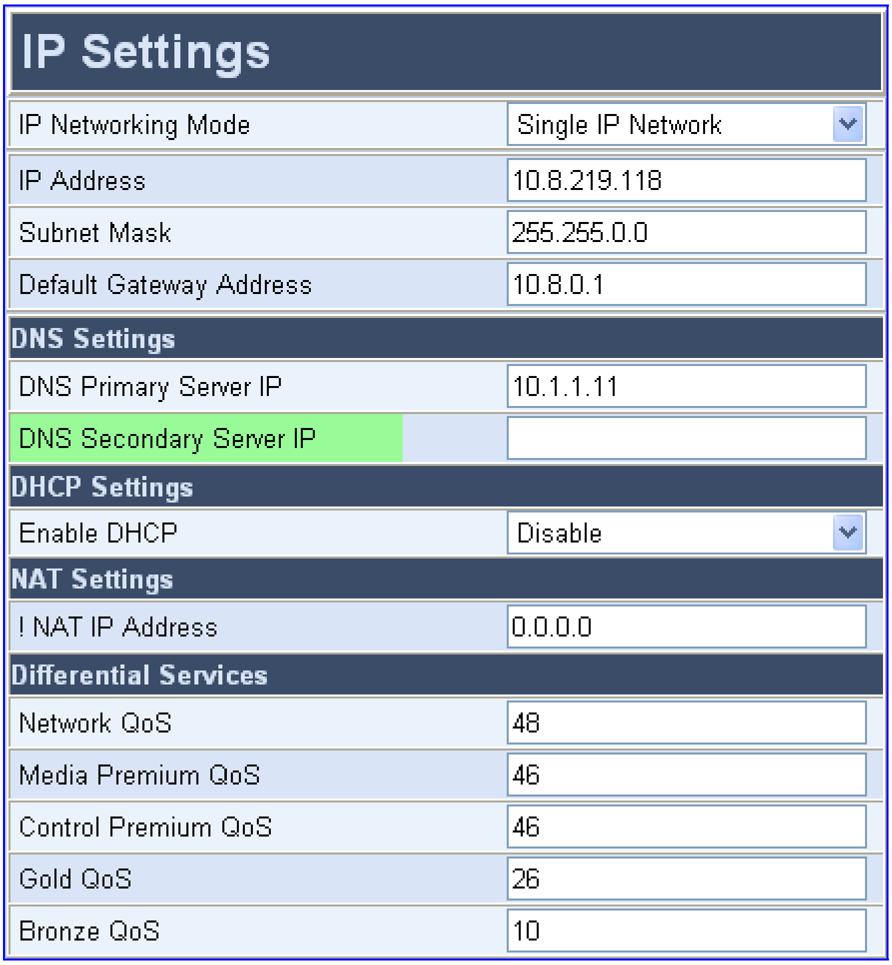
Figure 5-3: Searched Result Screen

Each searched result displays the following:

- Parameter name (hyperlinked to its location in the Embedded Web Server)
- Brief description of the parameter
- Hyperlink in green displaying the URL path to its location in the Embedded Web Server location

3. In the searched result list, click the required parameter to open the screen in which the parameter appears. The relevant parameter is highlighted in green in the screen for easy viewing.

Figure 5-4: Searched Parameter Highlighted in Screen



Note: If the searched parameter is not located, the "No Matches Found For This String" message is displayed.

5.5 Protocol Management

Use this menu to configure the gateway's SIP parameters and tables.

> **Note:** Those parameters contained within square brackets are the names used to configure the parameters via the *ini* file.

5.5.1 Protocol Definition Parameters

Use this submenu to configure the gateway's specific SIP protocol parameters.

5.5.1.1 General Parameters

Use this screen to configure general SIP parameters.

➢ To configure the general parameters under Protocol Definition, take these 4 steps:

1. Open the ‘General Parameters’ screen (**Protocol Management** menu > **Protocol Definition** submenu > **General Parameters** option); the ‘General Parameters’ screen is displayed.

Figure 5-5: Protocol Definition, General Parameters Screen

2. Configure the general parameters under Protocol Definition according to Table 5-3.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-3: Protocol Definition, General Parameters (continues on pages 58 to 64)

Parameter	Description
PRACK Mode **[PRACKMode]**	PRACK mechanism mode for 1xx reliable responses: Disable **[0]**. Supported **[1]** (default). Required **[2]**. **Note 1:** The Supported and Required headers contain the '100rel' parameter. **Note 2:** MediaPack sends PRACK message if 180/183 response is received with '100rel' in the Supported or the Required headers.
Channel Select Mode **[ChannelSelectMode]**	Port allocation algorithm for IP to Tel calls. You can select one of the following methods: ▪ By Dest Phone Number **[0]** = Select the gateway port according to the called number (called number is defined in the 'Endpoint Phone Number' table). ▪ Cyclic Ascending **[1]** = Select the next available channel in an ascending cycle order. Always select the next higher channel number in the hunt group. When the gateway reaches the highest channel number in the hunt group, it selects the lowest channel number in the hunt group and then starts ascending again. ▪ Ascending **[2]** = Select the lowest available channel. Always start at the lowest channel number in the hunt group and if that channel is not available, select the next higher channel. ▪ Cyclic Descending **[3]** = Select the next available channel in descending cycle order. Always select the next lower channel number in the hunt group. When the gateway reaches the lowest channel number in the hunt group, it selects the highest channel number in the hunt group and then starts descending again. ▪ Descending **[4]** = Select the highest available channel. Always start at the highest channel number in the hunt group and if that channel is not available, select the next lower channel. ▪ Dest Number + Cyclic Ascending **[5]** = First select the gateway port according to the called number (called number is defined in the 'Endpoint Phone Number' table). If the called number isn't found, then select the next available channel in ascending cyclic order. Note that if the called number is found, but the port associated with this number is busy, the call is released. ▪ By Source Phone Number **[6]** = Select the gateway port according to the calling number. The default method is 'By Phone Number'.
Enable Early Media **[EnableEarlyMedia]**	Disable **[0]** = Early Media is disabled (default). Enable **[1]** = Enable Early Media. If enabled, the gateway sends 183 Session Progress response with SDP (instead of 180 Ringing), allowing the media stream to be set up prior to the answering of the call. Note that to send 183 response you must also set the parameter 'ProgressIndicator2IP' to 1. If it is equal to 0, 180 Ringing response is sent. **Note:** Generally, this parameter is set to 1.
Session-Expires Time **[SIPSessionExpires]**	Determines the timeout (in seconds) for keeping a re-INVITE message alive within a SIP session. The SIP session is refreshed each time this timer expires. The SIP method used for session-timer updates is determined according to the parameter SessionExpiresMethod. The default is 0 (not activated).
Minimum Session-Expires **[MINSE]**	Defines the time (in seconds) that is used in the Min-SE header. This header defines the minimum time that the user agent supports for session refresh. The valid range is 10 to 100000. The default value is 90.

Table 5-3: Protocol Definition, General Parameters (continues on pages 58 to 64)

Parameter	Description
Session Expires Method **[SessionExpiresMethod]**	Defines the SIP method used for session-timer updates. Re-INVITE **[0]** = Use Re-INVITE messages for session-timer updates (default). UPDATE **[1]** = Use UPDATE messages. **Note 1:** The gateway can receive session-timer refreshes using both methods. **Note 2:** The UPDATE message used for session-timer purposes is not included in the SDP body.
Asserted Identity Mode **[AssertedIdMode]**	Disable **[0]** = None (default). Adding PAsserted Identity **[1]**. Adding PPreferred Identity **[2]**. The Asserted ID mode defines the header that is used in the generated INVITE request. The header also depends on the calling Privacy: allowed or restricted. The P-asserted (or P-preferred) headers are used to present the originating party's Caller ID. The Caller ID is composed of a Calling Number and (optionally) a Calling Name. P-asserted (or P-preferred) headers are used together with the Privacy header. If Caller ID is restricted, the 'Privacy: id' is included. Otherwise for allowed Caller ID the 'Privacy: none' is used. If Caller ID is restricted (received from Tel or configured in the gateway), the From header is set to <anonymous@anonymous.invalid>.
Fax Signaling Method **[IsFaxUsed]**	Determines the SIP signaling method used to establish and convey a fax session after a fax is detected. No Fax **[0]** = No fax negotiation using SIP signaling (default). T.38 Relay **[1]** = Initiates T.38 fax relay. G.711 Transport **[2]** = Initiates fax using the coder G.711 A-law/μ-law with adaptations (refer to note 1). Fax Fallback **[3]** = Initiates T.38 fax relay. If the T.38 negotiation fails, the gateway re-initiates a fax session using the coder G.711 A-law/μ-law with adaptations (see note 1). **Note 1:** Fax adaptations: Echo Canceller = On Silence Compression = Off Echo Canceller Non-Linear Processor Mode = Off Dynamic Jitter Buffer Minimum Delay = 40 Dynamic Jitter Buffer Optimization Factor = 13 **Note 2:** If the gateway initiates a fax session using G.711 (option 2 and possibly 3), a 'gpmd' attribute is added to the SDP in the following format: For A-law: 'a=gpmd:0 vbd=yes;ecan=on'. For μ-law: 'a=gpmd:8 vbd=yes;ecan=on'. **Note 3:** When 'IsFaxUsed' is set to 1, 2 or 3 the parameter 'FaxTransportMode' is ignored. **Note 4:** When the value of IsFaxUsed is other than 1, T.38 might still be used without the control protocol's involvement. To completely disable T.38, set FaxTransportMode to a value other than 1.
Detect Fax on Answer Tone **[DetFaxOnAnswerTone]**	Initiate T.38 on Preamble **[0]** = Terminating fax gateway initiates T.38 session on receiving of HDLC preamble signal from fax (default) Initiate T.38 on CED **[1]** = Terminating fax gateway initiates T.38 session on receiving of CED answer tone from fax. **Note:** This parameters is applicable only if 'IsFaxUsed = 1'.
SIP Transport Type **[SIPTransportType]**	Determines the *default* transport layer used for outgoing SIP calls initiated by the gateway. UDP **[0]** (default). TCP **[1]**. TLS **[2]** (SIPS). **Note:** It is recommended to use TLS to communicate with a SIP Proxy and not for direct gateway-gateway communication.

Table 5-3: Protocol Definition, General Parameters (continues on pages 58 to 64)

Parameter	Description
SIP UDP Local Port **[LocalSIPPort]**	Local UDP port used to receive SIP messages. The valid range is 1 to 65534. The default value is 5060.
SIP TCP Local Port **[TCPLocalSIPPort]**	Local TCP port used to receive SIP messages. The default value is 5060.
SIP TLS Local Port **[TLSLocalSIPPort]**	Local TLS port used to receive SIP messages. The default value is 5061. **Note:** The value of 'TLSLocalSIPPort' must be different to the value of 'TCPLocalSIPPort'.
Enable SIPS **[EnableSIPS]**	Enables secured SIP (SIPS) connections over multiple hops. Disable **[0]** (default). Enable **[1]**. When SIPTransportType = 2 (TLS) and EnableSIPS is disabled, TLS is used for the next network hop only. When SIPTransportType = 2 (TLS) or 1 (TCP) and EnableSIPS is enabled, TLS is used through the entire connection (over multiple hops). **Note:** If SIPS is enabled and SIPTransportType = UDP, the connection fails.
Enable TCP Connection Reuse **[EnableTCPConnectionReuse]**	Enables the reuse of the same TCP connection for all calls to the same destination. Valid options include: **[0]** = Use a separate TCP connection for each call (default) **[1]** = Use the same TCP connection for all calls
SIP Destination Port **[SIPDestinationPort]**	SIP destination port for sending initial SIP requests. The valid range is 1 to 65534. The default port is 5060. **Note:** SIP responses are sent to the port specified in the Via header.
Use "user=phone" in SIP URL **[IsUserPhone]**	No **[0]** = 'user=phone' string isn't used in SIP URI. Yes **[1]** = 'user=phone' string is part of the SIP URI (default).
Use "user=phone" in From Header **[IsUserPhoneInFrom]**	No **[0]** = Doesn't use ';user=phone' string in From header (default). Yes **[1]** = ';user=phone' string is part of the From header.
Use Tel URI for Asserted Identity **[UseTelURIForAssertedID]**	Determines the format of the URI in the P-Asserted and P-Preferred headers. 0 = 'sip:' (default). 1 = 'tel:'.
Tel to IP No Answer Timeout **[IPAlertTimeout]**	Defines the time (in seconds) the gateway waits for a 200 OK response from the called party (IP side) after sending an INVITE message. If the timer expires, the call is released. The valid range is 0 to 3600. The default value is 180.
Enable Remote Party ID **[EnableRPIheader]**	Enable Remote-Party-ID (RPI) headers for calling and called numbers for Tel→IP calls. Disable **[0]** (default). Enable **[1]** = RPI headers are generated in SIP INVITE messages for both called and calling numbers.

Table 5-3: Protocol Definition, General Parameters (continues on pages 58 to 64)

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>Enable History-Info Header
[EnableHistoryInfo]</td><td>Enables usage of the History-Info header.
Valid options include:
[0] = Disable (default)
[1] = Enable

UAC Behavior:
▪ Initial request: The History-Info header is equal to the Request URI. If a PSTN Redirect number is received, it is added as an additional History-Info header with an appropriate reason.
▪ Upon receiving the final failure response, the gateway copies the History-Info as is, adds the reason of the failure response to the last entry, and concatenates a new destination to it (if an additional request is sent).
The order of the reasons is as follows:
1. Q.850 Reason
2. SIP Reason
3. SIP Response code
▪ Upon receiving the final (success or failure) response, the gateway searches for a Redirect reason in the History-Info (i.e., 3xx/4xx SIP Reason). If found, it is passed to ISDN, according to the following table:
<table>
<tr><th>SIP Reason Code</th><th>ISDN Redirecting Reason</th></tr>
<tr><td>302 – Moved Temporarily</td><td>Call Forward Universal (CFU)</td></tr>
<tr><td>408 – Request Timeout</td><td rowspan="3">Call Forward No Answer (CFNA)</td></tr>
<tr><td>480 – Temporarily Unavailable</td></tr>
<tr><td>487 – Request Terminated</td></tr>
<tr><td>486 – Busy Here</td><td rowspan="2">Call Forward Busy (CFB)</td></tr>
<tr><td>600 – Busy Everywhere</td></tr>
</table>
▪ If history reason is a Q.850 reason, it is translated to the SIP reason (according to the SIP-ISDN tables) and then to ISDN Redirect reason according to the table above.
UAS Behavior:
▪ History-Info is sent in the final response only.
Upon receiving a request with History-Info, the UAS checks the policy in the request. If 'session', 'header', or 'history' policy tag is found, the (final) response is sent without History-Info. Otherwise, it is copied from the request.</td></tr>
<tr><td>Use Source Number as Display Name
[UseSourceNumberAsDisplayName]</td><td>Applicable to Tel→IP calls.
No **[0]** = The Tel Source Number is used as the IP Source Number and the Tel Display Name is used as the IP Display Name (if Tel Display Name is received). If no Display Name is received from the Tel side, the IP Display Name remains empty (default).
Yes **[1]** = If a Tel Display Name is received, the Tel Source Number is used as the IP Source Number and the Tel Display Name is used as the IP Display Name. If no Display Name is received from the Tel side, the Tel Source Number is used as the IP Source Number and also as the IP Display Name.
Overwrite **[2]** = The Tel Source Number is used as the IP Source Number and also as the IP Display Name (even if the received Tel Display Name is not empty).</td></tr>
</table>

Table 5-3: Protocol Definition, General Parameters (continues on pages 58 to 64)

Parameter	Description
Use Display Name as Source Number **[UseDisplayNameAsSourceNumber]**	Applicable to IP→Tel calls. No **[0]** = The IP Source Number is used as the Tel Source Number and the IP Display Name is used as the Tel Display Name (if IP Display Name is received). If no Display Name is received from IP, the Tel Display Name remains empty (default). Yes **[1]** = If an IP Display Name is received, it is used as the Tel Source Number and also as the Tel Display Name, the Presentation is set to Allowed (0). If no Display Name is received from IP, the IP Source Number is used as the Tel Source Number and the Presentation is set to Restricted (1). For example: When the following is received 'from: 100 <sip:200@201.202.203.204>', the outgoing Source Number and Display Name are set to '100' and the Presentation is set to Allowed (0). When the following is received 'from: <sip:100@101.102.103.104>', the outgoing Source Number is set to '100' and the Presentation is set to Restricted (1).
Play Ringback Tone to IP **[PlayRBTone2IP]**	Don't Play **[0]** = Ringback tone isn't played to the IP side of the call (default). Play **[1]** = Ringback tone is played to the IP side of the call after SIP 183 session progress response is sent (applies only to FXS gateways, in FXO gateways the Ringback tone isn't played). **Note 1:** To enable the gateway to send a 183 response, set 'EnableEarlyMedia' to 1. **Note 2:** If 'EnableDigitDelivery = 1', the gateway doesn't play a Ringback tone to IP and doesn't send a 183 response.
Play Ringback Tone to Tel **[PlayRBTone2Tel]**	Don't Play **[0]** = Ringback Tone isn't played. Always Play **[1]** = Ringback Tone is played to the Tel side of the call when 180/183 response is received. Play According to 180/183 **[2]** = Ringback Tone is played to the Tel side of the call if no SDP is received in 180/183 responses. If 180/183 with SDP message is received, the gateway cuts through the voice channel and doesn't play Ringback tone (default). Play According to PI **[3]** = N/A.

Table 5-3: Protocol Definition, General Parameters (continues on pages 58 to 64)

Parameter	Description
Enable GRUU **[EnableGRUU]**	Determines whether or not the GRUU mechanism is used. Valid options include: Disable **[0]** (default) Enable **[1]** The gateway obtains a GRUU by generating a normal REGISTER request. This request contains a Supported header field with the value “gruu”. The gateway includes a “+sip.instance” Contact header field parameter for each contact for which the GRUU is desired. This Contact parameter contains a globally unique ID that identifies the gateway instance. The global unique id is as follows: ▪ If registration is per endpoint (AuthenticationMode=0), it is the MAC address of the gateway concatenated with the phone number of the endpoint. ▪ If the registration is per gateway (AuthenticationMode=1) it is only the MAC address. ▪ When the “User Information” mechanism is used, the globally unique ID is the MAC address concatenated with the phone number of the endpoint (defined in the User-Info file). If the Registrar/Proxy supports GRUU, the REGISTER responses contain the “gruu” parameter in each Contact header field. The Registrar/Proxy provides the same GRUU for the same AOR and instance-id in case of sending REGISTER again after expiration of the registration. The gateway places the GRUU in any header field which contains a URI. It uses the GRUU in the following messages: INVITE requests, 2xx responses to INVITE, SUBSCRIBE requests, 2xx responses to SUBSCRIBE, NOTIFY requests, REFER requests, and 2xx responses to REFER. **Note:** If the GRUU contains the "opaque" URI parameter, the gateway obtains the AOR for the user by stripping the parameter. The resulting URI is the AOR. For example: AOR: sip:alice@example.com GRUU: sip:alice@example.com;opaque="kjh29x97us97d"
Use Tgrp Information **[UseSIPTgrp]**	Disable **[0]** = Tgrp parameter isn’t used (default). Send Only **[1]** = The hunt group number is added as the ‘tgrp’ parameter to the Contact header of outgoing SIP messages. If a hunt group number is not associated with the call, the ‘tgrp’ parameter isn’t included. If a ‘tgrp’ value is specified in incoming messages, it is ignored. Send and Receive **[2]** = The functionality of outgoing SIP messages is identical to the functionality described in option (1). In addition, for incoming SIP messages, if the Request-URI includes a ‘tgrp’ parameter, the gateway routes the call according to that value (if possible). If the Contact header includes a ‘tgrp’ parameter, it is copied to the corresponding outgoing messages in that dialog.
User-Agent Information **[UserAgentDisplayInfo]**	Defines the string that is used in the SIP request header ‘User-Agent’ and SIP response header ‘Server’. If not configured, the default string ‘AudioCodes product-name s/w-version’ is used (e.g., User-Agent: Audiocodes-Sip-Gateway-MP-118 FXS/v.4.80.004.008). When configured, the string ‘UserAgentDisplayInfo s/w-version’ is used (e.g., User-Agent: MyNewOEM/v.4.80.004.008). Note that the version number can't be modified. The maximum string length is 50 characters.
Subject **[SIPSubject]**	Defines the value of the Subject header in outgoing INVITE messages. If not specified, the Subject header isn’t included (default). The maximum length of the subject is limited to 50 characters.
Enable Reason Header **[EnableReasonHeader]**	Enables or disables the usage of the SIP Reason header. Valid options include: **[0]** = Disable **[1]** = Enable (default)

Table 5-3: Protocol Definition, General Parameters (continues on pages 58 to 64)

Parameter	Description
Enable Semi-Attended Transfer **[EnableSemiAttendedTransfer]**	Determines the gateway behavior when Transfer is initiated while still in Alerting state. Valid options include: Disable **[0]** = Send REFER with Replaces (default). Enable **[1]** = Send CANCEL, and after a 487 response is received, send REFER without Replaces.
3xx Behavior **[3xxBehavior]**	Determines the gateway's behavior when a 3xx response is received for an outgoing INVITE request. The gateway can either use the same call identifiers (CallID, branch, to and from tags) or change them in the new initiated INVITE. 0 (forward) = Use different call identifiers for a redirected INVITE message (default). 1 (redirect) = Use the same call identifiers.
Multiple Packetization Time Format **[MultiPtimeFormat]**	Determines whether the 'mptime' attribute is included in the outgoing SDP. Valid options include: 0 = Disable (default) 1 = Enable (includes the mptime attribute in the outgoing SDP -- PacketCable defined format) The 'mptime' attribute enables the IP gateway to define a separate Packetization period for each negotiated coder in the SDP. The 'mptime' attribute is only included if this parameter is enabled, even if the remote side includes it in the SDP offer. Upon reception, each coder receives its 'ptime' value in the following precedence: ▪ From 'mptime' attribute ▪ From 'ptime' attribute ▪ Default value
Retransmission Parameters	
SIP T1 Retransmission Timer [msec] **[SipT1Rtx]**	The time interval (in msec) between the first transmission of a SIP message and the first retransmission of the same message. The default is 500. **Note:** The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx. For example (assuming that SipT1Rtx = 500 and SipT2Rtx = 4000): The first retransmission is sent after 500 msec. The second retransmission is sent after 1000 (2*500) msec. The third retransmission is sent after 2000 (2*1000) msec. The fourth retransmission and subsequent retransmissions until SIPMaxRtx are sent after 4000 (2*2000) msec.
SIP T2 Retransmission Timer [msec] **[SipT2Rtx]**	The maximum interval (in msec) between retransmissions of SIP messages. The default is 4000. **Note:** The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx.
SIP Maximum Rtx **[SIPMaxRtx]**	Number of UDP transmissions (first transmission + retransmissions) of SIP messages. The range is 1 to 7. The default value is 7.
EnableReasonHeader [Enable Reason Header]	Enables / disables the usage of the SIP Reason header. 0 = Disable. 1 = Enable (default).

5.5.1.2 Proxy & Registration Parameters

Use this screen to configure parameters that are associated with Proxy and Registration.

➢ **To configure the Proxy & Registration parameters, take these 4 steps:**

1. Open the 'Proxy & Registration' parameters screen (**Protocol Management** menu > **Protocol Definition** submenu > **Proxy & Registration** option); the 'Proxy & Registration' parameters screen is displayed.

Figure 5-6: Proxy & Registration Parameters Screen

Proxy & Registration

Enable Proxy	Use Proxy
Proxy Name	
Proxy IP Address	10.2.1.2
First Redundant Proxy IP Address	0.0.0.0
Second Redundant Proxy IP Address	0.0.0.0
Third Redundant Proxy IP Address	0.0.0.0
Redundancy Mode	Parking
Proxy Load Balancing Method	Disable
Proxy IP List Refresh Time	60
Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Disable
Send All Invite to Proxy	No
Enable Proxy Hot-Swap	Disable
Enable Registration	Disable
Gateway Name	audiocodes.com
Gateway Registration Name	
DNS Query Type	A-Record
Proxy DNS Query Type	A-Record
Subscription Mode	Per Endpoint
Use Gateway Name for OPTIONS	No
Number of RTX Before Hot-Swap	3
User Name	
Password	••••••
Cnonce	0a123bcf
Authentication Mode	Per Endpoint

2. Configure the Proxy & Registration parameters according to Table 5-4.
3. Click the **Submit** button to save your changes, or click the **Register** or **Un-Register** buttons to save your changes and to register / unregister to a Proxy / Registrar.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-4: Proxy & Registration Parameters (continues on pages 66 to 71)

Parameter	Description
Enable Proxy **[IsProxyUsed]**	Don't Use Proxy **[0]** = Proxy isn't used, the internal routing table is used instead (default). Use Proxy **[1]** = Proxy is used. If you are using a Proxy server, enter the IP address of the primary Proxy server in the **Proxy IP address** field. If you are not using a Proxy server, you must configure the Tel to IP Routing table on the gateway (described in Section 5.5.5.2 on page 100).
Proxy Name **[ProxyName]**	Defines the Home Proxy Domain Name. If specified, the Proxy Name is used as Request-URI in REGISTER, INVITE and other SIP messages. If not specified, the Proxy IP address is used instead.
Proxy IP Address **[ProxyIP]**	IP address (and optionally port number) of the primary Proxy server you are using. Enter the IP address as FQDN or in dotted format notation (for example 201.10.8.1). You can also specify the selected port in the format: <IP Address>:<port>. This parameter is applicable only if you select 'Yes' in the 'Is Proxy Used' field. If you enable Proxy Redundancy (by setting EnableProxyKeepAlive=1 or 2), the gateway can work with up to four Proxy servers. If there is no response from the primary Proxy, the gateway tries to communicate with the redundant Proxies. When a redundant Proxy is found, the gateway either continues working with it until the next failure occurs or reverts to the primary Proxy (refer to the 'Redundancy Mode' parameter). If none of the Proxy servers respond, the gateway goes over the list again. The gateway also provides real time switching (hotswap mode), between the primary and redundant proxies ('IsProxyHotSwap=1'). If the first Proxy doesn't respond to INVITE message, the same INVITE message is immediately sent to the second Proxy. **Note 1:** If 'EnableProxyKeepAlive=1 or 2', the gateway monitors the connection with the Proxies by using keep-alive messages (OPTIONS or REGISTER). **Note 2:** To use Proxy Redundancy, you must specify one or more redundant Proxies using multiple 'ProxyIP= <IP address>' definitions. **Note 3:** When port number is specified (e.g., domain.com:5080), DNS SRV queries aren't performed, even if 'ProxyDNSQueryType' is set to 1.
Gateway Name **[SIPGatewayName]**	Use this parameter to assign a name to the gateway (e.g., 'gateway1.com'). Ensure that the name you choose is the one that the Proxy is configured with to identify your media gateway. **Note:** If specified, the gateway Name is used as the host part of the SIP URI in the From header. If not specified, the gateway IP address is used instead (default).
Gateway Registration Name **[GWRegistrationName]**	Defines the user name that is used in From and To headers of REGISTER messages. Applicable only to single registration per gateway ('AuthenticationMode = 1). If 'GWRegistrationName' isn't specified (default), the 'Username' parameter is used instead. **Note:** If 'AuthenticationMode=0', all the gateway's endpoints are registered with a user name that equals to the endpoint's phone number.

Table 5-4: Proxy & Registration Parameters (continues on pages 66 to 71)

Parameter	Description
First Redundant Proxy IP Address **[ProxyIP]**	IP addresses of the first redundant Proxy you are using. Enter the IP address as FQDN or in dotted format notation (for example 192.10.1.255). You can also specify the selected port in the format: <IP Address>:<port>. **Note 1:** This parameter is available only if you select 'Use Proxy' in the 'Enable Proxy' field. **Note 2:** When port number is specified, DNS NAPTR/SRV queries aren't performed, even if 'ProxyDNSQueryType' is set to 1. ***ini* file note:** The IP address of the first redundant Proxy is defined by the second repetition of the *ini* file parameter 'ProxyIP'.
Second Redundant Proxy IP Address **[ProxyIP]**	IP addresses of the second redundant Proxy you are using. Enter the IP address as FQDN or in dotted format notation (for example 192.10.1.255). You can also specify the selected port in the format: <IP Address>:<port>. **Note 1:** This parameter is available only if you select 'Use Proxy' in the 'Enable Proxy' field. **Note 2:** When port number is specified, DNS NAPTR/SRV queries aren't performed, even if 'ProxyDNSQueryType' is set to 1. ***ini* file note:** The IP address of the second redundant Proxy is defined by the third repetition of the *ini* file parameter 'ProxyIP'.
Third Redundant Proxy IP Address **[ProxyIP]**	IP addresses of the third redundant Proxy you are using. Enter the IP address as FQDN or in dotted format notation (for example 192.10.1.255). You can also specify the selected port in the format: <IP Address>:<port>. **Note 1:** This parameter is available only if you select 'Use Proxy' in the 'Enable Proxy' field. **Note 2:** When port number is specified, DNS NAPTR/SRV queries aren't performed, even if 'ProxyDNSQueryType' is set to 1. ***ini* file note:** The IP addresses of the third redundant Proxy is defined by the fourth repetition of the *ini* file parameter 'ProxyIP'.

Table 5-4: Proxy & Registration Parameters (continues on pages 66 to 71)

Parameter	Description
Proxy Load Balancing Method **[ProxyLoadBalancingMethod]**	Enables the usage of the Proxy Load Balancing mechanism. Valid options include: Disable **[0]** = Load Balancing is disabled (default). Round Robin **[1]** = Round Robin algorithm. Random Weights **[2]** = Random Weights. When Round Robin (1) algorithm is used, a list of all possible Proxy IP addresses is compiled. This list includes all entries in the ProxyIP table after necessary DNS resolutions (including NAPTR and SRV, if configured). This list can handle up to 15 entries. After this list is compiled, the Proxy Keep-Alive mechanism (according to EnableProxyKeepAlive and ProxyKeepAliveTime) is used to mark each entry as Offline or Online. The balancing is only performed on Proxy servers that are marked as Online. All outgoing messages are equally distributed across the Proxy IP list. REGISTER messages are also distributed unless a RegistrarIP is configured. The Proxy IP list is refreshed according to ProxyIPListRefreshTime. If a change in the order of the entries in the list occurs, all load statistics are erased and balancing starts over again. When Random Weights (2) algorithm is used, the outgoing requests are not distributed equally among the Proxies. The weights are received from the DNS server by using SRV records. The gateway sends the requests in such a fashion that each Proxy receives a percentage of the requests according to its assigned weight. Load Balancing is not used in the following scenarios: ▪ The ProxyIP table includes more than one entry. ▪ The only Proxy defined is an IP address and not an FQDN. ▪ SRV usage is not enabled (DNSQueryType). ▪ The SRV response includes several records with a different Priority value.
Proxy IP List Refresh Time **[ProxyIPListRefreshTime]**	Defines the time interval (in seconds) between refreshes of the Proxy IP list. This parameter is used only when ProxyLoadBalancingMethod = 1. The interval range is 5 to 2,000,000. The default interval is 60.
DNS Query Type **[DNSQueryType]**	Enables the use of DNS Naming Authority Pointer (NAPTR) and Service Record (SRV) queries to resolve Proxy and Registrar servers and to resolve all domain names that appear in the Contact and Record-Route headers. Valid options include: ▪ 0 = A-Record (default) ▪ 1 = SRV ▪ 2 = NAPTR If set to A-Record, no NAPTR or SRV queries are performed. If set to SRV and the Proxy / Registrar IP address parameter or the domain name in the Contact / Record-Route headers contains a domain name without port definition, an SRV query is performed. The gateway uses the first host name received from the SRV query. The gateway then performs DNS A-record query for the host name to locate an IP address. If set to NAPTR, an NAPTR query is performed. If it is successful, an SRV query is sent according to the information received in the NAPTR response. If the NAPTR query fails, an SRV query is performed according to the configured transport type. If the Proxy / Registrar IP address parameter or the domain name in the Contact / Record-Route headers contains a domain name with port definition, the gateway performs a regular DNS A-record query. **Note:** To enable NAPTR/SRV queries only for Proxy servers, use the parameter ProxyDNSQueryType.
Enable SRV Queries **[EnableSRVQuery]**	This parameter is obsolete. Please use the parameter DNSQueryType.

Table 5-4: Proxy & Registration Parameters (continues on pages 66 to 71)

Parameter	Description
Proxy DNS Query Type **[ProxyDNSQueryType]**	Enables the use of DNS Naming Authority Pointer (NAPTR) and Service Record (SRV) queries to discover Proxy servers. Valid options include: **[0]** = A-Record (default) **[1]** = SRV **[2]** = NAPTR If set to A-Record, no NAPTR or SRV queries are performed. If set to SRV and the Proxy IP address parameter contains a domain name without port definition (e.g., ProxyIP = domain.com), an SRV query is performed. The SRV query returns up to four Proxy host names and their weights. The gateway then performs DNS A-record queries for each Proxy host name (according to the received weights) to locate up to four Proxy IP addresses. Therefore, if the first SRV query returns two domain names, and the A-record queries return two IP addresses each, no more searches are performed. If set to NAPTR, an NAPTR query is performed. If it is successful, an SRV query is sent according to the information received in the NAPTR response. If the NAPTR query fails, an SRV query is performed according to the configured transport type. If the Proxy IP address parameter contains a domain name with port definition (e.g., ProxyIP = domain.com:5080), the gateway performs a regular DNS A-record query. **Note:** When enabled, NAPTR/SRV queries are used to discover Proxy servers even if the parameter DNSQueryType is disabled.
Enable Proxy SRV Queries **[EnableProxySRVQuery]**	This parameter is now obsolete. Please use the parameter ProxyDNSQueryType.
Redundancy Mode **[ProxyRedundancyMode]**	Parking **[0]** = Gateway continues working with the last active Proxy until the next failure (default). Homing **[1]** = Gateway always tries to work with the primary Proxy server (switches back to the main Proxy whenever it is available). **Note:** To use ProxyRedundancyMode, enable Keep-alive with Proxy option (EnableProxyKeepAlive=1 or 2).
Is Proxy Trusted **[IsTrustedProxy]**	This parameter isn't applicable and must always be set to 'Yes' [1]. The parameter 'AssertedIdMode' should be used instead.
Enable Registration **[IsRegisterNeeded]**	No **[0]** = Gateway doesn't register to Proxy / Registrar (default). Yes **[1]** = Gateway registers to Proxy / Registrar when the device is powered up and every RegistrationTime seconds. **Note:** The gateway sends a REGISTER request for each channel or for the entire gateway (according to the AuthenticationMode parameter).
Registrar Name **[RegistrarName]**	Registrar Domain Name. If specified, the name is used as Request-URI in REGISTER messages. If isn't specified (default), the Registrar IP address or Proxy name or Proxy IP address is used instead.
Registrar IP Address **[RegistrarIP]**	IP address and optionally port number of Registrar server. Enter the IP address in dotted format notation, for example 201.10.8.1:<5080>. **Note 1:** If not specified, the REGISTER request is sent to the primary Proxy server (refer to 'Proxy IP address' parameter). **Note 2**: When port number is specified, DNS NAPTR/SRV queries aren't performed, even if 'DNSQueryType' is set to 1.
Registration Time **[RegistrationTime]**	Defines the time (in seconds) for which registration to a Proxy server is valid. The value is used in the header 'Expires'. In addition, this parameter defines the time interval between Keep-Alive messages when EnableProxyKeepAlive = 2 (REGISTER). Typically, a value of 3600 should be assigned for one hour registration. The gateway resumes registration according to the parameter RegistrationTimeDivider. The default value is 180. The valid range is 10 to 2000000.

Table 5-4: Proxy & Registration Parameters (continues on pages 66 to 71)

Parameter	Description
Re-registration Timing (%) **[RegistrationTimeDivider]**	Defines the re-registration timing (in percentage). The timing is a percentage of the re-register timing set by the Registration server. The valid range is 50 to 100. The default value is 50. For example: If 'RegistrationTimeDivider = 70' (%) and Registration Expires time = 3600, the gateway resends its registration request after 3600 x 70% = 2520 sec.
Registration Retry Time **[RegistrationRetryTime]**	Defines the time period (in seconds) after which a Registration request is resent if registration fails with 4xx, or there is no response from the Proxy/Registrar. The default is 30 seconds. The range is 10 to 3600.
Subscription Mode **[SubscriptionMode]**	Determines the method the gateway uses to subscribe to an MWI server. Per Endpoint **[0]** = Each endpoint subscribes separately. This method is usually used for FXS gateways (default). Per Gateway **[1]** = Single subscription for the entire gateway. This method is usually used for FXO gateways.
Enable Proxy Keep Alive **[EnableProxyKeepAlive]**	Disable **[0]** = Disable (default). Using OPTIONS **[1]** = Enable Keep alive with Proxy using OPTIONS. Using REGISTER **[2]** = Enable Keep alive with Proxy using REGISTER. If EnableProxyKeepAlive = 1, SIP OPTIONS message is sent every ProxyKeepAliveTime. If EnableProxyKeepAlive = 2, SIP REGISTER message is sent every RegistrationTime. Any response from the Proxy, either success (200 OK) or failure (4xx response) is considered as if the Proxy is correctly communicating. **Note 1:** This parameter must be set to 1 (OPTIONS) when Proxy redundancy is used. **Note 2:** When EnableProxyKeepAlive = 2 (REGISTER), the homing redundancy mode is disabled. **Note 3:** When the active proxy does not respond to INVITE messages sent by the gateway, the proxy is marked as offline. The behavior is similar to a Keep-Alive (OPTIONS or REGISTER) failure.
Proxy Keep Alive Time **[ProxyKeepAliveTime]**	Defines the Proxy keep-alive time interval (in seconds) between Keep-Alive messages. The default value is 60 seconds. **Note:** This parameter is applicable only if EnableProxyKeepAlive = 1 (OPTIONS). When EnableProxyKeepAlive = 2 (REGISTER), the time interval between Keep-Alive messages is determined by the parameter RegistrationTime.
Use Gateway Name for OPTIONS **[UseGatewayNameForOptions]**	No **[0]** = Use the gateway's IP address in keep-alive OPTIONS messages (default). Yes **[1]** = Use 'GatewayName' in keep-alive OPTIONS messages. The OPTIONS Request-URI host part contains either the gateway's IP address or a string defined by the parameter 'Gatewayname'. The gateway uses the OPTIONS request as a keep-alive message to its primary and redundant Proxies (EnableProxyKeepAlive = 1).
Enable Fallback to Routing Table **[IsFallbackUsed]**	No **[0]** = Gateway fallback is not used (default). Yes **[1]** = Internal Tel to IP Routing table is used when Proxy servers are not available. When the gateway falls back to the internal Tel to IP Routing table, the gateway continues scanning for a Proxy. When the gateway finds an active Proxy, it switches from internal routing back to Proxy routing. **Note:** To enable the redundant Proxies mechanism set 'EnableProxyKeepAlive' to 1 or 2.
Prefer Routing Table **[PreferRouteTable]**	Determines if the local Tel to IP routing table takes precedence over a Proxy for routing calls. No **[0]** = Only Proxy is used to route calls (default). Yes **[1]** = The gateway checks the 'Dest Phone Prefix' and/or 'Source Phone Prefix' field in the 'Tel to IP Routing' table for a match with the outgoing call. Only if a match is not found, a Proxy is used. **Note:** Applicable only if Proxy is not always used ('AlwaysSendToProxy' = 0, 'SendInviteToProxy' = 0).

Table 5-4: Proxy & Registration Parameters (continues on pages 66 to 71)

Parameter	Description
Use Routing Table for Host Names and Profiles **[AlwaysUseRouteTable]**	Use the internal Tel to IP routing table to obtain the URI Host name and (optionally) an IP profile (per call), even if Proxy server is used. No **[0]** = Don't use (default). Yes **[1]** = Use. **Note:** This domain name is used, instead of Proxy name or Proxy IP address, in the INVITE SIP URI.
Always Use Proxy **[AlwaysSendToProxy]**	No **[0]** = Use standard SIP routing rules (default). Yes **[1]** = All SIP messages and Responses are sent to Proxy server. **Note:** Applicable only if Proxy server is used.
Send All INVITE to Proxy **[SendInviteToProxy]**	No **[0]** = INVITE messages, generated as a result of Transfer or Redirect, are sent directly to the URI (according to the Refer-To header in the REFER message or Contact header in 30x response) (default). Yes **[1]** = All INVITE messages, including those generated as a result of Transfer or Redirect are sent to Proxy. **Note:** Applicable only if Proxy server is used and 'AlwaysSendtoProxy=0'.
Enable Proxy Hot-Swap **[IsProxyHotSwap]**	Enable Proxy Hot-Swap redundancy mode. No **[0]** = Disabled (default). Yes **[1]** = Enabled. If Hot Swap is enabled, SIP INVITE message is first sent to the primary Proxy server. If there is no response from the primary Proxy server for 'Number of RTX before Hot-Swap' retransmissions, the INVITE message is resent to the redundant Proxy server.
Number of RTX Before Hot-Swap **[ProxyHotSwapRtx]**	Number of retransmitted INVITE messages before call is routed (hot swapped) to another Proxy. The valid range is 1 to 30. The default value is 3. **Note:** This parameter is also used for alternative routing using the Tel to IP Routing table. If a domain name in the routing table is resolved into 2 IP addresses, and if there is no response for 'ProxyHotSwapRtx' retransmissions to the INVITE message that is sent to the first IP address, the gateway immediately initiates a call to the second IP address.
User Name **[UserName]** **Note:** The Authentication table can be used instead.	Username used for Registration and for Basic/Digest authentication process with Proxy / Registrar. Parameter doesn't have a default value (empty string). **Note:** Applicable only if single gateway registration is used ('Authentication Mode = Authentication Per gateway').
Password **[Password]**	Password used for Basic/Digest authentication process with Proxy / Registrar. Single password is used for all gateway ports. The default is 'Default_Passwd'. **Note:** The Authentication table can be used instead.
Cnonce **[Cnonce]**	String used by the server and client to provide mutual authentication. (Free format i.e., 'Cnonce = 0a4f113b'). The default is 'Default_Cnonce'.
Authentication Mode **[AuthenticationMode]**	Per Endpoint **[0]** = Registration & Authentication separately for each endpoint (default). Per gateway **[1]** = Single Registration & Authentication for the gateway. Per Ch. Select Mode **[2]** = N/A. Usually Authentication on a per endpoint basis is used for FXS gateways, in which each endpoint registers (and authenticates) separately with its own username and password. Single Registration and Authentication (Authentication Mode=1) is usually defined for FXO gateways.

5.5.1.3 Coders

From the Coders screen you can configure the first to fifth preferred coders (and their attributes) for the gateway. The first coder is the highest priority coder and is used by the gateway whenever possible. If the far end gateway cannot use the coder assigned as the first coder, the gateway attempts to use the next coder and so forth.

➢ To configure the gateway's coders, take these 9 steps:

1. Open the 'Coders' screen (**Protocol Management** menu > **Protocol Definition** submenu > **Coders** option); the 'Coders' screen is displayed.

Figure 5-7: Coders Screen

Coders

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.729	20	8	18	Disabled
G.723.1	30	5.3	4	Disabled
G.711A-law	20	64	8	Disabled

2. From the 'Coder Name' drop-down list, select the coder you want to use. For the full list of available coders and their corresponding attributes, refer to Table 5-5.
3. From the 'Packetization Time' drop-down list, select the packetization time (in msec) for the coder you selected. The packetization time determines how many coder payloads are combined into a single RTP packet.
4. From the 'Rate' drop-down list, select the bit rate (in kbps) for the coder you selected.
5. In the 'Payload Type' field, if the payload type for the coder is dynamic, enter a value from 0 to 120 (payload types of 'well-known' coders cannot be modified). The payload type identifies the format of the RTP payload.
6. From the 'Silence Suppression' drop-down list, enable or disable the silence suppression option for the coder you selected.
7. Repeat steps 2 through 6 for the second to fifth coders (optional).
8. Click the **Submit** button to save your changes.
9. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Notes:

- Each coder can appear only once.
- If not specified, the ptime gets a default value. The ptime specifies packetization time the gateway expects to receive. The gateway always uses the ptime requested by the remote side for sending RTP packets. Only the ptime of the first coder in the list is declared in INVITE / 200 OK SDP, even if multiple coders are defined.
- If payload type is not specified, a default is used.
- For G.729 it's also possible to select silence suppression without adaptations
- If coder G.729 is selected and silence suppression disabled, the gateway includes the string 'annexb=no' in the SDP of the relevant SIP messages. If silence suppression is enabled or set to 'Enable w/o Adaptations', 'annexb=yes' is included. An exception to this logic is when the remote gateway is a Cisco device (IsCiscoSCEMode).

Table 5-5: Supported Coders and their Attributes

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711 A-law [g711Alaw64k]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Always 8	Disable **[0]** Enable **[1]**
G.711 μ-law [g711Ulaw64k]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Always 0	Disable **[0]** Enable **[1]**
G.729 [g729]	10, 20 (default), 30, 40, 50, 60	Always 8	Always 18	Disable **[0]** Enable **[1]** Enable w/o Adaptations **[2]**
G.723.1 [g7231]	30 (default), 60, 90	5.3 **[0]**, 6.3 **[1]** (default)	Always 4	Disable **[0]** Enable **[1]**
G.726 [g726]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	16 [0], 24 [1], 32 [2] (default), 40 [3]	Dynamic (0-120)	Disable **[0]** Enable **[1]**
T.38 [t38fax]	N/A	N/A	N/A	N/A
G.711A-law_VBD [g711AlawVbd]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Dynamic (0-120)	N/A
G.711U-law_VBD [g711UlawVbd]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Dynamic (0-120)	N/A
Transparent	N/A.			

Table 5-6: *ini* File Coder Parameter

Parameter	Description
CoderName	Defines the gateway's coder list (up to five coders can be configured). Enter coders in the following format: CoderName=<Coder Name>,<Ptime>,<Rate>,<Payload Type>,<Silence Suppression Mode>. **Note 1:** The coder name is case-sensitive. **Note 2:** If silence suppression is not defined (for a specific coder), the value defined by the parameter EnableSilenceCompression is used. **Note 3:** The value of several fields is hard-coded according to well-known standards (e.g., the payload type of G.711 U-law is always 0). Other values can be set dynamically. If no value is specified for a dynamic field, a default value is assigned. If a value is specified for a hard-coded field, the value is ignored. For example: CoderName = g711Alaw64k,20,,,0 CoderName = g711Ulaw64k,40 CoderName = g7231,90,1,,1 CoderName = g726,$$,$$,,0

Notes: For an explanation on V.152 support (and implementation of T.38 and VBD coders), refer to Section 8.3.5 on page 222.

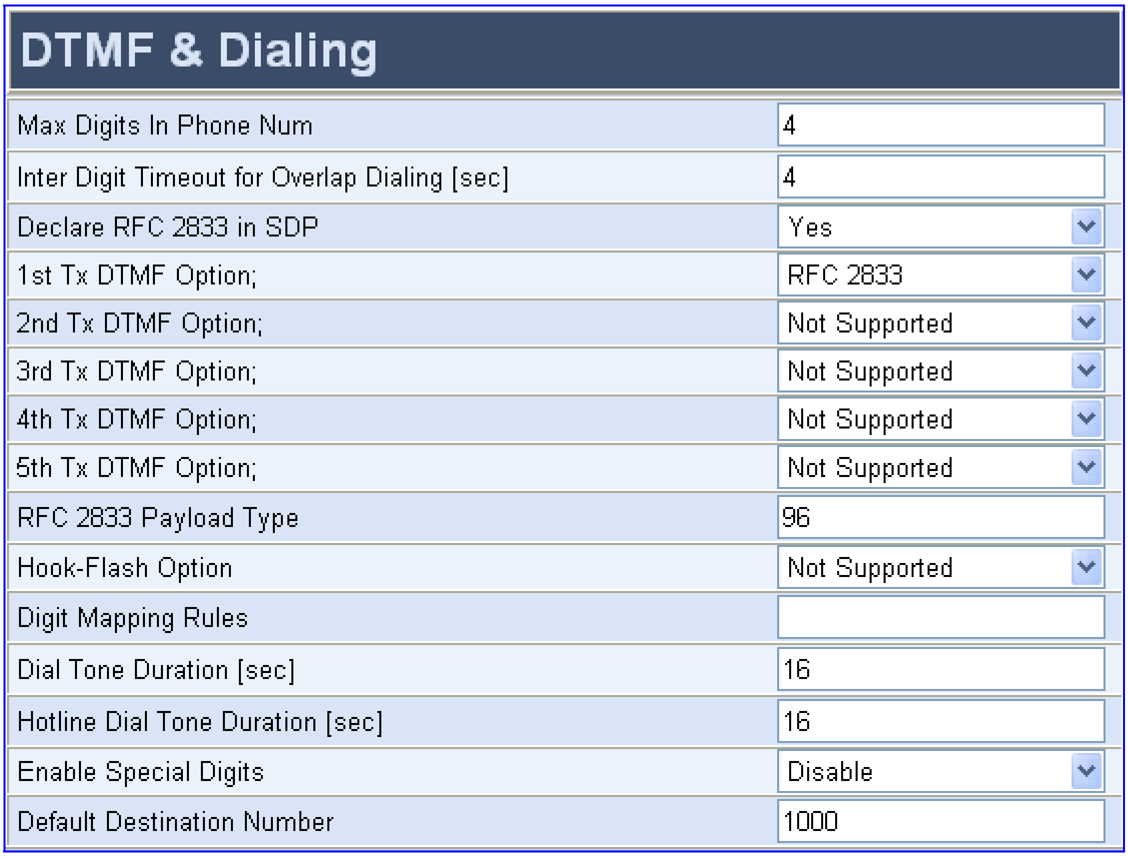
5.5.1.4 DTMF & Dialing Parameters

Use this screen to configure parameters that are associated with DTMF and dialing.

➢ **To configure the dialing parameters, take these 4 steps:**

1. Open the 'DTMF & Dialing' screen (**Protocol Management** menu > **Protocol Definition** submenu > **DTMF & Dialing** option); the 'DTMF & Dialing' screen is displayed.

Figure 5-8: DTMF & Dialing Screen



2. Configure the DTMF & Dialing parameters according to Table 5-7.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-7: DTMF & Dialing Parameters (continues on pages 74 to 76)

Parameter	Description
Max Digits in Phone Num **[MaxDigits]** **Note:** Digit Mapping Rules can be used instead.	Maximum number of digits that can be dialed. The valid range is 1 to 49. The default value is 5. **Note:** Dialing ends when the maximum number of digits is dialed, the Interdigit Timeout expires, the '#' key is dialed, or a digit map pattern is matched.
Inter Digits Timeout for Overlap Dialing [sec] **[TimeBetweenDigits]**	Time in seconds that the gateway waits between digits dialed by the user. When the Interdigit Timeout expires, the gateway attempts to dial the digits already received. The valid range is 1 to 10 seconds. The default value is 4 seconds.

Table 5-7: DTMF & Dialing Parameters (continues on pages 74 to 76)

Parameter	Description
Declare RFC 2833 in SDP **[RxDTMFOption]**	Defines the supported Receive DTMF negotiation method. No **[0]** = Don't declare RFC 2833 telephony-event parameter in SDP Yes **[3]** = Declare RFC 2833 telephony-event parameter in SDP (default) The MediaPack is designed to always be receptive to RFC 2833 DTMF relay packets. Therefore, it is always correct to include the 'telephony-event' parameter as a default in the SDP. However some gateways use the absence of the 'telephony-event' from the SDP to decide to send DTMF digits in-band using G.711 coder, if this is the case you can set 'RxDTMFOption=0'.
1st to 5th Tx DTMF Option **[TxDTMFOption]**	Determines a single or several preferred transmit DTMF negotiation methods. 0 (Not Supported) = No negotiation, DTMF digits are sent according to the parameters 'DTMFTransportType' and 'RFC2833PayloadType' (default). 1 (INFO Nortel) = Sends DTMF digits according to IETF <draft-choudhuri-sip-info-digit-00>. 2 (NOTIFY) = Sends DTMF digits according to <draft-mahy-sipping-signaled-digits-01>. 3 (INFO Cisco) = Sends DTMF digits according to Cisco format. 4 (RFC 2833). **Note 1:** DTMF negotiation methods are prioritized according to the order of their appearance. **Note 2:** When out-of-band DTMF transfer is used (1, 2 or 3), the parameter 'DTMFTransportType' is automatically set to 0 (DTMF digits are erased from the RTP stream). **Note 3:** When RFC 2833 (4) is selected, the gateway: ▪ Negotiates RFC 2833 Payload Type (PT) using local and remote SDPs. ▪ Sends DTMF packets using RFC 2833 PT according to the PT in the received SDP. ▪ Expects to receive RFC 2833 packets with the same PT as configured by the parameter 'RFC2833PayloadType'. ▪ Uses the same PT for send and receive if the remote party doesn't include the RFC 2833 DTMF PT in its SDP. **Note 4:** When TxDTMFOption is set to 0, the RFC 2833 PT is set according to the parameter 'RFC2833PayloadType' for both transmit and receive. ***ini* file note:** The DTMF transmit methods are defined using a repetition of the same (TxDTMFOption) parameter (up to five options can be provided).
RFC 2833 Payload Type **[RFC2833PayloadType]**	The RFC 2833 DTMF relay dynamic payload type. Range: 96 to 99, 106 to 127; Default = 96 The 100, 102 to 105 range is allocated for proprietary usage. **Note 1:** Cisco is using payload type 101 for RFC 2833. **Note 2:** When RFC 2833 payload type (PT) negotiation is used (TxDTMFOption=4), this payload type is used for the received DTMF packets. If negotiation isn't used, this payload type is used for receive and for transmit.
Hook-flash Option **[HookFlashOption]**	Supported hook-flash Transport Type (method by which hook-flash is sent and received). Valid options include: Not Supported **[0]** = Hook-Flash indication isn't sent (default). INFO **[1]** = Send proprietary INFO message with Hook-Flash indication. RFC 2833 **[4]** = RFC 2833. **Note:** FXO gateways support the receiving of RFC 2833 Hook-Flash signals.
[IsHookFlashUsed]	This parameter is obsolete; use instead the parameter HookFlashOption

Table 5-7: DTMF & Dialing Parameters (continues on pages 74 to 76)

Parameter	Description
Digit Mapping Rules **[DigitMapping]**	Digit map pattern. If the digit string (dialed number) has matched one of the patterns in the digit map, the gateway stops collecting digits and starts to establish a call with the collected number The digit map pattern contains up to 52 options separated by a vertical bar (|) and enclosed in parenthesis. The maximum length of the entire digit pattern is limited to 152 characters. Available notations: ▪ [n-m] represents a range of numbers ▪ '.' (single dot) represents repetition ▪ 'x' represents any single digit ▪ 'T' represents a dial timer (configured by TimeBetweenDigits parameter) ▪ 'S' should be used when a specific rule, that is part of a general rule, is to be applied immediately. For example, if you enter the general rule x.T and the specific rule 11x, you should append 'S' to the specific rule 11xS. For example: (11xS|00T|[1-7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)
Dial Tone Duration [sec] **[TimeForDialTone]**	Time in seconds that the dial tone is played. The default time is 16 seconds. FXS gateway ports play the dial tone after phone is picked up; while FXO gateway ports play the dial tone after port is seized in response to ringing. **Note 1:** During play of dial tone, the gateway waits for DTMF digits. **Note 2:** 'TimeForDialTone' is not applicable when Automatic Dialing is enabled.
Hotline Dial Tone Duration **[HotLineToneDuration]**	Duration (in seconds) of the Hotline dial tone. If no digits are received during the Hotline dial tone duration, the gateway initiates a call to a preconfigured number (set in the automatic dialing table). The valid range is 0 to 60. The default time is 16 seconds. Applicable to FXS and FXO gateways.
Enable Special Digits **[IsSpecialDigits]**	Disable **[0]** = '*' or '#' terminate number collection (default). Enable **[1]** = if you want to allow '*' and '#' to be used for telephone numbers dialed by a user or entered for the endpoint telephone number. **Note:** The # and * can always be used as first digit of a dialed number, even if you select 'Disable' for this parameter.
Default Destination Number **[DefaultNumber]**	Defines the telephone number that the gateway uses if the parameter 'TrunkGroup_x' doesn't include a phone number. The parameter is used as a starting number for the list of channels comprising all hunt groups in the gateway.

5.5.2 Configuring the Advanced Parameters

Use this submenu to configure the gateway's advanced control protocol parameters.

5.5.2.1 General Parameters

Use this screen to configure general control protocol parameters.

➢ To configure the general parameters under Advanced Parameters, take these 4 steps:

1. Open the 'General Parameters' screen (**Protocol Management** menu > **Advanced Parameters** submenu > **General Parameters** option); the 'General Parameters' screen is displayed.

Figure 5-9: Advanced Parameters, General Parameters Screen

General Parameters	
IP Security	Disable
Filter Calls to IP	Don't Filter
! Enable Digit Delivery to Tel	Disable
! Enable Digit Delivery to IP	Disable
RTP Only Mode	Disable
Enable DID Wink	Disable
Delay Before DID Wink	0
Reanswer Time	0
Disconnect and Answer Supervision	
Enable Polarity Reversal	Disable
Enable Current Disconnect	Disable
Disconnect on Broken Connection	Yes
Broken Connection Timeout [100 msec]	100
Disconnect Call on Silence Detection	No
Silence Detection Period [sec]	120
Silence Detection Method	Voice/Energy Detectors
Send Digit Pattern on Connect	
CDR and Debug	
CDR Server IP Address	
CDR Report Level	None
Debug Level	5
Misc. Parameters	
Progress Indicator to IP	Not Configured
Enable X-Channel Header	Disable
Enable Busy Out	Disable
Default Release Cause	3
Delay After Reset [sec]	7
Max Number of Active Calls	8
Max Call Duration [min]	0
Enable LAN Watchdog	Disable
Enable Calls Cut Through	Disable
Enable User-Information Usage	Disable
Out-Of-Service Behavior	Reorder Tone

2. Configure the general parameters under 'Advanced Parameters' according to Table 5-8.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-8: Advanced Parameters, General Parameters (continues on pages 78 to 82)

Parameter	Description
Signaling DiffServ **[ControlIPDiffServ]**	Obsolete parameter, use PremiumServiceClassControlDiffServ instead.
IP Security **[SecureCallsFromIP]**	No **[0]** = Gateway accepts all SIP calls (default). Yes **[1]** = Gateway accepts SIP calls only from IP addresses defined in the Tel to IP routing table. The gateway rejects all calls from unknown IP addresses. For detailed information on the Tel to IP Routing table, refer to Section 5.5.5.2 on page 100. **Note:** Specifying the IP address of a Proxy server in the Tel to IP Routing table enables the gateway to only accept calls originating in the Proxy server and rejects all other calls.
Filter Calls to IP **[FilterCalls2IP]**	Don't Filter **[0]** = Disabled (default) Filter **[1]** = Enabled If the filter calls to IP feature is enabled, then when a Proxy is used, the gateway first checks the Tel→IP routing table before making a call through the Proxy. If the number is not allowed (number isn't listed or a Call Restriction routing rule, IP=0.0.0.0, is applied), the call is released.
Enable Digit Delivery to IP **[EnableDigitDelivery2IP]**	Disable **[0]** = Disabled (default). Enable **[1]** = Enable digit delivery to IP. The digit delivery feature enables sending of DTMF digits to the destination IP address after the Tel→IP call was answered. To enable this feature, modify the called number to include at least one 'p' character. The gateway uses the digits before the 'p' character in the initial INVITE message. After the call was answered the gateway waits for the required time (# of 'p' * 1.5 seconds) and then sends the rest of the DTMF digits using the method chosen (in-band, out-of-band). **Note:** The called number can include several 'p' characters (1.5 seconds pause). For example, the called number can be as follows: 1001pp699, 8888p9p300.
Enable Digit Delivery to Tel **[EnableDigitDelivery]**	Disable **[0]** = Disabled (default). Enable **[1]** = Enable Digit Delivery feature for MediaPack/FXO & FXS. The digit delivery feature enables sending of DTMF digits to the gateway's port after the line is offhooked (FXS) or seized (FXO). For IP→Tel calls, after the line is offhooked / seized, the MediaPack plays the DTMF digits (of the called number) towards the phone line. **Note 1:** The called number can also include the characters 'p' (1.5 seconds pause) and 'd' (detection of dial tone). If the character 'd' is used, it must be the first 'digit' in the called number. The character 'p' can be used several times. For example, the called number can be as follows: d1005, dpp699, p9p300. To add the 'd' and 'p' digits, use the usual number manipulation rules. **Note 2:** To use this feature with FXO gateways, configure the gateway to work in one stage dialing mode. **Note 3:** If the parameter 'EnableDigitDelivery' is enabled, it is possible to configure the gateway to wait for dial tone per destination phone number (before or during dialing of destination phone number), therefore the parameter 'IsWaitForDialTone' (that is configurable for the entire gateway) is ignored. **Note 4:** The FXS gateway sends 200 OK messages only after it finishes playing the DTMF digits to the phone line.

Table 5-8: Advanced Parameters, General Parameters (continues on pages 78 to 82)

Parameter	Description
Enable DID Wink **[EnableDIDWink]**	Disable **[0]** = DID is disabled (default). Enable **[1]** = Enable DID. If enabled, the MediaPack can be used for connection to EIA/TIA-464B DID Loop Start lines. Both FXO (detection) and FXS (generation) are supported. An FXO gateway dials DTMF digits after a Wink signal is detected (instead of a Dial tone). An FXS gateway generates the Wink signal after the detection of offhook (instead of playing a Dial tone).
Reanswer Time **[RegretTime]**	The time period (in seconds) after user hangs up the phone and before call is disconnected (FXS). Also called regret time. The default time is 0 seconds.
Delay Before DID Wink **[DelayBeforeDIDWink]**	Defines the time interval (in seconds) between detection of offhook and generation of DID Wink. Applicable only to FXS gateways. The valid range is 0 to 1,000. The default value is 0.
Disconnect and Answer Supervision	
Enable Polarity Reversal **[EnableReversalPolarity]**	Disable **[0]** = Disable the polarity reversal service (default). Enable **[1]** = Enable the polarity reversal service. If the polarity reversal service is enabled, then the FXS gateway changes the line polarity on call answer and changes it back on call release. The FXO gateway sends a 200 OK response when polarity reversal signal is detected (applicable to one stage dialing only), and releases a call when a second polarity reversal signal is detected.
Enable Current Disconnect **[EnableCurrentDisconnect]**	Disable **[0]** = Disable the current disconnect service (default). Enable **[1]** = Enable the current disconnect service. If the current disconnect service is enabled, the FXO gateway releases a call when current disconnect signal is detected on its port, while the FXS gateway generates a ‘Current Disconnect Pulse’ after a call is released from IP. The current disconnect duration is determined by the parameter ‘CurrentDisconnectDuration’. The current disconnect threshold (FXO only) is determined by the parameter ‘CurrentDisconnectDefaultThreshold’. The frequency at which the analog line voltage is sampled is determined by the parameter ‘TimeToSampleAnalogLineVoltage’.
Disconnect on Broken Connection **[DisconnectOnBrokenConnection]**	No **[0]** = Don’t release the call. Yes **[1]** = Call is released if RTP packets are not received for a predefined timeout (default). **Note 1:** If enabled, the timeout is set by the parameter ‘BrokenConnectionEventTimeout’, in 100 msec resolution. The default timeout is 10 seconds: (BrokenConnectionEventTimeout =100). **Note 2:** This feature is applicable only if RTP session is used without Silence Compression. If Silence Compression is enabled, the gateway doesn’t detect that the RTP connection is broken. **Note 3:** During a call, if the source IP address (from where the RTP packets were sent) is changed without notifying the gateway, the gateway filters these RTP packets. To overcome this issue, set ‘DisconnectOnBrokenConnection=0’; the gateway doesn’t detect RTP packets arriving from the original source IP address, and switches (after 300 msec) to the RTP packets arriving from the new source IP address.
Broken Connection Timeout **[BrokenConnectionEventTimeout]**	The amount of time (in 100 msec units) an RTP packet isn’t received, after which a call is disconnected. The valid range is 1 to 1000. The default value is 100 (10 seconds). **Note 1:** Applicable only if ‘DisconnectOnBrokenConnection = 1’. **Note 2:** Currently this feature works only if Silence Suppression is disabled.

Table 5-8: Advanced Parameters, General Parameters (continues on pages 78 to 82)

Parameter	Description
Disconnect Call on Silence Detection **[EnableSilenceDisconnect]**	Yes **[1]** = The MediaPack disconnect calls in which silence occurs in both (call) directions for more than 120 seconds. No **[0]** = Call is not disconnected when silence is detected (default). The silence duration can be set by the 'FarEndDisconnectSilencePeriod' parameter (default 120).
Silence Detection Period [sec] **[FarEndDisconnectSilencePeriod]**	Duration of silence period (in seconds) prior to call disconnection. The range is 10 to 28800 (8 hours). The default is 120 seconds. Applicable to gateways, that use DSP templates 2 or 3.
Silence Detection Method **[FarEndDisconnectSilenceMethod]**	Silence detection method. None **[0] =** Silence detection option is disabled. Packets Count **[1]** = According to packet count. Voice/Energy Detectors **[2]** = According to energy and voice detectors (default). All **[3]** = According to packet count and energy / voice detectors.
TelConnectCode **[Send Digit Pattern on Connect]**	Defines a digit pattern that is sent to the Tel side after 200 OK is received from the IP side. The digit pattern is a predefined DTMF sequence that is used to indicate an answer signal (e.g., for billing purposes). Applicable only to FXS gateways. The valid range is 1 to 8 characters.
CDR and Debug	
CDR Server IP Address **[CDRSyslogServerIP]**	Defines the destination IP address for CDR logs. The default value is a null string that causes the CDR messages to be sent with all Syslog messages. **Note:** The CDR messages are sent to UDP port 514 (default Syslog port).
CDR Report Level **[CDRReportLevel]**	None **[0]** = Call Detail Recording (CDR) information isn't sent to the Syslog server (default). End Call **[1]** = CDR information is sent to the Syslog server at end of each Call. Start & End Call **[2]** = CDR information is sent to the Syslog server at the start and at the end of each Call. The CDR Syslog message complies with RFC 3161 and is identified by: Facility = 17 (local1) and Severity = 6 (Informational).
Debug Level **[GwDebugLevel]**	Syslog logging level. One of the following debug levels can be selected: 0 **[0]** = Debug is disabled (default) 1 **[1]** = Flow debugging is enabled 2 **[2]** = Flow and device interface debugging are enabled 3 **[3]** = Flow, device interface and stack interface debugging are enabled 4 **[4]** = Flow, device interface, stack interface and session manager debugging are enabled 5 **[5]** = Flow, device interface, stack interface, session manager and device interface expanded debugging are enabled. **Note:** Usually set to 5 if debug traces are needed.
Misc. Parameters	
Progress Indicator to IP **[ProgressIndicator2IP]**	No PI **[0]** = For Tel→IP calls, the gateway sends '180 Ringing' SIP response to IP after placing a call to phone (FXS) or to PBX (FXO). PI = 1, PI = 8 **[1]**, **[8]** = For Tel→IP calls, if 'EnableEarlyMedia=1', the gateway sends '183 session in progress' message + SDP immediately after a call is placed to Phone/PBX. This is used to cut through the voice path, before remote party answers the call, enabling the originating party to listen to network Call Progress Tones (such as Ringback tone or other network announcements). Not Configured **[-1]** = Default values are used. The default for FXO gateways is 1; The default for FXS gateways is 0.

Table 5-8: Advanced Parameters, General Parameters (continues on pages 78 to 82)

Parameter	Description
Enable Busy Out **[EnableBusyOut]**	No **[0]** = 'Busy out' feature is not used (default). Yes **[1]** = 'Busy out' feature is enabled. When Busy out is enabled, the MediaPack gateway performs a specific behavior (e.g., plays a reorder tone when the phone is offhooked) due to one of the following reasons: ▪ Physically disconnected from the network (i.e., Ethernet cable is disconnected). ▪ The Ethernet cable is connected, but the gateway can't communicate with any host. Note that LAN Watch-Dog must be activated (EnableLANWatchDog = 1). ▪ The gateway can't communicate with the gatekeeper/proxy and no other alternative exists to send the call. **Note:** The FXSOOSBehavior parameter is used to control the behavior of the FXS endpoints of the gateway when a Busy out or Graceful Lock occurs. **Note:** FXO endpoints during Busy out and Lock are inactive. **Note:** Refer to LifeLineType parameter for complementary optional behavior.
Default Release Cause **[DefaultReleaseCause]**	Default Release Cause (to IP) for IP→Tel calls, used when the gateway initiates a call release, and if an explicit matching cause for this release isn't found, a default release cause can be configured: The default release cause is: NO_ROUTE_TO_DESTINATION (3). Other common values are: NO_CIRCUIT_AVAILABLE (34), DESTINATION_OUT_OF_ORDER (27), etc. **Note:** The default release cause is described in the Q.931 notation, and is translated to corresponding SIP 40x or 50x value. For example: 404 for 3, 503 for 34 and 502 for 27.
Delay After Reset [sec] **[GWAppDelayTime]**	Defines the amount of time (in seconds) the gateway's operation is delayed after a reset cycle. The valid range is 0 to 45. The default value is 7 seconds. **Note:** This feature helps to overcome connection problems caused by some LAN routers or IP configuration parameters change by a DHCP Server.
Max Number of Active Calls **[MaxActiveCalls]**	Defines the maximum number of calls that the gateway can have active at the same time. If the maximum number of calls is reached, new calls are not established. The default value is max available channels (no restriction on the maximum number of calls). The valid range is 1 to max number of channels.
Max Call Duration (sec) **[MaxCallDuration]**	Defines the maximum call duration in seconds. If this time expires, both sides of the call are released (IP and Tel). The valid range is 0 to 120. The default is 0 (no limitation).
Enable LAN Watchdog **[EnableLanWatchDog]**	Disable **[0]** = Disable LAN Watch-Dog (default). Enable **[1]** = Enable LAN Watch-Dog. When LAN Watch-Dog is enabled, the gateway's overall communication integrity is checked periodically. If no communication for about 3 minutes is detected, the gateway performs a self test. - If the self test succeeds, the problem is logical link down (i.e. Ethernet cable disconnected on the switch side), and the Busy out mechanism is activated if enabled (EnableBusyOut = 1). LifeLine is activated if enabled. - If the self test fails, the gateway restarts to overcome internal fatal communication error. **Note:** Enable LAN Watchdog is relevant only if the Ethernet connection is full duplex. **Note:** EnableLanWatchDog is not applicable to MP-118.

Table 5-8: Advanced Parameters, General Parameters (continues on pages 78 to 82)

Parameter	Description
Enable Calls Cut Through **[CutThrough]**	Enables users to receive incoming IP calls while the port is in an offhooked state. Disable **[0]** = Disabled (default). Enable **[1]** = Enabled. If enabled, FXS gateways answer the call and 'cut through' the voice channel, if there is no other active call on that port, even if the port is in offhooked state. When the call is terminated (by the remote party), the gateway plays a reorder tone for 'TimeForReorderTone' seconds and is then ready to answer the next incoming call, without onhooking the phone. The waiting call is automatically answered by the gateway when the current call is terminated (EnableCallWaiting=1). **Note:** This option is applicable only to FXS gateways.
EnableUserInfoUsage **[Enable User-Information Usage]**	Enables or disables usage of the User Information loaded to the gateway via the User Information auxiliary file. 0 = Disable (default). 1 = Enable.
FXSOOSBehavior **[Out-Of-Service Behavior]**	Determines the behavior of FXS endpoints that are not defined (in the Endpoint Phone Number table), and the behavior of all FXS endpoints when a Busy-Out condition exists. 0 (None) = Normal operation: No response is provided to undefined endpoints. Dial tone is played to FXS endpoints when a Busy-Out condition exists. 1 (Reorder Tone) = The gateway plays a reorder tone to the connected phone/PBX (default). 2 (Polarity Reversal) = The gateways reverses the polarity of the endpoint, marking it as unusable (relevant, for example, to PBX DID lines). This option can't be configured on-the-fly. 3 (Polarity Reversal + Reorder Tone) = Same as 2 and 3 combined. This option can't be configured on-the-fly.

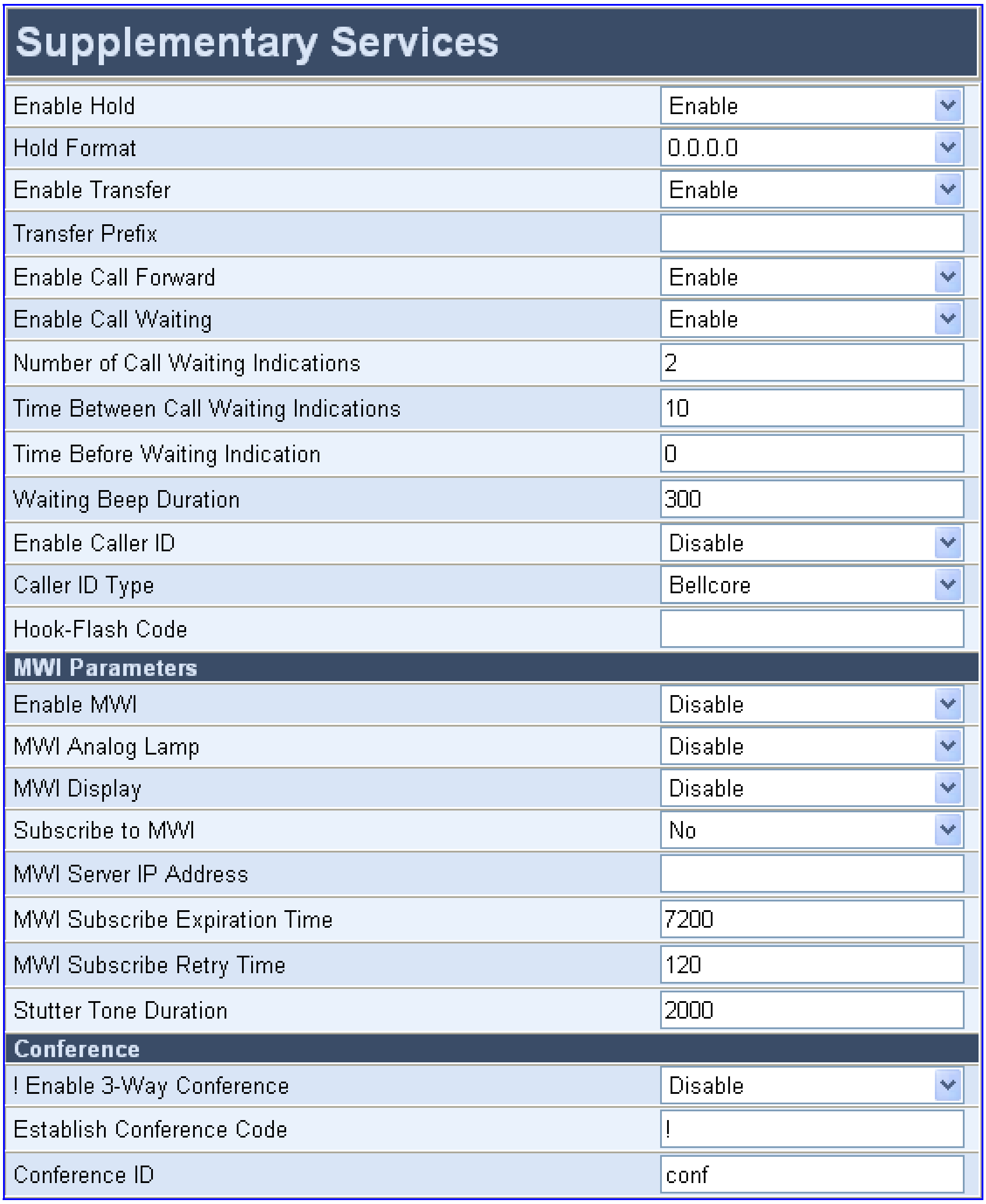
5.5.2.2 Supplementary Services

Use this screen to configure parameters that are associated with supplementary services. For detailed information on the supplementary services, refer to Section 8.1 on page 215.

➢ **To configure the supplementary services' parameters, take these 4 steps:**

1. Open the 'Supplementary Services' screen (**Protocol Management** menu > **Advanced Parameters** submenu > **Supplementary Services** option); the 'Supplementary Services' screen is displayed.

Figure 5-10: Supplementary Services Parameters Screen



Supplementary Services	
Enable Hold	Enable
Hold Format	0.0.0.0
Enable Transfer	Enable
Transfer Prefix	
Enable Call Forward	Enable
Enable Call Waiting	Enable
Number of Call Waiting Indications	2
Time Between Call Waiting Indications	10
Time Before Waiting Indication	0
Waiting Beep Duration	300
Enable Caller ID	Disable
Caller ID Type	Bellcore
Hook-Flash Code	
MWI Parameters	
Enable MWI	Disable
MWI Analog Lamp	Disable
MWI Display	Disable
Subscribe to MWI	No
MWI Server IP Address	
MWI Subscribe Expiration Time	7200
MWI Subscribe Retry Time	120
Stutter Tone Duration	2000
Conference	
! Enable 3-Way Conference	Disable
Establish Conference Code	!
Conference ID	conf

2. Configure the supplementary services parameters according to Table 5-9.
3. Click the **Submit** button to save your changes, or click the **Subscribe for MWI** or **Un-Subscribe for MWI** buttons to save your changes and to subscribe / unsubscribe to the MWI server.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-9: Supplementary Services Parameters (continues on pages 84 to 86)

Parameter	Description
Enable Hold **[EnableHold]**	No **[0]** = Disable the Hold service. Yes **[1]** = Enable the Hold service (default). If the Hold service is enabled, a user can activate Hold (or Unhold) using the hook-flash. On receiving a Hold request, the remote party is put on-hold and hears the hold tone. **Note:** To use this service, the gateways at both ends must support this option.
Hold Format **[HoldFormat]**	Determines the format of the hold request. 0.0.0.0 **[0]** = The connection IP address in SDP is 0.0.0.0 (default). Send Only **[1]** = The last attribute of the SDP contains the following 'a=sendonly'.
Enable Transfer **[EnableTransfer]**	No **[0]** = Disable the call transfer service. Yes **[1]** = Enable the call transfer service (using REFER) (default). If the Transfer service is enabled, the user can activate Transfer using hook-flash signaling. If this service is enabled, the remote party performs the call transfer. **Note 1:** To use this service, the gateways at both ends must support this option. **Note 2:** To use this service, set the parameter 'Enable Hold' to 'Yes'.
Transfer Prefix **[xferPrefix]**	Defined string that is added, as a prefix, to the transferred / forwarded called number, when REFER / 3xx message is received. **Note 1:** The number manipulation rules apply to the user part of the 'REFER-TO / Contact' URI before it is sent in the INVITE message. **Note 2:** The 'xferprefix' parameter can be used to apply different manipulation rules to differentiate the transferred / forwarded number from the original dialed number.
Enable Call Forward **[EnableForward]**	No **[0]** = Disable the Call Forward service. Yes **[1]** = Enable Call Forward service (using REFER) (default). For FXS gateways a Call Forward table must be defined to use the Call Forward service. To define the Call Forward table, refer to Section 5.5.9.4 on page 122. **Note:** To use this service, the gateways at both ends must support this option.
Enable Call Waiting **[EnableCallWaiting]**	No **[0]** = Disable the Call Waiting service. Yes **[1]** = Enable the Call Waiting service (default). If enabled, when an FXS gateway receives a call on a busy endpoint, it responds with a 182 response (and not with a 486 busy). The gateway plays a call waiting indication signal. When hook-flash is detected, the gateway switches to the waiting call. The gateway that initiated the waiting call plays a Call Waiting Ringback tone to the calling party after a 182 response is received. **Note 1:** The gateway's Call Progress Tones file must include a 'call waiting Ringback' tone (caller side) and a 'call waiting' tone (called side, FXS only). **Note 2:** The 'Enable Hold' parameter must be enabled on both the calling and the called sides. For information on the Call Waiting feature, refer to Section 8.1.5 on page 217. For information on the Call Progress Tones file, refer to Section 15.1 on page 325.
Number of Call Waiting Indications **[NumberOfWaitingIndications]**	Number of waiting indications that are played to the receiving side of the call (FXS only) for Call Waiting. This parameter is used to control the Registration / Subscription rate. The default value is 2.

Table 5-9: Supplementary Services Parameters (continues on pages 84 to 86)

Parameter	Description
Time Between Call Waiting Indications **[TimeBetweenWaitingIndications]**	Difference (in seconds) between call waiting indications (FXS only) for call waiting. The default value is 10 seconds.
Time before Waiting Indication **[TimeBeforeWaitingIndication]**	Defines the interval (in seconds) before a call waiting indication is played to the port that is currently in a call (FXS only). The valid range is 0 to 100. The default time is 0 seconds.
Waiting Beep Duration **[WaitingBeepDuration]**	Duration (in msec) of waiting indications that are played to the receiving side of the call (FXS only) for Call Waiting. The default value is 300.
Enable Caller ID **[EnableCallerID]**	No **[0]** = Disable the Caller ID service (default). Yes **[1]** = Enable the Caller ID service. If the Caller ID service is enabled, then, for FXS gateways, calling number and Display text are sent to gateway port. For FXO gateways, the Caller ID signal is detected and is sent to IP in SIP INVITE message (as 'Display' element). For information on the Caller ID table, refer to Section 5.5.9.3 on page 121. To disable/enable caller ID generation per port, refer to Section 5.5.9.4 on page 122.
Caller ID Type **[CallerIDType]**	Defines one of the following standards for detection (FXO) and generation (FXS) of Caller ID and detection (FXO) of MWI (when specified) signals. Bellcore **[0]** (Caller ID and MWI) (default). ETSI **[1]** (Caller ID and MWI) NTT **[2]** British **[4]** DTMF ETSI **[16]** Denmark **[17]** (Caller ID and MWI) India **[18]** Brazil **[19]** **Note 1:** The Caller ID signals are generated/detected between the first and the second rings. **Note 2:** To select the Bellcore Caller ID sub standard, use the parameter 'BellcoreCallerIDTypeOneSubStandard'. To select the ETSI Caller ID sub standard, use the parameter 'ETSICallerIDTypeOneSubStandard'. **Note 3:** To select the Bellcore MWI sub standard, use the parameter 'BellcoreVMWITypeOneStandard'. To select the ETSI MWI sub standard, use the parameter 'ETSIVMWITypeOneStandard'.
Hook-Flash Code **[HookFlashCode]**	Determines a digit pattern which, when received from the Tel side, indicates a Hook Flash event. The valid range is a 25-character string.
MWI Parameters	
Enable MWI **[EnableMWI]**	Enable MWI (message waiting indication). Disable **[0]** = Disabled (default). Enable **[1]** = MWI service is enabled. This parameter is applicable only to FXS gateways. **Note:** The MediaPack only supports reception of MWI. For detailed information on MWI, refer to Section 8.1.6 on page 218.
MWI Analog Lamp **[MWIAnalogLamp]**	Disable **[0]** = Disable (default). Enable **[1]** = Enable visual Message Waiting Indication, supplies line voltage of approximately 100 VDC to activate the phone's lamp. This parameter is applicable only to FXS gateways.
MWI Display **[MWIDisplay]**	Disable **[0]** = MWI information isn't sent to display (default). Enable **[1]** = MWI information is sent to display. If enabled, the gateway generates an MWI FSK message that is displayed on the MWI display. This parameter is applicable only to FXS gateways.

Table 5-9: Supplementary Services Parameters (continues on pages 84 to 86)

Parameter	Description
Subscribe to MWI **[EnableMWISubscription]**	Disable **[0]** = Disable MWI subscription (default). Enable **[1]** = Enable subscription to MWI (to MWIServerIP address). **Note:** Use the parameter 'SubscriptionMode' (described in Table 5-35 on page 132) to determine whether the gateway subscribes separately per endpoint of for the entire gateway.
MWI Server IP Address **[MWIServerIP]**	MWI server IP address. If provided, the gateway subscribes to this IP address. Can be configured as a numerical IP address or as a domain name. If not configured, the Proxy IP address is used instead.
MWI Subscribe Expiration Time **[MWIExpirationTime]**	MWI subscription expiration time in seconds. The default is 7200 seconds. The range is 10 to 72000.
MWI Subscribe Retry Time **[SubscribeRetryTime]**	Subscription retry time in seconds. The default is 120 seconds. The range is 10 to 7200.
Stutter Tone Duration **[StutterToneDuration]**	Duration (in msec) of the played Stutter dial tone, which indicates that Call Forwarding is enabled or that there is a waiting message(s). The default is 2,000 (i.e., 2 seconds). The range is 1,000 to 60,000. The Stutter tone is played (instead of a regular Dial tone), when a Call Forward is enabled on the specific port or when MWI is received. The tone is composed of a 'Confirmation tone', which is played for a user-defined duration (StutterToneDuration), followed by a 'Stutter tone'. Both tones are defined in the CPT file. **Note 1:** This parameter is applicable only to FXS gateways. **Note 2:** The message waiting notification (MWI) tone takes precedence over the call forwarding reminder tone. For detailed information on Message Waiting Indication (MWI), refer to Section 8.1.6 on page 218.
Conference Parameters	
Enable 3-Way Conference **[Enable3WayConference]**	Enables or disables the 3-Way Conference feature. Valid options include: Disable **[0]** = Disables 3-way conference feature (default) Enable **[1]** = Enables 3-way conference feature
Establish Conference Code **[ConferenceCode]**	Defines the digit pattern that once detected, generates the Conference-initiating INVITE when Enable3WayConference is set to 1. The valid range is a 25-character string. The default is "!" (Hook-Flash).
Conference ID **[ConferenceID]**	Defines the Conference Identification string (up to 16 characters). The gateway uses this identifier in the Conference-initiating INVITE that is sent to the media server when Enable3WayConferenceis set to 1. The default value is 'conf'. For example: ConferenceID = MyConference.

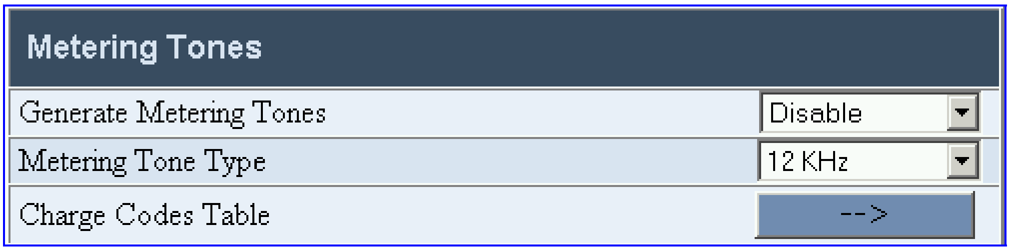
5.5.2.3 Metering Tones

FXS gateways can generate 12/16 KHz metering pulses towards the Tel side (e.g., for connection to a payphone or private meter). Tariff pulse rate is determined according to an internal table. This capability enables users to define different tariffs according to the Source / Destination numbers and the Time-of-Day. The tariff rate includes the time interval between the generated pulses and the number of pulses generated on answer.

➢ To configure the Metering Tones, take these 6 steps:

1. Open the 'Metering Tones' screen (**Protocol Management** menu > **Advanced Parameters** submenu > **Metering Tones** option); the 'Metering Tones' screen is displayed.

Figure 5-11: Metering Tones Parameters Screen



2. From the 'Generate Metering Tones' drop-down list, select the method used to configure the metering tones that are generated to the Tel side (refer to Table 5-10). If you selected 'Internal Table', you must configure the 'Charge Codes Table'. To configure the 'Charge Codes Table', refer to Section 5.5.2.3.1 below. Continue with Step 4.
3. From the 'Metering Tone Type' drop-down list, select the type of the metering tone according to your requirements (refer to Table 5-10).
4. In the Tel to IP Routing table (Section 5.5.5.2 on page 100) assign a charge code rule to the routing rules you require. When a new call is established, the Tel to IP Routing table is searched for the destination IP addresses. Once a route is found, the Charge Code (configured for that route) is used to associate the route with an entry in the Charge Codes table.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-10: Metering Tones Parameters

Parameter	Description
Generate Metering Tones **[PayPhoneMeteringMode]**	Determines the method used to configure the metering tones that are generated to the Tel side (FXS gateways only). Disable **[0]** = Metering tones aren't generated (default). Internal Table **[1]** = Metering tones are generated according to the internal table configured by the parameter ChargeCode. RADIUS **[2]** = N/A. **Note:** This parameter is not applicable to the Metering Tones Relay mechanism (described in Section 11 on page 277).
Metering Tones Type **[MeteringType]**	Defines the metering tone (12 kHz or 16 kHz) that is detected by FXO gateways and generated by FXS gateways. 12 kHz **[0]** = 12 kHz metering tone (default). 16 kHz **[1]** = 16 kHz metering tone. **Note:** Suitable (12 kHz or 16 KHz) *coeff* must be used for both FXS and FXO gateways.
Charge Codes Table	For detailed information on configuring the Charge Codes Table, refer to Section 5.5.2.3.1 below.

5.5.2.3.1 Charge Codes Table

The Charge Codes table is used to configure the metering tones (and their time interval) that the FXS gateway generates to the Tel side. To associate a charge code to an outgoing Tel to IP call, use the Tel to IP Routing table.

➢ To configure the Charge Codes table, take these 6 steps:

1. Access the 'Metering Tones' screen (**Protocol Management** menu > **Advanced Parameters** submenu > **Metering Tones** option); the 'Metering Tones' screen is displayed (Figure 5-11).
2. Open the 'Charge Codes Table' screen by clicking the arrow sign (-->) to the right of the Charge Codes Table label; the Charge Codes Table is displayed.

Figure 5-12: Charge Codes Table Screen

Charge Codes Table

Index	Time Period 1			Time Period 2			Time Period 3			Time Period 4		
	End Time	Puls Interval	Pulses On Answer	End Time	Puls Interval	Pulses On Answer	End Time	Puls Interval	Pulses On Answer	End Time	Puls Interval	Pulses On Answer
1	07	30	1	14	20	2	20	15	1	00	60	1
2	05	60	1	14	20	1	00	60	1			
3	00	60	1									
4												

3. Use the table to define up to 25 different charge codes (each charge code is defined in a single row). Each code can include from a single and up to four different time periods in a day (24 hours). Each time period is composed of:
   - The end (in a 24 rounded-hour's format) of the time period.
   - The time interval between pulses (in seconds).
   - The number of pulses sent on answer.
4. The first time period always starts at midnight (00). It's mandatory that the last time period per rule ends at midnight (00). This prevents undefined time frames in a day. The gateway selects the time period by comparing the gateway's current time to the end time of each time period of the selected Charge Code. The gateway generates the Number of Pulses on Answer once the call is connected and from that point on, it generates a pulse each Pulse Interval. If a call starts at a certain time period and crosses to the next, the information of the next time period is used.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-11: Charge Codes Table *ini* File Parameter

Parameter Name in *ini* File	Parameter Format
ChargeCode	ChargeCode_<Charge Code ID> = <1st period end time>,<1st period pulse interval>,<1st period pulses on answer>, <2nd period end time>,<2nd period pulse interval>,<2nd period pulses on answer>, <3rd period end time>,<3rd period pulse interval>,<3rd period pulses on answer>, <4th period end time>,<4th period pulse interval>,<4th period pulses on answer> For example: ChargeCode_1 = 07,30,1,14,20,2,20,15,1,00,60,1 ChargeCode_2 = 05,60,1,14,20,1,00,60,1 ChargeCode_3 = 00,60,1 **Note:** Up to 25 different metering rules can be defined by repeating the parameter 25 times.

5.5.2.4 Keypad Features

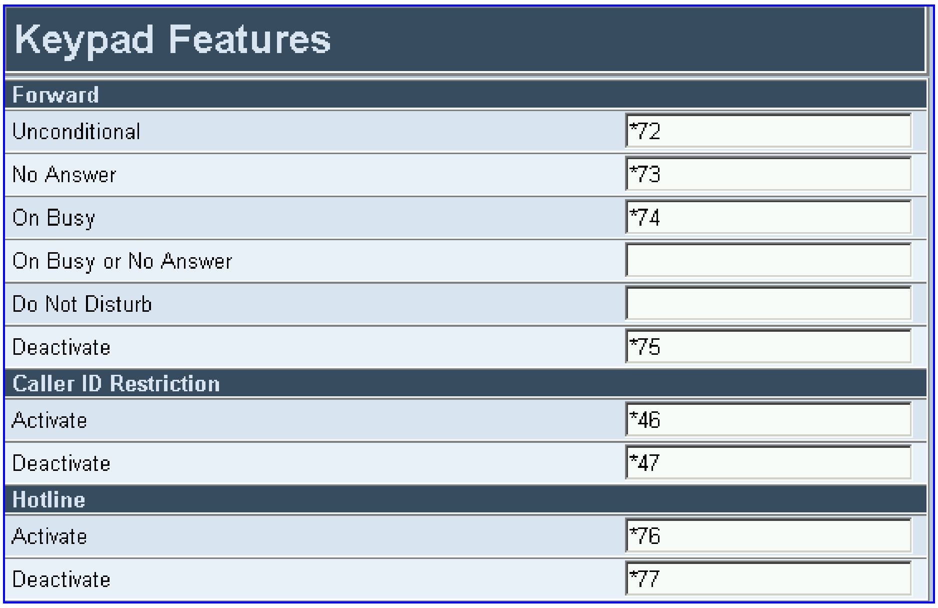
The Keypad Features screen (applicable only to FXS gateways) enables you to activate / deactivate the following features directly from the connected telephone's keypad:

- Hotline (refer to Section 5.5.9.2 on page 120).
- Caller ID Restriction (refer to Section 5.5.9.3 on page 121).
- Call Forward (refer to Section 5.5.9.4 on page 122).

➢ To configure the keypad features, take these 4 steps:

1. Open the 'Keypad Features' screen (**Protocol Management** menu > **Advanced Parameters** submenu > **Keypad Features** option); the 'Keypad Features' screen is displayed.

Figure 5-13: Keypad Features Screen



2. Configure the Keypad Features according to Table 5-12.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Notes:

- The method used by the gateway to collect dialed numbers is identical to the method used during a regular call (i.e., max digits, interdigit timeout, digit map, etc.).
- The activation of each feature remains in effect until it is deactivated (i.e., it is not per call).

Table 5-12: Keypad Features Parameters

Parameter	Description
Forward Note that the forward type and number can be viewed in the Call Forward Table (refer to Section 5.5.9.5 on page 124)	
Unconditional **[KeyCFUnCond]**	Keypad sequence that activates the immediate forward option.
No Answer **[KeyCFNoAnswer]**	Keypad sequence that activates the forward on no answer option.
On Busy **[KeyCFBusy]**	Keypad sequence that activates the forward on busy option.
On Busy or No Answer **[KeyCFBusyOrNoAnswer]**	Keypad sequence that activates the forward on 'busy or no answer' option.
Do Not Disturb **[KeyCFDoNotDisturb]**	Keypad sequence that activates the Do Not Disturb option (immediately reject incoming calls).
To activate the required forward method from the telephone: ▪ Dial the preconfigured sequence number on the keypad; a dial tone is heard. ▪ Dial the telephone number to which the call is forwarded (terminate the number with #); a confirmation tone is heard.	
Deactivate **[KeyCFDeact]**	Keypad sequence that deactivates any of the forward options. After the sequence is pressed a confirmation tone is heard.
Caller ID Restriction Note that the caller ID presentation can be viewed in the Caller Display Information table (refer to Section 5.5.9.3 on page 121)	
Activate **[KeyCLIR]**	Keypad sequence that activates the restricted Caller ID option. After the sequence is pressed a confirmation tone is heard.
Deactivate **[KeyCLIRDeact]**	Keypad sequence that deactivates the restricted Caller ID option. After the sequence is pressed a confirmation tone is heard.
Hotline Note that the destination phone number and the auto dial status can be viewed in the Automatic Dialing table (refer to Section 5.5.9.2 on page 120)	
Activate **[KeyHotLine]**	Keypad sequence that activates the delayed hotline option. To activate the delayed hotline option from the telephone: ▪ Dial the preconfigured sequence number on the keypad; a dial tone is heard. ▪ Dial the telephone number to which the phone automatically dials after a configurable delay (terminate the number with #); a confirmation tone is heard.
Deactivate **[KeyHotLineDeact]**	Keypad sequence that deactivates the delayed hotline option. After the sequence is pressed a confirmation tone is heard.

5.5.3 Configuring the Number Manipulation Tables

The VoIP gateway provides four Number Manipulation tables for incoming and outgoing calls. These tables are used to modify the destination and source telephone numbers so that the calls can be routed correctly.

The Manipulation Tables are:

- Destination Phone Number Manipulation Table for IP→Tel calls
- Destination Phone Number Manipulation Table for Tel→IP call
- Source Phone Number Manipulation Table for IP→Tel calls
- Source Phone Number Manipulation Table for Tel→IP calls

> **Note:** Number manipulation can occur either before or after a routing decision is made. For example, you can route a call to a specific hunt group according to its original number, and then you can remove / add a prefix to that number before it is routed. To control when number manipulation is done, set the IP to Tel Routing Mode (described in Table 5-18) and the Tel to IP Routing Mode (described in Table 5-17) parameters.

Possible uses for number manipulation can be as follows:

- To strip/add dialing plan digits from/to the number. For example, a user could dial 9 in front of each number in order to indicate an external line. This number (9) can be removed here before the call is setup.
- Allow / disallow Caller ID information to be sent according to destination / source prefixes. For detailed information on Caller ID, refer to Section 5.5.9.3 on page 121.

➢ To configure the Number Manipulation tables, take these 5 steps:

1. Open the Number Manipulation screen you want to configure (**Protocol Management** menu > **Manipulation Tables** submenu); the relevant Manipulation table screen is displayed. Figure 5-14 shows the 'Source Phone Number Manipulation Table for Tel→IP calls'.

Figure 5-14: Source Phone Number Manipulation Table for Tel→IP calls

	Dest. Prefix	Source Prefix	Num of Stripped Digits	Prefix (Suffix) to Add	Number of Digits to Leave	Presentation
1	03	201	0	972		Allowed
2		1001	4	5(23)		Restricted
3		123451001#	0	(8)	4	Not Configured
4		[30-40]xx	(1)	2		Not Configured
5	[6,7,8]	2001	5	3		Not Configured
6						Not Configured
7						Not Configured

2. From the 'Table Index' drop-down list, select the range of entries that you want to edit (up to 20 entries can be configured for Source Number Manipulation and 50 entries for Destination Number Manipulation).
3. Configure the Number Manipulation table according to Table 5-13.

4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-13: Number Manipulation Parameters (continues on pages 92 to 93)

Parameter	Description
Dest. Prefix	Each entry in the Destination Prefix fields represents a destination telephone number prefix. An asterisk (*) represents any number.
Source Prefix	Each entry in the Source Prefix fields represents a source telephone number prefix. An asterisk (*) represents any number.
Source IP	Each entry in the Source IP fields represents the source IP address of the call (obtained from the Contact header in the INVITE message). This column only applies to the 'Destination Phone Number Manipulation Table for IP to Tel'. **Note:** The source IP address can include the 'x' wildcard to represent single digits. For example: 10.8.8.xx represents all the addresses between 10.8.8.10 to 10.8.8.99.
The manipulation rules are applied to any incoming call whose: ▪ Destination number prefix matches the prefix defined in the 'Destination Number' field. ▪ Source number prefix matches the prefix defined in the 'Source Prefix' field. ▪ Source IP address matches the IP address defined in the 'Source IP' field (if applicable). Note that number manipulation can be performed using a combination of each of the above criteria, or using each criterion independently. **Note:** For available notations that represent multiple numbers, refer to Section 5.5.3.1 on page 95.	
Number of Stripped Digits	▪ Enter the number of digits that you want to remove from the left of the telephone number prefix. For example, if you enter 3 and the phone number is 5551234, the new phone number is 1234. ▪ Enter the number of digits (in brackets) that you want to remove from the right of the telephone number prefix. **Note:** A combination of the two options is allowed (e.g., 2(3)).
Prefix (Suffix) to Add	▪ Prefix - Enter the number / string you want to add to the front of the phone number. For example, if you enter 9 and the phone number is 1234, the new number is 91234. ▪ Suffix - Enter the number / string (in brackets) you want to add to the end of the phone number. For example, if you enter (00) and the phone number is 1234, the new number is 123400. **Note:** You can enter a prefix and a suffix in the same field (e.g., 9(00)).
Number of Digits to Leave	Enter the number of digits that you want to leave from the right.
Note: The manipulation rules are executed in the following order: ▪ Num of stripped digits ▪ Number of digits to leave ▪ Prefix / suffix to add Figure 5-14 on the previous page exemplifies the use of these manipulation rules in the 'Source Phone Number Manipulation Table for Tel→IP Calls': ▪ When destination number equals 035000 and source number equals 20155, the source number is changed to 97220155. ▪ When source number equals 1001876, it is changed to 587623. ▪ Source number 1234510012001 is changed to 20018. ▪ Source number 3122 is changed to 2312.	

Table 5-13: Number Manipulation Parameters (continues on pages 92 to 93)

Parameter	Description
Presentation	Select 'Allowed' to send Caller ID information when a call is made using these destination / source prefixes. Select 'Restricted' if you want to restrict Caller ID information for these prefixes. When set to 'Not Configured', the privacy is determined according to the Caller ID table (refer to Section 5.5.9.3 on page 121). **Note:** If 'Presentation' is set to 'Restricted' and 'Asserted Identity Mode' is set to 'P-Asserted', the From header in INVITE message is: From: 'anonymous' <sip: anonymous@anonymous.invalid> and 'privacy: id' header is included in the INVITE message.

Table 5-14: Number Manipulation *ini* File Parameters (continues on pages 93 to 95)

Parameter	Description
NumberMapTel2IP	Manipulates the destination number for Tel to IP calls. NumberMapTel2IP = a,b,c,d,e,f,g a = Destination number prefix b = Number of stripped digits from the left, or (if brackets are used) from the right. A combination of both options is allowed. c = String to add as prefix, or (if brackets are used) as suffix. A combination of both options is allowed. d = Number of remaining digits from the right e = Number Plan used in RPID header f = Number Type used in RPID header g = Source number prefix The 'b' to 'f' manipulation rules are applied if the called and calling numbers match the 'a' and 'g' conditions. The manipulation rules are executed in the following order: 'b', 'd' and 'c'. Parameters can be skipped by using the sign '$$', for example: NumberMapTel2IP=01,2,972,$$,0,0,$$ NumberMaPTel2IP=03,(2),667,$$,0,0,22 **Note:** Number Plan & Type can optionally be used in Remote Party ID (RPID) header by using the 'EnableRPIHeader' parameter.

Table 5-14: Number Manipulation *ini* File Parameters (continues on pages 93 to 95)

Parameter	Description
NumberMapIP2Tel	Manipulate the destination number for IP to Tel calls. NumberMapIP2Tel = a,b,c,d,e,f,g,h,i a = Destination number prefix. b = Number of stripped digits from the left, or (if brackets are used) from the right. A combination of both options is allowed. c = String to add as prefix, or (if brackets are used) as suffix. A combination of both options is allowed. d = Number of remaining digits from the right. e = Not applicable, set to $$. f = Not applicable, set to $$. g = Source number prefix. h = Not applicable, set to $$. i = Source IP address (obtained from the Contact header in the INVITE message). The 'b' to 'd' manipulation rules are applied if the called and calling numbers match the 'a', 'g' and 'i' conditions. The manipulation rules are executed in the following order: 'b', 'd' and 'c'. Parameters can be skipped by using the sign '$$', for example: NumberMapIP2Tel =01,2,972,$$,$$,$$,034,$$,10.13.77.8 NumberMapIP2Tel =03,(2),667,$$,$$,$$,22 **Note:** The Source IP address can include the 'x' wildcard to represent <u>single</u> digits. For example: 10.8.8.xx represents all the addresses between 10.8.8.10 to 10.8.8.99. The '*' wildcard represents any number between 0 and 255, e.g. 10.8.8.* represents all the addresses between 10.8.8.0 and 10.8.8.255.
SourceNumberMapTel2IP	SourceNumberMapTel2IP = a,b,c,d,e,f,g,h a = Source number prefix b = Number of stripped digits from the left, or (if in brackets are used) from right. A combination of both options is allowed. c = String to add as prefix, or (if in brackets are used) as suffix. A combination of both options is allowed. d = Number of remaining digits from the right e = Number Plan used in RPID header f = Number Type used in RPID header g = Destination number prefix h = Calling number presentation (0 to allow presentation, 1 to restrict presentation) The 'b' to 'f' and 'h' manipulation rules are applied if the called and calling numbers match the 'a' and 'g' conditions. The manipulation rules are executed in the following order: 'b', 'd' and 'c'. Parameters can be skipped by using the sign '$$', for example: SourceNumberMapTel2IP=01,2,972,$$,0,0,$$,1 SourceNumberMapTel2IP=03,(2),667,$$,0,0,22 **Note 1:** 'Presentation' is set to 'Restricted' only if 'Asserted Identity Mode' is set to 'P-Asserted'. **Note 2:** Number Plan & Type can optionally be used in Remote Party ID (RPID) header by using the 'EnableRPIHeader' parameter.

Table 5-14: Number Manipulation *ini* File Parameters **(continues on pages 93 to 95)**

Parameter	Description
SourceNumberMapIP2Tel	Manipulate the source number for IP to Tel calls. NumberMapIP2Tel = a,b,c,d,e,f,g,h,i a = Source number prefix b = Number of stripped digits from the left, or (if brackets are used) from the right. A combination of both options is allowed. c = String to add as prefix, or (if brackets are used) as suffix. A combination of both options is allowed. d = Number of remaining digits from the right e = Not in use, should be set to $$ f = Not in use, should be set to $$ g = Destination number prefix h = Not in use, should be set to $$ I = Source IP address (obtained from the Request-URI in the INVITE message). The 'b' to 'd' manipulation rules are applied if the called and calling numbers match the 'a', 'g' and 'I' conditions. The manipulation rules are executed in the following order: 'b', 'd' and 'c'. Parameters can be skipped by using the sign '$$', for example: NumberMapIP2Tel =01,2,972,$$,$$,$$,034 NumberMapIP2Tel =03,(2),667,$$,$$,$$,22 **Note:** The Source IP address can include the 'x' wildcard to represent single digits. For example: 10.8.8.xx represents all the addresses between 10.8.8.10 to 10.8.8.99. The '*' wildcard represents any number between 0 and 255, e.g. 10.8.8.* represents all the addresses between 10.8.8.0 and 10.8.8.255.

5.5.3.1 Dialing Plan Notation

The dialing plan notation applies, in addition to the four Manipulation tables, also to Tel→IP Routing table and to IP→Hunt Group Routing table.

When entering a number in the destination and source 'Prefix' columns, you can create an entry that represents multiple numbers using the following notation:

- [n-m] represents a range of numbers
- [n,m] represents multiple numbers. Note that this notation only supports single digit numbers.
- x represents any single digit
- # (that terminates the number) represents the end of a number
- A single asterisk (*) represents any number

For example:

- [5551200-5551300]# represents all numbers from 5551200 to 5551300
- [2,3,4]xxx# represents four-digit numbers that start with 2, 3 or 4
- 54324 represents any number that starts with 54324
- 54324xx# represents a 7 digit number that starts with 54324
- 123[100-200]# represents all numbers from 123100 to 123200.

The VoIP gateway matches the rules starting at the top of the table. For this reason, enter more specific rules above more generic rules. For example, if you enter 551 in entry 1 and 55 in entry 2, the VoIP gateway applies rule 1 to numbers that starts with 551 and applies rule 2 to numbers that start with 550, 552, 553, 554, 555, 556, 557, 558 and 559. However if you enter 55 in entry 1 and 551 in entry 2, the VoIP gateway applies rule 1 to all numbers that start with 55 including numbers that start with 551.

5.5.4 Mapping NPI/TON to Phone-Context

The Phone-Context table is used to configure the mapping of NPI and TON to the Phone-Context SIP parameter. When a call is received from the ISDN, the NPI and TON are compared against the table and the Phone-Context value is used in the outgoing SIP INVITE message. The same mapping occurs when an INVITE with a Phone-Context attribute is received. The Phone-Context parameter appears in the standard SIP headers where a phone number is used (Request-URI, To, From, Diversion).

You can also configure the Phone Context table using the *ini* file parameter PhoneContext (refer to Section 5.5.13 on page 132).

➢ To configure the Phone-Context tables, take these 6 steps:

1. Open the 'Phone Context Table' screen (**Protocol Management** menu > **Manipulation Tables** submenu > **Phone Context Table** option); the 'Phone Context Table' screen appears, as shown below.

Figure 5-15: Phone Context Table Screen

2. From the 'Add Phone Context As Prefix' drop-down list, select 'Enable' to add the received Phone-Context parameter as a prefix to outgoing ISDN SETUP Called and Calling numbers, if necessary.
3. From the 'Phone Context Index' drop-down list, select the index number.

4. Configure the Phone Context table according to Table 5-15.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Notes:

- Several rows with the same NPI-TON or Phone-Context are allowed. In such a scenario, a Tel-to-IP call uses the first match.
- Phone-Context '+' is a unique case as it doesn't appear in the Request-URI as a Phone-Context parameter. Instead, it's added as a prefix to the phone number. The '+' isn't removed from the phone number in the IP-to-Tel direction.

Table 5-15: Phone-Context Parameters

Parameter	Description
Add Phone Context As Prefix **[AddPhoneContextAsPrefix]**	Determines whether or not the received Phone-Context parameter is added as a prefix to the outgoing ISDN SETUP Called and Calling numbers. Valid options include: 0 = Disable (default). 1 = Enable.
NPI	Select the Number Plan assigned to this entry. You can select the following: 0 = Unknown (default) 1 = E.164 Public 9 = Private
TON	Select the Number Type assigned to this entry. ▪ If you selected Unknown as the NPI, you can select Unknown (0) ▪ If you selected Private as the NPI, you can select Unknown (0), Level 2 Regional (1), Level 1 Regional (2), PSTN Specific (3), or Level 0 Regional (Local) (4). ▪ If you selected E.164 Public as the NPI, you can select Unknown (0), International (1), National (2), Network Specific (3), Subscriber (4), or Abbreviated (6).
Phone Context	The Phone-Context SIP URI parameter.

5.5.5 Configuring the Routing Tables

Use this submenu to configure the gateway's IP→Tel and Tel→IP routing tables and their associated parameters.

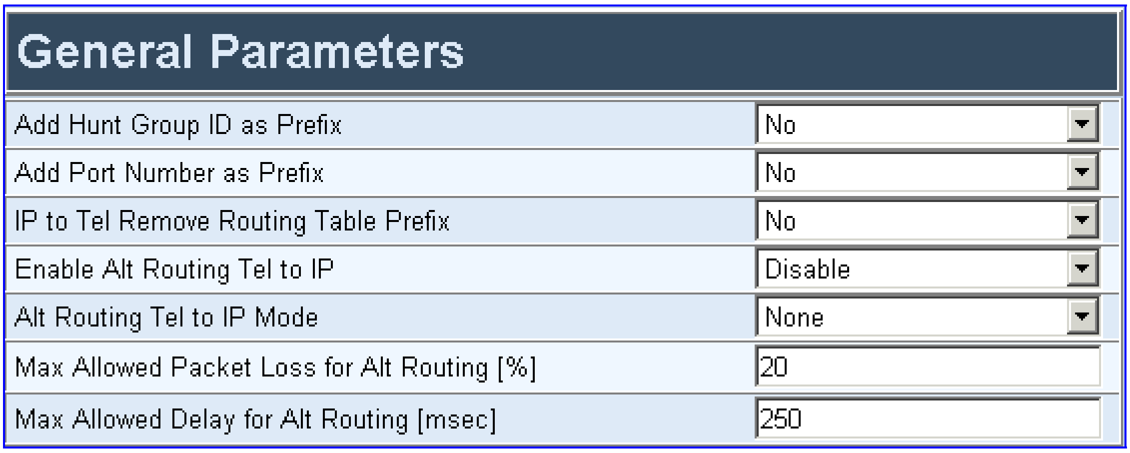
5.5.5.1 General Parameters

Use this screen to configure the gateway's IP→Tel and Tel→IP routing parameters.

➢ **To configure the general parameters under Routing Tables, take these 4 steps:**

1. Open the 'General Parameters' screen (**Protocol Management** menu > **Routing Tables** submenu > **General** option); the 'General Parameters' screen is displayed.

Figure 5-16: Routing Tables, General Parameters Screen



2. Configure the general parameters under 'Routing Tables' according to Table 5-16.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-16: Routing Tables, General Parameters (continues on pages 98 to 99)

Parameter	Description
Add Hunt Group ID as Prefix **[AddTrunkGroupAsPrefix]**	No **[0]** = Don't add hunt group ID as prefix (default). Yes **[1]** = Add hunt group ID as prefix to called number. If enabled, then the hunt group ID is added as a prefix to the destination phone number for Tel→IP calls. **Note 1:** This option can be used to define various routing rules. **Note 2:** To use this feature you must configure the hunt group IDs.
Add Port Number as Prefix **[AddPortAsPrefix]**	No **[0]** = Disable the add port as prefix service (default). Yes **[1]** = Enable the add port as prefix service. If enabled, then the gateway's port number (single digit in the range 1 to 8 for 8-port gateways, two digits in the range 01 to 24 in MP-124) is added as a prefix to the destination phone number for Tel→IP calls. **Note:** This option can be used to define various routing rules.

Table 5-16: Routing Tables, General Parameters (continues on pages 98 to 99)

Parameter	Description
IP to Tel Remove Routing Table Prefix **[RemovePrefix]**	No **[0]** = Don't remove prefix (default) Yes **[1]** = Remove the prefix (defined in the IP to Hunt Group Routing table) from a telephone number for an IP→Tel call, before forwarding it to Tel. For example: To route an incoming IP→Tel Call with destination number 21100, the IP to Hunt Group Routing table is scanned for a matching prefix. If such prefix is found, 21 for instance, then before the call is routed to the corresponding hunt group the prefix (21) is removed from the original number, so that only 100 is left. **Note 1:** Applicable only if number manipulation is performed after call routing for IP→Tel calls (refer to 'IP to Tel Routing Mode' parameter). **Note 2:** Similar operation (of removing the prefix) is also achieved by using the usual number manipulation rules.
Enable Alt Routing Tel to IP **[AltRoutingTel2IPEnable]**	No **[0]** = Disable the Alternative Routing feature (default). Yes **[1]** = Enable the Alternative Routing feature. Status Only **[2]** = The Alternative Routing feature is disabled. A read-only information on the quality of service of the destination IP addresses is provided. For information on the Alternative Routing feature, refer to Section 8.7 on page 231.
Alt Routing Tel to IP Mode **[AltRoutingTel2IPMode]**	None **[0]** = Alternative routing is not used. Conn **[1]** = Alternative routing is performed if ping to initial destination failed. QoS **[2]** = Alternative routing is performed if poor quality of service was detected. Both **[3]** = Alternative routing is performed if, either ping to initial destination failed, or poor quality of service was detected, or DNS host name is not resolved (default). **Note:** QoS (Quality of Service) is quantified according to delay and packet loss, calculated according to previous calls. QoS statistics are reset if no new data is received for two minutes. For information on the Alternative Routing feature, refer to Section 8.7 on page 231.
Max Allowed Packet Loss for Alt Routing [%] **[IPConnQoSMaxAllowedPL]**	Packet loss percentage at which the IP connection is considered a failure. The range is 1% to 20%. The default value is 20%.
Max Allowed Delay for Alt Routing [msec] **[IPConnQoSMaxAllowedDelay]**	Transmission delay (in msec) at which the IP connection is considered a failure. The range is 100 to 1000. The default value is 250 msec.

5.5.5.2 Tel to IP Routing Table

The Tel to IP Routing Table is used to route incoming Tel calls to IP addresses. This routing table associates a called / calling telephone number's prefixes with a destination IP address or with an FQDN (Fully Qualified Domain Name). When a call is routed through the VoIP gateway (Proxy isn't used), the called and calling numbers are compared to the list of prefixes on the IP Routing Table (up to 50 prefixes can be configured); Calls that match these prefixes are sent to the corresponding IP address. If the number dialed does not match these prefixes, the call is not made.

When using a Proxy server, you do not need to configure the Tel to IP Routing Table. However, if you want to use fallback routing when communication with Proxy servers is lost, or to use the 'Filter Calls to IP' and 'IP Security' features, or to obtain different SIP URI host names (per called number) or to assign IP profiles, you need to configure the IP Routing Table.

Note that for the Tel to IP Routing table to take precedence over a Proxy for routing calls, set the parameter 'PreferRouteTable' to 1. The gateway checks the 'Destination IP Address' field in the 'Tel to IP Routing' table for a match with the outgoing call. Only if a match is not found, a Proxy is used.

Possible uses for Tel to IP Routing can be as follows:

- Can fallback to internal routing table if there is no communication with the Proxy servers.
- Call Restriction – (when Proxy isn't used), reject all outgoing Tel→IP calls that are associated with the destination IP address: 0.0.0.0.
- IP Security – When the IP Security feature is enabled (SecureCallFromIP = 1), the VoIP gateway accepts only those IP→Tel calls with a source IP address identical to one of the IP addresses entered in the Tel to IP Routing Table.
- Filter Calls to IP – When a Proxy is used, the gateway checks the Tel→IP routing table before a telephone number is routed to the Proxy. If the number is not allowed (number isn't listed or a Call Restriction routing rule was applied), the call is released.
- Always Use Routing Table – When this feature is enabled (AlwaysUseRouteTable = 1), even if a Proxy server is used, the SIP URI host name in the sent INVITE message is obtained from this table. Using this feature users are able to assign a different SIP URI host name for different called and/or calling numbers.
- Assign Profiles to destination address (also when a Proxy is used).
- Alternative Routing – (When Proxy isn't used) an alternative IP destination for telephone number prefixes is available. To associate an alternative IP address to called telephone number prefix, assign it with an additional entry (with a different IP address), or use an FQDN that resolves to two IP addresses. Call is sent to the alternative destination when one of the following occurs:
  - No ping to the initial destination is available, or when poor QoS (delay or packet loss, calculated according to previous calls) is detected, or when a DNS host name is not resolved. For detailed information on Alternative Routing, refer to Section 8.7 on page 231.
  - When a release reason that is defined in the 'Reasons for Alternative Tel to IP Routing' table is received. For detailed information on the 'Reasons for Alternative Routing Tables', refer to Section 5.5.5.5 on page 106.

  Alternative routing (using this table) is commonly implemented when there is no response to an INVITE message (after INVITE retransmissions). The gateway then issues an internal 408 'No Response' implicit release reason. If this reason is included in the 'Reasons for Alternative Routing' table, the gateway immediately initiates a call to the redundant destination using the next matched entry in the 'Tel to IP Routing' table. Note that if a domain name in this table is resolved to two IP addresses, the timeout for INVITE retransmissions can be reduced by using the parameter 'Number of RTX Before Hotswap'.

Tip:	Tel to IP routing can be performed either before or after applying the number manipulation rules. To control when number manipulation is done, set the Tel to IP Routing Mode parameter (described in Table 5-17).

➢ To configure the Tel to IP Routing table, take these 6 steps:

1. Open the 'Tel to IP Routing' screen (**Protocol Management** menu > **Routing Tables** submenu > **Tel to IP Routing** option); the 'Tel to IP Routing' screen is displayed (shown in Figure 5-17).
2. In the 'Tel to IP Routing Mode' field, select the Tel to IP routing mode (refer to Table 5-17).
3. In the 'Routing Index' drop-down list, select the range of entries that you want to edit.
4. Configure the Tel to IP Routing table according to Table 5-17.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Figure 5-17: Tel to IP Routing Table Screen

	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Profile ID	Status	Charge Code
1	10	100	10.33.45.63	1	OK	1
2	20	*	10.33.45.60	1	QOS Low	1
3	[3,4,6]	*	10.33.45.64	1	OK	2
4	54324	[1,2]	domain.com	1	Dns Error	3
5	9	*	0.0.0.0	2	n/a	3
6	8xx#	*	10.13.77.7	1	Ping Error	4
7	*	*	10.13.77.7	1	OK	5
8						
9						
10						

Table 5-17: Tel to IP Routing Table (continues on pages 101 to 102)

Parameter	**Description**
Tel to IP Routing Mode **[RouteModeTel2IP]**	Route calls before manipulation **[0]** = Tel→IP calls are routed before the number manipulation rules are applied (default). Route calls after manipulation **[1]** = Tel→IP calls are routed after the number manipulation rules are applied. **Note:** Not applicable if Proxy routing is used.
Destination Phone Prefix	Each entry in the Destination Phone Prefix fields represents a called telephone number prefix. The prefix can be 1 to 19 digits long. An asterisk (*) represents all numbers.
Source Phone Prefix	Each entry in the Source Phone Prefix fields represents a calling telephone number prefix. The prefix can be 1 to 19 digits long. An asterisk (*) represents all numbers.
Any telephone number whose destination number matches the prefix defined in the 'Destination Phone Prefix' field *and* its source number matches the prefix defined in the adjacent 'Source Phone Prefix' field, is sent to the IP address entered in the 'IP Address' field. Note that Tel to IP routing can be performed according to a combination of source and destination phone prefixes, or using each independently. **Note 1:** An additional entry of the same prefixes can be assigned to enable alternative routing. **Note 2:** For available notations that represent multiple numbers, refer to Section 5.5.3.1 on page 95.	

Table 5-17: Tel to IP Routing Table (continues on pages 101 to 102)

Parameter	Description
Destination IP Address	In each of the IP Address fields, enter the IP address (and optionally port number) that is assigned to these prefixes. Domain names, such as domain.com, can be used instead of IP addresses. For example: <IP Address>:<Port> To discard outgoing IP calls, enter 0.0.0.0 in this field. **Note:** When using domain names, you must enter a DNS server IP address, or alternatively define these names in the 'Internal DNS Table'.
Profile ID	Enter the number of the IP profile that is assigned to the destination IP address defined in the 'Destination IP Address' field.
Status	A read only field representing the quality of service of the destination IP address. N/A = Alternative Routing feature is disabled. OK = IP route is available Ping Error = No ping to IP destination, route is not available QoS Low = Bad QoS of IP destination, route is not available DNS Error = No DNS resolution (only when domain name is used instead of an IP address).
Charge Code	An optional Charge Code (1 to 25) can be applied to each routing rule to associate it with an entry in the Charge Code table (refer to Section 5.5.2.3.1 on page 88).
Parameter Name in *ini* File	**Parameter Format**
Prefix	Prefix = <Destination Phone Prefix>, <IP Address>,<Src Phone Prefix>,<IP Profile ID>,<Charge Code> For example: Prefix = 20,10.2.10.2,202,1,15 Prefix = 10[340-451]xxx#,10.2.10.6,*,1,1 Prefix = *,gateway.domain.com,*,20 **Note 1:** <destination / source phone prefix> can be single number or a range of numbers. For available notations, refer to Section 5.5.3.1 on page 95. **Note 2:** This parameter can appear up to 50 times. **Note 3:** Parameters can be skipped by using the sign '$$', for example: Prefix = $$,10.2.10.2,202,1 **Note 4:** An optional IP ProfileID (1 to 9) can be applied to each routing rule.

5.5.5.3 IP to Hunt Group Routing

The IP to Hunt Group Routing Table is used to route incoming IP calls to groups of channels called hunt groups. Calls are assigned to hunt groups according to any combination of the following three options (or using each independently):

- Destination phone prefix
- Source phone prefix
- Source IP address

The call is then sent to the VoIP gateway channels assigned to that hunt group. The specific channel, within a hunt group, that is assigned to accept the call is determined according to the hunt group's channel selection mode which is defined in the Hunt Group Settings table (Section 5.5.8 on page 117) or according to the global parameter 'ChannelSelectMode' (refer to Table 5-8 on page 78). Hunt groups can be used on both FXO and FXS gateways; however, usually they are used with FXO gateways.

Note: When a release reason that is defined in the 'Reasons for Alternative IP to Tel Routing' table is received for a specific IP→Tel call, an alternative hunt group for that call is available. To associate an alternative hunt group to an incoming IP call, assign it with an additional entry in the 'IP to Hunt Group Routing' table (repeat the same routing rules with a different hunt group ID). For detailed information on the 'Reasons for Alternative Routing Tables', refer to Section 5.5.5.5 on page 106.

To use hunt groups you must also do the following.

- You must assign a hunt group ID to the VoIP gateway channels on the Endpoint Phone Number screen. For information on how to assign a hunt group ID to a channel, refer to Section 5.5.7 on page 115.
- You can configure the Hunt Group Settings table to determine the method in which new calls are assigned to channels within the hunt groups (a different method for each hunt group can be configured). For information on how to enable this option, refer to Section 5.5.8 on page 117. If a Channel Select Mode for a specific hunt group isn't specified, then the global 'Channel Select Mode' parameter (defined in 'General Parameters' screen under 'Advanced Parameters') applies.

➢ To configure the IP to Hunt Group Routing table, take these 6 steps:

1. Open the 'IP to Hunt Group Routing' screen (**Protocol Management** menu > **Routing Tables** submenu > **IP to Hunt Group Routing** option); the 'IP to Hunt Group Routing' table screen is displayed (shown in Figure 5-18).

Figure 5-18: IP to Hunt Group Routing Table Screen

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Hunt Group ID	Profile ID
1	10	*	0	1	2
2	20	101	0	1	2
3					
4					
5	[5010-5020]	*	0	3	1
6	6xx	*	0	3	1
7	71234#	*	0	3	1
8	*	*	0	4	3
9					

2. In the 'IP to Tel Routing Mode' field, select the IP to Tel routing mode (refer to Table 5-18).
3. In the 'Routing Index' drop-down list, select the range of entries that you want to edit (up to 24 entries can be configured).
4. Configure the IP to Hunt Group Routing table according to Table 5-18.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-18: IP to Hunt Group Routing Table (continues on pages 103 to 104)

Parameter	Description
IP to Tel Routing Mode **[RouteModeIP2Tel]**	Route calls before manipulation **[0]** = IP→Tel calls are routed before the number manipulation rules are applied (default). Route calls after manipulation **[1]** = IP→Tel calls are routed after the number manipulation rules are applied.
Destination Phone Prefix	Each entry in the Destination Phone Prefix fields represents a called telephone number prefix. The prefix can be 1 to 49 digits long. An asterisk (*) represents all numbers.
Source Phone Prefix	Each entry in the Source Phone Prefix fields represents a calling telephone number prefix. The prefix can be 1 to 49 digits long. An asterisk (*) represents all numbers.

Table 5-18: IP to Hunt Group Routing Table (continues on pages 103 to 104)

Parameter	**Description**
Source IP Address	Each entry in the Source IP Address fields represents the source IP address of an IP→Tel call (obtained from the Contact header in the INVITE message). **Note:** The source IP address can include the 'x' wildcard to represent single digits. For example: 10.8.8.xx represents all the addresses between 10.8.8.10 to 10.8.8.99.
Any SIP incoming call whose destination number matches the prefix defined in the 'Destination Phone Prefix' field *and* its source number matches the prefix defined in the adjacent 'Source Phone Prefix' field *and* its source IP address matches the address defined in the 'Source IP Address' field, is assigned to the hunt group entered in the field to the right of these fields. Note that IP to hunt group routing can be performed according to any combination of source / destination phone prefixes and source IP address, or using each independently. **Note:** For available notations that represent multiple numbers (used in the prefix columns), refer to Section 5.5.3.1 on page 95.	
Hunt Group ID	In each of the Hunt Group ID fields, enter the hunt group ID to which calls that match these prefixes are assigned.
Profile ID	Enter the number of the IP profile that is assigned to the routing rule.
Parameter Name in *ini* File	**Parameter Format**
PSTNPrefix	PSTNPrefix = a,b,c,d,e a = Destination Number Prefix b = Hunt Group ID c = Source Number Prefix d = Source IP address (obtained from the Contact header in the INVITE message) e = IP Profile ID Selection of hunt groups (for IP to Tel calls) is according to destination number, source number and source IP address. **Note 1:** To support the 'in call alternative routing' feature, users can use two entries that support the same call, but assigned it with a different hunt groups. The second entree functions as an alternative selection if the first rule fails as a result of one of the release reasons listed in the AltRouteCauseIP2Tel table. **Note 2:** An optional IP ProfileID (1 to 4) can be applied to each routing rule. **Note 3:** The Source IP Address can include the 'x' wildcard to represent single digits. For example: 10.8.8.xx represents all IP addresses between 10.8.8.10 to 10.8.8.99. The '*' wildcard represents any number between 0 and 255, e.g., 10.8.8.* represents all addresses between 10.8.8.0 and 10.8.8.255. **Note 4:** For available notations that represent multiple numbers, refer to Section 5.5.3.1 on page 95. **Note 5:** This parameter can appear up to 24 times.

5.5.5.4 Internal DNS Table

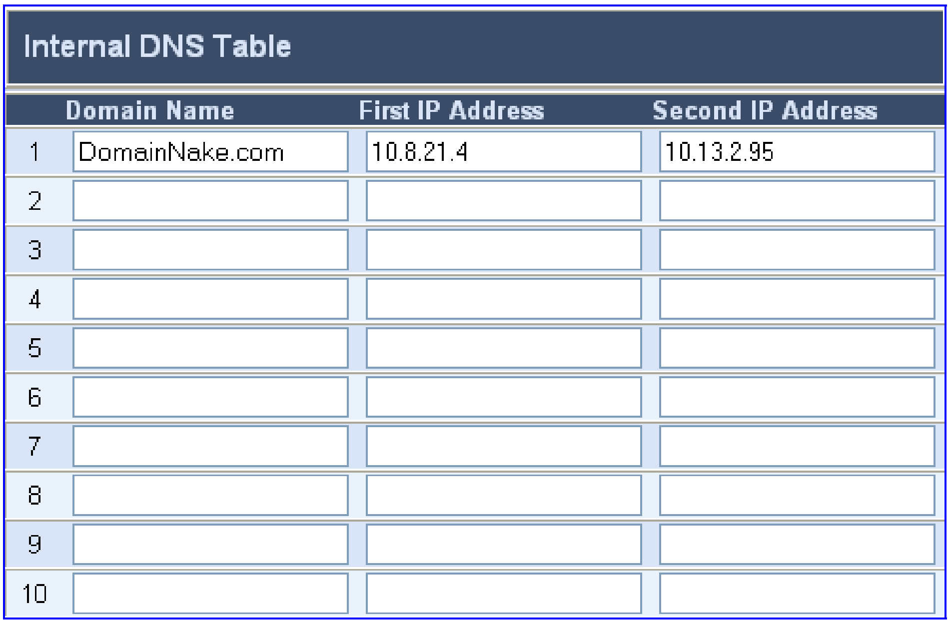
The internal DNS table, similar to a DNS resolution, translates hostnames into IP addresses. This table is used when hostname translation is required (e.g., 'Tel to IP Routing' table). Two different IP addresses can be assigned to the same hostname. If the hostname isn't found in this table, the gateway communicates with an external DNS server.

Assigning two IP addresses to hostname can be used for alternative routing (using the 'Tel to IP Routing' table).

➢ To configure the internal DNS table, take these 7 steps:

1. Open the 'Internal DNS Table' screen (**Protocol Management** menu > **Routing Tables** submenu > **Internal DNS Table** option); the 'Internal DNS Table' screen is displayed.

Figure 5-19: Internal DNS Table Screen



2. In the 'Domain Name' field, enter the hostname to be translated. You can enter a string up to 31 characters long.
3. In the 'First IP Address' field, enter the first IP address that the hostname is translated to.
4. In the 'Second IP Address' field, enter the second IP address that the hostname is translated to.
5. Repeat steps 2 to 4, for each Internal DNS Table entry.
6. Click the **Submit** button to save your changes.
7. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-19: Internal DNS *ini* File Parameter

Parameter Name in *ini* File	Parameter Format
DNS2IP	DNS2IP = <Hostname>, <first IP address>, <second IP address> For example: DNS2IP = Domainname.com, 10.8.21.4, 10.13.2.95 **Note:** This parameter can appear up to 10 times.

5.5.5.5 Internal SRV Table

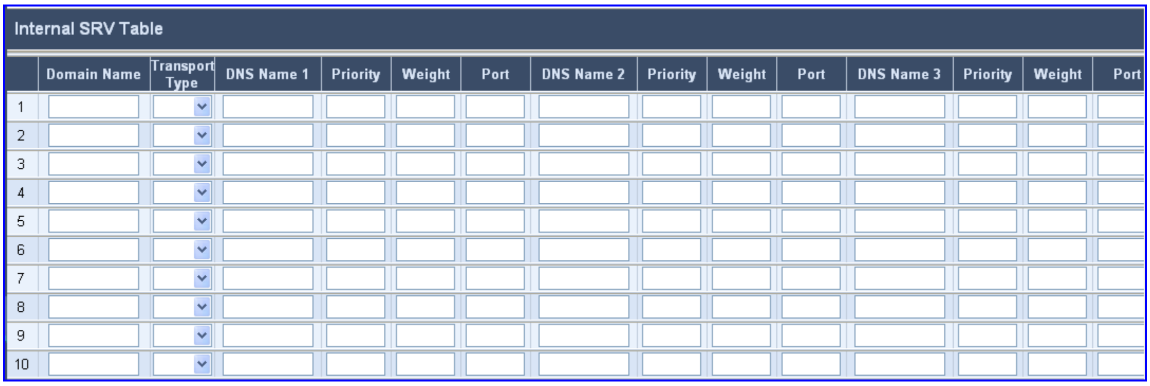
The Internal SRV table is used for resolving host names to DNS A-Records. Three different A-Records can be assigned to a hostname. Each A-Record contains the host name, priority, weight, and port.

> **Note:** If the Internal SRV table is configured, the gateway first tries to resolve a domain name using this table. If the domain name isn't found, the gateway performs an SRV resolution using an external DNS server.

➢ To configure the Internal SRV table, take these 9 steps:

1. Open the 'Internal SRV Table' screen (**Protocol Management** menu > **Routing Tables** submenu > **Internal SRV Table** option); the 'Internal SRV Table' screen is displayed.

Figure 5-20: Internal SRV Table Screen



2. In the 'Domain Name' field, enter the hostname to be translated. You can enter a string up to 31 characters long.
3. From the 'Transport Type' drop-down list, select a transport type.
4. In the 'DNS Name 1' field, enter the first DNS A-Record to which the hostname is translated.
5. In the 'Priority', 'Weight' and 'Port' fields, enter the relevant values\
6. Repeat steps 4 to 5, for the second and third DNS names, if required.
7. Repeat steps 2 to 6, for each Internal SRV Table entry.
8. Click the **Submit** button to save your changes.
9. To save the changes so they are available after a hardware reset or power fail, refer to Section 5.10.2 on page 205.

Table 5-20: Internal SRV *ini* File Parameter

Parameter Name in *ini* File	Parameter Format
SRV2IP	SRV2IP = <Internal Domain Name>, <Transport Type>, <DNS Name 1>, <Priority 1>, <Weight 1>, <Port 1>, <DNS Name 2>, <Priority 2>, <Weight 2>, <Port 2>, <DNS Name 3>, <Priority 3>, <Weight 3>, <Port 3> **Note 1**: If the internal SRV table is configured, the gateway first tries to resolve a domain name using this table. If the domain name isn't found, the gateway performs an SRV resolution using an external DNS server. **Note 2**: This parameter can appear up to 10 times.

5.5.5.6 Reasons for Alternative Routing

The Reasons for Alternative Routing screen includes two tables (Tel→IP and IP→Tel). Each table enables you to define up to 4 different release reasons. If a call is released as a result of one of these reasons, the gateway tries to find an alternative route to that call. The release reason for IP→Tel calls is provided in Q.931 notation. The release reason for Tel→IP calls is provided in SIP 4xx, 5xx and 6xx response codes. For Tel→IP calls an alternative IP address, for IP→Tel calls an alternative hunt group.

Refer to 5.5.5.2 on page 100 for information on defining an alternative IP address. Refer to the 5.5.5.3 on page 102 for information on defining an alternative hunt group.

You can use this table for example:

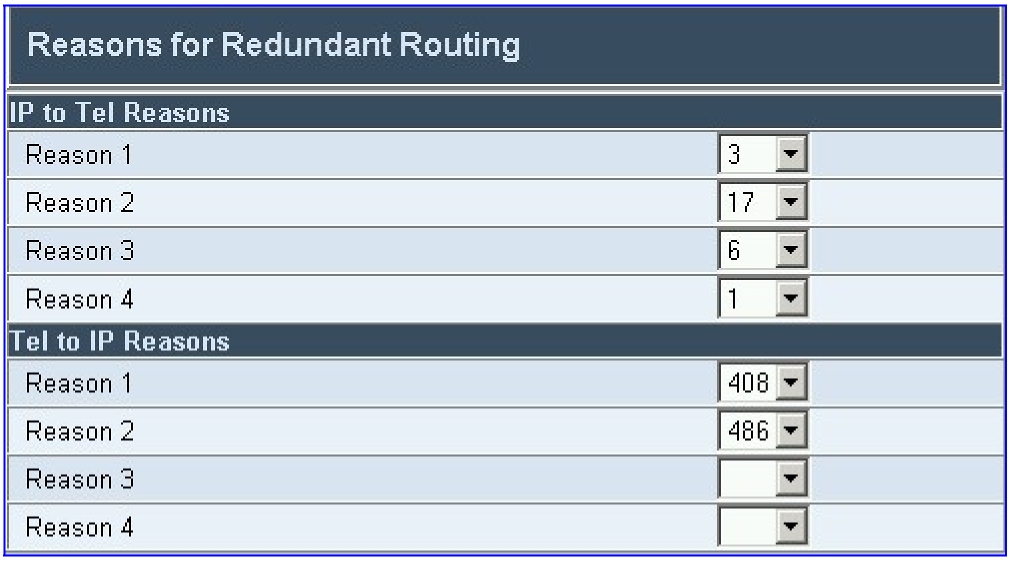
- For Tel→IP calls, when there is no response to an INVITE message (after INVITE retransmissions), and the gateway then issues an internal 408 'No Response' implicit release reason.
- For IP→Tel calls, when the destination is busy, and release reason #17 is issued or for other call releases that issue the default release reason (#3). Refer to 'DefaultReleaseCause' in Table 5-8.

Note: The reasons for alternative routing option for Tel→IP calls, only applies when Proxy isn't used.

➢ **To configure the reasons for alternative routing, take these 5 steps:**

1. Open the 'Reasons for Alternative Routing' screen (**Protocol Management** menu > **Routing Tables** submenu > **Reasons for Alternative Routing** option); the 'Reasons for Alternative Routing' screen is displayed.

Figure 5-21: Reasons for Alternative Routing Screen



2. In the 'IP to Tel Reasons' table, from the drop-down list select up to 4 different call failure reasons that invoke an alternative IP to Tel routing.
3. In the 'Tel to IP Reasons' table, from the drop-down list select up to 4 different call failure reasons that invoke an alternative Tel to IP routing.

4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-21: Reasons for Alternative Routing *ini* File Parameter

Parameter Name in *ini* File	Parameter Format
AltRouteCauseTel2IP	AltRouteCauseTel2IP = <SIP Call failure reason from IP> For example: AltRouteCauseTel2IP = 408 (Response timeout). AltRouteCauseTel2IP = 486 (User is busy). **Note:** This parameter can appear up to 4 times.
AltRouteCauseIP2Tel	AltRouteCauseIP2Tel = <Call failure reason from Tel> For example: AltRouteCauseIP2Tel = 3 (No route to destination). AltRouteCauseIP2Tel = 17 (Busy here). **Note:** This parameter can appear up to 4 times.

5.5.6 Configuring the Profile Definitions

Utilizing the Profiles feature, the MediaPack provides high-level adaptation when connected to a variety of equipment (from both Tel and IP sides) and protocols, each of which requires a different system behavior. Using Profiles, users can assign different Profiles (behavior) on a per-call basis, using the Tel to IP and IP to Hunt Group Routing tables, or associate different Profiles to the gateway's endpoint(s). The Profiles contain parameters such as Coders, T.38 Relay, Voice and DTMF Gains, Silence Suppression, Echo Canceler, RTP DiffServ, Current Disconnect and more. The Profiles feature allows users to tune these parameters or turn them on or off, per source or destination routing and/or the specific gateway or its ports. For example, specific ports can be designated to have a profile which always uses G.711.

Each call can be associated with one or two Profiles, Tel Profile and (or) IP Profile. If both IP and Tel profiles apply to the same call, the coders and other common parameters of the preferred Profile (determined by the Preference option) are applied to that call. If the Preference of the Tel and IP Profiles is identical, the Tel Profile parameters are applied.

> **Note:** The default values of the parameters in the Tel and IP Profiles are identical to the Web/*ini* file parameter values. If a value of a parameter is changed in the Web/*ini* file, it is automatically updated in the Profiles correspondingly. After any parameter in the Profile is modified by the user, modifications to parameters in the Web/*ini* file no longer impact that Profile.

5.5.6.1 Coder Group Settings

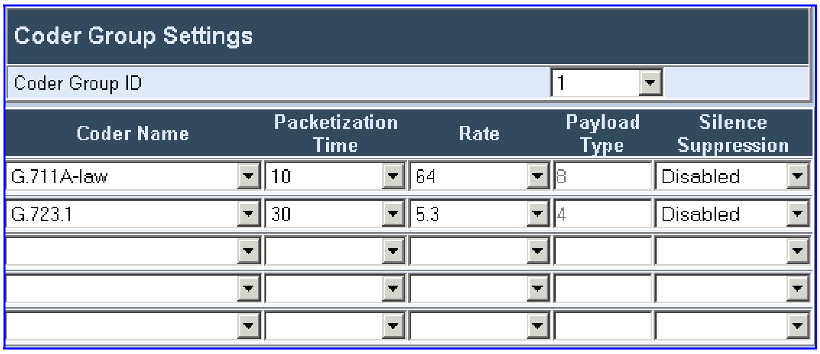
Use the Coder Group Settings screen to define up to four different coder groups. These coder groups are used in the Tel and IP Profile Settings screens to assign different coders to Profiles.

For each group you can define the first to fifth preferred coders (and their attributes) for the gateway. The first coder is the highest priority coder and is used by the gateway whenever possible. If the far end gateway cannot use the coder assigned as the first coder, the gateway attempts to use the next coder and so forth.

➢ To configure the coder group settings, take these 11 steps:

1. Open the 'Coder Group Settings' screen (**Protocol Management** menu > **Profile Definitions** submenu > **Coder Group Settings** option); the 'Coder Group Settings' screen is displayed.

Figure 5-22: Coder Group Settings Screen



2. From the 'Coder Group ID' drop-down list, select the coder group you want to edit (up to four coder groups can be configured).
3. From the 'Coder Name' drop-down list, select the coder you want to use. For the full list of available coders and their corresponding attributes, refer to Table 5-5.
4. From the 'Packetization Time' drop-down list, select the packetization time (in msec) for the coder you selected. The packetization time determines how many coder payloads are combined into a single RTP packet. The ptime specifies the packetization time the gateway expects to receive. The gateway always uses the ptime requested by the remote side for sending RTP packets.
5. From the 'Rate' drop-down list, select the bit rate (in kbps) for the coder you selected.
6. In the 'Payload Type' field, if the payload type for the coder you selected is dynamic, enter a value from 0 to 120 (payload types of 'well-known' coders cannot be modified). The payload type identifies the format of the RTP payload.
7. From the 'Silence Suppression' drop-down list, enable or disable the silence suppression option for the coder you selected.
8. Repeat steps 3 to 7 for the second to fifth coders (optional).
9. Repeat steps 2 to 8 for the second to fourth coder groups (optional).
10. Click the **Submit** button to save your changes.
11. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Notes:

- Each coder can appear only once.
- If not specified, the ptime gets a default value. The ptime specifies the packetization time the gateway expects to receive. The gateway always uses the ptime requested by the remote side for sending RTP packets.
- If payload type is not specified, a default is used.
- For G.729 it is also possible to select silence suppression without adaptations
- Only the ptime of the first coder in the defined coder list is declared in INVITE / 200 OK SDP, even if multiple coders are defined.
- If coder G.729 is selected and silence suppression disabled, the gateway includes the string 'annexb=no' in the SDP of the relevant SIP messages. If silence suppression is enabled or set to 'Enable w/o Adaptations', 'annexb=yes' is included. An exception to this logic is when the remote gateway is a Cisco device (IsCiscoSCEMode).

Table 5-22: *ini* File Coder Group Parameter

Parameter	Description
CoderName_ID	Defines groups of coders that can be associated with IP or Tel profiles (up to five coders in each group). Enter coder groups in the following format: CoderName_<coder group ID from 1 to 4>=<Coder Name>,<Ptime>,<Rate>,<Payload Type>,<Silence Suppression Mode>. **Note 1:** This parameter (CoderName_ID) can appear up to 20 times (five coders in four coder groups). **Note 2:** The coder name is case-sensitive. **Note 3:** If silence suppression is not defined (for a specific coder), the value defined by the parameter EnableSilenceCompression is used. **Note 4:** The value of several fields is hard-coded according to well-known standards (e.g., the payload type of G.711 U-law is always 0). Other values can be set dynamically. If no value is specified for a dynamic field, a default value is assigned. If a value is specified for a hard-coded field, the value is ignored. For example: CoderName_1 = g711Alaw64k,20,,,0 CoderName_1 = g711Ulaw64k,40 CoderName_1 = g7231,90,1,,1 CoderName_2 = g726,$$,2,,0

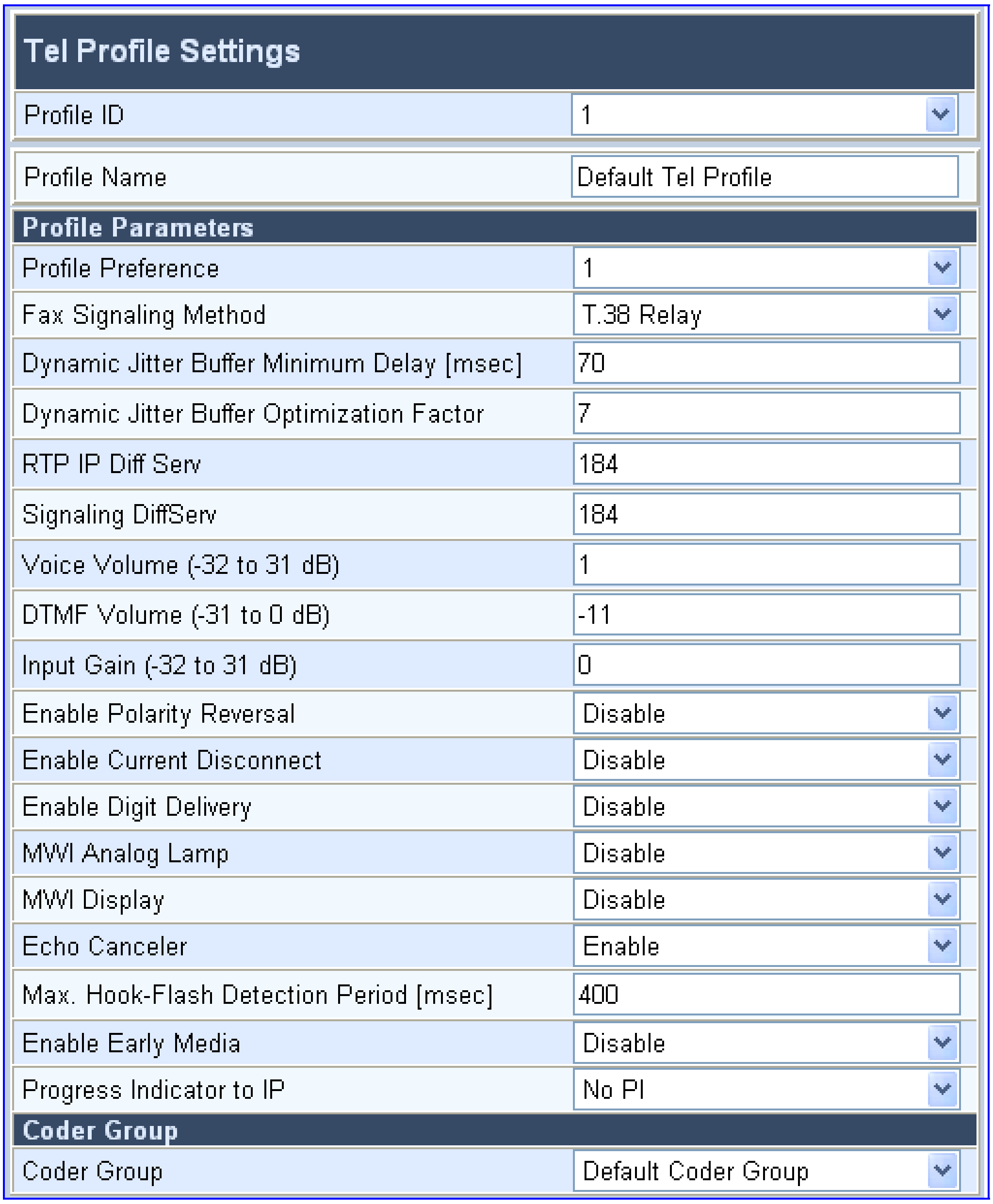
5.5.6.2 Tel Profile Settings

Use the Tel Profile Settings screen to define up to four different Tel Profiles. These Profiles are used in the ‘Endpoint Phone Number’ table to associate different Profiles to gateway’s endpoints, thereby applying different behavior to different MediaPack ports.

➢ **To configure the Tel Profile settings, take these 9 steps:**

1. Open the ‘Tel Profile Settings’ screen (**Protocol Management** menu > **Profile Definitions** submenu > **Tel Profile Settings** option); the ‘Tel Profile Settings’ screen is displayed.

Figure 5-23: Tel Profile Settings Screen



2. From the ‘Profile ID’ drop-down list, select the Tel Profile you want to edit (up to four Tel Profiles can be configured).
3. In the ‘Profile Name’ field, enter a name that enables you to identify the Profile intuitively and easily.

4. From the 'Profile Preference' drop-down list, select the preference (1-20) of the current Profile. The preference option is used to determine the priority of the Profile. Where '20' is the highest preference value. If both IP and Tel profiles apply to the same call, the coders and other common parameters (noted by an asterisk in the description of the parameter TelProfile_ID) of the preferred Profile are applied to that call. If the Preference of the Tel and IP Profiles is identical, the Tel Profile parameters are applied.
   Note: If the coder lists of both IP and Tel Profiles apply to the same call, an intersection of the coders is performed (i.e., only common coders remain). The order of the coders is determined by the preference.
5. Configure the Profile's parameters according to your requirements. For detailed information on each parameter, refer to the description of the screen in which it is configured as an individual parameter.
6. From the 'Coder Group' drop-down list, select the coder group you want to assign to that Profile. You can select the gateway's default coders (refer to Section 5.5.1.3 on page 72) or one of the coder groups you defined in the Coder Group Settings screen (refer to Section 5.5.6.1 on page 108).
7. Repeat steps 2 to 6 for the second to fifth Tel Profiles (optional).
8. Click the **Submit** button to save your changes.
9. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-23: *ini* File Tel Profile Settings

Parameter	Description
TelProfile_ID	TelProfile_<Profile ID> = <Profile Name>,<Preference>,<Coder Group ID>,<IsFaxUsed *>,<DJBufMinDelay *>, <DJBufOptFactor *>,<IPDiffServ *>,<ControlIPDiffServ*>,<DtmfVolume>,<InputGain>, <VoiceVolume>,<EnableReversePolarity>,<EnableCurrentDisconnect>, <EnableDigitDelivery>, <ECE>, <MWIAnalogLamp>, <MWIDisplay>, <FlashHookPeriod>, <EnableEarlyMedia*>, <ProgressIndicator2IP*> For examples: TelProfile_1 = FaxProfile,1,1,1,40,13,22,33,$$,$$,$$,0,0,0,1,0,0,$$,0,$$ TelProfile_2 = ModemProfile,2,2,0,40,13,$$,$$,$$,$$,$$,$$,$$,$$,$$,0,0,0,$$,0,$$ $$ = Not configured, the default value of the parameter is used. (*) = Common parameter used in both IP and Tel profiles. **Note:** This parameter can appear up to 9 times (ID = 1 to 9).

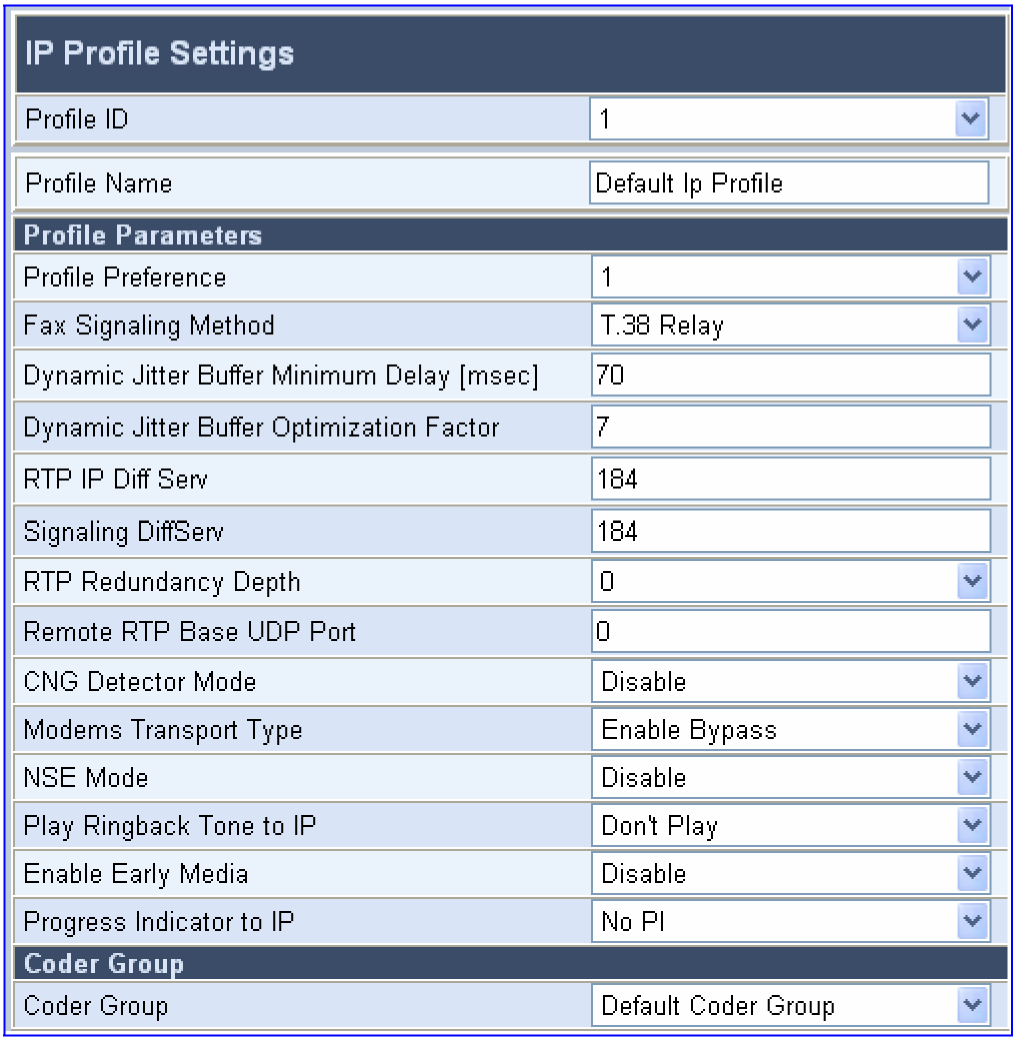
5.5.6.3 IP Profile Settings

Use the IP Profile Settings screen to define up to four different IP Profiles. These Profiles are used in the Tel to IP and IP to Hunt Group Routing tables to associate different Profiles to routing rules. IP Profiles can also be used when working with Proxy server (set 'AlwaysUseRouteTable' to 1).

➢ To configure the IP Profile settings, take these 9 steps:

1. Open the 'IP Profile Settings' screen (**Protocol Management** menu > **Profile Definitions** submenu > **IP Profile Settings** option); the 'IP Profile Settings' screen is displayed.

Figure 5-24: IP Profile Settings Screen



2. From the 'Profile ID' drop-down list, select the IP Profile you want to edit (up to four IP Profiles can be configured).
3. In the 'Profile Name' field, enter a name that enables you to identify the Profile intuitively and easily.

4. From the ‘Profile Preference’ drop-down list, select the preference (1-20) of the current Profile. The preference option is used to determine the priority of the Profile. Where ‘20’ is the highest preference value. If both IP and Tel profiles apply to the same call, the coders and other common parameters (noted by an asterisk in the description of the parameter IPProfile_ID) of the preferred Profile are applied to that call. If the Preference of the Tel and IP Profiles is identical, the Tel Profile parameters are applied.
   Note: If the coder lists of both IP and Tel Profiles apply to the same call, an intersection of the coders is performed (i.e., only common coders remain). The order of the coders is determined by the preference.
5. Configure the Profile’s parameters according to your requirements. For detailed information on each parameter, refer to the description of the screen in which it is configured as an individual parameter.
6. From the ‘Coder Group’ drop-down list, select the coder group you want to assign to that Profile. You can select the gateway’s default coders (refer to Section 5.5.1.3 on page 72) or one of the coder groups you defined in the Coder Group Settings screen (refer to Section 5.5.6.1 on page 108).
7. Repeat steps 2 to 6 for the second to fifth IP Profiles (optional).
8. Click the **Submit** button to save your changes.
9. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-24: *ini* File IP Profile Settings

Parameter	Description
IPProfile_ID	IPProfile_<Profile ID> = <Profile Name>,<Preference>,<Coder Group ID>,<IsFaxUsed *>,<DJBufMinDelay *>, <DJBufOptFactor *>,<IpDiffServ *>,<ControlIPDiffServ *>,<N/A use $$ instead>, <RTPRedundancyDepth>, <RemoteBaseUDPPort>, <CNGmode>, <VxxTransportType>, <NSEMode>, <PlayRBTone2IP>, <EnableEarlyMedia*>, <ProgressIndicator2IP*> For example: IPProfile_1 = name1,2,1,0,10,13,15,44,1,1,6000,0,2,0,0,1,0 IPProfile_2 = name2,$$,$$,$$,$$,$$,$$,$$,$$,1,$$,$$,$$,$$,$$,$$,$$ $$ = Not configured, the default value of the parameter is used. (*) = Common parameter used in both IP and Tel profiles. **Note:** This parameter can appear up to 9 times (ID = 1 to 9).

5.5.7 Configuring the Endpoint Phone Numbers

From the 'Endpoint Phone Number Table' screen you can enable and assign telephone numbers, hunt groups (optional) and profiles to the VoIP gateway ports.

➢ To configure the Endpoint Phone Number table, take these 4 steps:

1. Open the ‘Endpoint Phone Number Table’ screen (**Protocol Management** menu > **Endpoint Phone Numbers**); the ‘Endpoint Phone Number Table’ screen is displayed.

Figure 5-25: Endpoint Phone Number Table Screen

Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Profile ID
1 FXS	1-4	201	1	1
2 FXS				
3 FXS				
4 FXS				
5 FXO	5	301		
6 FXO				
7 FXO				
8 FXO				

2. Configure the endpoint phone numbers according to Table 5-25. You must enter a number in the' Phone Number' fields for each port that you want to use.
3. Click the **Submit** button to save your changes, or click the **Register** or **Un-Register** buttons to save your changes and to register / unregister to a Proxy / Registrar.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-25: Endpoint Phone Number Table (continues on pages 115 to 116)

Parameter	**Description**
Channel(s)	The numbers (1-8) in the Channel(s) fields represent the ports on the back of the VoIP gateway. To enable a VoIP gateway channel, you **must** enter the port number on this screen. [n-m] represents a range of ports. For example, enter [1-4] to specify the ports from 1 to 4.
Phone Number	In each of the Phone Number fields, enter the telephone number that is assigned to that channel. For a range of channels enter the first number in an ordered sequence. These numbers are also used for port allocation for IP to Tel calls, if the hunt group’s ‘Channel Select Mode’ is set to ‘By Phone Number’.

Table 5-25: Endpoint Phone Number Table (continues on pages 115 to 116)

Parameter	Description
Hunt Group ID	In each of the Hunt Group ID fields, enter the hunt group ID (1-99) assigned to the channel(s). The same hunt group ID can be used for more than one channel and in more than one field. The hunt group ID is an optional field that is used to define a group of common behavior channels that are used for routing IP to Tel calls. If an IP to Tel call is assigned to a hunt group, the call is routed to the channel or channels that correspond to the hunt group ID. You can configure the Hunt Group Settings table to determine the method in which new calls are assigned to channels within the hunt groups (refer to Section 5.5.8 on page 117). **Note:** If you enter a hunt group ID, you must configure the IP to Hunt Group Routing Table (assigns incoming IP calls to the appropriate hunt group). If you do not configure the IP to Hunt Group Routing Table, calls don't complete. For information on how to configure this table, refer to Section 5.5.5.3.
Profile ID	Enter the number of the Tel profile that is assigned to the endpoints defined in the 'Channel(s)' field.
Parameter Name in *ini* File	**Parameter Format**
TrunkGroup_x	TrunkGroup_<Hunt Group ID> = <Starting channel> - <Ending channel>, <Phone Number>, <Tel Profile ID> For example: TrunkGroup_1 = 1-4,100 TrunkGroup_2 = 5-8,200,1 **Note 1:** The numbering of channels starts with 1. **Note 2:** 'Hunt Group ID' can be set to any number in the range 1 to 99. **Note 3:** This parameter can appear up to 8 times for 8-port gateways and up to 24 times for MP-124 gateways. **Note 4:** An optional Tel ProfileID (1 to 4) can be applied to each group of channels.
ChannelList	This parameter is obsolete; use instead TrunkGroup_x.

5.5.8 Configuring the Hunt Group Settings

The Hunt Group Settings Table is used to determine the method in which new calls are assigned to channels within each hunt group. If such a rule doesn't exist (for a specific hunt group), the global rule, defined by the Channel Select Mode parameter (Protocol Definition > General Parameters), applies.

➢ To configure the Hunt Group Settings table, take these 8 steps:

1. Open the 'Hunt Group Settings' screen (**Protocol Management** menu > **Hunt Group Settings**); the 'Hunt Group Settings' screen is displayed.

Figure 5-26: Hunt Group Settings screen

	Hunt Group ID	Channel Select Mode	Registration Mode
1	1	Cyclic Ascending	Per Endpoint
2			
3			
4			
5			
6			
7			
8			
9			
10			
11			
12			

2. From the 'Routing Index' drop-down list, select the range of entries that you want to edit (up to 24 entries can be configured).
3. In the 'Hunt Group ID' field, enter the hunt group ID number.
4. From the 'Channel Select Mode' drop-down list, select the Channel Select Mode that determines the method in which new calls are assigned to channels within the hunt groups entered in the field to the right of this field. For information on available Channel Select Modes, refer to Table 5-26 on page 118.
5. From the 'Registration Mode' drop-down list, select the registration mode.
6. Repeat steps 4 and 5, for each defined hunt group.
7. Click the **Submit** button to save your changes.
8. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-26: Channel Select Modes

Mode	**Description**
By Dest Phone Number	Select the gateway port according to the called number (refer to the note below).
Cyclic Ascending	Select the next available channel in ascending cycle order. Always select the next higher channel number in the hunt group. When the gateway reaches the highest channel number in the hunt group, it selects the lowest channel number in the hunt group and then starts ascending again.
Ascending	Select the lowest available channel. Always start at the lowest channel number in the hunt group and if that channel is not available, select the next higher channel.
Cyclic Descending	Select the next available channel in descending cycle order. Always select the next lower channel number in the hunt group. When the gateway reaches the lowest channel number in the hunt group, it selects the highest channel number in the hunt group and then starts descending again.
Descending	Select the highest available channel. Always start at the highest channel number in the hunt group and if that channel is not available, select the next lower channel.
Dest Number + Cyclic Ascending	First select the gateway port according to the called number (refer to the note below). If the called number isn't found, then select the next available channel in ascending cyclic order. Note that if the called number is found, but the port associated with this number is busy, the call is released.
By Source Phone Number	Select the gateway port according to the calling number.
Parameter Name in *ini* File	**Parameter Format**
TrunkGroupSettings	Defines rules for port allocation for specific Hunt Groups. If no rule exists, the global rule defined by ChannelSelectMode applies. TrunkGroupSettings = <Hunt group ID>, <Channel select Mode>, <Registration Mode> The format is: a, b, c Where, a = Hunt Group ID number b = Channel select mode for that Hunt Group. c = Registration mode for that Hunt Group (Per Endpoint [0] or Per Hunt Group [1]). If not configured [-1], the value of AuthenticationMode is used. For example: TrunkGroupSettings = 1,5 **Note:** This parameter can appear up to 24 times.

Note: The gateway's port numbers are defined in the 'Endpoint Phone Numbers' table under the 'Phone Number' column. For detailed information on the 'Endpoint Phone Numbers' table, refer to Section 5.5.7 on page 115).

5.5.9 Configuring the Endpoint Settings

The Endpoint Settings screens enable you to configure port-specific parameters.

5.5.9.1 Authentication

The Authentication Table (normally used with FXS gateways) defines a username and password combination for authentication for each MediaPack port.

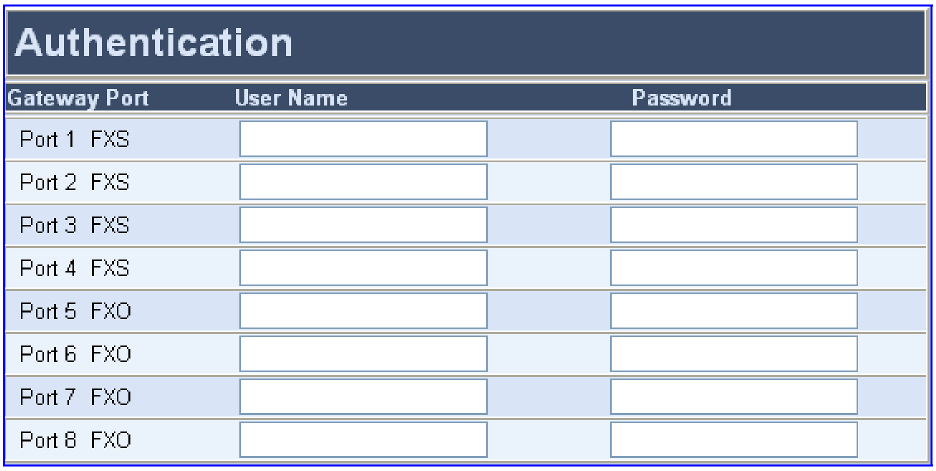
The 'Authentication Mode' parameter (described in Table 5-4) determines if authentication is performed per port or for the entire gateway. If authentication is performed for the entire gateway, this table is ignored.

Note: If either the username or password field is omitted, the port's phone number (defined in Table 5-25) and global password (refer to the parameter 'Password' described in Table 5-4) are used instead.

➢ To configure the Authentication Table, take these 6 steps:

1. Set the 'Authentication Mode' parameter to 'Authentication per Endpoint'.
2. Open the 'Authentication' screen (**Protocol Management** menu > **Endpoint Settings** > **Authentication** option); the 'Authentication' screen is displayed.

Figure 5-27: Authentication Screen



3. In the 'User Name' and 'Password' fields for a port, enter the username and password combination respectively.
4. Repeat Step 4 for each port.
5. Click the **Submit** button to save your changes.
6. To save the changes, refer to Section 5.10.2 on page 205.

Table 5-27: Authentication *ini* File Parameter

Parameter Name in *ini* File	Parameter Format
Authentication_x	Authentication_<Port> = <Username>,<Password> For example: Authentication_0 = david,14325 Authentication_1 = Alex,18552 **Note:** Using the sign '$$' enables the user to omit either the username or the password. For instance, Authentication_5 = $$, 152. In this case, endpoint 5's phone number is used instead of username.

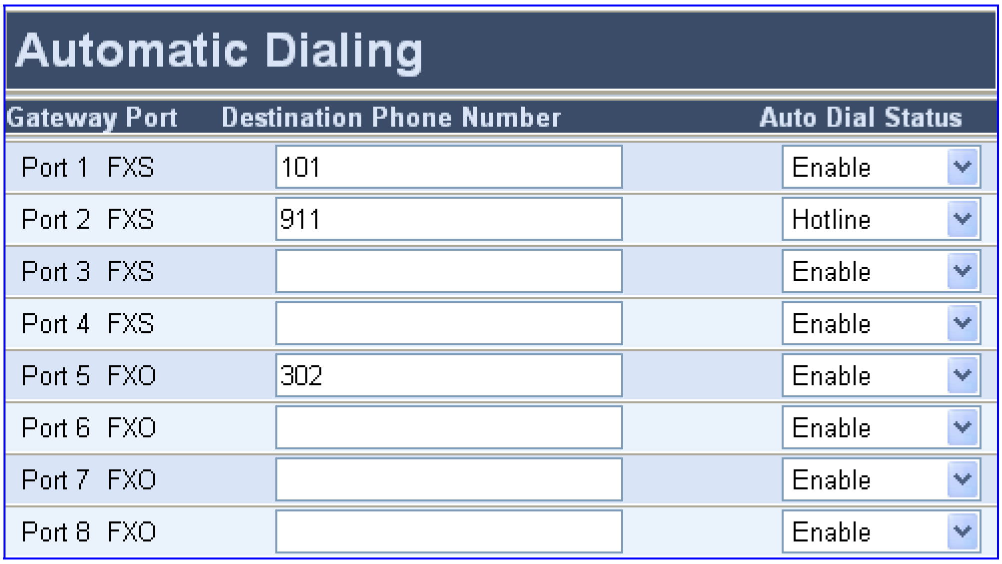
5.5.9.2 Automatic Dialing

Use the Automatic Dialing Table to define telephone numbers that are automatically dialed when a specific port is used.

➢ **To configure the Automatic Dialing table, take these 6 steps:**

1. Open the 'Automatic Dialing' screen (**Protocol Management** menu > **Endpoint Settings** submenu > **Automatic Dialing** option); the 'Automatic Dialing' screen is displayed.

Figure 5-28: Automatic Dialing Table Screen



Automatic Dialing

Gateway Port	Destination Phone Number	Auto Dial Status
Port 1 FXS	101	Enable
Port 2 FXS	911	Hotline
Port 3 FXS		Enable
Port 4 FXS		Enable
Port 5 FXO	302	Enable
Port 6 FXO		Enable
Port 7 FXO		Enable
Port 8 FXO		Enable

2. In the 'Destination Phone Number' field for a port, enter the telephone number to dial.
3. From the 'Auto Dial Status' drop-down list, select one of the following:
   - Enable **[1]:** When a port is selected, when making a call, the number in the Destination Phone Number field is automatically dialed if phone is offhooked (for FXS gateways) or ring signal is applied to port (FXO gateways).
   - Disable **[0]:** The automatic dialing option on the specific port is disabled (the number in the Destination Phone Number field is ignored).
   - Hotline **[2]:** When a phone is offhooked and no digit is pressed for 'HotLineToneDuration', the number in the Destination Phone Number field is automatically dialed (applies to FXS and FXO gateways).
4. Repeat step 3 for each port you want to use for Automatic Dialing.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Notes:

- After a ring signal is detected, on an 'Enabled' FXO port, the gateway initiates a call to the destination number without seizing the line. The line is seized only after the call is answered.
- After a ring signal is detected on a 'Disabled' or 'Hotline' FXO port, the gateway seizes the line.

Table 5-28: Automatic Dialing *ini* File Parameter

Parameter Name in *ini* File	Parameter Format
TargetOfChannelX	TargetOfChannel<Port> = <Phone>,<Mode> For example: TargetOfChannel0 = 1001,1 TargetOfChannel3 = 911,2 **Note 1:** The numbering of channels starts with 0. **Note 2:** Define this parameter for each gateway port you want to use for Automatic Dialing. **Note 3:** This parameter can appear up to 8 times for 8-port gateways and up to 24 times for MP-124 gateways.

5.5.9.3 Caller ID

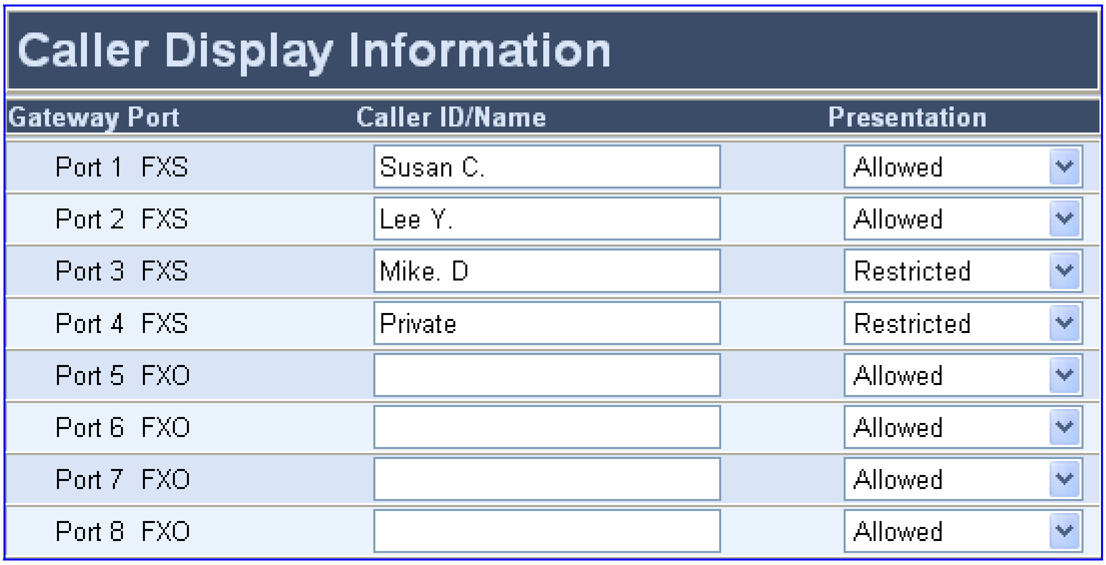
Use the Caller Display Information screen to send (to IP) Caller ID information when a call is made using the VoIP gateway (relevant to both FXS and FXO). The person receiving the call can use this information for caller identification. The information on this table is sent in an INVITE message in the 'From' header. For information on Caller ID restriction according to destination / source prefixes, refer to Section 5.5.3 on page 91.

Note: If Caller ID name is detected on an FXO line (EnableCallerID = 1), it is used instead of the Caller ID name defined in this table (FXO gateways only).

➢ To configure the Caller ID table, take these 6 steps:

1. Open the 'Caller Display Information' screen (**Protocol Management** menu > **Endpoint Settings** submenu > **Caller ID** option); the 'Caller Display Information' screen is displayed.

Figure 5-29: Caller Display Information Screen



2. In the 'Caller ID/Name' field, enter the Caller ID string. The Caller ID string can contain up to 18 characters. Note that when the FXS gateway receives 'Private' or 'Anonymous' strings in the From header, it doesn't send the calling name or number to the Caller ID display.

3. From the ‘Presentation’ drop-down list, select:
   - ‘Allowed’ **[0]** to send the string in the Caller ID/Name field when a (Tel→IP) call is made using this VoIP gateway port.
   - ‘Restricted’ **[1]** if you don’t want to send this string.

Notes:

- When the ‘Presentation’ field is set to ‘Restricted’, the caller identity is passed to the remote side using only the P-Asserted-Identity and P-Preferred-Identity headers (AssertedIdMode).
- The value of the ‘Presentation’ field can (optionally) be overridden by configuring the ‘Presentation’ parameter in the ‘Source Number Manipulation’ table.
- To maintain backward compatibility, when the strings ‘Private’ or ‘Anonymous’ are set in the Caller ID/Name field, the Caller ID is restricted and the value in the 'Presentation' field is ignored.

4. Repeat steps 2 and 3 for each VoIP gateway port.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-29: Caller ID *ini* File Parameter

Parameter Name in *ini* File	Parameter Format
CallerDisplayInfoX	CallerDisplayInfo<Port> = <Caller ID string>,<Restriction> 0 = Not restricted (default). 1 = Restricted. For example: CallerDisplayInfo0 = Susan C.,0 CallerDisplayInfo2 = Mark M.,1 **Note 1:** The numbering of channels starts with 0. **Note 2:** This parameter can appear up to eight times for 8-port gateways, and up to 24 times for MP-124.

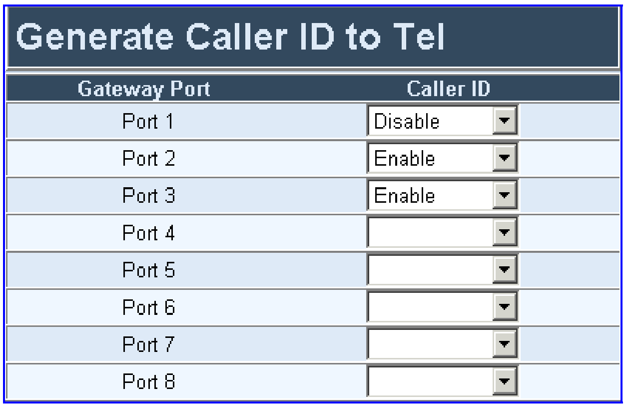
5.5.9.4 Generate Caller ID to Tel

The 'Generate Caller ID to Tel' screen for FXS and 'Caller ID Permissions' screen for FXO are used to enable or disable (per port) the Caller ID generation (for FXS gateways) and detection (for FXO gateways). If a port isn’t configured, its Caller ID generation / detection are determined according to the global parameter ‘EnableCallerID’ (described in Table 5-9).

➢ To configure the Generate Caller ID to Tel Table, take these 5 steps:

1. Open the ‘Generate Caller ID to Tel’ screen (**Protocol Management** menu > **Endpoint Settings** > **Generate Caller ID to Tel** option); the ‘Generate Caller ID to Tel’ screen is displayed.

Figure 5-30: MediaPack FXS Generate Caller ID to Tel Screen



2. In the ‘Caller ID’ field, select one of the following:
   - Enable: Enables Caller ID generation (FXS) or detection (FXO) for the specific port.
   - Disable: Caller ID generation (FXS) or detection (FXO) for the specific port is disabled.
   - Empty: Caller ID generation (FXS) or detection (FXO) for the specific port is determined according to the parameter ‘EnableCallerID’ (described in Table 5-9).
3. Repeat step 2 for each port.
4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-30: Generate Caller ID to Tel *ini* File Parameter

Parameter Name in *ini* File	Parameter Format
EnableCallerID_X	EnableCallerID_<Port> = <Caller ID> Caller ID: 0 = Disabled (default). 1 = Enabled. If not configured, use the global parameter ‘EnableCallerID’. **Note 1:** The numbering of ports starts with 0. **Note 2:** This parameter can appear up to eight times for 8-port gateways, and up to 24 times for MP-124.

5.5.9.5 Call Forward

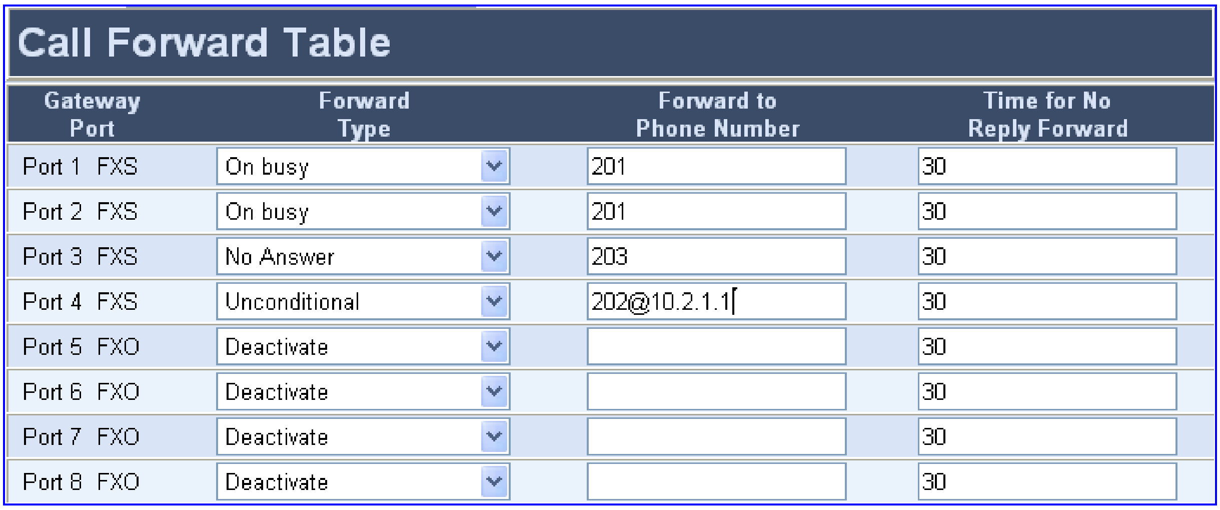
The VoIP gateway allows you to forward incoming IP→Tel calls (using 302 response) based on the VoIP gateway port to which the call is routed (applicable only to FXS gateways).

The Call Forwarding Table is applicable only if the Call Forward feature is enabled. To enable Call Forward set ‘Enable Call Forward’ to ‘Enable’ in the ‘Supplementary Services’ screen, or ‘EnableForward=1’ in the *ini* file (refer to Table 5-9).

➢ To configure the Call Forward table, take these 4 steps:

1. Open the ‘Call Forward Table’ screen (**Protocol Management** menu > **Endpoint Settings** submenu > **Call Forward** option); the ‘Call Forward Table’ screen is displayed.

Figure 5-31: Call Forward Table Screen



Call Forward Table

Gateway Port	Forward Type	Forward to Phone Number	Time for No Reply Forward
Port 1 FXS	On busy	201	30
Port 2 FXS	On busy	201	30
Port 3 FXS	No Answer	203	30
Port 4 FXS	Unconditional	202@10.2.1.1	30
Port 5 FXO	Deactivate		30
Port 6 FXO	Deactivate		30
Port 7 FXO	Deactivate		30
Port 8 FXO	Deactivate		30

2. Configure the Call Forward parameters for each port according to the table below.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-31: Call Forward Table (continues on pages 124 to 125)

Parameter	Description
Forward Type	▪ Deactivate **[0]** = Don’t forward incoming calls (default). ▪ On Busy **[1]** = Forward incoming calls when the gateway port is busy. ▪ Unconditional **[2]** = Forward any incoming call to the Phone number specified. ▪ No Answer **[3]** = Forward incoming calls that are not answered with the time specified in the ‘Time for No Reply Forward’ field. ▪ On Busy or No Answer **[4]** = Forward incoming calls when the port is busy or when calls are not answered after a configurable period of time. ▪ Do Not Disturb **[5]** = Immediately reject incoming calls.
Forward to Phone Number	Enter the telephone number or URI (number@IP address) to which the call is forwarded. **Note:** If this field only contains telephone number and Proxy isn’t used, the ‘forward to’ phone number must be specified in the ‘Tel to IP Routing’ table of the forwarding gateway.

Table 5-31: Call Forward Table (continues on pages 124 to 125)

Parameter	Description
Time for No Reply Forward	If you have set the Forward Type for this port to **no reply**, enter the number of seconds the VoIP gateway waits before forwarding the call to the phone number specified.
Parameter Name in *ini* File	**Parameter Format**
FwdInfo_x	FwdInfo_<Port (0 to 23)> = <Forward Type>, <Forwarded SIP User Identification>, <Timeout (in seconds) for No Reply> For example: FwdInfo_0 = 1,1001 FwdInfo_1 = 1,2003@10.5.1.1 FwdInfo_2 = 3,2005,30 **Note 1:** The numbering of gateway ports starts with 0. **Note 2:** This parameter can appear up to 24 times for MP-124.

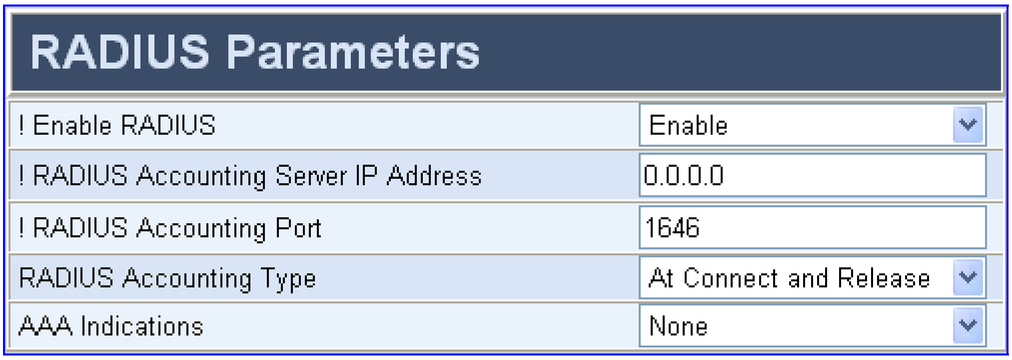
5.5.10 Configuring RADIUS Accounting Parameters

The RADIUS Parameters screen is used for configuring the Remote Authentication Dial In User Service (RADIUS) accounting parameters.

➢ To configure the FXO parameters, take these 4 steps:

1. Open the 'RADIUS Parameters' screen (**Protocol Management** menu > **RADIUS Parameters**); the 'RADIUS Parameters' screen is displayed.

Figure 5-32: RADIUS Parameters Screen



2. Configure the RADIUS accounting parameters according to Table 5-33.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-32: RADIUS Parameters

Parameter	Description
Enable RADIUS **EnableRADIUS**	Enables or disables the RADIUS application. Valid options include: Disables **[0]** = disables RADIUS application (default) Enable **[1]** = enables RADIUS application
RADIUS Accounting Server IP Address **[RADIUSAccServerIP]**	IP address of the RADIUS accounting server.
RADIUS Accounting Port **[RADIUSAccPort]**	Port of the RADIUS accounting server. The default value is 1646.
RADIUS Accounting Type **[RADIUSAccountingType]**	Determines when the RADIUS accounting messages are sent to the RADIUS accounting server. Valid options include: At Call Release **[0]** = At the release of the call only (default). At Connect and Release **[1]** = At the connect and release of the call. At Setup and Release **[2]** = At the setup and release of the call.
AAA Indications [**AAAIndications**]	Determines which Authentication, Authorization and Accounting (AAA) indications to use. Valid options include: None **[0]** = No indications (default) Accounting Only **[3]** = Only accounting indications are used.

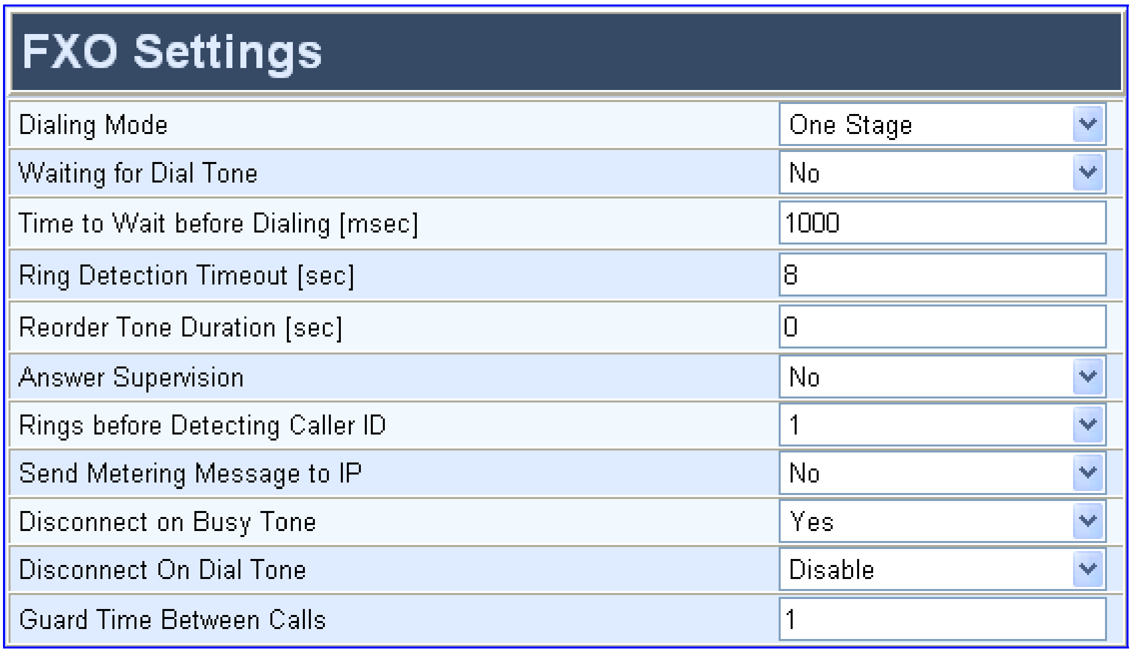
5.5.11 Configuring the FXO Parameters

Use this screen to configure the gateway's specific FXO parameters.

➢ **To configure the FXO parameters, take these 4 steps:**

1. Open the 'FXO Settings' screen (**Protocol Management** menu > **FXO Settings** > **FXO Settings** option); the 'FXO Settings' screen is displayed.

Figure 5-33: FXO Settings Screen



2. Configure the FXO parameters according to Table 5-33.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-33: FXO Parameters (continues on pages 127 to 130)

Parameter	Description
Dialing Mode **[IsTwoStageDial]**	One Stage **[0]** = One-stage dialing. Two Stage **[1]** = Two-stage dialing (default). Used for IP→FXO gateways calls. If two-stage dialing is enabled, then the FXO gateway seizes one of the PSTN/PBX lines without performing any dial, the remote user is connected over IP to PSTN/PBX, and all further signaling (dialing and Call Progress Tones) is performed directly with the PBX without the gateway's intervention. If one-stage dialing is enabled, then the FXO gateway seizes one of the available lines (according to Channel Select Mode parameter), and dials the destination phone number received in INVITE message. Use the 'Waiting For Dial Tone' parameter to specify whether the dialing should come after detection of dial tone, or immediately after seizing of the line.

Table 5-33: FXO Parameters (continues on pages 127 to 130)

Parameter	**Description**
Waiting For Dial Tone **[IsWaitForDialTone]**	No **[0]** = Don't wait for dial tone. Yes **[1]** = Wait for dial tone (default). Used for IP→MediaPack/FXO gateways, when 'One Stage Dialing' is enabled. If 'wait for dial tone' is enabled, the FXO gateway dials the phone number (to the PSTN/PBX line) only after it detects a dial tone. **Note 1:** The correct dial tone parameters should be configured in the Call Progress Tones file. **Note 2:** It can take the gateway 1 to 3 seconds to detect a dial tone (according to the dial tone configuration in the Call Progress Tones file). If 'Waiting For Dial Tone' is disabled, the FXO gateway immediately dials the phone number after seizing the PSTN/PBX line, without 'listening' to dial tone.
Time to Wait before Dialing [msec] **[WaitForDialTime]** **Note:** Replaces the obsolete parameter FXOWaitForDialTime.	Determines the delay before the gateway starts dialing on the FXO line in the following scenarios (applicable only to FXO gateways): 1. The delay between the time the line is seized and dialing is begun, during the establishment of an IP→Tel call. **Note:** Applicable only to FXO gateways for single stage dialing, when waiting for dial tone (IsWaitForDialTone) is disabled. 2. The delay between the time when Wink is detected and dialing is begun, during the establishment of an IP→Tel call (for DID lines, EnableDIDWink = 1). 3. For call transfer. The delay after hook-flash is generated and dialing is begun. The valid range (in milliseconds) is 0 to 20000 (20 seconds). The default value is 1000 (1 second).
Ring Detection Timeout [sec] **[FXOBetweenRingTime]**	**Note:** Applicable only to FXO gateways for Tel→IP calls. The Ring Detection timeout is used differently for normal and for automatic dialing. If automatic dialing is not used, and if Caller ID is enabled, then the FXO gateway seizes the line after detection of the second ring signal (allowing detection of caller ID, sent between the first and the second rings). If the second ring signal doesn't arrive for 'Ring Detection Timeout' the gateway doesn't initiate a call to IP. When automatic dialing is used, the FXO gateway initiates a call to IP when ringing signal is detected. The FXO line is seized only if the remote IP party answers the call. If the remote party doesn't answer the call and the ringing signal stops for 'Ring Detection Timeout', the FXO gateway Releases the IP call. Usually set to a value between 5 and 8. The default is 8 seconds.
Reorder Tone Duration [sec] **[TimeForReorderTone]**	Busy or Reorder tone duration (seconds) the FXO gateway plays before releasing the line. The valid range is 0 to 100. The default is 0 seconds. Usually, after playing a Reorder / Busy tone for the specified duration, the FXS gateway, starts playing an Offhook Warning tone. **Note 1:** Selection of Busy or Reorder tone is performed according to the release cause received from IP. **Note 2:** Refer also to the parameter 'CutThrough' (described in Table 5-8).
Answer Supervision **[EnableVoiceDetection]**	Yes **[1]** = FXO gateway sends 200 OK (to INVITE) messages when speech/fax/modem is detected. No **[0]** = 200 OK is sent immediately after the FXO gateway finishes dialing (default). Usually this feature is used only with early media establish voice path before the call is answered. **Note:** This feature is applicable only to 'One Stage' dialing.
Rings before Detecting Caller ID **[RingsBeforeCallerID]**	Sets the number of rings before the gateway starts detection of Caller ID (FXO only). 0 **[0]** = Before first ring. 1 **[1]** = After first ring (default). 2 **[2]** = After second ring.

Table 5-33: FXO Parameters (continues on pages 127 to 130)

Parameter	Description
Send Metering Message to IP **[SendMetering2IP]**	No **[0]** = Disabled (default). Yes **[1]** = FXO gateways send a metering tone INFO message to IP on detection of 12/16 kHz metering pulse. FXS gateways generate the 12/16 kHz metering tone on reception of a metering message. **Note 1:** Suitable (12 kHz or 16 kHz) *coeff* must be used for both FXS and FXO gateways. The 'MeteringType' parameter must be defined in both FXS/FXO gateways. **Note 2:** The proprietary metering tone INFO message is shown in Section 11 on page 277.
Disconnect on Busy Tone **[DisconnectOnBusyTone]**	No **[0]** = Call isn't released (FXO gateway). Yes **[1]** = Call is released (on FXO gateways) if busy or reorder (fast busy) tones are detected on the gateway's FXO port (default).
Guard Time Between Calls **[GuardTimeBetweenCalls]**	Defines the time interval (in seconds) after a call has ended and a new call can be accepted for IP to Tel calls. Applicable only to FXO gateways. The valid range is 0 to 10. The default value is 1 second. **Note:** Occasionally, after a call is ended and onhook is applied, a delay is required before placing a new call (and performing offhook). This is necessary to prevent wrong hook-flash detection or other glare phenomena.
Disconnect on Dial Tone **[DisconnectOnDialTone]**	FXO gateways can disconnect a call after a dial tone from the PBX is detected. No **[0]** = Call isn't released. Yes **[1]** = Call is released if dial tone is detected on the gateway's FXO port (default). **Note:** This option is in addition to the mechanism that disconnects a call when either busy or reorder tones are detected.

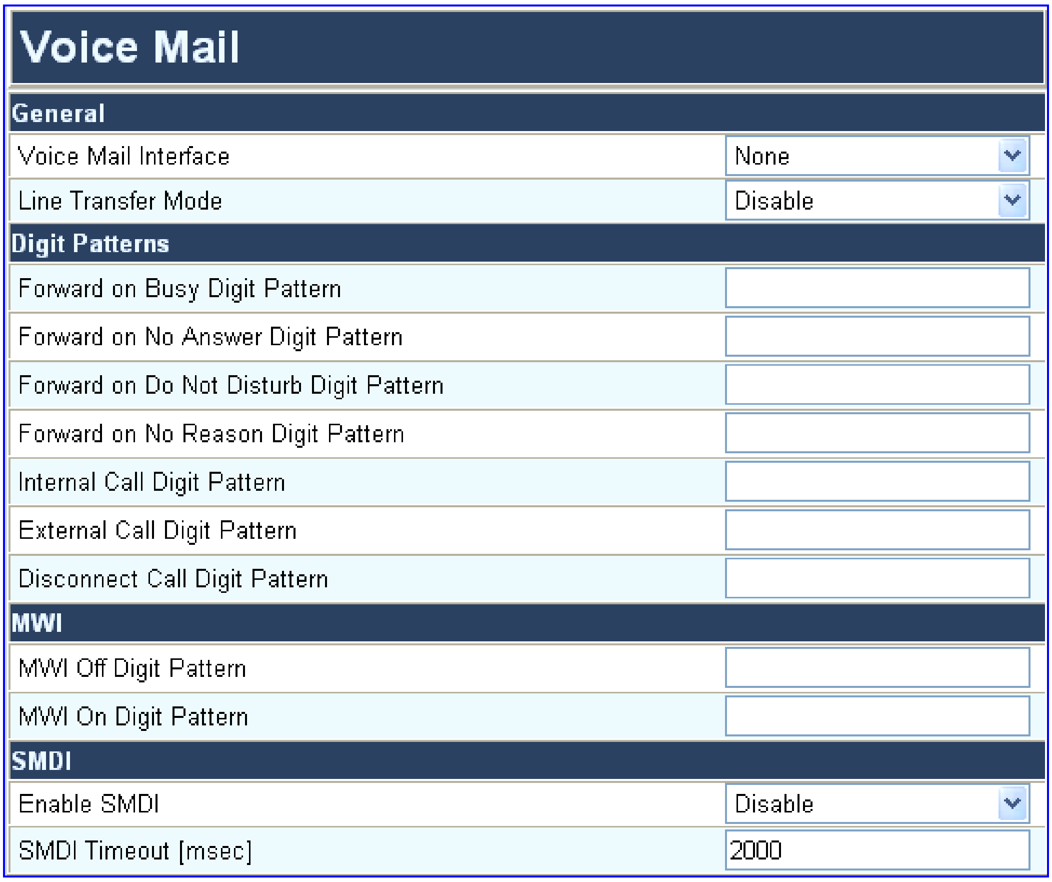
5.5.12 Configuring Voice Mail (VM) Parameters

Use this screen to configure the VM parameters. The VM application applies only to FXO gateways. For detailed information on VM, refer to the CPE Configuration Guide for Voice Mail.

➢ To configure the VM parameters, take these 4 steps:

1. Open the 'Voice Mail' screen (**Protocol Management** menu > **FXO Settings** > **Voice Mail** option); the 'Voice Mail' screen is displayed.

Figure 5-34: Voice Mail Screen



2. Configure the Voice Mail parameters according to Table 5-34.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-34: Voice Mail Parameters ((continues on pages 130 to 131)

Parameter	Description
General	
Voice Mail Interface **[VoiceMailInterface]**	Enables the voice mail (VM) application on the gateway and determines the communication method used between the PBX and the gateway. Valid options include: None (default) **[0]** DTMF **[1]** SMDI **[2]**

Table 5-34: Voice Mail Parameters ((continues on pages 130 to 131)

Parameter	Description
Line Transfer Mode **[LineTransferMode]**	Determines the transfer method used by the gateway. Disable **[0]** = IP (default). Blind Transfer **[1]** = PBX blind transfer. In this mode, after receiving a REFER message from the IP side, the FXO: sends a hook-flash to the PBX, dials the digits (that are received in the Refer-To header), and then immediately drops the line (on-hook). The PBX performs the transfer internally.
Digit Patterns The following digit pattern parameters apply only to VM applications that use the DTMF communication method. For the available patterns' syntaxes, refer to the CPE Configuration Guide for Voice Mail.	
Forward on Busy Digit Pattern **[DigitPatternForwardOnBusy]**	Determines the digit pattern used by the PBX to indicate 'call forward on busy'. The valid range is a 120-character string.
Forward on No Answer Digit Pattern **[DigitPatternForwardOnNoAnswer]**	Determines the digit pattern used by the PBX to indicate 'call forward on no answer'. The valid range is a 120-character string.
Forward on Do Not Disturb Digit Pattern **[DigitPatternForwardOnDND]**	Determines the digit pattern used by the PBX to indicate 'call forward on do not disturb'. The valid range is a 120-character string.
Forward on No Reason Digit Pattern **[DigitPatternForwardNoReason]**	Determines the digit pattern used by the PBX to indicate 'call forward with no reason'. The valid range is a 120-character string.
Internal Call Digit Pattern **[DigitPatternInternalCall]**	Determines the digit pattern used by the PBX to indicate an internal call. The valid range is a 120-character string.
External Call Digit Pattern **[DigitPatternExternalCall]**	Determines the digit pattern used by the PBX to indicate an external call. The valid range is a 120-character string.
Disconnect Call Digit Pattern **[TelDisconnectCode]**	Determines a digit pattern that, when received from the Tel side, indicates the gateway to disconnect the call. The valid range is a 25-character string.
MWI	
MWI Off Digit Pattern **[MWIOffCode]**	Determines a digit code used by the gateway to notify the PBX that there aren't any messages waiting for a specific extension. This code is added as prefix to the dialed number. The valid range is a 25-character string.
MWI On Digit Pattern **[MWIOnCode]**	Determines a digit code used by the gateway to notify the PBX of messages waiting for a specific extension. This code is added as prefix to the dialed number. The valid range is a 25-character string.
SMDI	
Enable SMDI **[SMDI]**	Enables the Simplified Message Desk Interface (SMDI) on the gateway. Disable **[0]** = Normal serial (default). Enable **[1]** = Enable RS-232 SMDI interface. **Note:** When the RS-232 connection is used for SMDI messages (Serial SMDI) it cannot be used for other applications, for example, to access the Command Line Interface.
SMDI Timeout **[SMDITimeOut]**	Determines the time (in msec) that the gateway waits for an SMDI Call Status message before or after a Setup message is received. This parameter is used to synchronize the SMDI and analog interfaces. If the timeout expires and only an SMDI message was received, the SMDI message is dropped. If the timeout expires and only a Setup message was received, the call is established. The valid range is 0 to 10000 (10 seconds). The default value is 2000.

5.5.13 Protocol Management *ini* File Parameters

Table 5-35 describes the SIP Protocol Management parameters that can only be configured via the *ini* file.

Table 5-35: Protocol Management, *ini* File Parameters (continues on pages 132 to 137)

ini File Parameter Name	Valid Range and Description
EnablePtime	0 = Remove the ptime header from SDP. 1 = Include the ptime header in SDP (default).
IsUseToHeaderAsCalledNumber	0 = Sets the destination number to the user part of the Request-URI for IP→Tel calls, and sets the 'Contact' header to the source number for Tel→ IP calls (default). 1 = Sets the destination number to the user part of the 'To' header for IP→Tel calls, and sets the 'Contact' header to the username parameter for Tel→IP calls.
SIPSRequireClientCertificate	0 = The gateway doesn't require client certificate (default). 1 = The gateway (when acting as a server for the TLS connection) requires reception of client certificate to establish the TLS connection. **Note:** The SIPS certificate files can be changed using the parameters 'HTTPSCertFileName' and 'HTTPSRootFileName'.
Send180ForCallWaiting	0 = Use 182 Queued response to indicate a call waiting (default). 1 = Use 180 Ringing response to indicate a call waiting.
NumberOfActiveDialogs	Defines the maximum number of active SIP dialogs that are not call related (i.e., REGISTER and SUBSCRIBE). The valid range is 1 to 5. The default value is 5.
EnableDID	Enables Japan NTT 'Modem' Direct Inward Dialing (DID) support. FXS gateways can be connected to Japan's NTT PBX using 'Modem' DID lines. These DID lines are used to deliver a called number to the PBX (applicable to FXS gateways). The DID signal can be sent alone or combined with an NTT Caller ID signal.
EnableDID_X	Enables generation of Japan NTT Modem DID signal per port. The format is: EnableDID_<Port> = <Modem DID> Modem DID: 0 = Disabled (default). 1 = Enabled. If not configured, use the global parameter 'EnableDID'. **Note:** Applicable only to MediaPack/FXS gateways.
NTTDIDSignallingForm	Determines the type of Direct Inward Dialing (DID) signaling support for Japan NTT modem: DTMF- or Frequency Shift Keying (FSK)-based signaling. Gateways can be connected to Japan's NTT PBX using 'Modem' DID lines. These DID lines are used to deliver a called number to the PBX. 0 = FSK-based signaling (default). 1 = DTMF-based signaling. **Note:** Applicable only to FXS gateways.
FarEndDisconnectSilenceThreshold	Threshold of the packet count (in percents), below which is considered silence by the media gateway. The valid range is 1 to 100. The default is 8%. Note: Applicable only if silence is detected according to packet count (FarEndDisconnectSilenceMethod = 1).
T38UseRTPPort	Defines that the T.38 packets are sent / received using the same port as RTP packets. 0 = Use the RTP port +2 to send / receive T.38 packets (default). 1 = Use the same port as the RTP port to send / receive T.38 packets.

Table 5-35: Protocol Management, *ini* File Parameters (continues on pages 132 to 137)

ini File Parameter Name	Valid Range and Description
DisableAutoDTMFMute	Enables / disables the automatic mute of DTMF digits when out-of-band DTMF transmission is used. 0 = Auto mute is used (default). 1 = No automatic mute of in-band DTMF. When 'DisableAutoDTMFMute=1', the DTMF transport type is set according to the parameter 'DTMFTransportType' and the DTMF digits aren't muted if out-of-band DTMF mode is selected ('TxDTMFOption = 1, 2 or 3'). This enables the sending of DTMF digits in-band (transparent of RFC 2833) in addition to out-of-band DTMF messages. **Note:** Usually this mode is not recommended.
FirstCallWaitingToneID	Determines the index of the first Call Waiting Tone in the CPT file. This feature enables the called party to distinguish between four different call origins (e.g., external vs. internal calls). The gateway plays the tone received in the 'play tone CallWaitingTone#' parameter of an INFO message + the value of this parameter - 1. The valid range is -1 to 100. The default value is -1 (not used). **Note 1:** It is assumed that all Call Waiting Tones are defined in sequence in the CPT file. **Note 2:** This feature is relevant only to Broadsoft's application servers (the tone is played using INFO message).
CountryCoefficients	Determines the FXO line characteristics (AC and DC) according to country of origin. Argentina = 0, Australia = 1, Austria = 2, Bahrain = 3, Belgium = 4, Brazil = 5, Bulgaria = 6, Canada = 7, Chile = 8, China = 9, Colombia = 10, Croatia = 11, Cyprus = 12, Czech_Republic = 13, Denmark = 14, Ecuador = 15, Egypt = 16, El_Salvador = 17, Finland = 18, France = 19, Germany = 20, Greece = 21, Guam = 22, Hong_Kong = 23, Hungary = 24, Iceland = 25, India = 26, Indonesia = 27, Ireland = 28, Israel = 29, Italy = 30, Japan = 31, Jordan = 32, Kazakhstan = 33, Kuwait = 34, Latvia = 35, Lebanon = 36, Luxembourg = 37, Macao = 38, Malaysia = 39, Malta = 40, Mexico = 41, Morocco = 42, Netherlands = 43, New_Zealand = 44, Nigeria = 45, Norway = 46, Oman = 47, Pakistan = 48, Peru = 49, Philippines = 50, Poland = 51, Portugal = 52, Romania = 53, Russia = 54, Saudi_Arabia = 55, Singapore = 56, Slovakia = 57, Slovenia = 58, South_Africa = 59, South_Korea = 60, Spain = 61, Sweden = 62, Switzerland = 63, Syria = 64, Taiwan = 65, TBR21 = 66, Thailand = 67, UAE = 68, United_Kingdom = 69, UnitedStates = 70, Yemen = 71 The default value is 70 (United States).
EnableCallerIDTypeTwo	Disables the generation of Caller ID type 2 when the phone is offhooked. 0 = Caller ID type 2 isn't played. 1 = Caller ID type 2 is played (default).
MeteringType	Defines the metering tone (12 kHz or 16 kHz) that is detected by FXO gateways and generated by FXS gateways. 0 = 12 kHz metering tone (default). 1 = 16 kHz metering tone. **Note:** Suitable (12 kHz or 16 KHz) *coeff* must be used for both FXS and FXO gateways.
PolarityReversalType	Defines the voltage change slope during polarity reversal or wink. 0 = Soft (default). 1 = Hard. **Note 1:** Some Caller ID signals use reversal polarity and/or wink signals. In these cases it is recommended to set PolarityReversalType to 1 (Hard). **Note 2:** Applicable only to FXS gateways.

Table 5-35: Protocol Management, *ini* File Parameters (continues on pages 132 to 137)

ini File Parameter Name	Valid Range and Description
CurrentDisconnectDuration	Duration of the current disconnect pulse (in msec). The default is 900 msec, The range is 200 to 1500 msec. Applicable for both FXS and FXO gateways. **Note:** The FXO gateways' detection range is +/-200 msec of the parameter's value + 100. For example if CurrentDisconnectDuration = 200, the detection range is 100 to 500 msec.
CurrentDisconnectDefaultThreshold	Determines the line voltage threshold which, when reached, is considered a current disconnect detection. **Note:** Applicable only to FXO gateways. The valid range is 0 to 20 Volts. The default value is 4 Volts.
TimeToSampleAnalogLineVoltage	Determines the frequency at which the analog line voltage is sampled (after offhook), for detection of the current disconnect threshold. **Note:** Applicable only to FXO gateways. The valid range is 100 to 2500 msec. The default value is 1000 msec.
AnalogCallerIDTimimgMode	0 = Caller ID is generated between the first two rings (default). 1 = The gateway attempts to find an optimized timing to generate the Caller ID according to the selected Caller ID type. Note that when used with distinctive ringing, the Caller ID signal doesn't change the distinctive ringing timing. **Note:** Applicable only to FXS gateways.
EnableRAI	0 = Disable RAI (Resource Available Indication) service (default). 1 = Enable RAI service. If RAI is enabled, an SNMP 'acBoardCallResourcesAlarm' Alarm Trap is sent if gateway resources fall below a predefined (configurable) threshold.
RAIHighThreshold	High Threshold (in percentage) that defines the gateway's busy endpoints. The range is 0 to 100. The default value is 90%. When the percentage of the gateway's busy endpoints exceeds the value configured in High Threshold, the gateway sends an SNMP 'acBoardCallResourcesAlarm' Alarm Trap with a 'major' Alarm Status. **Note:** The gateway's available Resources are calculated by dividing the number of busy endpoints by the total number of available gateway endpoints.
RAILowThreshold	Low Threshold (in percentage) that defines the gateway's busy endpoints. The range is 0 to 100. The default value is 90%. When the percentage of the gateway's busy endpoints falls below the value defined in Low Threshold, the gateway sends an SNMP 'acBoardCallResourcesAlarm' Alarm Trap with a 'cleared' Alarm Status.
RAILoopTime	Time interval (in seconds) that the gateway checks for resource availability. The default is 10 seconds.

Table 5-35: Protocol Management, *ini* File Parameters (continues on pages 132 to 137)

ini File Parameter Name	Valid Range and Description
3WayConferenceMode	Defines the mode of operation when the 3-Way Conference feature is used. Valid options include: **[0]** = Conference-initiating INVITE (sent by the gateway), uses the ConferenceID concatenated with a unique identifier as the Request-UR (default) **[1]** = Conference-initiating INVITE (sent by the gateway), uses only the ConferenceID as the Reques-URI If 3wayConferenceMode is set to 0, the Conference-initiating INVITE sent by the gateway, uses the ConferenceID concatenated with a unique identifier as the Request-URI. This same Request-URI is set as the Refer-To header value in the REFER messages that are sent to the two remote parties. If 3wayConferenceMode is set to 1, the Conference-initiating INVITE sent by the gateway, only uses the ConferenceID as the Reques-URI. The media server sets the Contact header of the 200 OK response to the actual unique identifier (Conference URI) to be used by the participants. This Conference URI is included (by the gateway), in the Refer-To header value in the REFER messages sent by the gateway to the remote parties. The remote parties join the conference by sending INVITE messages to the media server using this conference URI.
WarningToneDuration	Defines the duration (in seconds) for which Off-Hook Warning Tone is played to the user. The valid range is -1 to 2,147,483,647 seconds. The default is 600 seconds. **Note**: A negative value indicates that the tone is played infinitely.
FXONumberOfRings	Defines the number of rings before the FXO gateway answers a call. The valid range is 0 to 255. The default is 0 seconds.
PhoneContext	Defines the Phone Context table. When a call is received from the ISDN, the NPI and TON are compared against the table, and the Phone-Context value is used in the outgoing SIP INVITE message. The same mapping occurs when an INVITE with a Phone-Context attribute is received. The Phone-Context parameter appears in the standard SIP headers where a phone number is used (Request-URI, To, From, Diversion). PhoneContext = <Number Plan>,<Number Type>,<Phone-Context> For example: PhoneContext = 0,0,unknown.com PhoneContext = 1,1,host.com PhoneContext = 9,1,na.e164.host.com **Note 1:** This parameter can appear up to 20 times. **Note 2:** Several rows with the same NPI-TON or Phone-Context are allowed. In this scenario, a Tel-to-IP call uses the first match. **Note 3:** Phone-Context '+' is a unique case as it doesn't appear in the Request-URI as a Phone-Context parameter. Instead, it's added as a prefix to the phone number. The '+' isn't removed from the phone number in the IP-to-Tel direction. **Note 4:** To configure Phone Context table using the Web interface, refer to Section 5.5.4 on page 96.

Table 5-35: Protocol Management, *ini* File Parameters (continues on pages 132 to 137)

ini File Parameter Name	Valid Range and Description
EnableRport	Enables / disables the usage of the 'rport' parameter in the Via header. **[0]** = Enabled. **[1]** = Disabled (default). The gateway adds an 'rport' parameter to the Via header field of each outgoing SIP message. The first Proxy that receives this message sets the 'rport' value of the response to the actual port from which the request was received. This method is used, for example, to enable the gateway to identify its port mapping outside a NAT. If the Via doesn't include 'rport' tag, the destination port of the response will be taken from the host part of the VIA. If the Via includes 'rport' tag with no port value, the destination port of the response will be the source port of the incoming request. If the Via includes 'rport' tag with a port value (rport=1001), the destination port of the response will be the port indicated in the 'rport' tag.
Serial parameters (applicable only to the VM SMDI application)	
SerialBaudRate	Determines the value of the RS-232 baud rate. The valid range is: any value. It is recommended to use the following standard values: 1200, 2400, 9600 (default), 14400, 19200, 38400, 57600, 115200.
SerialData	Determines the value of the RS-232 data bit. 7 = 7-bit. 8 = 8-bit (default).
SerialParity	Determines the value of the RS-232 polarity. 0 = None (default). 1 = Odd. 2 = Even.
SerialStop	Determines the value of the RS-232 stop bit. 1 = 1-bit (default). 2 = 2-bit.
SerialFlowControl	Determines the value of the RS-232 flow control. 0 = None (default). 1 = Hardware.

5.6 Advanced Configuration

Use this menu to set the gateway's advanced configuration parameters.

> **Note:** Those parameters contained within square brackets are the names used to configure the parameters via the *ini* file.

5.6.1 Configuring the Network Settings

From the Network Settings you can:

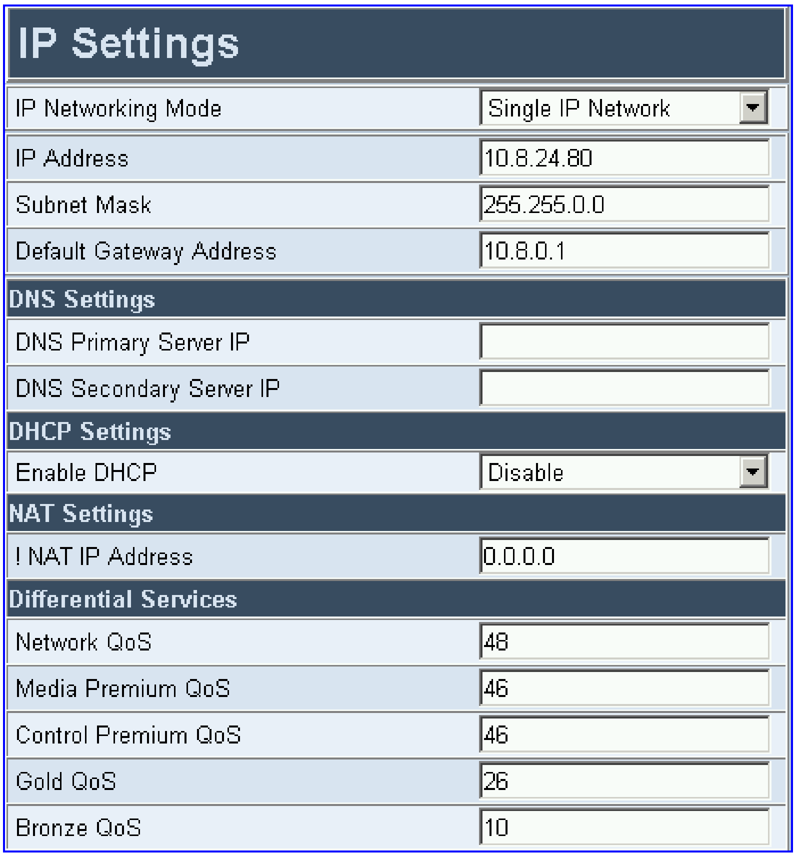
- Configure the IP Settings (refer to Section 5.6.1.1).
- Configure the Application Settings (refer to Section 5.6.1.2 on page 141).
- Configure the NFS Settings (refer to Section 5.6.1.3 on page 143).
- Configure the IP Routing Table (refer to Section 5.6.1.4 on page 145).
- Configure the VLAN Settings (refer to Section 5.6.1.5 on page 147).
- Configure the PPPoE Settings (refer to Section 5.6.1.6 on page 149).

5.6.1.1 Configuring the IP Settings

➢ To configure the IP Settings parameters, take these 4 steps:

1. Open the ‘IP Settings’ screen (**Advanced Configuration** menu > **Network Settings** > **IP Settings** option); the ‘IP Settings’ screen is displayed.

Figure 5-35: IP Settings Screen



2. Configure the IP Settings according to Table 5-36.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-36: Network Settings, IP Settings Parameters **(continues on pages 138 to 140)**

Parameter	Description
IP Networking Mode **[EnableMultipleIPs]**	Enables / disables the Multiple IPs mechanism. Single IP Network **[0]** (default). Multiple IP Network **[1]**. For detailed information on Multiple IPs, refer to Section 9.8 on page 253.
IP Address	IP address of the gateway. Enter the IP address in dotted format notation, for example 10.8.201.1. **Note 1:** A warning message is displayed (after pressing the button ‘Submit’) if the entered value is incorrect. **Note 2:** After changing the IP address and pressing the button ‘Submit’, a prompt appears indicating that for the change to take effect, the gateway is to be reset.

Table 5-36: Network Settings, IP Settings Parameters (continues on pages 138 to 140)

Parameter	Description
Subnet Mask	Subnet mask of the gateway. Enter the subnet mask in dotted format notation, for example 255.255.0.0 **Note 1:** A warning message is displayed (after pressing the button 'Submit') if the entered value is incorrect. **Note 2:** After changing the subnet mask and pressing the button 'Submit', a prompt appears indicating that for the change to take effect, the gateway is to be reset.
Default Gateway Address	IP address of the default gateway used by the gateway. Enter the IP address in dotted format notation, for example 10.8.0.1. **Note 1:** A warning message is displayed (after pressing the button 'Submit') if the entered value is incorrect. **Note 2:** After changing the default gateway IP address and pressing the button 'Submit', a prompt appears indicating that for the change to take effect, the gateway is to be reset. For detailed information on multiple routers support, refer to Section 9.6 on page 252.
OAM Network Settings (available only in Multiple IPs mode)	
IP Address **[LocalOAMIPAddress]**	The gateway's source IP address in the OAM network. The default value is 0.0.0.0.
Subnet Mask **[LocalOAMSubnetMask]**	The gateway's subnet mask in the OAM network. The default subnet mask is 0.0.0.0.
Default Gateway Address **[LocalOAMDefaultGW]**	N/A. Use the IP Routing table instead (Advanced Configuration > Network Settings).
Control Network Settings (available only in Multiple IPs mode)	
IP Address **[LocalControlIPAddress]**	The gateway's source IP address in the Control network. The default value is 0.0.0.0.
Subnet Mask **[LocalControlSubnetMask]**	The gateway's subnet mask in the Control network. The default subnet mask is 0.0.0.0.
Default Gateway Address **[LocalControlDefaultGW]**	N/A. Use the IP Routing table instead (Advanced Configuration > Network Settings).
Media Network Settings (available only in Multiple IPs mode)	
IP Address **[LocalMediaIPAddress]**	The gateway's source IP address in the Media network. The default value is 0.0.0.0.
Subnet Mask **[LocalMediaSubnetMask]**	The gateway's subnet mask in the Media network. The default subnet mask is 0.0.0.0.
Default Gateway Address **[LocalMediaDefaultGW]**	The gateway's default gateway IP address in the Media network. The default value is 0.0.0.0.
DNS Settings	
DNS Primary Server IP **[DNSPriServerIP]**	IP address of the primary DNS server. Enter the IP address in dotted format notation, for example 10.8.2.255. **Note:** To use Fully Qualified Domain Names (FQDN) in the Tel to IP Routing table, you must define this parameter.
DNS Secondary Server IP **[DNSSecServerIP]**	IP address of the second DNS server. Enter the IP address in dotted format notation, for example 10.8.2.255.

Table 5-36: Network Settings, IP Settings Parameters **(continues on pages 138 to 140)**

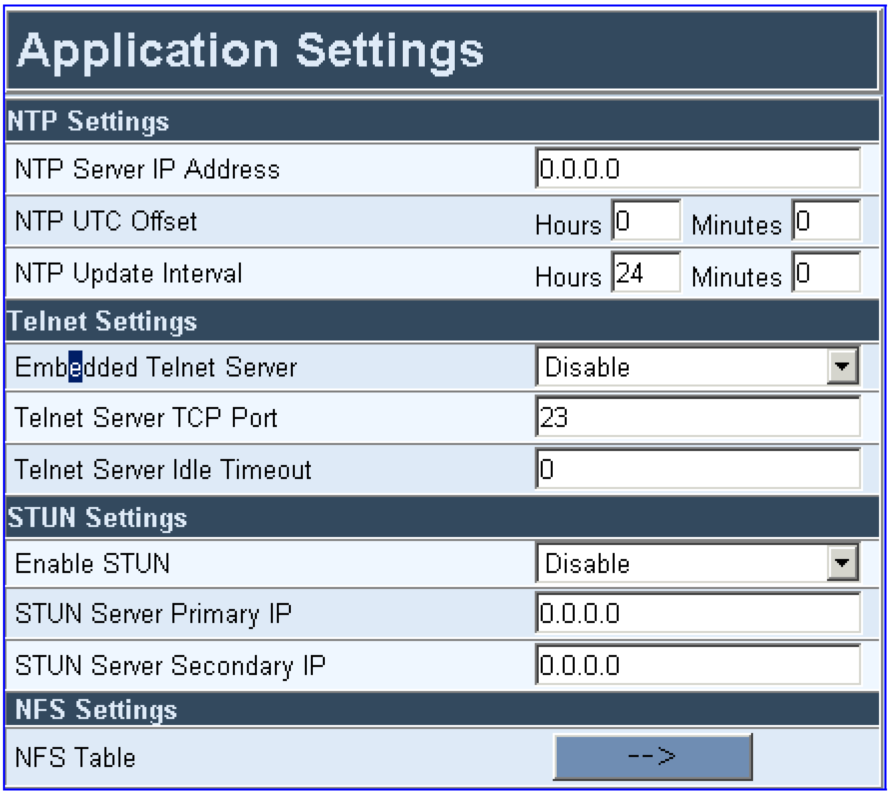
Parameter	Description
DHCP Settings	
Enable DHCP **[DHCPEnable]**	Disable **[0]** = Disable DHCP support on the gateway (default). Enable **[1]** = Enable DHCP support on the gateway. After the gateway is powered up, it attempts to communicate with a BootP server. If a BootP server is not responding and if DHCP is enabled, then the gateway attempts to get its IP address and other network parameters from the DHCP server. **Note:** After you enable the DHCP Server (from the Web browser) follow this procedure: ▪ Click the Submit button. ▪ Save the configuration using the 'Maintenance' button (before you reset the gateway). For information on how to save the configuration, refer to Section 5.10.2 on page 205. ▪ Reset the gateway *directly* (Web reset doesn't trigger the BootP/DHCP procedure and the parameter DHCPEnable reverts to '0'). Note that throughout the DHCP procedure the BootP/TFTP application must be deactivated. Otherwise, the MediaPack receives a response from the BootP server instead of the DHCP server. **Note:** For additional information on DHCP, refer to Section 7.2 on page 212. ***ini* file note:** The DHCPEnable is a special 'Hidden' parameter. Once defined and saved in flash memory, its assigned value doesn't revert to its default even if the parameter doesn't appear in the *ini* file.
NAT Settings	
NAT IP Address **[StaticNatIP]**	Global gateway IP address. Define if static Network Address Translation (NAT) device is used between the gateway and the Internet.
Differential Services. For detailed information on IP QoS via Differentiated Services, refer to Section 9.8 on page 253.	
Network QoS **[NetworkServiceClassDiffServ]**	Sets the DiffServ value for Network service class content. The valid range is 0 to 56. The default value is 48.
Media Premium QoS **[PremiumServiceClassMediaDiffServ]**	Sets the DiffServ value for Premium Media service class content (only if IPDiffServ is not set in the selected IP Profile). The valid range is 0 to 56. The default value is 46. **Note:** The value for the Premium Control DiffServ is determined by (according to priority): (1) IPDiffServ value in the selected IP Profile. (2) PremiumServiceClassMediaDiffServ.
Control Premium QoS **[PremiumServiceClassControlDiffServ]**	Sets the DiffServ value for Premium Control service class content (only if ControlIPDiffserv is not set in the selected IP Profile). The valid range is 0 to 56. The default value is 46. **Note:** The value for the Premium Control DiffServ is determined by (according to priority): (1) ControlPDiffserv value in the selected IP Profile. (2) PremiumServiceClassControlDiffServ.
Gold QoS **[GoldServiceClassDiffServ]**	Sets the DiffServ value for the Gold service class content. The valid range is 0 to 56. The default value is 26.
Bronze QoS **[BronzeServiceClassDiffServ]**	Sets the DiffServ value for the Bronze service class content. The valid range is 0 to 56. The default value is 10.

5.6.1.2 Configuring the Application Settings

➢ **To configure the Application Settings parameters, take these 4 steps:**

1. Open the 'Application Settings' screen (**Advanced Configuration** menu > **Network Settings** > **Application Settings** option); the 'Application Settings' screen is displayed.

Figure 5-36: Application Settings Screen



2. Configure the Application Settings according to Table 5-37.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-37: Network Settings, Application Settings Parameters

Parameter	Description
NTP Settings	
For detailed information on NTP, refer to Section 9.7 on page 253.	
NTP Server IP Address **[NTPServerIP]**	IP address (in dotted format notation) of the NTP server. The default IP address is 0.0.0.0 (the internal NTP client is disabled).
NTP UTC Offset **[NTPServerUTCOffset]**	Defines the UTC (Universal Time Coordinate) offset (in seconds) from the NTP server. The default offset is 0. The offset range is –43200 to 43200 seconds.
NTP Update Interval **[NTPUpdateInterval]**	Defines the time interval (in seconds) the NTP client requests for a time update. The default interval is 86400 seconds (24 hours). The range is 0 to 214783647 seconds. **Note:** It isn't recommended to be set beyond one month (2592000 seconds).
Telnet Settings	
Embedded Telnet Server **[TelnetServerEnable]**	Enables or disables the embedded Telnet server. Telnet is disabled by default for security reasons. Disable **[0]** (default). Enable (Unsecured) **[1]**. Enable Secured (SSL) **[2]** = N/A.
Telnet Server TCP Port **[TelnetServerPort]**	Defines the port number for the embedded Telnet server. The valid range = valid port numbers. The default port is 23.
Telnet Server Idle Timeout **[TelnetServerIdleDisconnect]**	Sets the timeout for disconnection of an idle Telnet session (in minutes). When set to zero, idle sessions are not disconnected. The valid range is any value. The default value is 0.
STUN Settings	
Enable STUN **[EnableSTUN]**	Disable **[0]** = STUN protocol is disabled (default). Enable **[1]** = STUN protocol is enabled. When enabled, the gateway functions as a STUN client and communicates with a STUN server located in the open internet. STUN is used to discover whether the gateway is located behind a NAT and the type of that NAT. In addition, it is used to determine the IP addresses and port numbers that the NAT assigns to outgoing signaling messages (using SIP) and media streams (using RTP, RTCP and T.38). STUN works with many existing NAT types, and does not require any special behavior from them. This parameter cannot be changed on-the-fly and requires a gateway reset. For detailed information on STUN, refer to Section 9.2.1 on page 248.
STUN Server Primary IP **[STUNServerPrimaryIP]**	Defines the IP address of the primary STUN server. The valid range is the legal IP addresses. The default value is 0.0.0.0. **Note:** Instead of using this parameter, you can define the STUN server's domain name (using the *ini* file parameter StunServerDomainName).
STUN Server Secondary IP **[STUNServerSecondaryIP]**	Defines the IP address of the secondary STUN server. The valid range is the legal IP addresses. The default value is 0.0.0.0.
NFS Settings	
NFS Table	For detailed information on configuring the NFS table, refer to Section 5.6.1.3 on page 143.

5.6.1.3 Configuring the NFS Settings

Network File System (NFS) enables the MediaPack to access a remote server's shared files and directories and to handle them as if they're located locally. A file system, the NFS is independent of machine types, OSs, and network architectures. Up to five different NFS file systems can be configured.

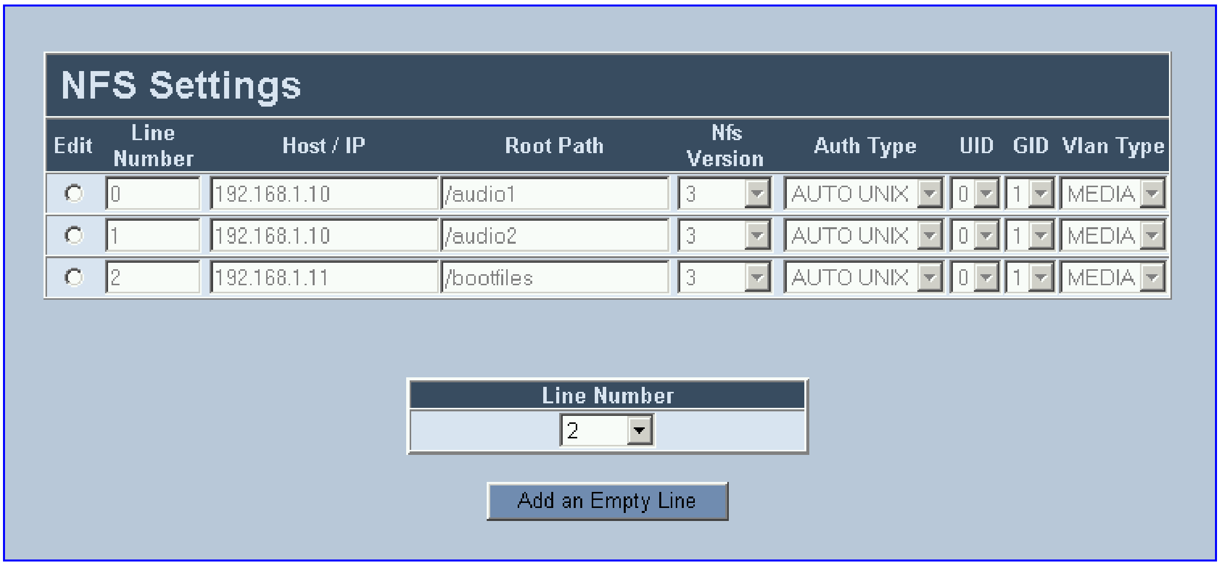
NFS is utilized by the MediaPack to load the *cmp*, *ini* and configuration files via the Automatic Update mechanism (refer to Section 10.3 on page 263).

Note that an NFS file server can share multiple file systems. There must be a separate row for each remote file system shared by the NFS file server that needs to be accessed by the MediaPack.

➢ To configure the NFS Settings parameters, take these 7 steps:

1. Open the 'Application Settings' screen (**Advanced Configuration** menu > **Network Settings** > **Application Settings** option); the 'Application Settings' screen is displayed (Figure 5-36).
2. Open the NFS Table screen by clicking the arrow sign (-->) to the right of the NFS label; the NFS Table screen is displayed (Figure 5-37).

Figure 5-37: NFS Settings Table Screen



3. To add a remote NFS file system, select an available line number from the 'Line Number' drop-down list.
4. Click the **Add an Empty Line** button; an empty line appears.
5. Configure the NFS Settings according to Table 5-38.
6. Click the **Apply New Settings** button; the remote NFS file system is mounted immediately. Check the Syslog server for the 'NFS mount was successful' message.
7. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Note: To avoid terminating calls in progress, a row must not be deleted or modified while the board is currently accessing files on that remote NFS file system.

➢ To delete a remote NFS file system, take these 3 steps:

1. Select the 'Edit' radio button for the row to be deleted.
2. Click the **Delete Line** button; the row is deleted.
3. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

➢ To modify an existing remote NFS file system, take these 4 steps:

1. Select the 'Edit' radio button for the row to be modified.
2. Change the values on the selected row according to your requirements.
3. Click the **Apply New Settings** button; the remote NFS file system is mounted using the new settings. Check the Syslog server for the 'NFS mount was successful' message.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-38: Network Settings, NFS Settings Parameters

Parameter	Description
Line Number **[NFSServers_Index]**	The row index of the remote file system. The valid range is 0 to 4.
Host / IP **[NFSServers_HostOrIP]**	The domain name or IP address of the NFS server. If a domain name is provided, a DNS server must be configured.
Root Path **[NFSServers_RootPath]**	Path to the root of the remote file system. In the format: '/' + [path] For example, /audio
The combination of Host / IP and Root Path must be unique for each row in the table. For example, there must be only one row in the table with a Host / IP of 192.168.1.1 and Root Path of /audio.	
NFS Version **[NFSServers_NfsVersion]**	NFS version to use with the remote file system, 2 or 3 (default).
Auth Type **[NFSServers_AuthType]**	Identifies the authentication method used with the remote file system. AUTH_NULL **[0]**. AUTH_UNIX **[1]** (default).
UID **[NFSServers_UID]**	User ID used in authentication if using AUTH_UNIX. The valid range is 0 to 65537. The default is 0.
GID **[NFSServers_GID]**	Group ID used in authentication if using AUTH_UNIX. The valid range is 0 to 65537. The default is 1
VLAN Type **[NFSServers_VlanType]**	The VLAN, OAM **[0]** or Media **[1]**, to use when accessing the remote file system. The default is to use the media VLAN. This parameter applies only if VLANs are enabled or if Multiple IPs is configured (refer to Section 9.8 on page 253).

Figure 5-38 below shows an example of an NFS table definition via *ini* file using parameter tables (for information on parameter tables, refer to Section 10.5 on page 267).

Figure 5-38: NFS *ini* File Example

```
[ NFSServers ]
FORMAT NFSServers_Index = NFSServers_HostOrIP, NFSServers_RootPath,
NFSServers_NfsVersion, NFSServers_AuthType, NFSServers_UID, NFSServers_GID,
NFSServers_VlanType;
NFSServers 1 = 101.1.13, /audio1, 3, 1, 0, 1, 1;
[ \NFSServers ]
```

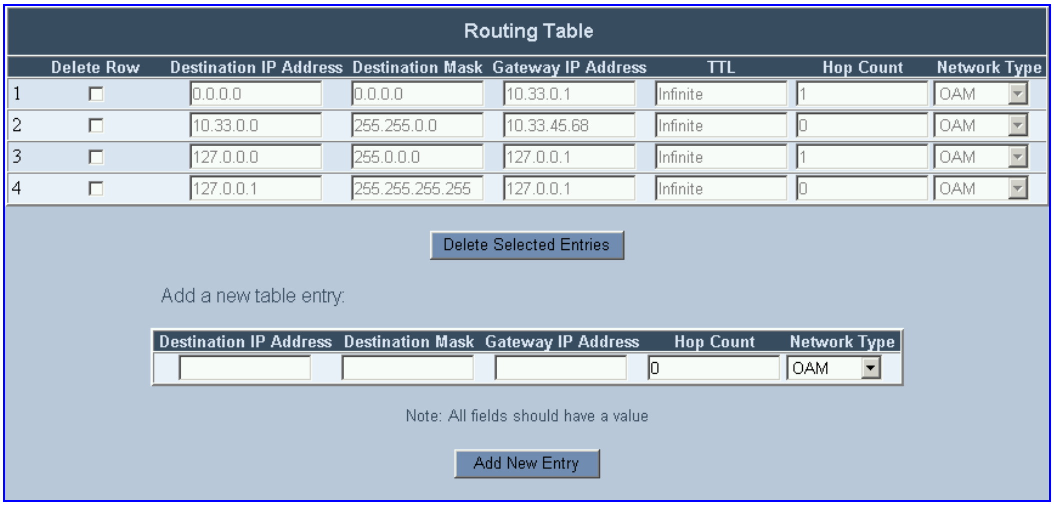
5.6.1.4 Configuring the IP Routing Table

The IP routing table is used by the gateway to determine IP routing rules. It can be used, for example, to define static routing rules for the OAM and Control networks since a default gateway isn't supported for these networks (refer to Section 9.9.1 on page 254). Before sending an IP packet, the gateway searches this table for an entry that matches the requested destination host / network. If such entry is found, the gateway sends the packet to the indicated router. If no explicit entry is found, the packet is sent to the default gateway (configured in **Network Settings** > 'IP Settings' screen). Up to 50 routing entries are available.

➢ To configure the IP Routing table, take these 3 steps:

1. Open the 'IP Routing Table' screen (**Advanced Configuration** menu > **Network Settings** > **Routing Table** option); the 'IP Routing Table' screen is displayed.

Figure 1-3: IP Routing Table Screen



2. Use the 'Add a new table entry' pane to add a new routing rule. Each field in the IP routing table is described in Table 5-39.
3. Click the button **Add New Entry**; the new routing rule is added to the IP routing table.

Table 5-39: IP Routing Table Column Description (continues on pages 145 to 146)

Column Name **[*ini* File Parameter Name]**	**Description**
Delete Row	To delete IP routing rules from the IP Routing Table, check the Delete Row checkbox in the rows of the routing rules you want to delete and click the button **Delete Selected Entries**; the routing rules are removed from the table.
Destination IP Address **[RoutingTableDestinationsColumn]**	Specifies the IP address of the destination host / network.
Destination Mask **[RoutingTableDestinationMasksColumn]**	Specifies the subnet mask of the destination host / network.

Table 5-39: IP Routing Table Column Description (continues on pages 145 to 146)

<table>
<tr><th>Column Name
[ini File Parameter Name]</th><th>Description</th></tr>
<tr><td colspan="2">The address of the host / network you want to reach is determined by an AND operation that is applied on the fields ‘Destination IP Address’ and ‘Destination Mask’.
For example:
To reach the network 10.8.x.x, enter 10.8.0.0 in the field ‘Destination IP Address’ and 255.255.0.0 in the field ‘Destination Mask’. As a result of the AND operation, the value of the last two octets in the field ‘Destination IP Address’ is ignored.
To reach a specific host, enter its IP address in the field ‘Destination IP Address’ and 255.255.255.255 in the field ‘Destination Mask’.</td></tr>
<tr><td>Gateway IP Address
[RoutingTableGatewaysColumn]</td><td>Specifies the IP address of the router to which the packets are sent if their destination matches the rules in the adjacent columns.</td></tr>
<tr><td>TTL</td><td>A read-only field that indicates the time period for which the specific routing rule is valid. The lifetime of a static route is infinite.</td></tr>
<tr><td>Hop Count
[RoutingTableHopsCountColumn]</td><td>The maximum number of allowed routers between the gateway and destination.</td></tr>
<tr><td>Network Type
[RoutingTableInterfacesColumn]</td><td>Specifies the network type the routing rule is applied to.
OAM [0] (default).
Control [1].
Media [2].
For detailed information on the network types, refer to Section 9.8 on page 253.</td></tr>
<tr><th colspan="2">ini File Example</th></tr>
<tr><td colspan="2">The IP routing ini file parameters are array parameters. Each parameter configures a specific column in the IP routing table. The first entry in each parameter refers to the first row in the IP routing table, the second entry to the second row and so forth.
In the following example two rows are configured when the gateway is in network 10.31.x.x:
RoutingTableDestinationsColumn = 130.33.4.6, 83.4.87.6
RoutingTableDestinationMasksColumn = 255.255.255.255, 255.255.255.0
RoutingTableGatewaysColumn = 10.31.0.1, 10.31.0.112
RoutingTableInterfacesColumn = 0, 1
RoutingTableHopsCountColumn = 20, 20</td></tr>
</table>

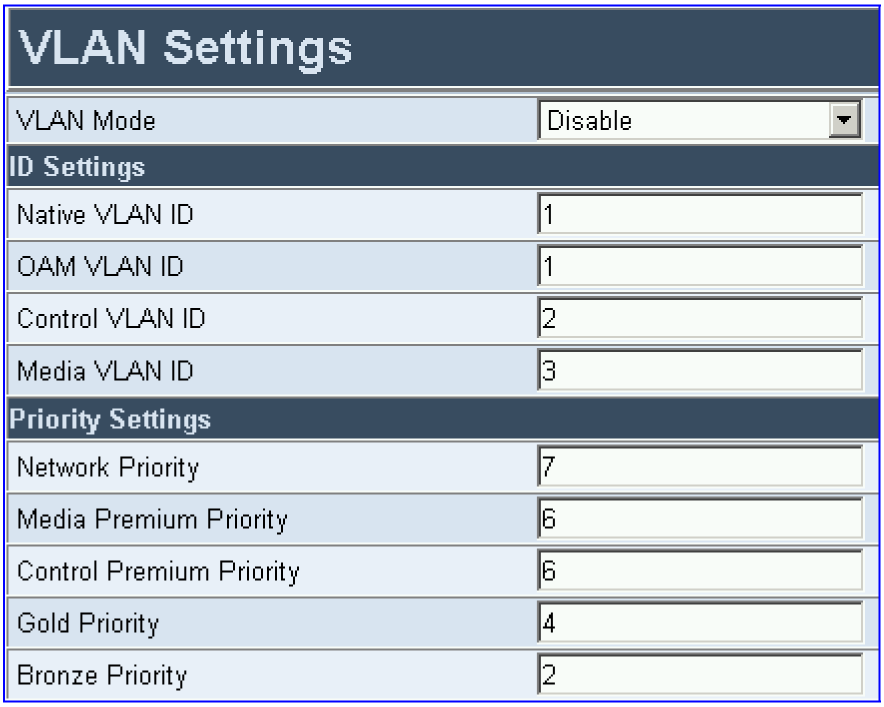
5.6.1.5 Configuring the VLAN Settings

For detailed information on the MediaPack VLAN implementation, refer to Section 9.8 on page 253.

➢ **To configure the VLAN Settings parameters, take these 4 steps:**

1. Open the ‘VLAN Settings’ screen (**Advanced Configuration** menu > **Network Settings** > **VLAN Settings** option); the ‘VLAN Settings’ screen is displayed.

Figure 5-39: VLAN Settings Screen



2. Configure the VLAN Settings according to Table 5-40.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-40: Network Settings, VLAN Settings Parameters

Parameter	Description
VLAN Mode **[VlanMode]**	Sets the VLAN functionality. Disable **[0]** (default). Enable **[1]**. PassThrough **[2]** = N/A. **Note:** This parameter cannot be changed on-the-fly and requires a gateway reset.
IP Settings	
Native VLAN ID **[VlanNativeVlanID]**	Sets the native VLAN identifier (PVID, Port VLAN ID). The valid range is 1 to 4094. The default value is 1.
OAM VLAN ID **[VlanOamVlanID]**	Sets the OAM (Operation, Administration and Management) VLAN identifier. The valid range is 1 to 4094. The default value is 1.
Control VLAN ID **[VlanControlVlanID]**	Sets the control VLAN identifier. The valid range is 1 to 4094. The default value is 2.
Media VLAN ID **[VlanMediaVlanID]**	Sets the media VLAN identifier. The valid range is 1 to 4094. The default value is 3.
Priority Settings	
Network Priority **[VlanNetworkServiceClassPriority]**	Sets the priority for Network service class content. The valid range is 0 to 7. The default value is 7.
Media Premium Priority **[VlanPremiumServiceClassMediaPriority]**	Sets the priority for the Premium service class content and media traffic. The valid range is 0 to 7. The default value is 6.
Control Premium Priority **[VlanPremiumServiceClassControlPriority]**	Sets the priority for the Premium service class content and control traffic. The valid range is 0 to 7. The default value is 6.
Gold Priority **[VlanGoldServiceClassPriority]**	Sets the priority for the Gold service class content. The valid range is 0 to 7. The default value is 4.
Bronze Priority **[VlanBronzeServiceClassPriority]**	Sets the priority for the Bronze service class content. The valid range is 0 to 7. The default value is 2.
***ini* File Parameters**	
EnableDNSasOAM	Determines the traffic type for DNS services. **[1]** = OAM VLAN (default). **[0]** = Control VLAN.
EnableNTPasOAM	Determines the traffic type for NTP services. **[1]** = OAM VLAN (default). **[0]** = Control VLAN.
VlanSendNonTaggedOnNative	Specify whether to send non-tagged packets on the native VLAN. **[0]** = Sends priority tag packets (default). **[1]** = Sends regular packets (with no VLAN tag).

5.6.1.6 Network Settings *ini* File Parameters

Table 5-41 describes the Network parameters that can only be configured via the *ini* file.

Table 5-41: Network Settings, *ini* File Parameters (continues on pages 149 to 151)

ini File Parameter Name	Valid Range and Description
EnablePPPoE	Enables the PPPoE (Point-to-Point Protocol over Ethernet) feature. **[0]** = Disable (default) **[1]**= Enable
PPPoEUserName	User Name for PAP or Host Name for CHAP authentication. The valid range is a string of up to 47 characters. The default value is 0.
PPPoEPassword	Password for PAP or Secret for CHAP authentication. The valid range is a string of up to 47 characters. The default value is 0.
PPPoEServerName	Server Name for CHAP authentication. The valid range is a string of up to 47 characters. The default value is 0.
PPPoEStaticIPAddress	IP address to use in a static configuration setup. If set, used during PPP negotiation to request this specific IP address from the PPP server. If approved by the server, this IP address is used during the session. The valid IP address range is in dotted notation xxx.xxx.xxx.xxx. The default value is 0.0.0.0.
PPPoERecoverIPAddress	IP address to use when booting from the flash to non-PPPoE (Point-to-Point Protocol over Ethernet) environments. The valid IP address range is in dotted notation xxx.xxx.xxx.xxx. The default value is 10.4.10.4.
PPPoERecoverSubnetMask	Subnet Mask to use when booting from the flash to non-PPPoE (Point-to-Point Protocol over Ethernet) environments. The valid IP address range is in dotted notation xxx.xxx.xxx.xxx. The default value is 255.255.0.0.
PPPoERecoverDfgwAddress	Default GW address to use when booting from the flash to non-PPPoE (Point-to-Point Protocol over Ethernet) environments. The valid IP address range is in dotted notation xxx.xxx.xxx.xxx. The default value is 10.4.10.1.
PPPoELCPEchoEnable	Enables or disables the Point-to-Point Protocol over Ethernet (PPPoE) disconnection auto-detection feature. Valid options include: **[0]** = Disable **[1]** = Enable (default) By default, the PPPoE Client (i.e., embedded in the gateway) sends LCP Echo packets to the server to check that the PPPoE connection is open. Some Access Concentrators (PPPoE servers) don't reply to these LCP Echo requests, resulting in a disconnection. By disabling the LCP disconnection auto-detection feature, the PPPoE Client does not send LCP Echo packets to the server (and does not detect PPPoE disconnections).
EthernetPhyConfiguration	**[0]** = 10 Base-T half-duplex. **[1]** = 10 Base-T full-duplex. **[2]** = 100 Base-TX half-duplex. **[3]** = 100 Base-TX full-duplex. **[4]** = Auto-Negotiate (default). For detailed information on Ethernet interface configuration, refer to Section 9.1 on page 247.

Table 5-41: Network Settings, *ini* File Parameters (continues on pages 149 to 151)

ini File Parameter Name	Valid Range and Description
DisableNAT	Enables / disables the Network Address Translation (NAT) mechanism. **[0]** = Enabled. **[1]**= Disabled (default). **Note:** The compare operation that is performed on the IP address is enabled by default and is controlled by the parameter 'EnableIPAddrTranslation'. The compare operation that is performed on the UDP port is disabled by default and is controlled by the parameter 'EnableUDPPortTranslation'.
EnableIPAddrTranslation	**[0]** = Disable IP address translation. **[1]** = Enable IP address translation for RTP, RTCP and T.38 packets (default). **[2]** = Enable IP address translation for ThroughPacket™. **[3]** = Enable IP address translation for all protocols (RTP, RTCP, T38 and ThroughPacket™). When enabled, the gateway compares the source IP address of the first incoming packet, to the remote IP address stated in the opening of the channel. If the two IP addresses don't match, the NAT mechanism is activated. Consequently, the remote IP address of the outgoing stream is replaced by the source IP address of the first incoming packet. **Note:** The NAT mechanism must be enabled for this parameter to take effect (DisableNAT = 0).
EnableUDPPortTranslation	**[0]** = Disable UDP port translation (default). **[1]** = Enable UDP port translation. When enabled, the gateway compares the source UDP port of the first incoming packet, to the remote UDP port stated in the opening of the channel. If the two UDP ports don't match, the NAT mechanism is activated. Consequently, the remote UDP port of the outgoing stream is replaced by the source UDP port of the first incoming packet. **Note:** The NAT mechanism and the IP address translation must be enabled for this parameter to take effect (DisableNAT = 0, EnableIpAddrTranslation = 1).
NoOperationSendingMode	Enables or disables the transmission of RTP or T.38 No-Op packets. Valid options include: **[0]** = Disable (default) **[1]** = Enable This mechanism ensures that the NAT binding remains open during RTP or T.38 silence periods.
RTPNoOpEnable	This parameter is now obsolete. Please use the parameter NoOperationSendingMode.

Table 5-41: Network Settings, *ini* File Parameters (continues on pages 149 to 151)

ini File Parameter Name	Valid Range and Description
NoOpInterval	Defines the time interval in which RTP or T.38 No-Op packets are sent in the case of silence (no RTP / T.38 traffic) when No-Op packet transmission is enabled. The valid range is 20 to 65,000 msec. The default is 10,000. **Note:** To enable No-Op packet transmission, use the NoOperationSendingMode parameter.
RTPNoOpInterval	This parameter is now obsolete. Please use the parameter NoOpInterval.
RTPNoOpPayloadType	Determines the payload type of No-Op packets. The valid range is 96 to 127. The default value is 120.
EnableDetectRemoteMACChange	Changes the RTP packets according to the MAC address of received RTP packets and according to Gratuitous Address Resolution Protocol (GARP) messages. Valid options include: **[0]** = nothing is changed. **[1]** = If the gateway receives RTP packets with a different source MAC address (than the MAC address of the transmitted RTP packets), then it sends RTP packets to this MAC address and removes this IP entry from the gateway's ARP cache table. **[2]** = The gateway uses the received GARP packets to change the MAC address of the transmitted RTP packets. **[3]** = both 1 and 2 options above are used (default).

5.6.2 Configuring the Media Settings

From the Media Settings page you can define:

- Voice Settings (refer to Section 5.6.2.1 below).
- Fax / Modem / CID Settings (refer to Section 5.6.2.2 on page 154).
- RTP/RTCP Settings (refer to Section 5.6.2.3 on page 157).
- Hook-Flash Settings (refer to Section 5.6.2.4 on page 160).
- General Media Settings (refer to Section 5.6.2.5 on page 161).

These parameters are applied to all MediaPack channels.

Notes:

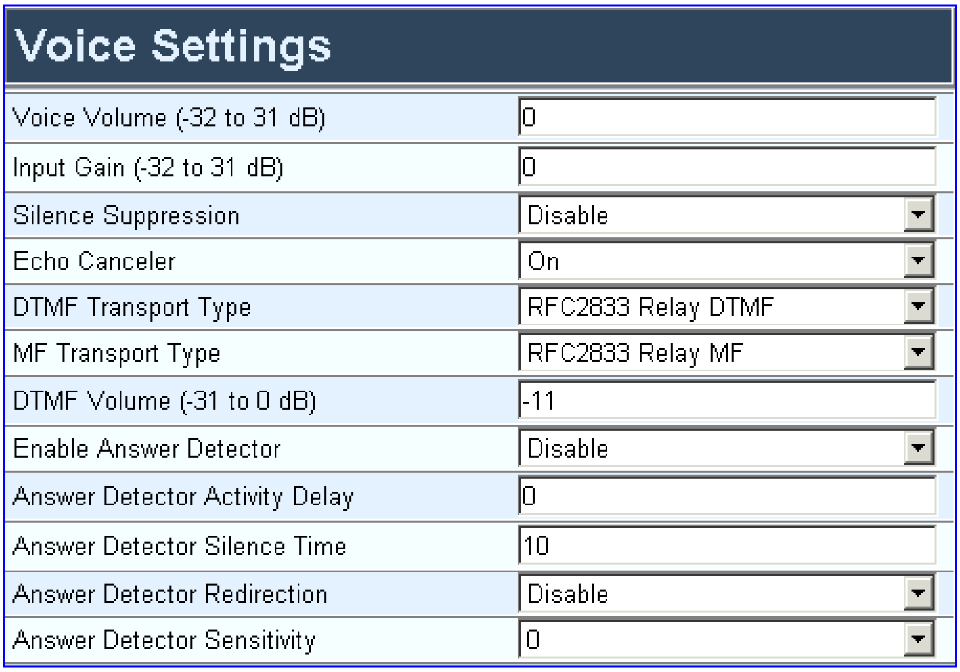
- Several Channels Settings parameters can be configured per call using profiles (refer to Section 5.5.6 on page 108
- Those parameters contained within square brackets are the names used to configure the parameters via the *ini* file.
- Channel parameters are changeable on-the-fly. Changes take effect from next call.

5.6.2.1 Configuring the Voice Settings

➢ **To configure the Voice Settings parameters, take these 4 steps:**

1. Open the 'Voice Settings' screen (**Advanced Configuration** menu > **Media Settings** > **Voice Settings** option); the 'Voice Settings' screen is displayed.

Figure 5-40: Voice Settings Screen



Voice Settings	
Voice Volume (-32 to 31 dB)	0
Input Gain (-32 to 31 dB)	0
Silence Suppression	Disable
Echo Canceler	On
DTMF Transport Type	RFC2833 Relay DTMF
MF Transport Type	RFC2833 Relay MF
DTMF Volume (-31 to 0 dB)	-11
Enable Answer Detector	Disable
Answer Detector Activity Delay	0
Answer Detector Silence Time	10
Answer Detector Redirection	Disable
Answer Detector Sensitivity	0

2. Configure the Voice Settings according to Table 5-42.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-42: Media Settings, Voice Settings Parameters

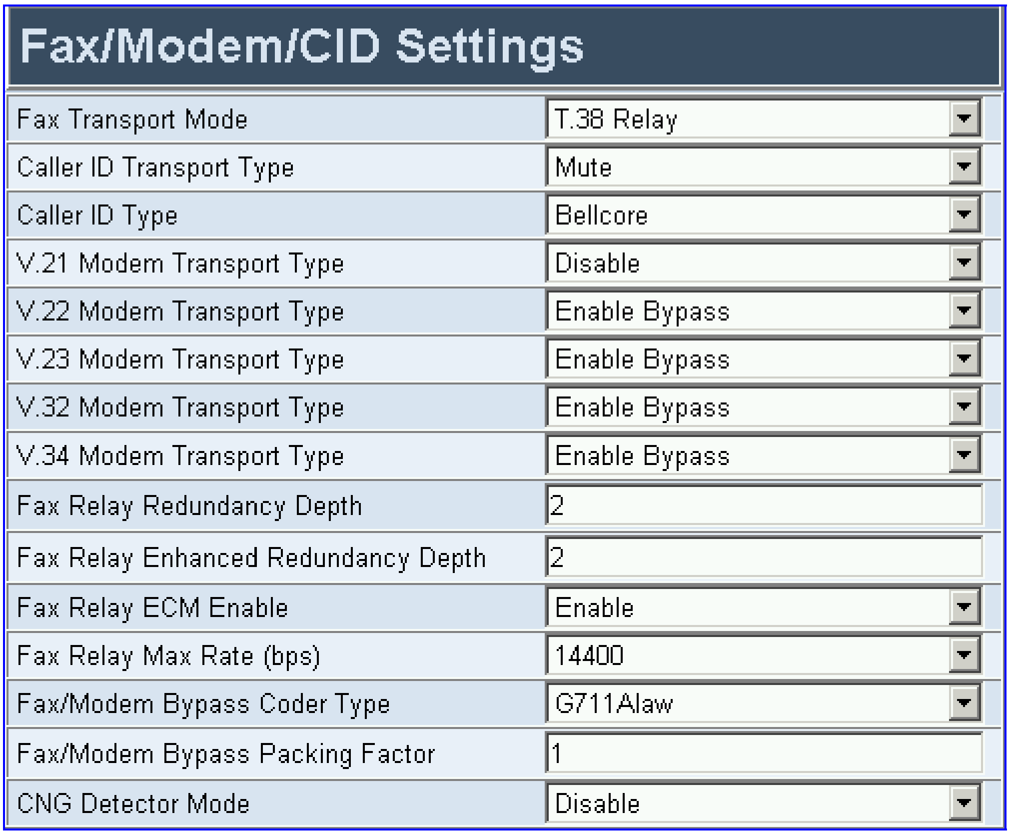
Parameter	Description
Voice Volume **[VoiceVolume]**	Voice gain control in dB. This parameter sets the level for the transmitted (IP→Tel) signal. The valid range is -32 to 31 dB. The default value is 0 dB.
Input Gain **[InputGain]**	PCM input gain control in dB. This parameter sets the level for the received (Tel→IP) signal. The valid range is -32 to 31 dB. The default value is 0 dB. **Note:** This parameter is intended for advanced users. Changing it affects other gateway functionalities.
Silence Suppression **[EnableSilenceCompression]** The parameter is used to maintain backward compatibility.	Disable **[0]** = Silence Suppression disabled (default). Enable **[1]** = Silence Suppression enabled. Enable without adaptation **[2]** = A single silence packet is sent during silence period (applicable only to G.729). Silence Suppression is a method conserving bandwidth on VoIP calls by not sending packets when silence is detected. **Note:** If the selected coder is G.729, the following rules determine the value of the 'annexb' parameter of the fmtp attribute in the SDP. EnableSilenceCompression = 0 → 'annexb=no'. EnableSilenceCompression = 1 → 'annexb=yes'. EnableSilenceCompression = 2 and IsCiscoSCEMode = 0 → 'annexb=yes'. EnableSilenceCompression = 2 and IsCiscoSCEMode = 1 → 'annexb=no'.
Echo Canceler **[EnableEchoCanceller]** The parameter is used to maintain backward compatibility.	Off **[0]** = Echo Canceler disabled. On **[1]** = Echo Canceler enabled (default).
DTMF Transport Type **[DTMFTransportType]**	DTMF Mute **[0]** = Erase digits from voice stream, do not relay to remote. Transparent DTMF **[2]** = Digits remain in voice stream. RFC 2833 Relay DTMF **[3]** = Erase digits from voice stream, relay to remote according to RFC 2833 (default). **Note:** This parameter is automatically updated if one of the following parameters is configured: TxDTMFOption or RxDTMFOption.
MF Transport Type **[MFTransportType]**	N/A.
DTMF Volume (-31 to 0 dB) **[DTMFVolume]**	DTMF gain control value in dB (to the analog side). The valid range is -31 to 0 dB. The default value is -11 dB.
Enable Answer Detector **[EnableAnswerDetector]**	N/A.
Answer Detector Activity Delay **[AnswerDetectorActivityDelay]**	N/A.
Answer Detector Silence Time **[AnswerDetectorSilenceTime]**	N/A.
Answer Detector Redirection **[AnswerDetectorRedirection]**	N/A.
Answer Detector Sensitivity **[AnswerDetectorSensitivity]**	Determines the Answer Detector sensitivity. The range is 0 (most sensitive) to 2 (least sensitive). The default is 0.

5.6.2.2 Configuring the Fax / Modem / CID Settings

➢ **To configure the Fax / Modem / CID Settings parameters, take these 4 steps:**

1. Open the 'Fax / Modem / CID Settings' screen (**Advanced Configuration** menu > **Media Settings** > **Fax / Modem / CID Settings** option); the 'Fax / Modem / CID Settings' screen is displayed.

Figure 5-41: Fax / Modem / CID Settings Screen



2. Configure the Fax / Modem / CID Settings according to Table 5-43.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-43: Media Settings, Fax/Modem/CID Parameters (continues on pages 154 to 157)

Parameter	Description
Fax Transport Mode **[FaxTransportMode]**	Fax Transport Mode that the gateway uses. You can select: Disable **[0]** (transparent mode). T.38 Relay **[1]** (default). Bypass **[2]**. Events Only **[3]**. **Note:** If parameter IsFaxUsed = 1, then FaxTransportMode is always set to 1 (T.38 relay).
Caller ID Transport Type **[CallerIDTransportType]**	N/A.

Table 5-43: Media Settings, Fax/Modem/CID Parameters (continues on pages 154 to 157)

Parameter	Description
Caller ID Type **[CallerIDType]**	Defines one of the following standards for detection (FXO) and generation (FXS) of Caller ID and detection (FXO) of MWI (when specified) signals. Bellcore **[0]** (Caller ID and MWI) (default). ETSI **[1]** (Caller ID and MWI) NTT **[2]**. Backward Compatible **[3]** British **[4]** DTMF ETSI **[16]** Denmark **[17]** (Caller ID and MWI) India **[18]** Brazil **[19]** **Note 1:** The Caller ID signals are generated/detected between the first and the second rings. **Note 2:** To select the Bellcore Caller ID sub standard, use the parameter 'BellcoreCallerIDTypeOneSubStandard'. To select the ETSI Caller ID sub standard, use the parameter 'ETSICallerIDTypeOneSubStandard'. **Note 3:** To select the Bellcore MWI sub standard, use the parameter 'BellcoreVMWITypeOneStandard'. To select the ETSI MWI sub standard, use the parameter 'ETSIVMWITypeOneStandard'. **Note 4:** If you define NTT (i.e., 2) for the caller ID type, you need to define the NTT DID signaling form (FSK or DTMF) using NTTDIDSignallingForm.
V.21 Modem Transport Type **[V21ModemTransportType]**	V.21 Modem Transport Type that the gateway uses. You can select: **[0]** = Disable (Transparent) -- default **[1]** = Enable Relay – N/A **[2]** = Enable Bypass 3 = Events Only (Transparent with Events)
V.22 Modem Transport Type **[V22ModemTransportType]**	V.22 Modem Transport Type that the gateway uses. You can select: **[0]** = Disable (Transparent) **[1]** = Enable Relay -- N/A **[2]** = Enable Bypass -- default **[3]** = Events Only (Transparent with Events)
V.23 Modem Transport Type **[V23ModemTransportType]**	V.23 Modem Transport Type that the gateway uses. You can select: **[0]** = Disable (Transparent) **[1]** = Enable Relay -- N/A **[2]** = Enable Bypass -- default **[3]** = Events Only (Transparent with Events)
V.32 Modem Transport Type **[V32ModemTransportType]**	V.32 Modem Transport Type that the gateway uses. You can select: **[0]** = Disable (Transparent) **[1]** = Enable Relay -- N/A **[2]** = Enable Bypass -- default **[3]** = Events Only (Transparent with Events) **Note:** This option applies to V.32 and V.32bis modems.
V.34 Modem Transport Type **[V34ModemTransportType]**	V.90 / V.34 Modem Transport Type that the gateway uses. You can select: **[0]** = Disable (Transparent) **[1]** = Enable Relay -- N/A **[2]** = Enable Bypass -- default **[3]** = Events Only (Transparent with Events)
Fax Relay Redundancy Depth **[FaxRelayRedundancyDepth]**	Number of times that each fax relay payload is retransmitted to the network. The valid range is 0 to 2. The default value is 0.

Table 5-43: Media Settings, Fax/Modem/CID Parameters (continues on pages 154 to 157)

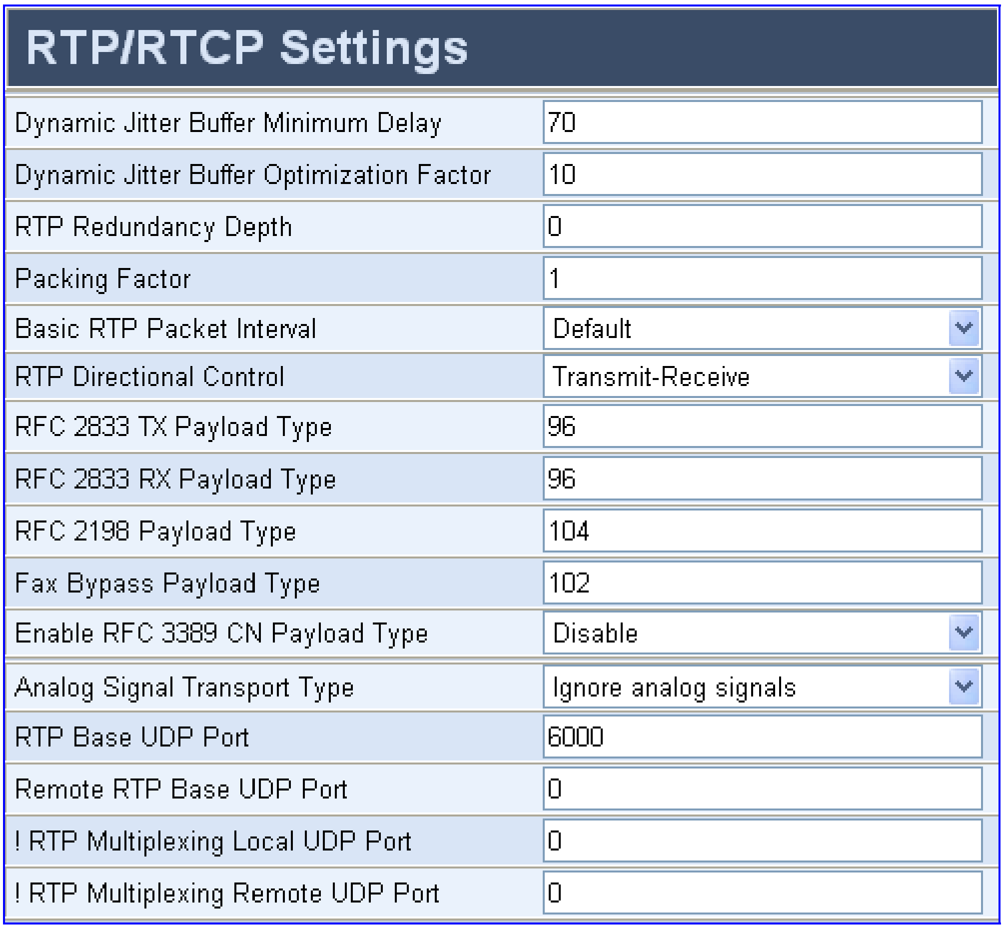
Parameter	Description
Fax Relay Enhanced Redundancy Depth **[FaxRelayEnhancedRedundancyDepth]**	Number of times that control packets are retransmitted when using the T.38 standard. The valid range is 0 to 4. The default value is 2.
Fax Relay ECM Enable **[FaxRelayECMEnable]**	Disable **[0]** = Error Correction Mode (ECM) mode is not used during fax relay. Enable **[1]** = ECM mode is used during fax relay (default).
Fax Relay Max Rate (bps) **[FaxRelayMaxRate]**	Maximum rate, in bps, at which fax relay messages are transmitted (outgoing calls). You can select: 2400 **[0]** = 2.4 kbps. 4800 **[1]** = 4.8 kbps. 7200 **[2]** = 7.2 kbps. 9600 **[3]** = 9.6 kbps. 12000 **[4]** = 12.0 kbps. 14400 **[5]** = 14.4 kbps (default). **Note:** The rate is negotiated between the sides, i.e., the media server adapts to the capabilities of the remote side.
Fax/Modem Bypass Coder Type **[FaxModemBypassCoderType]**	Coder the gateway uses when performing fax/modem bypass. Usually, high-bit-rate coders such as G.711 should be used. You can select: G.711 A-law 64 **[0]** (default). G.711 μ-law **[1]**.
Fax/Modem Bypass Packing Factor **[FaxModemBypassM]**	Number of (20 msec) coder payloads that are used to generate a fax/modem bypass packet. The valid range is 1, 2 or 3 coder payloads. The default value is 1 coder payload.
CNG Detector Mode **[CNGDetectorMode]**	**[0]** = Disable (default). Don't detect CNG. **[1]** = Relay. CNG is detected on the originating side. CNG packets are sent to the remote side according to T.38 (if IsFaxUsed=1) and the fax session is started. **[2]** = Events Only. CNG is detected on the originating side. The CNG signal passes transparently to the remote side and fax session is started. Usually T.38 fax session starts when the 'preamble' signal is detected by the answering side. Some SIP gateways don't support the detection of this fax signal on the answering side, thus, for these cases it is possible to configure the gateways to start the T.38 fax session when the CNG tone is detected by the originating side. However, this mode is not recommended.

5.6.2.3 Configuring the RTP / RTCP Settings

➢ **To configure the RTP / RTCP Settings parameters, take these 4 steps:**

1. Open the 'RTP / RTCP Settings' screen (**Advanced Configuration** menu > **Media Settings** > **RTP / RTCP Settings** option); the 'RTP / RTCP Settings' screen is displayed.

Figure 5-42: RTP / RTCP Settings Screen



2. Configure the RTP / RTCP Settings according to Table 5-44.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-44: Media Settings, RTP / RTCP Parameters (continues on pages 158 to 159)

Parameter	Description
Dynamic Jitter Buffer Minimum Delay **[DJBufMinDelay]**	Minimum delay for the Dynamic Jitter Buffer. The valid range is 0 to 150 milliseconds. The default delay is 10 milliseconds. **Note:** For more information on the Jitter Buffer, refer to Section 8.6 on page 230.
Dynamic Jitter Buffer Optimization Factor **[DJBufOptFactor]**	Dynamic Jitter Buffer frame error / delay optimization factor. The valid range is 0 to 13. The default factor is 10. **Note 1:** Set to 13 for data (fax & modem) calls. **Note 2:** For more information on the Jitter Buffer, refer to Section 8.6 on page 230.
RTP Redundancy Depth **[RTPRedundancyDepth]**	Enter **[0]** to disable the generation of redundant packets (default). Enter **[1]** to enable the generation of RFC 2198 redundancy packets.
Packing Factor **[RTPPackingFactor]**	N/A. Controlled internally by the gateway according to the selected coder.
Basic RTP Packet Interval **[BasicRTPPacketInterval]** **Note:** This parameter should not be used. Use the 'Coders' screen under 'Protocol Definition' instead.	N/A. Controlled internally by the gateway according to the selected coder.
RTP Directional Control **[RTPDirectionControl]**	N/A. Controlled internally by the gateway according to the selected coder.
RFC 2833 TX Payload Type **[RFC2833TxPayloadType]**	N/A. Use the *ini* file parameter RFC2833PayloadType instead.
RFC 2833 RX Payload Type **[RFC2833RxPayloadType]**	N/A. Use the *ini* file parameter RFC2833PayloadType instead.
RFC 2198 Payload Type **[RFC2198PayloadType]**	RTP redundancy packet payload type, according to RFC 2198. The range is 96-127. The default is 104. Applicable if 'RTP Redundancy Depth=1'
Fax Bypass Payload Type **[FaxBypassPayloadType]**	Determines the fax bypass RTP dynamic payload type. The valid range is 96 to 120. The default value is 102.
Enable RFC 3389 CN Payload Type **[EnableStandardSIDPayloadType]**	Determines whether Silence Indicator (SID) packets that are sent and received are according to RFC 3389. Disable **[0]** = G.711 SID packets are sent in a proprietary method (default). Enable **[1]** = SID (comfort noise) packets are sent with the RTP SID payload type according to RFC 3389. Applicable to G.711 and G.726 coders.
Analog Signal Transport Type **[AnalogSignalTransportType]**	Ignore analog signals **[0]** = Hook-flash isn't transferred to the remote side (default). RFC 2833 analog signal relay **[1]** = Hook-flash is transferred via RFC 2833.

Table 5-44: Media Settings, RTP / RTCP Parameters (continues on pages 158 to 159)

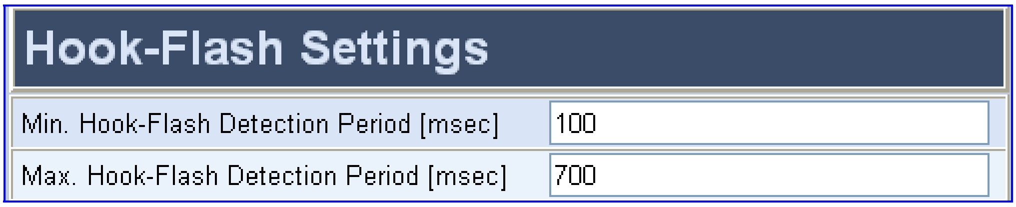
Parameter	Description
RTP Base UDP Port **[BaseUDPPort]**	Lower boundary of UDP port used for RTP, RTCP (Real-Time Control Protocol) (RTP port + 1) and T.38 (RTP port + 2). The upper boundary is the Base UDP Port + 10 * (number of gateway's channels). The range of possible UDP ports is 6,000 to 64,000. The default base UDP port is 6000. For example: If the Base UDP Port is set to 6000 (the default) then: The first channel uses the following ports: RTP 6000, RTCP 6001 and T.38 6002, the second channel uses: RTP 6010, RTCP 6011 and T.38 6012, etc. **Note:** If RTP Base UDP Port is not a factor of 10, the following message is generated: 'invalid local RTP port'. For detailed information on the default RTP/RTCP/T.38 port allocation, refer to the Section D.3 on page 362.
Remote RTP Base UDP Port **[RemoteBaseUDPPort]**	Determines the lower boundary of UDP ports used for RTP, RTCP and T.38 by a remote gateway. If this parameter is set to a non-zero value, ThroughPacket™ is enabled. Note that the value of 'RemoteBaseUDPPort' on the local gateway must equal the value of 'BaseUDPPort' of the remote gateway. The gateway uses these parameters to identify and distribute the payloads from the received multiplexed IP packet to the relevant channels. The valid range is the range of possible UDP ports: 6,000 to 64,000. The default value is 0 (ThroughPacket™ is disabled). **Note:** To enable ThroughPacket™ the parameters 'L1L1ComplexTxUDPPort' and 'L1L1ComplexRxUDPPort' must be set to a non-zero value.
RTP Multiplexing Local UDP Port **[L1L1ComplexTxUDPPort]**	Determines the local UDP port used for outgoing multiplexed RTP packets (applies to the ThroughPacket™ mechanism). The valid range is the range of possible UDP ports: 6,000 to 64,000. The default value is 0 (ThroughPacket™ is disabled). This parameter cannot be changed on-the-fly and requires a gateway reset.
RTP Multiplexing Remote UDP Port **[L1L1ComplexRxUDPPort]**	Determines the remote UDP port the multiplexed RTP packets are sent to, and the local UDP port used for incoming multiplexed RTP packets (applies to the ThroughPacket™ mechanism). The valid range is the range of possible UDP ports: 6,000 to 64,000. The default value is 0 (ThroughPacket™ is disabled). This parameter cannot be changed on-the-fly and requires a gateway reset. **Note:** All gateways that participate in the same ThroughPacket™ session must use the same 'L1L1ComplexRxUDPPort'.
Comfort Noise Generation Negotiation **[ComfortNoiseNegotiation]**	Enables negotiation and usage of Comfort Noise (CN). Valid options include: **[0]** = Disable (default) **[1]** = Enable Comfort Noise negotiation The use of CN is indicated by including a payload type for CN on the media description line of the SDP. The gateway can use CN with a codec whose RTP timestamp clock rate is 8,000 Hz (G.711/G.726). The static payload type 13 is used. The use of CN is negotiated between sides; therefore, if the remote side doesn't support CN, it is not used. **Note:** Silence Suppression must be enabled to generate CN.

5.6.2.4 Configuring the Hook-Flash Settings

➢ **To configure the Hook-Flash Settings parameters, take these 4 steps:**

1. Open the 'Hook-Flash Settings' screen (**Advanced Configuration** menu > **Media Settings** > **Hook-Flash Settings** option); the 'Hook-Flash Settings' screen is displayed.

Figure 5-43: Hook-Flash Settings Screen



2. Configure the Hook-Flash Settings according to Table 5-45.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-45: Media Settings, Hook-Flash Settings Parameters

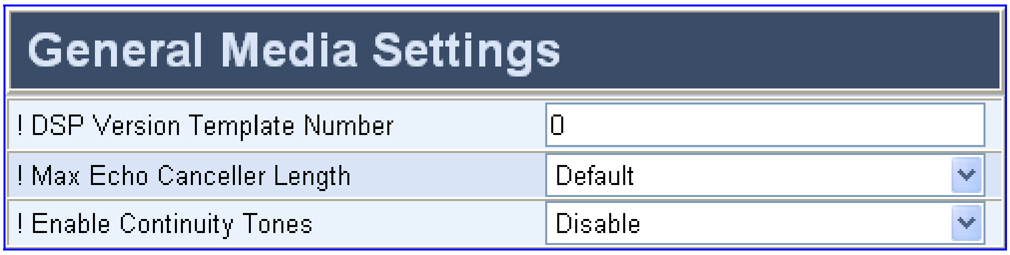
Parameter	Description
Min. Flash-Hook Detection Period [msec] **[MinFlashHookTime]**	Sets the minimal time (in msec) for detection of a flash-hook event (for FXS only). The valid range is 25 to 300. The default value is 300 msec. Detection is guaranteed for flash hook periods of at least 60 msec (when setting the minimal time to 25). Flash-hook signals that last a shorter period of time are ignored. **Note:** It's recommended to reduce the detection time by 50 msec from the desired value (e.g. if you set the value to 200 msec, then enter 150 msec (i.e. 200 minus 50).
Max. Flash-Hook Detection Period [msec] **[FlashHookPeriod]**	Defines the flash-hook period (in msec) for both analog and IP sides. For the analog side it defines: - The maximal hook-flash detection period (for FXS gateways). A longer signal is considered offhook / onhook event. - The hook-flash generation period (for FXO gateways). For the IP side it defines the flash-hook period that is reported to IP. The valid range is 25 to 1500. The default value is 700 msec. **Note:** For FXO gateways, a constant of 90 msec must be added to the required hook-flash period. For example, to generate a 450 msec hook-flash, set 'FlashHookPeriod' to 540.

5.6.2.5 Configuring the General Media Settings

➢ **To configure the General Media Settings parameters, take these 4 steps:**

1. Open the 'General Media Settings' screen (**Advanced Configuration** menu > **Media Settings** > **General Media Settings** option); the 'General Media Settings' screen is displayed.

Figure 5-44: General Media Settings Screen



2. Configure the General Media Settings according to Table 5-46.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-46: Media Settings, General Media Settings Parameters

Parameter	Description
DSP Version Template Number	N/A.
Max Echo Canceller Length **[MaxEchoCancellerLength]**	Maximum Echo Canceller Length in msec: **[0]** = based on internal gateway settings (default) **[4]** = 32 msec **[11]** = 64 msec **Note 1:** The gateway must be reset after the value of 'MaxEchoCancellerLength' is changed. **Note 2:** It is unnecessary to configure the parameter EchoCancellerLength as it automatically acquires its value from the parameter MaxEchoCancellerLength.
Enable Continuity Tones	N/A.

5.6.2.6 Media Settings *ini* File Parameters

Table 5-47 describes the Media Settings parameters that can only be configured via the *ini* file.

Table 5-47: Media Settings, *ini* File Parameters (continues on pages 162 to 163)

ini File Parameter Name	Valid Range and Description
RTPSIDCoeffNum	Determines the number of spectral coefficients added to an SID packet being sent according to RFC 3389. Valid only if 'EnableStandardSIDPayloadType' is set to 1. The valid values are 0 (default), 4, 6, 8 and 10.
ECHybridLoss	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. **[0]** = 6 dB (default) **[1]** = 9 dB **[2]** = 0 dB **[3]** = 3 dB
FaxModemRelayVolume	-18 to -3, corresponding to -18 dBm to -3 dBm in 1 dB steps. (Default = -12 dBm) fax gain control.
MGCPDTMFDetectionPoint	**[0]** = DTMF event is reported on the end of a detected DTMF digit. **[1]** = DTMF event is reported on the start of a detected DTMF digit (default).
DTMFDigitLength	Time in msec for generating DTMF tones to the PSTN side (if TxDTMFOption = 1, 2 or 3). The default value is 100 msec. The valid range is 0 to 32767.
DTMFInterDigitInterval	Time in msec between generated DTMF digits to PSTN side (if TxDTMFOption = 1, 2 or 3). The default value is 100 msec. The valid range is 0 to 32767.
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).
ModemBypassPayloadType	Modem Bypass dynamic payload type. The valid range is 0 to 127. The default value is 103.
BellModemTransportType	Determines the Bell modem transport method. **[0]** = Transparent (default). **[2]** = Bypass. **[3]** = Transparent with events.
FaxModemBypassBasicRtpPacketInterval	Determines the basic frame size that is used during fax / modem bypass sessions. **[0]** = set internally (default) **[1]** = 5 msec **[2]** = 10 msec **[3]** = 20 msec **Note:** When set for 5 msec (1), the maximum number of simultaneous channels supported is 120.
FaxModemBypassDJBufMinDelay	Determines the Jitter Buffer delay during fax and modem bypass session. The valid range is 0 to 150 msec. The default is 40.
EchoCancellerAggressiveNLP	Enables or disables the Aggressive NLP in the first 0.5 second of the call. **[0]** = Disabled (default) **[1]** = Enabled

Table 5-47: Media Settings, *ini* File Parameters (continues on pages 162 to 163)

ini File Parameter Name	Valid Range and Description
NSEMode	Cisco compatible fax and modem bypass mode 0 = NSE disabled (default) 1 = NSE enabled **Note 1:** This feature can be used only if VxxModemTransportType=2 (Bypass) **Note 2:** If NSE mode is enabled the SDP contains the following line: 'a=rtpmap:100 X-NSE/8000' **Note 3:** To use this feature: ▪ The Cisco gateway must include the following definition: 'modem passthrough nse payload-type 100 codec g711alaw'. ▪ Set the Modem transport type to Bypass mode ('VxxModemTransportType = 2') for all modems. ▪ Configure the gateway parameter NSEPayloadType= 100 In NSE bypass mode the gateway starts using G.711 A-Law (default) or G.711μ-Law, according to the parameter 'FaxModemBypassCoderType'. The payload type used with these G.711 coders is a standard one (8 for G.711 A-Law and 0 for G.711 μ-Law). The parameters defining payload type for the 'old' AudioCodes' Bypass mode. 'FaxBypassPayloadType' and 'ModemBypassPayloadType' are not used with NSE Bypass. The bypass packet interval is selected according to the parameter 'FaxModemBypassBasicRtpPacketInterval'.
NSEPayloadType	NSE payload type for Cisco Bypass compatible mode. The valid range is 96-127. The default value is 105. **Note:** Cisco gateways usually use NSE payload type of 100.
IsCiscoSCEMode	**[0]** = There isn't a Cisco gateway at the remote side (default). **[1]** = There is a Cisco gateway at the remote side. When there is a Cisco gateway at the remote side, the local gateway must set the value of the 'annexb' parameter of the fmtp attribute in the SDP to 'no'. This logic should be used if 'EnableSilenceCompression = 2' (enable without adaptation). In this case, Silence Suppression should be used on the channel but not declared in the SDP.
BellcoreCallerIDTypeOneSubStandard	Selects the Bellcore Caller ID sub-standard. **[0]** = Between rings (default). **[1]** = Not ring related.
ETSICallerIDTypeOneSubStandard	Selects the ETSI FSK Caller ID Type 1 sub-standard (FXS only). **[0]** = ETSI between rings (default). **[1]** = ETSI before ring DT_AS. **[2]** = ETSI before ring RP_AS. **[3]** = ETSI before ring LR_DT_AS. **[4]** = ETSI not ring related DT_AS. **[5]** = ETSI not ring related RP_AS. **[6]** = ETSI not ring related LR_DT_AS.
ETSIVMWITypeOneStandard	Selects the ETSI Visual Message Waiting Indication (VMWI) Type 1 sub-standard. **[0]** = ETSI VMWI between rings (default) **[1]** = ETSI VMWI before ring DT_AS **[2]** = ETSI VMWI before ring RP_AS **[3]** = ETSI VMWI before ring LR_DT_AS **[4]** = ETSI VMWI not ring related DT_AS **[5]** = ETSI VMWI not ring related RP_AS **[6]** = ETSI VMWI not ring related LR_DT_AS
BellcoreVMWITypeOneStandard	Selects the Bellcore VMWI sub-standard. **[0]** = Between rings (default). **[1]** = Not ring related.

Table 5-47: Media Settings, *ini* File Parameters (continues on pages 162 to 163)

ini File Parameter Name	Valid Range and Description
StunServerDomainName	Defines the domain name for the STUN server's address (used for retrieving all STUN servers with an SRV query). The STUN client can perform the required SRV query to resolve this domain name to an IP address and port, sort the server list, and use the servers according to the sorted list. **Note:** You can either use the STUNServerPrimaryIP or the STUNServerDomainName parameter, with priority to the former one.
NATBindingDefaultTimeout	Defines the NAT binding lifetime in seconds. STUN refreshes the binding information after this time expires. The valid range is 0 to 2592000. The default value is 30.
TxDTMFHangOverTime	Defines the Voice Silence time (in msec units) after detecting the end of DTMF or MF digits at the Tel / PSTN side when the DTMF Transport Type is either Relay or Mute. The Valid range is 0 to 2,000 msec. The default is 100 msec.
RxDTMFHangOverTime	Defines the Voice Silence time (in msec) after playing DTMF or MF digits to the Tel / PSTN side that arrive as Relay from the IP side. The valid range is 0 to 2,000 msec. The default is 1,000 msec.

5.6.3 Restoring and Backing up the Gateway Configuration

The Configuration File screen enables you to restore (load a new *ini* file to the gateway) or to back up (make a copy of the VoIP gateway *ini* file and store it in a directory on your computer) the current configuration the gateway is using.

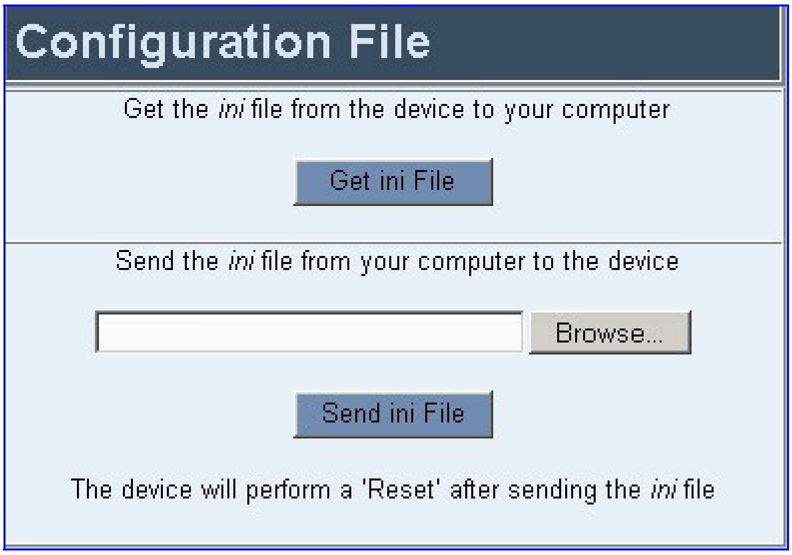
Back up your configuration if you want to protect your VoIP gateway programming. The backup *ini* file includes only those parameters that were modified and contain other than default values.

Restore your configuration if the VoIP gateway has been replaced or has lost its programming information, you can restore the VoIP gateway configuration from a previous backup or from a newly created *ini* file. To restore the VoIP gateway configuration from a previous backup you must have a backup of the VoIP gateway information stored on your computer.

➢ To restore or back up the *ini* file, take the following step:

- Open the 'Configuration File' screen (**Advanced Configuration** menu > **Configuration File**); the 'Configuration File' screen is displayed.

Figure 5-45: Configuration File Screen



➢ To back up the *ini* file, take these 4 steps:

1. Click the **Get ini File** button; the 'File Download' window opens.
2. Click the **Save** button; the 'Save As' window opens.
3. Navigate to the folder where you want to save the *ini* file.
4. Click the **Save** button; the VoIP gateway copies the *ini* file into the folder you selected.

➢ To restore the *ini* file, take these 4 steps:

1. Click the **Browse** button.
2. Navigate to the folder that contains the *ini* file you want to load.
3. Select the file, and then click the **Open** button; the name and path of the file appear in the field beside the Browse button.
4. Click the **Send *ini* File** button, and then click **OK** at the prompt; the gateway is automatically reset (from the *cmp* version stored on the flash memory).

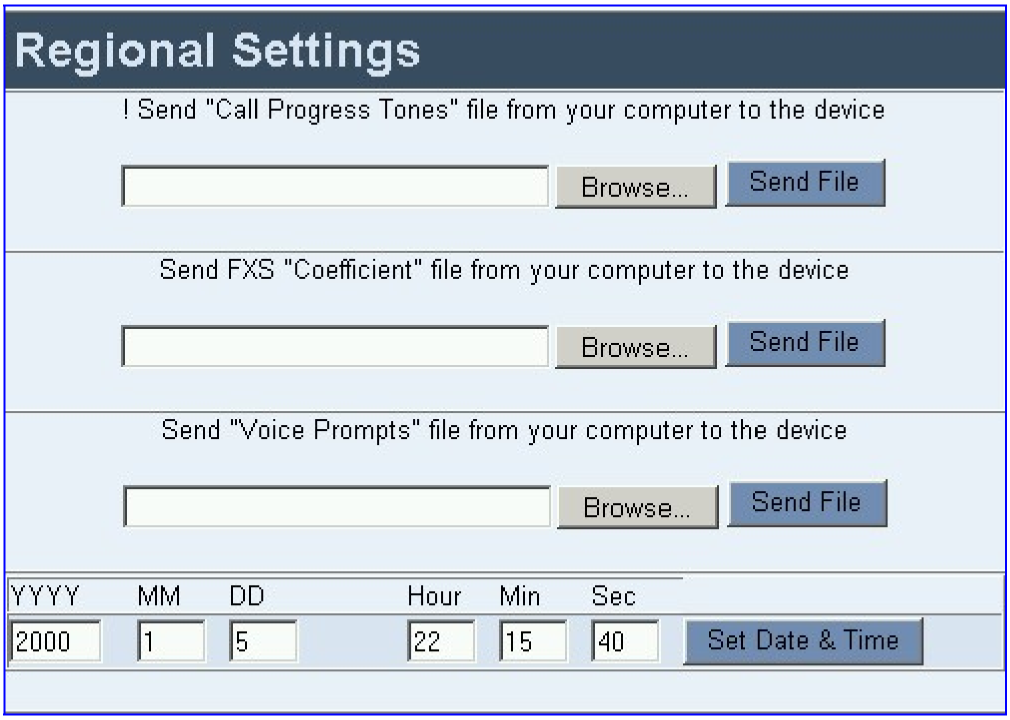
5.6.4 Regional Settings

The 'Regional Settings' screen enables you to set and view the gateway's internal date and time and to load to the gateway the following configuration files: Call Progress Tones, coefficient and Voice Prompts (currently not applicable to MediaPack gateways). For detailed information on the configuration files, refer to Chapter 6 on page 209.

➢ **To configure the date and time of the MediaPack, take these 3 steps:**

1. Open the 'Regional Settings' screen (**Advanced Configuration** menu > **Regional Settings**); the 'Regional Settings' screen is displayed.

Figure 5-46: Regional Settings Screen



2. Enter the time and date where the gateway is installed.
3. Click the **Set Date & Time** button; the date and time are automatically updated.

Note: After performing a hardware reset, the date and time are returned to their defaults and should be updated.

➢ To load a configuration file to the gateway, take these 8 steps:

1. Open the ‘Regional Settings’ screen (**Advanced Configuration** menu > **Regional Settings**); the ‘Regional Settings’ screen is displayed (shown in Figure 5-46).
2. Click the **Browse** button adjacent to the file you want to load.
3. Navigate to the folder that contains the file you want to load.
4. Click the file and click the **Open** button; the name and path of the file appear in the field beside the **Browse** button.
5. Click the **Send File** button that is next to the field that contains the name of the file you want to load. An exclamation mark in the screen section indicates that the file’s loading doesn’t take effect on-the-fly (e.g., CPT file).
6. Repeat steps 2 to 5 for each file you want to load.

Notes:

- Saving a configuration file to flash memory may disrupt traffic on the MediaPack. To avoid this, disable all traffic on the device before saving to flash memory by performing a graceful lock (refer to Section 5.10.1 on page 204).
- A device reset is required to activate a loaded CPT file.

7. To save the loaded auxiliary files so they are available after a power fail, refer to Section 5.10.2 on page 205.
8. To reset the MediaPack, refer to Section 5.10.3 on page 206.

5.6.5 Security Settings

From the Security Settings you can:

- Configure the Web User Accounts (refer to Section 5.6.5.1 below).
- Configure the Web & Telnet Access List (refer to Section 5.6.5.2 on page 170).
- Configure the Firewall Settings (refer to Section 5.6.5.3 on page 171).
- Configure the Certificates (refer to Section 5.6.5.4 on page 172).
- Configure the General Security Settings (refer to Section 5.6.5.5 on page 173).
- Configure the IPSec Table (refer to Section 5.6.5.6 on page 175).
- Configure the IKE Table (refer to Section 5.6.5.7 on page 176).

5.6.5.1 Configuring the Web User Accounts

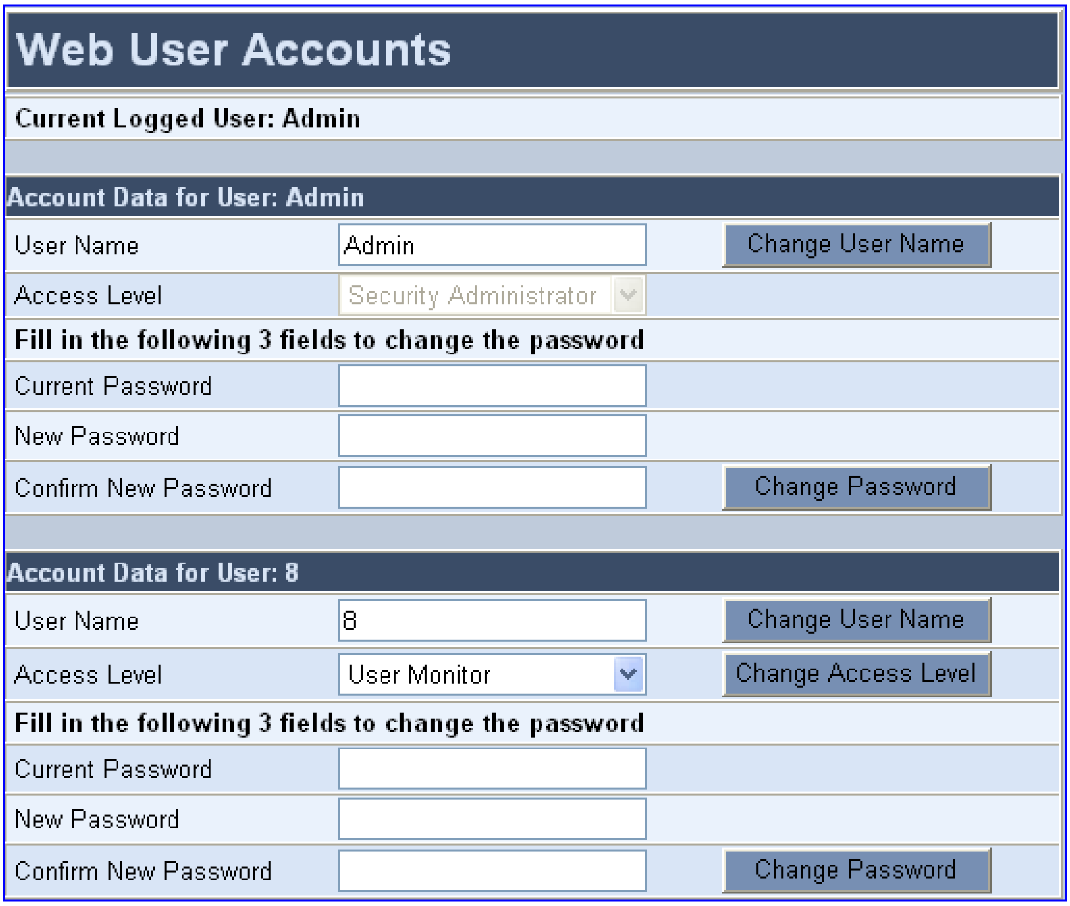
To prevent unauthorized access to the Embedded Web Server, two user accounts are available, a primary and secondary. Each account is composed of three attributes: username, password and access level. For detailed information on the user account mechanism, refer to Section 5.2.1 on page 49.

It is recommended that you change the default username and password of the account you use to access the Embedded Web Server.

➢ To change the Web User Accounts attributes, take these 4 steps:

1. Open the 'Web User Accounts' screen (**Advanced Configuration** menu > **Security Settings** > **Web User Accounts** option); the 'Web User Accounts' screen is displayed.

Figure 5-47: Web User Accounts Screen (for Users with 'Security Administrator' Privileges)



2. To change the access level of the secondary account (the access level of the primary account cannot be changed), in the 'Access Level' drop-down list, select the new access level and click the button **Change Access Level**; the new access level is applied immediately.
3. To change the username of an account, enter the new username in the field 'User Name' and click the button **Change User Name**; the new username is applied immediately and the 'Enter Network Password' screen appears (shown in Figure 5-1 on page 51). Enter the updated username in the 'Enter Network Password' screen. Note that the username can be a maximum of 19 case-sensitive characters.
4. To change the password of an account, enter the current password in the field 'Current Password', the new password in the fields 'New Password' and 'Confirm New Password' and click the button **Change Password**; the new password is applied immediately and the 'Enter Network Password' screen appears (shown in Figure 5-1 on page 51). Enter the updated password in the 'Enter Network Password' screen. Note that the password can be a maximum of 19 case-sensitive characters.

Note: A user with a 'Security Administrator' access level can change all attributes for all accounts. Users with an access level other than 'Security Administrator' can only change their own password and username.

5.6.5.2 Configuring the Web and Telnet Access List

Use this screen to define up to ten IP addresses that are permitted to access the gateway's Web and Telnet interfaces. Access from an undefined IP address is denied. This security feature is inactive (the gateway can be accessed from any IP address) when the table is empty.

➢ To manage the Web & Telnet access list, take these 4 steps:

1. Open the 'Web & Telnet Access List' screen (**Advanced Configuration** menu > **Security Settings** > **Web & Telnet Access List** option); the 'Web & Telnet Access List' screen is displayed.

Figure 5-48: Web & Telnet Access List Screen

2. To add a new authorized IP address, in the 'New Authorized IP Address' field, enter the required IP address (refer to Note 1 below) and click **Add New Address**; the IP address you entered is added as a new entry to the Web & Telnet Access List table.
3. To delete authorized IP addresses, check the Delete Row checkbox in the rows of the IP addresses you want to delete (refer to Note 2 below) and click the button **Delete Selected Addresses**; the IP addresses are removed from the table and can no longer access the Web & Telnet interfaces.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Notes:

- The first authorized IP address you add must be your own terminal's IP address. If it isn't, further access from your terminal is denied.
- Delete your terminal's IP address from the Web & Telnet Access List last. If it is deleted before the last, access from your terminal is denied from the point of its deletion on.

Table 5-48: Web & Telnet Access List *ini* File Parameter

Parameter Name in *ini* File	Parameter Format
WebAccessList_x	WebAccessList_0 = 10.13.2.66 WebAccessList_1 = 10.13.77.7 The default value is 0.0.0.0 (the gateway can be accessed from any IP address). **Note:** This parameter can appear up to ten times.

5.6.5.3 Configuring the Firewall Settings

The MediaPack accommodates an internal Firewall, allowing the security administrator to define network traffic filtering rules. For detailed information on the internal Firewall, refer to Section 12.5 on page 297.

➢ To create a new access rule, take these 6 steps:

1. Open the 'Firewall Settings' screen (**Advanced Configuration** menu > **Security Settings** > **Firewall Settings** option); the 'Firewall Settings' screen is displayed.

Figure 5-49: Firewall Settings Screen

Firewall Settings

Selected Rule	Is Rule Active?	Source IP	Mask	Local Port Range	Protocol	Packet Size	Byte rate	Burst Bytes	Action Upon Match	Match Count
0	No	mgmt.customer.com	255.255.255.255	0-80	tcp	0	0	0	ALLOW	0
1	No	192.0.0.0	255.0.0.0	0-65535	Any	0	40000	50000	BLOCK	0
2	Yes	10.31.4.0	255.255.255.0	4000 - 9000	Any	0	0	0	Block	0
3	Yes	10.4.0.0	255.255.0.0	4000-9000	Any	0	0	0	BLOCK	0

2. In the 'New Rule Index' field, enter the index of the access rule that you want to add.
3. Click the **Add an Empty Rule** button; a new rule appears; alternatively, click the **Copy Selected Rule as a New Rule** button; a new rule that is an exact copy of the currently selected rule appears.
4. Configure the rule's parameters according to Table 5-49.
5. Click one of the following buttons:
   - **Apply Rule Settings** to save the new rule (the rule isn't active).
   - **Activate Rule** to save the new rule and activate it.
   - **Delete Rule** to delete the rule.
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

➢ To edit a rule, take these 5 steps:

1. Select the radio button of the entry you want to edit.
2. Click the **Make Rule Editable** button; the rule's fields can now be modified.
3. Modify the fields according to your requirements.
4. Click the **Apply Rule Settings** button to save the changes.
5. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

➢ To activate a de-activated rule, take these 2 steps:

1. Select the radio button of the entry you want to activate.
2. Click the **Activate Rule** button; the rule is active.

➢ To de-activate an activate rule, take these 2 steps:

1. Select the radio button of the entry you want to activate.
2. Click the **DeActivate Rule** button; the rule is de-activated.

➢ To delete a rule, take these 3 steps:

1. Select the radio button of the entry you want to activate.
2. Click the **Delete Rule** button; the rule is deleted.
3. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-49: Internal Firewall Fields

Parameter	Description
Is Rule Active	A read-only field that indicates whether the rule is active or not. **Note:** After reset all rules are active.
Source IP **[AccessList_Source_IP]**	IP address (or DNS name) of source network, or a specific host.
Mask **[AccessList_Net_Mask]**	IP network mask. 255.255.255.255 for a single host or the appropriate value for the source IP addresses. The IP address of the sender of the incoming packet is bitwise ANDed with this mask and then compared to the field 'Source IP'.
Local Port Range **[AccessList_Start_Port]** **[AccessList_End_Port]**	The destination UDP/TCP ports (on this device) to which packets are sent. The valid range is 0 to 65535. **Note:** When the protocol type isn't TCP or UDP, the entire range must be provided.
Protocol **[AccessList_Protocol]**	The protocol type (e.g., UDP, TCP, ICMP, ESP or 'Any'), or the IANA protocol number (in the range of 0 (Any) to 255). **Note:** The protocol field also accepts the abbreviated strings 'SIP', 'MGCP', 'MEGACO' and 'HTTP'. Specifying these strings implies selection of the TCP or UDP protocols, and the appropriate port numbers as defined on the device.
Packet Size **[AccessList_Packet_Size]**	Maximum allowed packet size. The valid range is 0 to 65535. **Note:** When filtering fragmented IP packets, the Packet Size field relates to the overall (reassembled) packet size, not to the size of each fragment.
Byte Rate **[AccessList_Byte_Rate]**	Expected traffic rate (bytes per second).
Burst Bytes **[AccessList_Byte_Burst]**	Tolerance of traffic rate limit (number of bytes)
Action Upon Match **[AccessList_Allow_Type]**	Action upon match (allow or block)
Match Count **[AccessList_MatchCount]**	A read-only field that provides the number of packets accepted / rejected by a specific rule.

5.6.5.4 Configuring the Certificates

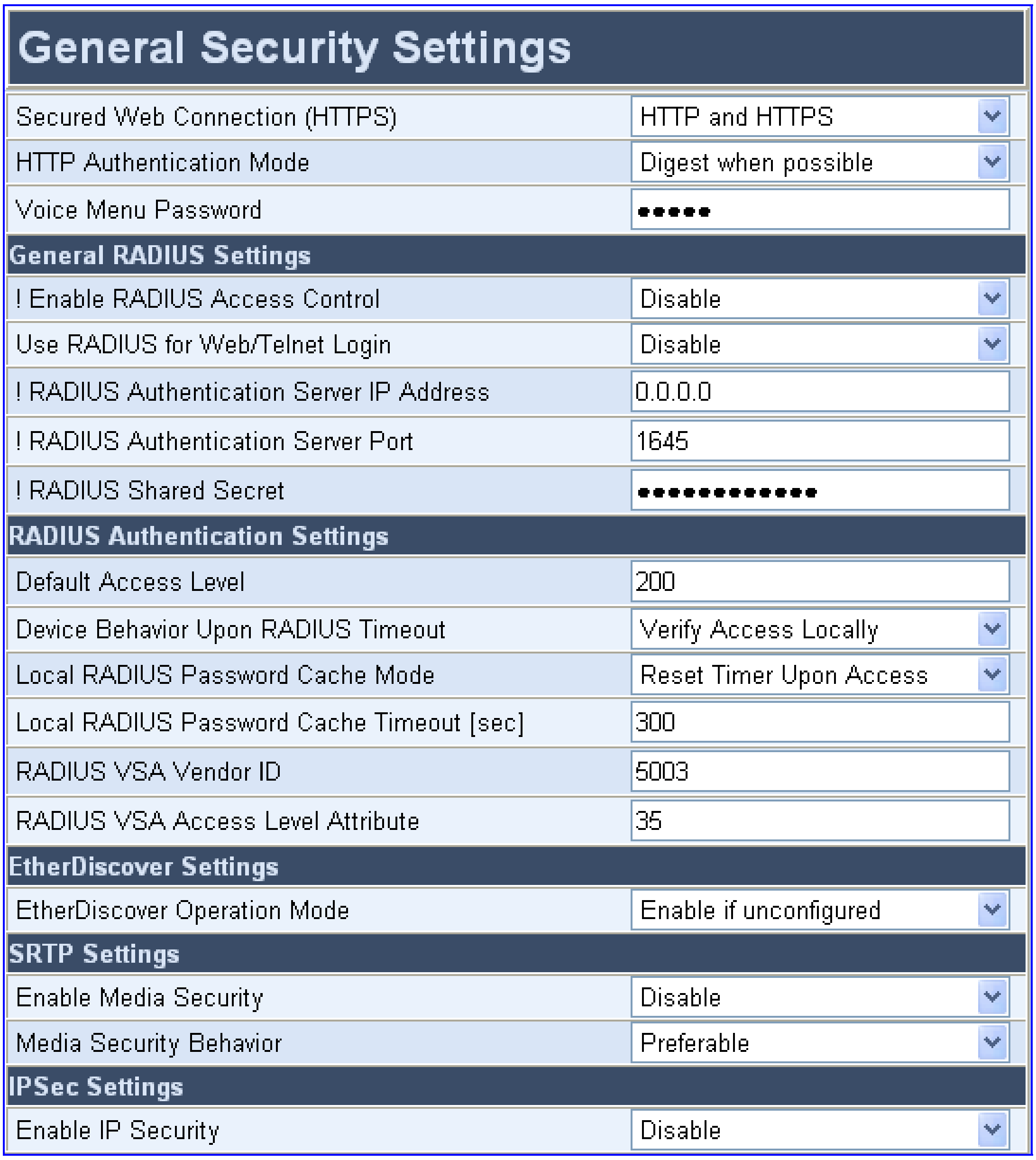
Use the Certificates screen to replace the server (refer to Section 12.2.4 on page 290) and client (refer to Section 12.2.5 on page 292) certificates and to update the private key (HTTPSPkeyFileName, described in Table 5-55 on page 182).

5.6.5.5 Configuring the General Security Settings

➢ **To configure the General Security Settings parameters, take these 4 steps:**

1. Open the 'General Security Settings' screen (**Advanced Configuration** menu > **Security Settings** > **General Security Settings** option); the 'General Security Settings' screen is displayed.

Figure 5-50: General Security Settings Screen



2. Configure the General Security Settings according to Table 5-50 below.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-50: Security Settings, General Security Settings Parameters
(continues on pages 174 to 175)

Parameter	Description
Secured Web Connection **[HTTPSOnly]**	Determines the protocol types used to access the Embedded Web Server. HTTP and HTTPS **[0]** (default). HTTPS only **[1]** (unencrypted HTTP packets are blocked).
HTTP Authentication Mode **[WebAuthMode]**	Determines the authentication mode for the Embedded Web Server. Basic **[0]** = Basic authentication (clear text) is used (default). Digest When Possible **[1]** = Digest authentication (MD5) is used. Basic if HTTPS, Digest if HTTP **[2]** = Digest authentication (MD5) is used for HTTP, and basic authentication is used for HTTPS. Note that when RADIUS login is enabled (WebRADIUSLogin = 1), basic authentication is forced.
Voice Menu Password **[VoiceMenuPassword]**	Password for the voice menu, used for configuration and status. To activate the menu, connect an analog telephone and dial *** (three stars) followed by the password. The default value is 12345. For detailed information on the voice menu, refer to Section 4.2.3 on page 43.
RADIUS General Settings	
Enable RADIUS Access Control **[EnableRADIUS]**	Enables / disables the RADIUS application. Disable **[0]** = RADIUS application is disabled (default). Enable **[1]** = RADIUS application is enabled. **Note:** In the current version RADIUS is used only for HTTP authentication (CDR over RADIUS isn't supported).
Use RADIUS for Web/Telnet Login **[WebRADIUSLogin]**	Uses RADIUS queries for Web and Telnet interface authentication. Disable **[0]** (default). Enable **[1]**. When enabled, logging to the gateway's Web and Telnet embedded servers is performed via a RADIUS server. The gateway contacts a predefined server and verifies the given username and password pair against a remote database, in a secure manner. **Note 1:** The parameter 'EnableRADIUS' must be set to 1. **Note 2:** RADIUS authentication requires HTTP basic authentication, meaning the username and password are transmitted in clear text over the network. Therefore, users are recommended to set the parameter 'HttpsOnly = 1' to force the use of HTTPS, since the transport is encrypted.
RADIUS Authentication Server IP Address **[RADIUSAuthServerIP]**	IP address of the RADIUS authentication server.
RADIUS Authentication Server Port **[RADIUSAuthPort]**	Port number of the RADIUS authentication server. The default value is 1645.
RADIUS Shared Secret **[SharedSecret]**	'Secret' used to authenticate the gateway to the RADIUS server. Should be a cryptographically strong password.
RADIUS Authentication Settings	
Default Access Level **[DefaultAccessLevel]**	Defines the default access level for the gateway when the RADIUS (authentication) response doesn't include an access level attribute. The valid range is 0 to 255. The default value is 200 (Security Administrator').
Device Behavior Upon RADIUS Timeout **[BehaviorUponRadiusTimeout]**	Defines the gateway's operation if a response isn't received from the RADIUS server after the 5 seconds timeout expires: Deny Access **[0]** = the gateway denies access to the Web and Telnet embedded servers. Verify Access Locally **[1]** = the gateway checks the local username and password (default).

Table 5-50: Security Settings, General Security Settings Parameters
(continues on pages 174 to 175)

Parameter	Description
Local RADIUS Password Cache Mode **[RadiusLocalCacheMode]**	Defines the gateway's mode of operation regarding the timer (configured by the parameter RadiusLocalCacheTimeout) that determines the validity of the username and password (verified by the RADIUS server). Absolute Expiry Timer **[0]** = when you access a Web screen, the timeout doesn't reset but rather continues decreasing. Reset Timer Upon Access **[1]** = upon each access to a Web screen, the timeout always resets (reverts to the initial value configured by RadiusLocalCacheTimeout).
Local RADIUS Password Cache Timeout **[RadiusLocalCacheTimeout]**	Defines the time (in seconds) the locally stored username and password (verified by the RADIUS server) are valid. When this time expires, the username and password becomes invalid and a must re-verified with the RADIUS server. The valid range is 1 to 0xFFFFFF. -1 = Never expires. 0 = Each request requires RADIUS authentication. The default value is 300 (5 minutes).
RADIUS VSA Vendor ID **[RadiusVSAVendorID]**	Defines the vendor ID the gateway accepts when parsing a RADIUS response packet. The valid range is 0 to 0xFFFFFFFF. The default value is 5003.
RADIUS VSA Access Level Attribute **[RadiusVSAAccessAttribute]**	Defines the code that indicates the access level attribute in the Vendor Specific Attributes (VSA) section of the received RADIUS packet. The valid range is 0 to 255. The default value is 35.
EtherDiscover Settings	
EtherDiscover Operation Mode	N/A.
SRTP Settings	
Enable Media Security **[EnableMediaSecurity]**	Enables or disables the Secure Real-Time Transport Protocol (SRTP). Disable (TGCP) **[0]** = SRTP is disabled (default). Enable (SRTP) **[1]** = SRTP is enabled. **Note:** Use of SRTP reduces the number of available channels. MP-124 18 available channels MP-118 6 available channels MP-114 3 available channels MP-112 no reduction.
Media Security Behavior **[MediaSecurityBehaviour]**	Determines the gateway's mode of operation when SRTP is used (EnableMediaSecurity = 1). Prefer **[0]** = The gateway initiates encrypted calls. If negotiation of the cipher suite fails, an unencrypted call is established. Incoming calls that don't include encryption information are accepted. Must **[1]** = The gateway initiates encrypted calls. If negotiation of the cipher suite fails, the call is terminated. Incoming calls that don't include encryption information are rejected (default).
IPSec Settings	
Enable IP Security **[EnableIPSec]**	Enables / disables the Secure Internet Protocol (IPSec) on the gateway. Disable **[0]** = IPSec is disabled (default). Enable **[1]** = IPSec is enabled.

5.6.5.6 Configuring the IPSec Table

Use the IPSec Table screen to configure the IPSec parameters. For detailed information on IPSec and IKE, refer to Section 12.1 on page 279.

5.6.5.7 Configuring the IKE Table

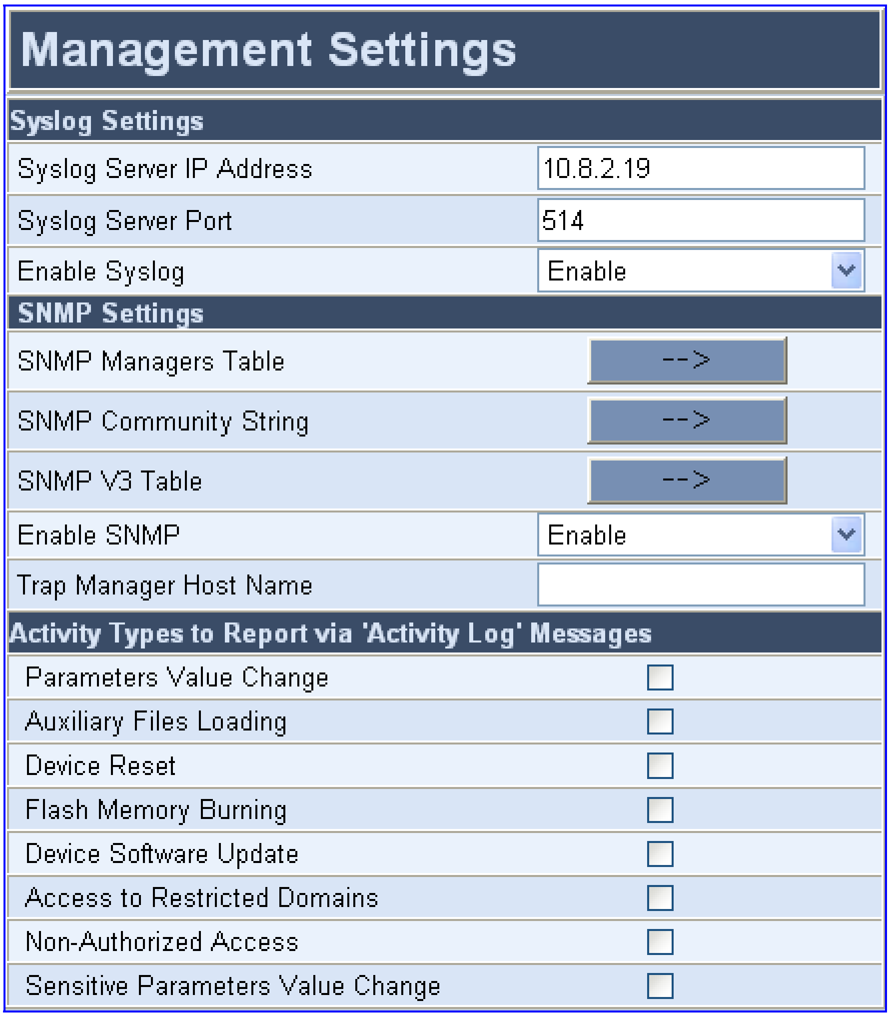
Use the IKE Table screen to configure the IKE parameters. For detailed information on IPSec and IKE, refer to Section 12.1 on page 279.

5.6.6 Configuring the Management Settings

➢ **To configure the Management Settings parameters, take these 4 steps:**

1. Open the 'Management Settings' screen (**Advanced Configuration** menu > **Management Settings**); the 'Management Settings' screen is displayed.

Figure 5-51: Management Settings Screen



2. Configure the Management Settings according to Table 5-51.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Table 5-51: Management Settings Parameters (continues on pages 177 to 178)

Parameter	Description
Syslog Settings	
Syslog Server IP address **[SyslogServerIP]**	IP address (in dotted format notation) of the computer you are using to run the Syslog Server. The Syslog Server is an application designed to collect the logs and error messages generated by the VoIP gateway. For information on the Syslog, refer to Section 13.2 on page 301.
Syslog Server Port **[SyslogServerPort]**	Defines the UDP port of the Syslog server. The default value (i.e., port) is 514.
Enable Syslog **[EnableSyslog]**	Enable **[1]** = Send the logs and error message generated by the gateway to the Syslog Server. If you select Enable, you must enter an IP address in the Syslog Server IP address field. Disable **[0]** = Logs and errors are not sent to the Syslog Server (default). **Note 1:** Syslog messages may increase the network traffic. **Note 2:** Logs are also sent to the RS-232 serial port (for information on establishing a serial communications link with the MediaPack, refer to Section 10.2 on page 262). **Note 3:** To configure the Syslog logging levels use the parameter 'Debug Level'.
SNMP Settings	
For detailed information on the SNMP parameters that can only be configured via the *ini* file, refer to Table 5-57 on page 186. For detailed information on developing an SNMP-based program to manage your devices, refer to Chapter 14 on page 305.	
SNMP Managers Table	Refer to Section 5.6.6.1 on page 178.
SNMP Community Strings	Refer to Section 5.6.6.2 on page 180.
SNMP V3 Table	Refer to Section 5.6.6.3 on page 181.
Enable SNMP **[DisableSNMP]**	Enable **[0]** = SNMP is enabled (default). Disable **[1]** = SNMP is disabled and no traps are sent.
Trap Manager Host Name **[SNMPTrapManagerHostName]**	Defines a FQDN of a remote host that is used as an SNMP Manager. The resolved IP address replaces the last entry in the trap manager table (defined by the parameter 'SNMPManagerTableIP_x') and the last trap manager entry of snmpTargetAddrTable in the snmpTargetMIB. For example: 'mngr.corp.mycompany.com'. The valid range is a 99-character string
Activity Types to Report via 'Activity Log' Messages The Activity Log mechanism enables the MediaPack to send log messages (to a Syslog server) that report certain types of Web actions according to a pre-defined filter. The following filters are available:	
Parameters Value Change **[ActivityListToLog = PVC]**	Changes made on-the-fly to parameters.
Auxiliary Files Loading **[ActivityListToLog = AFL]**	Loading of auxiliary files (e.g., via Certificate screen).
Device Reset **[ActivityListToLog = DR]**	Device reset via the 'Maintenance Actions' screen.
Flash Memory Burning **[ActivityListToLog = FB]**	Burning of files / parameters to flash (e.g., 'Maintenance Actions' screen).
Device Software Update **[ActivityListToLog = SWU]**	*cmp* loading via the Software Upgrade Wizard.

Table 5-51: Management Settings Parameters (continues on pages 177 to 178)

Parameter	Description
Access to Restricted Domains **[ActivityListToLog = ARD]**	Access to Restricted Domains. The following screens are restricted: (1) *ini* parameters (AdminPage) (2) General Security Settings (3) Configuration File (4) IPSec/IKE tables (5) Software Upgrade Key (6) Internal Firewall (7) Web Access List. (8) Web User Accounts
Non Authorized Access **[ActivityListToLog = NAA]**	Attempt to access the Embedded Web Server with a false / empty username or password.
Sensitive Parameters Value Change **[ActivityListToLog = SPC]**	Changes made to sensitive parameters: (1) IP Address (2) Subnet Mask (3) Default Gateway IP Address (4) ActivityListToLog
ini file example: ActivityListToLog = 'pvc', 'afl', 'dr', 'fb', 'swu', 'ard', 'naa', 'spc'	

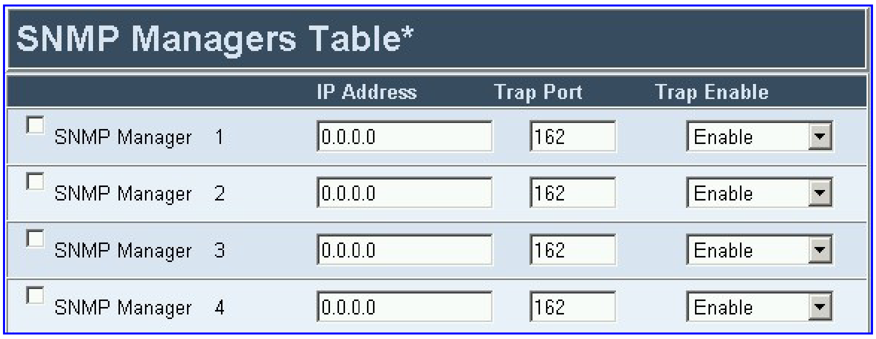
5.6.6.1 Configuring the SNMP Managers Table

The SNMP Managers table allows you to configure the attributes of up to five SNMP managers.

➢ **To configure the SNMP Managers Table, take these 5 steps:**

1. Access the 'Management Settings' screen (**Advanced Configuration** menu > **Management Settings**); the 'Management Settings' screen is displayed (Figure 5-51).
2. Open the SNMP Managers Table screen by clicking the arrow sign (-->) to the right of the SNMP Managers Table label; the SNMP Managers Table screen is displayed (Figure 5-52).
3. Configure the SNMP Managers parameters according to Table 5-52.
4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Figure 5-52: SNMP Managers Table Screen



Note: If you clear a checkbox and click **Submit**, all settings in the same row revert to their defaults.

Table 5-52: SNMP Managers Table Parameters

Parameter	Description
Checkbox **[SNMPManagerIsUsed_x]**	Up to five parameters, each determines the **validity** of the parameters (IP address and port number) of the corresponding SNMP Manager used to receive SNMP traps. Checkbox cleared **[0]** = Disabled (default) Checkbox selected **[1]** = Enabled
IP Address **[SNMPManagerTableIP_x]**	Up to five IP addresses of remote hosts that are used as SNMP Managers. The device sends SNMP traps to these IP addresses. Enter the IP address in dotted format notation, for example 108.10.1.255.
Trap Port **[SNMPManagerTrapPort_x]**	Up to five parameters used to define the Port numbers of the remote SNMP Managers. The device sends SNMP traps to these ports. **Note:** The first entry (out of the five) replaces the obsolete parameter SNMPTrapPort. The default SNMP trap port is 162 The valid SNMP trap port range is 100 to 4000.
Trap Enable **[SNMPManagerTrapSending Enable_x]**	Up to five parameters, each determines the activation/deactivation of sending traps to the corresponding SNMP Manager. Disable **[0]** = Sending is disabled Enable **[1]** = Sending is enabled (default)

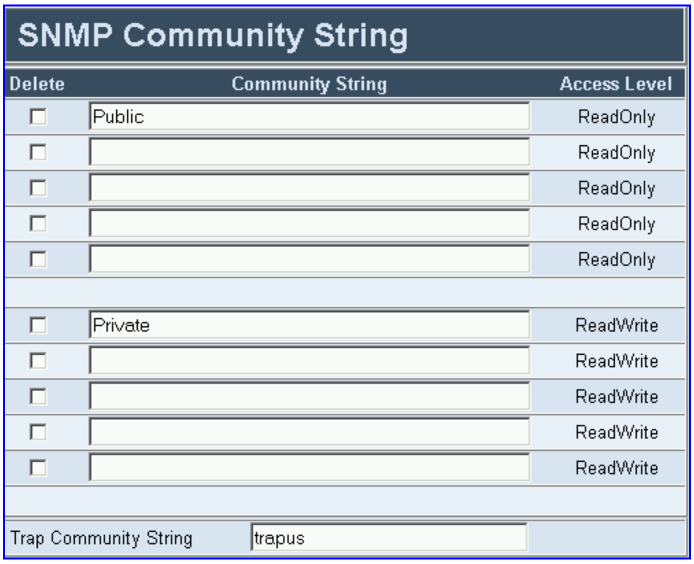
5.6.6.2 Configuring the SNMP Community Strings

Use the SNMP Community Strings table to configure up to five read-only and up to five read / write SNMP community strings, and to configure the community string that is used for sending traps. For detailed information on SNMP community strings, refer to Section 14.8.1 on page 313.

➢ **To configure the SNMP Community Strings, take these 5 steps:**

1. Access the 'Management Settings' screen (**Advanced Configuration** menu > **Management Settings**); the 'Management Settings' screen is displayed (Figure 5-51).
2. Open the SNMP Community Strings screen by clicking the arrow sign (-->) to the right of the SNMP Community Strings label; the SNMP Community Strings screen is displayed (Figure 5-53).
3. Configure the SNMP Community Strings parameters according to Table 5-53 below.
4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Figure 5-53: SNMP Community Strings Screen



Note: To delete a community string, check the **Delete** checkbox to the left of the community string you want to delete and click the button **Submit**.

Table 5-53: SNMP Community Strings Parameters

Parameter	Description
Read Only **[SNMPReadOnlyCommunityString_x]**	Up to five read-only community strings (up to 19 characters each). The default string is 'public'.
Read / Write **[SNMPReadWriteCommunityString_x]**	Up to five read / write community strings (up to 19 characters each). The default string is 'private'.
Trap Community String **[SNMPTrapCommunityString]**	Community string used in traps (up to 19 characters). The default string is 'trapuser'.

5.6.6.3 Configuring SNMP V3

Use the SNMP V3 Table to configure authentication and privacy for up to 10 SNMP V3 users.

For detailed information on SNMP community strings, refer to Section 14.8.1 on page 313.

➢ To configure the SNMP V3 Users, take these 5 steps:

1. Access the 'Management Settings' screen (**Advanced Configuration** menu > **Management Settings**); the 'Management Settings' screen is displayed (Figure 5-51).
2. Open the SNMP V3 Setting screen by clicking the arrow sign (-->) to the right of the SNMP V3 Table label; the SNMP V3 Setting screen is displayed (Figure 5-54).
3. Configure the SNMP V3 Setting parameters according to Table 5-54 below.
4. Click the **Apply Row Settings** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

Figure 5-54: SNMP V3 Setting Screen

Note: To delete an SNMP V3 user, select the Index radio button corresponding to the SNMP V3 user row entry to which you want to delete, and then click the **Delete Row** button.

Table 5-54: SNMP V3 Setting Parameters

Parameter	Description
Index **[Row number]**	The table index
Username **[SNMPUsers_Username]**	Name of the SNMP v3 user. This name must be unique.
AuthProtocol **[SNMPUsers_AuthProtocol]**	Authentication protocol to be used for the SNMP v3 user. 0 = none (default) 1 = MD5 2 = SHA-1
PrivProtocol **[SNMPUsers_PrivProtocol]**	Privacy protocol to be used for the SNMP v3 user. 0 = none (default) 1 = DES 2 = 3DES 3 = AES128 4 = AES192 5 = AES256
AuthKey **[SNMPUsers_AuthKey]**	Authentication key. Keys can be entered in the form of a text password or long hex string. Keys are always persisted as long hex strings and keys are localized.
PrivKey **[SNMPUsers_PrivKey]**	Privacy key. Keys can be entered in the form of a text password or long hex string. Keys are always persisted as long hex strings and keys are localized.
Group **[SNMPUsers_Group]**	The group with which the SNMP v3 user is associated. 0 = read-only group (default) 1 = read-write group 2 = trap group **Note:** all groups can be used to send traps.

5.6.6.4 Advanced Configuration *ini* File Parameters

Table 5-55 describes the board parameters that can only be configured via the *ini* file.

Table 5-55: Board, *ini* File Parameters (continues on pages 182 to 184)

ini File Parameter Name	Valid Range and Description
LifeLineType	The Lifeline is activated on: 0 = Power down (default) 1 = Power down or when link is down (physical disconnect) 2 = Power down or when link is down or on network failure (logical link disconnect) **Note:** To enable Lifeline switching on network failure, LAN watch dog must be activated (EnableLANWatchDog=1).
EnableDiagnostics	Checks the correct functionality of the different hardware components on the gateway. On completion of the check, if the test fails, the gateway sends information on the test results of each hardware component to the Syslog server. 0 = Rapid self-test mode (default). 1 = Detailed self-test mode (full test of DSPs, PCM, Switch, LAN, PHY and Flash). 2 = A quicker version of the Detailed self-test mode (full test of DSPs, PCM, Switch, LAN, PHY, but partial test of Flash). For detailed information, refer to Section 13.1 on page 301.
EnableParametersMonitoring	Obsolete parameter, use the parameter 'ActivityListToLog' instead.
WatchDogStatus	0 = Disable gateway's watch dog. 1 = Enable gateway's watch dog (default).
DisableRS232	0 = RS-232 serial port is enabled (default). 1 = RS-232 serial port is disabled. The RS-232 serial port can be used to change the networking parameters (Section 4.2.4 on page 44) and to view error / notification messages. For information on establishing a serial communications link with the MediaPack, refer to Section 10.2 on page 262).
DisableWebTask	0 = Enable Web management (default) 1 = Disable Web management
ResetWebPassword	Resets the username and password of the primary and secondary accounts to their defaults. 0 = Password and username retain their values (default). 1 = Password and username are reset (for the default username and password, refer to Table 4-1 on page 41). **Note:** The username and password cannot be reset from the Web (i.e., via AdminPage or by loading an *ini* file).
DisableWebConfig	0 = Enable changing parameters from Web (default) 1 = Operate Web server in 'read only' mode
HTTPport	HTTP port used for Web management (default = 80)
HeartBeatDestIP	Destination IP address (in dotted format notation) to which the gateway sends proprietary UDP 'ping' packets. The default IP address is 0.0.0.0.
HeartBeatDestPort	Destination UDP port to which the heartbeat packets are sent. The range is 0 to 64000. The default is 0.
HeartBeatIntervalmsec	Delay (in msec) between consecutive heartbeat packets. 10 = 100000. -1 = disabled (default).

Table 5-55: Board, *ini* File Parameters (continues on pages 182 to 184)

<table>
<tr><th>ini File Parameter Name</th><th colspan="2">Valid Range and Description</th></tr>
<tr><td>RADIUSRetransmission</td><td colspan="2">Determines the number of RADIUS retransmission retries for the same request.
The valid range is 1 to 10.
The default value is 3.</td></tr>
<tr><td>RADIUSTo</td><td colspan="2">Determines the time interval (measured in seconds) the gateway waits for a response before a RADIUS retransmission is issued.
The valid range is 1 to 30.
The default value is 10.</td></tr>
<tr><td colspan="3">HTTPS Parameters</td></tr>
<tr><td>HTTPSPort</td><td colspan="2">Determine the local Secured HTTPS port of the device.
The valid range is 1 to 65535 (other restrictions may apply within this range).
The default port is 443.</td></tr>
<tr><td>HTTPSRequireClientCertificate</td><td colspan="2">Requires client certificates for HTTPS connection. The client certificate must be preloaded to the gateway, and its matching private key must be installed on the managing PC. Time and date must be correctly set on the gateway, for the client certificate to be verified.
0 = Client certificates are not required (default).
1 = Client certificates are required.</td></tr>
<tr><td>HTTPSRootFileName</td><td colspan="2">Defines the name of the HTTPS trusted root certificate file to be loaded via TFTP. The file must be in base64-encoded PEM (Privacy Enhanced Mail) format.
The valid range is a 47-character string.
Note: This parameter is only relevant when the gateway is loaded via BootP/TFTP. For information on loading this file via the Embedded Web Server, refer to the Security section in the User’s Manual.</td></tr>
<tr><td>HTTPSPkeyFileName
[Security Settings > Certificates]</td><td colspan="2">Defines the name of a private key file (in unencrypted PEM format) to be loaded from the TFTP server.</td></tr>
<tr><td>HTTPSCertFileName</td><td colspan="2">Defines the name of the HTTPS server certificate file to be loaded via TFTP. The file must be in base64-encoded PEM format.
The valid range is a 47-character string.
Note: This parameter is only relevant when the gateway is loaded via BootP/TFTP. For information on loading this file via the Embedded Web Server, refer to the Security section in the User’s Manual.</td></tr>
<tr><td colspan="3">BootP and TFTP Parameters</td></tr>
<tr><td colspan="3">The BootP parameters are special ‘Hidden’ parameters. Once defined and saved in the flash memory, they are used even if they don't appear in the ini file.</td></tr>
<tr><td rowspan="2">BootPRetries</td><td colspan="2">This parameter is used to:
Note: This parameter only takes effect from the next reset of the gateway.</td></tr>
<tr><td>Set the number of BootP requests the gateway sends during start-up. The gateway stops sending BootP requests when either BootP reply is received or number of retries is reached.

1 = 1 BootP retry, 1 second.
2 = 2 BootP retries, 3 second.
3 = 3 BootP retries, 6 second (default).
4 = 10 BootP retries, 30 second.
5 = 20 BootP retries, 60 second.
6 = 40 BootP retries, 120 second.
7 = 100 BootP retries, 300 second.
15 = BootP retries indefinitely.</td><td>Set the number of DHCP packets the gateway sends.
After all packets were sent, if there's still no reply, the gateway loads from flash.

1 = 4 DHCP packets
2 = 5 DHCP packets
3 = 6 DHCP packets (default)
4 = 7 DHCP packets
5 = 8 DHCP packets
6 = 9 DHCP packets
7 = 10 DHCP packets
15 = 18 DHCP packets</td></tr>
</table>

Table 5-55: Board, *ini* File Parameters (continues on pages 182 to 184)

ini File Parameter Name	Valid Range and Description
BootPSelectiveEnable	Enables the Selective BootP mechanism. 1 = Enabled. 0 = Disabled (default). The Selective BootP mechanism (available from Boot version 1.92) enables the gateway's integral BootP client to filter unsolicited BootP/DHCP replies (accepts only BootP replies that contain the text 'AUDC' in the vendor specific information field). This option is useful in environments where enterprise BootP/DHCP servers provide undesired responses to the gateway's BootP requests. **Note:** When working with DHCP (DHCPEnable = 1) the selective BootP feature must be disabled.
BootPDelay	The interval between the device's startup and the first BootP/DHCP request that is issued by the device. 1 = 1 second (default). 2 = 3 second. 3 = 6 second. 4 = 30 second. 5 = 60 second. **Note:** This parameter only takes effect from the next reset of the device.
ExtBootPReqEnable	0 = Disable (default). 1 = Enable extended information to be sent in BootP request. If enabled, the device uses the vendor specific information field in the BootP request to provide device-related initial startup information such as board type, current IP address, software version, etc. For a full list of the vendor specific Information fields, refer to Section 7.3.2 on page 214. The BootP/TFTP configuration utility displays this information in the 'Client Info' column (refer to Figure C-1). **Note:** This option is not available on DHCP servers.

5.6.6.5 Automatic Updates Parameters

For detailed information on the automatic update mechanism, refer to Section 10.3 on page 263.

Table 5-56: Automatic Updates Parameters (continues on pages 184 to 185)

ini File Parameter Name	Description
CmpFileURL	Specifies the name of the *cmp* file and the location of the server (IP address or FQDN) from which the gateway loads a new *cmp* file and updates itself. The *cmp* file can be loaded using: HTTP, HTTPS, FTP, FTPS or NFS. For example: http://192.168.0.1/filename **Note 1:** When this parameter is set in the *ini* file, the gateway always loads the *cmp* file after it is reset. **Note 2:** The *cmp* file is validated before it is burned to flash. The checksum of the *cmp* file is also compared to the previously-burnt checksum to avoid unnecessary resets. **Note 3:** The maximum length of the URL address is 99 characters.

Table 5-56: Automatic Updates Parameters (continues on pages 184 to 185)

ini File Parameter Name	Description
IniFileURL	Specifies the name of the *ini* file and the location of the server (IP address or FQDN) from which the gateway loads the *ini* file. The *ini* file can be loaded using: HTTP, HTTPS, FTP, FTPS or NFS. For example: http://192.168.0.1/filename http://192.8.77.13/config<MAC> https://<username>:<password>@<IP address>/<file name> **Note 1:** When using HTTP or HTTPS, the date and time of the *ini* file are validated. Only more recently-dated *ini* files are loaded. **Note 2:** The optional string '<MAC>' is replaced with the gateway's MAC address. Therefore, the gateway requests an *ini* file name that contains its MAC address. This option enables loading different configurations for specific gateways. **Note 3:** The maximum length of the URL address is 99 characters.
PrtFileURL	Specifies the name of the Prerecorded Tones file and the location of the server (IP address or FQDN) from which it is loaded. http://server_name/file, https://server_name/file. **Note:** The maximum length of the URL address is 99 characters.
CptFileURL	Specifies the name of the CPT file and the location of the server (IP address or FQDN) from which it is loaded. http://server_name/file, https://server_name/file. **Note:** The maximum length of the URL address is 99 characters.
FXSCoeffFileURL	Specifies the name of the FXS coefficients file and the location of the server (IP address or FQDN) from which it is loaded. http://server_name/file, https://server_name/file. **Note:** The maximum length of the URL address is 99 characters.
UserInfoFileURL	Specifies the name of the User Information file and the location of the server (IP address or FQDN) from which it is loaded. http://server_name/file, https://server_name/file. **Note:** The maximum length of the URL address is 99 characters.
AutoUpdateCmpFile	Enables / disables the Automatic Update mechanism for the *cmp* file. 0 = The Automatic Update mechanism doesn't apply to the *cmp* file (default). 1 = The Automatic Update mechanism includes the *cmp* file.
AutoUpdateFrequency	Determines the number of minutes the gateway waits between automatic updates. The default value is 0 (the update at fixed intervals mechanism is disabled).
AutoUpdatePredefinedTime	Schedules an automatic update to a predefined time of the day. The range is 'HH:MM' (24-hour format). For example: 20:18. **Note:** The actual update time is randomized by five minutes to reduce the load on the Web servers.
ResetNow	Invokes an immediate restart of the gateway. This option can be used to activate offline (not on-the-fly) parameters that are loaded via IniFileUrl. 0 = The immediate restart mechanism is disabled (default). 1 = The gateway immediately restarts after an *ini* file with this parameter set to 1 is loaded.

5.6.6.6 SNMP *ini* File Parameters

Table 5-57 describes the SNMP parameters that can only be configured via the *ini* file.

Table 5-57: SNMP *ini* File Parameters

ini File Parameter Name	Description
SNMPPort	The device's local UDP port used for SNMP Get/Set commands. The range is 100 to 3999. The default port is 161.
SNMPTrustedMGR_x	Up to five IP addresses of remote trusted SNMP managers from which the SNMP agent accepts and processes get and set requests. **Note 1:** If no values are assigned to these parameters any manager can access the device. **Note 2:** Trusted managers can work with *all* community strings.
SNMPManagerTrapUser_x	This parameter can be set to the name of any configured SNMPV3 user to associate with this trap destination. This determines the trap format, authentication level, and encryption level. By default, the trap is associated with the SNMP trap community string.
AlarmHistoryTableMaxSize	Determines the maximum number of rows in the Alarm History table. The parameter can be controlled by the Config Global Entry Limit MIB (located in the Notification Log MIB). The valid range is 50 to 100. The default value is 100.

5.7 Status & Diagnostics

Use this menu to view and monitor the gateway's channels, Syslog messages, hardware / software product information, and to assess the gateway's statistics and IP connectivity information.

5.7.1 Gateway Statistics

Use the screens under Gateway Statistics to monitor real-time activity such as IP Connectivity information, call details and call statistics, including the number of call attempts, failed calls, fax calls, etc.

> **Notes:** The 'Gateway Statistics' screens doesn't refresh automatically. To view updated information, re-access the screen you require.

5.7.1.1 IP Connectivity

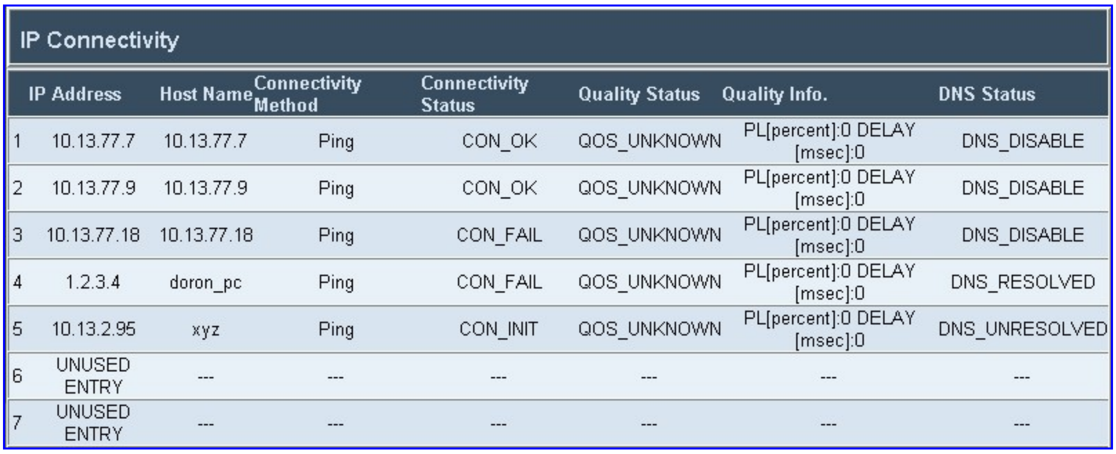
The IP Connectivity screen provides you with an online read-only network diagnostic connectivity information on all destination IP addresses configured in the Tel to IP Routing table.

> **Notes:**
>
> - This information is available only if the parameter 'AltRoutingTel2IPEnable' (described in Table 5-16) is set to 1 (Enable) or 2 (Status Only).
> - The information in columns 'Quality Status' and 'Quality Info.' (per IP address) is reset if two minutes elapse without a call to that destination.

➢ To view the IP connectivity information, take these 2 steps:

1. Set 'AltRoutingTel2IPEnable' to 1 or 2.
2. Open the 'IP Connectivity' screen (**Status & Diagnostics** menu > **Gateway Statistics** submenu > **IP Connectivity**); the 'IP Connectivity' screen is displayed (Figure 5-55).

Figure 5-55: IP Connectivity Screen



IP Connectivity

	IP Address	Host Name	Connectivity Method	Connectivity Status	Quality Status	Quality Info.	DNS Status
1	10.13.77.7	10.13.77.7	Ping	CON_OK	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_DISABLE
2	10.13.77.9	10.13.77.9	Ping	CON_OK	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_DISABLE
3	10.13.77.18	10.13.77.18	Ping	CON_FAIL	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_DISABLE
4	1.2.3.4	doron_pc	Ping	CON_FAIL	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_RESOLVED
5	10.13.2.95	xyz	Ping	CON_INIT	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_UNRESOLVED
6	UNUSED ENTRY	---	---	---	---	---	---
7	UNUSED ENTRY	---	---	---	---	---	---

Table 5-58: IP Connectivity Parameters

Column Name	Description
IP Address	IP address defined in the destination IP address field in the Tel to IP Routing table. or IP address that is resolved from the host name defined in the destination IP address field in the Tel to IP Routing table.
Host Name	Host name (or IP address) defined in the destination IP address field in the Tel to IP Routing table.
Connectivity Method	The method according to which the destination IP address is queried periodically (currently only by ping).
Connectivity Status	Displays the status of the IP address' connectivity according to the method in the 'Connectivity Method' field. Can be one of the following: ▪ OK = Remote side responds to periodic connectivity queries. ▪ Lost = Remote side didn't respond for a short period. ▪ Fail = Remote side doesn't respond. ▪ Init = Connectivity queries not started (e.g., IP address not resolved). ▪ Disable = Connectivity option is disabled ('AltRoutingTel2IPMode' equals 0 or 2).
Quality Status	Determines the QoS (according to packet loss and delay) of the IP address. Can be one of the following: ▪ Unknown = Recent quality information isn't available. ▪ OK ▪ Poor **Note 1:** This field is applicable only if the parameter 'AltRoutingTel2IPMode' is set to 2 or 3. **Note 2:** This field is reset if no QoS information is received for 2 minutes.
Quality Info.	Displays QoS information: delay and packet loss, calculated according to previous calls. **Note 1:** This field is applicable only if the parameter 'AltRoutingTel2IPMode' is set to 2 or 3. **Note 2:** This field is reset if no QoS information is received for 2 minutes.
DNS Status	Can be one of the following: ▪ DNS Disable ▪ DNS Resolved ▪ DNS Unresolved

5.7.1.2 Call Counters

The Call Counters screens provide you with statistic information on incoming (IP→Tel) and outgoing (Tel→IP) calls. The statistic information is updated according to the release reason that is received after a call is terminated (during the same time as the end-of-call CDR message is sent). The release reason can be viewed in the Termination Reason field in the CDR message. For detailed information on each counter, refer to Table 5-59 on page 189.

You can reset this information by clicking the **Reset Counters** button.

➢ To view the IP→Tel and Tel→IP Call Counters information, take this step:

- Open the Call Counters screen you want to view (**Status & Diagnostics** menu > **Gateway Statistics** submenu); the relevant Call Counters screen is displayed. Figure 5-56 shows the 'Tel→IP Call Counters' screen.

Figure 5-56: Tel→IP Call Counters Screen

Tel to IP Calls Count	
Number of Attempted Calls	10
Number of Established Calls	5
Percentage of Successful Calls	50.000000
Number of Failed Calls due to a Busy Line	1
Number of Failed Calls due to No Answer	3
Number of Failed Calls due to No Route	0
Number of Failed Calls due to No Matched Capabilities	0
Number of Failed Calls due to Other Failures	1
Average Call Duration [sec]	15
Attempted Fax Calls Counter	0
Successful Fax Calls Counter	0

Table 5-59: Call Counters Description (continues on pages 189 to 190)

Counter	Description
Number of Attempted Calls	This counter indicates the number of attempted calls. It is composed of established and failed calls. The number of established calls is represented by the 'Number of Established Calls' counter. The number of failed calls is represented by the five failed-call counters. Only one of the established / failed call counters is incremented every time.
Number of Established Calls	This counter indicates the number of established calls. It is incremented as a result of one of the following release reasons, if the duration of the call is bigger then zero: GWAPP_REASON_NOT_RELEVANT (0) GWAPP_NORMAL_CALL_CLEAR (16) GWAPP_NORMAL_UNSPECIFIED (31) And the internal reasons: RELEASE_BECAUSE_UNKNOWN_REASON RELEASE_BECAUSE_REMOTE_CANCEL_CALL RELEASE_BECAUSE_MANUAL_DISC RELEASE_BECAUSE_SILENCE_DISC RELEASE_BECAUSE_DISCONNECT_CODE **Note:** When the duration of the call is zero, the release reason GWAPP_NORMAL_CALL_CLEAR increments the 'Number of Failed Calls due to No Answer' counter. The rest of the release reasons increment the 'Number of Failed Calls due to Other Failures' counter.

Table 5-59: Call Counters Description (continues on pages 189 to 190)

Counter	Description
Number of Failed Calls due to a Busy Line	This counter indicates the number of calls that failed as a result of a busy line. It is incremented as a result of the following release reason: GWAPP_USER_BUSY (17)
Number of Failed Calls due to No Answer	This counter indicates the number of calls that weren't answered. It is incremented as a result of one of the following release reasons: GWAPP_NO_USER_RESPONDING (18) GWAPP_NO_ANSWER_FROM_USER_ALERTED (19) And (when the call duration is zero) as a result of the following: GWAPP_NORMAL_CALL_CLEAR (16)
Number of Failed Calls due to No Route	This counter indicates the number of calls whose destinations weren't found. It is incremented as a result of one of the following release reasons: GWAPP_UNASSIGNED_NUMBER (1) GWAPP_NO_ROUTE_TO_DESTINATION (3)
Number of Failed Calls due to No Matched Capabilities	This counter indicates the number of calls that failed due to mismatched gateway capabilities. It is incremented as a result of an internal identification of capability mismatch. This mismatch is reflected to CDR via the value of the parameter 'DefaultReleaseReason' (default is GWAPP_NO_ROUTE_TO_DESTINATION (3)), or by the GWAPP_SERVICE_NOT_IMPLEMENTED_UNSPECIFIED(79) reason.
Number of Failed Calls due to Other Failures	This counter is incremented as a result of calls that fail due to reasons not covered by the other counters.
Percentage of Successful Calls	The percentage of established calls from attempted calls.
Average Call Duration [sec]	The average call duration of established calls.
Attempted Fax Calls Counter	This counter indicates the number of attempted fax calls.
Successful Fax Calls Counter	This counter indicates the number of successful fax calls.

5.7.1.3 Call Routing Status

The Call Routing Status screen provides you with information on the current routing method used by the gateway. This information includes the IP address and FQDN (if used) of the Proxy server the gateway currently operates with.

Figure 5-57: Call Routing Status Screen

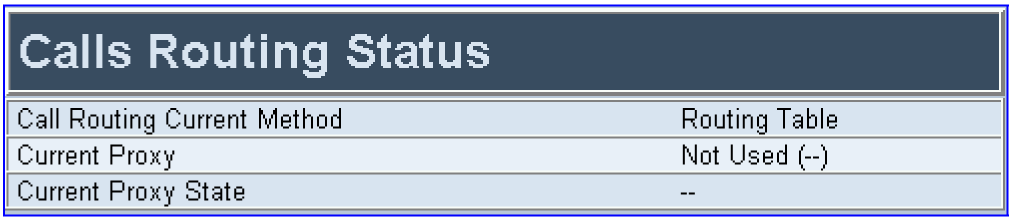


Table 5-60: Call Routing Status Parameters

Parameter	Description
Current Call-Routing Method	Proxy = Proxy server is used to route calls.
	Routing Table preferred to Proxy = The Tel to IP Routing table takes precedence over a Proxy for routing calls (PreferRouteTable = 1).
	Routing Table = The Tel to IP Routing table is used to route calls.
Current Proxy	Not Used = Proxy server isn't defined.
	IP address and FQDN (if exists) of the Proxy server the gateway currently operates with.
Current Proxy State	N/A = Proxy server isn't defined.
	OK = Communication with the Proxy server is in order.
	Fail = No response from any of the defined Proxies.

5.7.2 Activating the Internal Syslog Viewer

The Message Log screen displays Syslog debug messages sent by the gateway.

Note that it is not recommended to keep a ‘Message Log’ session open for a prolonged period (refer to the Note below). For prolong debugging use an external Syslog server, refer to Section 13.2 on page 301.

Refer to the Debug Level parameter ‘GwDebugLevel’ (described in Table 5-8) to determine the Syslog logging level.

➢ To activate the Message Log, take these 4 steps:

1. In the 'General Parameters' screen under **Advanced Parameters** submenu (accessed from the **Protocol Management** menu), set the parameter ‘Debug Level’ to 5. This parameter determines the Syslog logging level, in the range 0 to 5, where 5 is the highest level.
2. Open the ‘Message Log’ screen (**Status & Diagnostics** menu > **Message Log**); the ‘Message Log’ screen is displayed and the Log is activated.

Figure 5-58: Message Log Screen

```
Log is Activated

21d:23h:48m:23s (      lgr_flow)(380       )  #0:OFF_HOOK_EV

21d:23h:48m:23s (      lgr_flow)(381       )  |         #0:OFF_HOOK_EV

21d:23h:48m:23s (   lgr_psbrdif)(382       )  DigitMap for channel : 0 Not Activated

21d:23h:48m:23s (   lgr_psbrdif)(383       )  #0:PSOSBoardInterface::PlayTone - Called Tone=DIAL_TONE

21d:23h:48m:23s Short line was detected - going to Active Low [Code:36010] [CID:0]
```

3. Select the messages, copy them and paste them into a text editor such as Notepad. Send this *txt* file to our Technical Support for diagnosis and troubleshooting.
4. To clear the screen of messages, click on the submenu **Message Log**; the screen is cleared and new messages begin appearing.

Tip:	Do not keep the ‘Message Log’ screen minimized for a prolonged period as a prolonged session may cause the MediaPack to overload. As long as the screen is open (even if minimized), a session is in progress and messages are sent. Closing the screen (and accessing another) stops the messages and terminates the session.

5.7.3 Device Information

The Device Information screen displays specific hardware and software product information. This information can help you to expedite any troubleshooting process. Capture the screen and email it to 'our' Technical Support personnel to ensure quick diagnosis and effective corrective action. From this screen you can also view and remove any loaded files used by the MediaPack (stored in the RAM).

➢ To access the System Information screen:

- Open the 'Device Information' screen (**Status & Diagnostics** menu > **Device Information**); the 'Device Information' screen is displayed.

Figure 5-59: Device Information Screen

Device Information		
General		
MAC Address:	00908f084f99	
Serial Number:	544665	
Board Type:	56	
Device Up Time:	0d:17h:59m:4s:75th	
Device Administrative State:	Unlocked	
Device Operational State:	Enabled	
Flash Size [bytes]:	8388608	
RAM Size [bytes]:	33554432	
CPU Speed [MHz]:	40	
Versions		
Version ID:	5.00A.010	
DSP Type:	0	
DSP Software Version:	20912	
DSP Software Name:	204IM	
Flash Version:	195	
Loaded Files		
Call Progress Tones File Name:	usa_tones_12.dat	Delete
FXS Coefficient File Name:	MP11x-02-1-FXS_16KHZ.dat	Delete
Loaded Coder Table :	Default CODERTABLE	

➢ To delete any of the loaded files, take these 4 steps:

1. Click the **Delete** button to the right of the files you want to delete. Deleting a file takes effect only after the MediaPack is reset.
2. Click the **Maintenance** button on the main menu bar; the 'Maintenance Actions' screen is displayed.
3. In the 'Burn to FLASH' field, select 'Yes'.
4. Click the **Reset** button. The gateway is reset and the files you chose to delete are removed.

5.7.4 Viewing the Ethernet Port Information

The Ethernet Port Information screen provides read-only information on the Ethernet connection used by the MediaPack. The Ethernet Port Information parameters are displayed in Table 5-61. For detailed information on the Ethernet interface configuration, refer to Section 9.1 on page 247.

➢ **To view the Ethernet Port Information parameters, take this step:**

- Open the 'Ethernet Port Information' screen (**Advanced Configuration** menu > **Network Settings** > **Ethernet Port Information** option); the 'Ethernet Port Information' screen is displayed.

Figure 5-60: Ethernet Port Information Screen

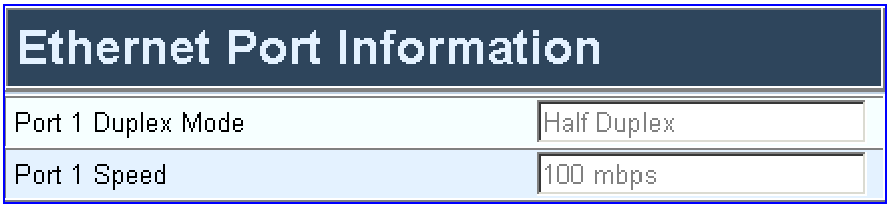


Table 5-61: Ethernet Port Information Parameters

Parameter	Description
Port 1 Duplex Mode	Shows the Duplex mode the Ethernet port is using (Half Duplex or Full Duplex).
Port 1 Speed	Shows the speed, in Mbps, that the Ethernet port is using (10 Mbps or 100 Mbps).

5.8 Monitoring the MediaPack Channels (Home Page)

The 'Channel Status' screen provides real-time monitoring on the current channels status. In addition, this screen allows you to assign a brief description or name to each port as well as releasing a channel.

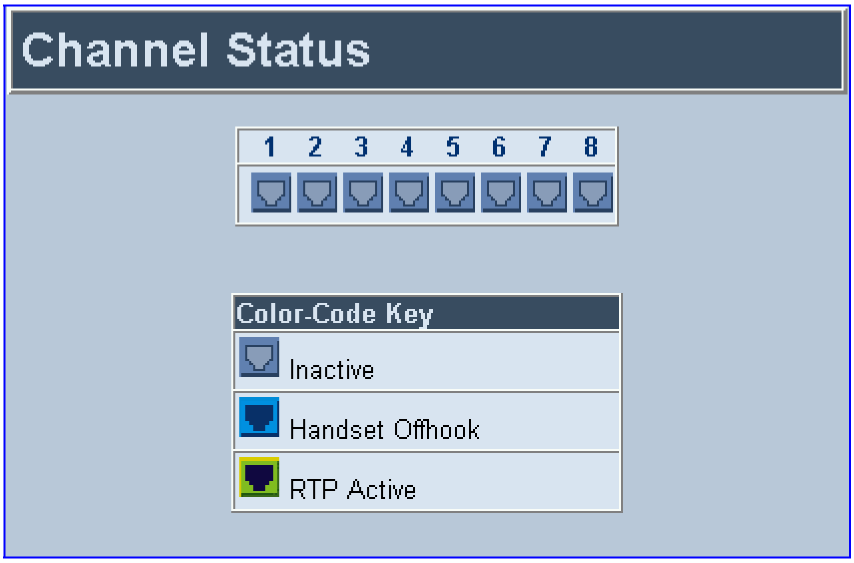
The Web interface provides the Home icon  for quick-and-easy access to this screen.

5.8.1 Viewing the Status of Channels

➢ To monitor the status of the MediaPack channels, take this step:

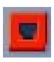
- Open the ‘Channel Status’ screen by clicking the Home icon; the ‘Channel Status’ screen is displayed (different screen for FXS and FXO).

Figure 5-61: MediaPack/FXS Channel Status Screen



The color of each channel shows the call status of that channel. Refer to Table 5-62 below for information on the different statuses a call can have.

Table 5-62: Channel Status Color Indicators

Indicator	Label	Description
	Inactive	Indicates this channel is currently onhook
	RTP Active	Indicates an active RTP stream.
	Not Connected (FXO only)	Indicates that no analog line is connected to this port.
	Handset Offhook	Indicates this channel is offhook but there is no active RTP session.

➢ To monitor the details of a specific channel, take these 3 steps:

1. Click the numbered port icon of the specific channel whose detailed status you need to check/monitor; a shortcut menu appears.
2. From the shortcut menu, choose **Port Settings**; the channel-specific Channel Status screen appears, shown in Figure 5-62.
3. Click the submenu links to check/view a specific channel's parameter settings.

Figure 5-62: Channel Status Details Screen

SIP Channel Status		
Static Information		
Endpoint Status :	ACTIVE	
Assigned Phone Number :	100	
Hunt Group :	default (0)	
MWI Information :	--	
Associated Calls Information		
Call ID :	265821508dMlu@10.8.58.1	--
Call Originator :	TEL	--
Source Tel Number :	100	--
Destination Tel Number :	200	--
Redirect Calling Number :		--
Remote Signaling IP :	10.8.58.2	--
Remote RTP (IP:Port) :	10.8.58.2: 4000	--
Call Establishment Duration :	2	--
Call Duration :	17	--
Call State :	SESSION	--
Fax State :	n/a	--
Coder + PTime :	g7231:30	--
Call Type :	Voice	--
Call Establishment Method :	Normal	--
DTMF Selected Method for Tx/Rx :	DTMF_NOT_SUPPORTED	--

5.8.2 Adding a Port Description

The 'Channel Status' screen allows you to add a brief text description or name for each port / channel.

➢ To add a port description, take these 3 steps:

1. Open the 'Channel Status' screen by clicking the Home icon.
2. Click a port / channel icon, and then from the shortcut menu, choose **Update Port Info**; a text box appears.
7. In the text box, type a brief description of this port, and then click **Apply Port Info**.

5.8.3 Resetting a Channel

The 'Channel Status' screen allows you to inactivate (*reset*) a channel. This is sometimes useful in cases, for example, when the gateway (FXO) is connected to a PBX and the communication between the two can't be disconnected (e.g., using reverse polarity).

➢ To reset channel, take these 2 steps:

1. Open the 'Channel Status' screen by clicking the **Home** icon.
2. Click a channel icon, and then from the shortcut menu, choose **Reset Channel**; the channel is changed to inactive.

5.9 Software Update

The ‘Software Update’ menu enables users to upgrade the MediaPack software by loading a new *cmp* file along with the *ini* and a suite of auxiliary files, or to update the existing auxiliary files.

The ‘Software Update’ menu comprises two submenus:

- Software Upgrade Wizard (refer to Section 5.9.1 below).
- Load Auxiliary Files (refer to Section 5.9.2 on page 202).

Note: When upgrading the MediaPack software you *must* load the new *cmp* file with all other related configuration files.

5.9.1 Software Upgrade Wizard

The Software Upgrade Wizard guides users through the process of software upgrade: selecting files and loading them to the gateway. The wizard also enables users to upgrade software while maintaining the existing configuration. Using the wizard obligates users to load and burn a *cmp* file. Users can choose to also use the Wizard to load the *ini* and auxiliary files (e.g., Call Progress Tones*)* but this option cannot be pursued without loading the *cmp* file. For the *ini* and each auxiliary file type, users can choose to reload an existing file, load a new file or not load a file at all.

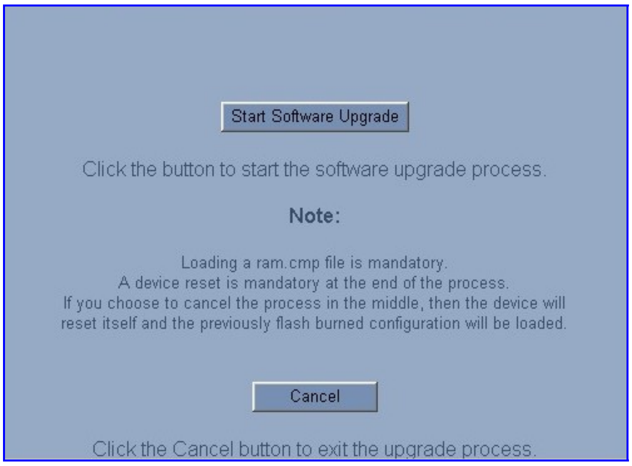
Warning 1: The Software Upgrade Wizard requires the MediaPack to be reset at the end of the process, disrupting any of its traffic. To avoid disruption, disable all traffic on the MediaPack before initiating the Wizard.

Warning 2: Verify, prior to clicking the **Start Software Upgrade** button that no traffic is running on the device. After clicking this button a device reset is mandatory. Even if you choose to cancel the process in the middle, the device resets itself and the previous configuration burned to flash is reloaded.

➢ To use the Software Upgrade Wizard, take these 11 steps:

1. Stop all traffic on the MediaPack (refer to the note above).
2. Open the 'Software Upgrade Wizard' (**Software Update** menu > **Software Upgrade Wizard**); the 'Start Software Upgrade' screen appears.

Figure 5-63: Start Software Upgrade Screen

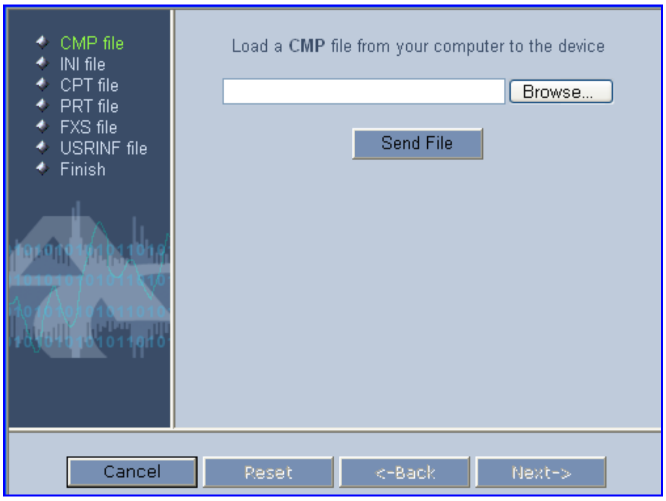


Note:	At this point, the process can be canceled with no consequence to the MediaPack (click the **Cancel** button). If you continue the process (by clicking the **Start Software Upgrade** button, the process must be followed through and completed with a MediaPack reset at the end. If you click the **Cancel** button in any of the subsequent screens, the MediaPack is automatically reset with the configuration that was previously burned in flash memory.

3. Click the **Start Software Upgrade** button; the 'Load a cmp file' screen appears (Figure 5-64).

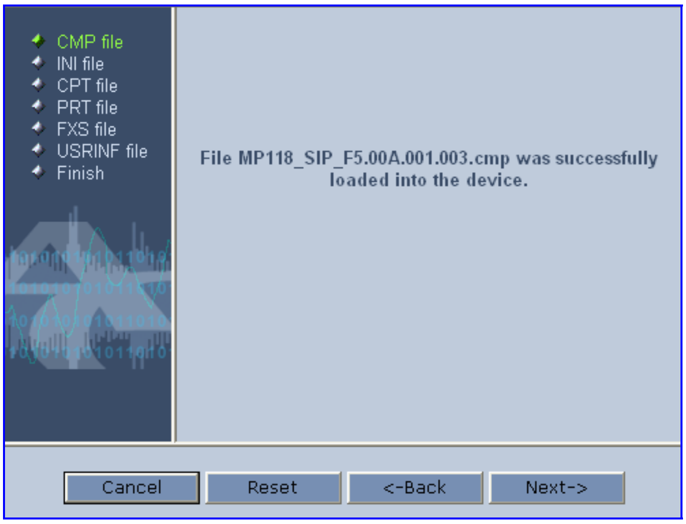
Note:	When in the Wizard process, the rest of the Web application is unavailable and the background Web screen is disabled. After the process is completed, access to the full Web application is restored.

Figure 5-64: Load a *cmp* File Screen



4. Click the **Browse** button, navigate to the *cmp* file and click the button **Send File**; the *cmp* file is loaded to the MediaPack and you're notified as to a successful loading (refer to Figure 5-65).

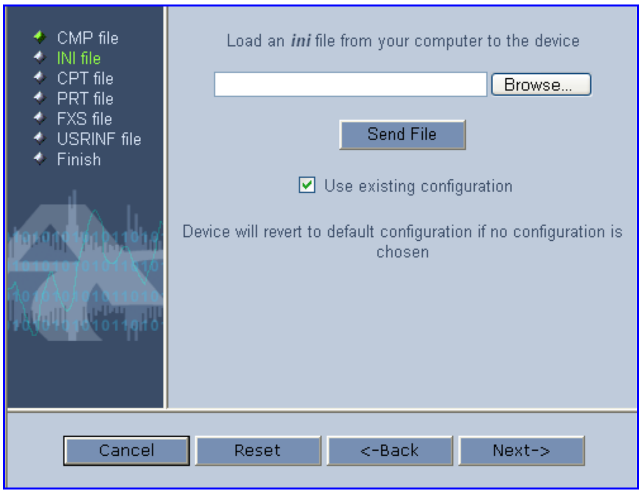
Figure 5-65: *cmp* File Successfully Loaded into the MediaPack Notification



5. Note that the four action buttons (**Cancel**, **Reset**, **Back**, and **Next**) are now activated (following *cmp* file loading).
6. You can now choose to either:
   - Click **Reset**; the MediaPack resets, utilizing the new *cmp* you loaded and utilizing the current configuration files.
   - Click **Cancel**; the MediaPack resets utilizing the *cmp, ini* and all other configuration files that were previously stored in flash memory. Note that these are NOT the files you loaded in the previous Wizard steps.
   - Click **Back**; the 'Load a *cmp* File' screen is reverted to; refer to Figure 5-64.
   - Click **Next**; the 'Load an *ini* File' screen opens; refer to Figure 5-66. Loading a new *ini* file or any other auxiliary file listed in the Wizard is optional.

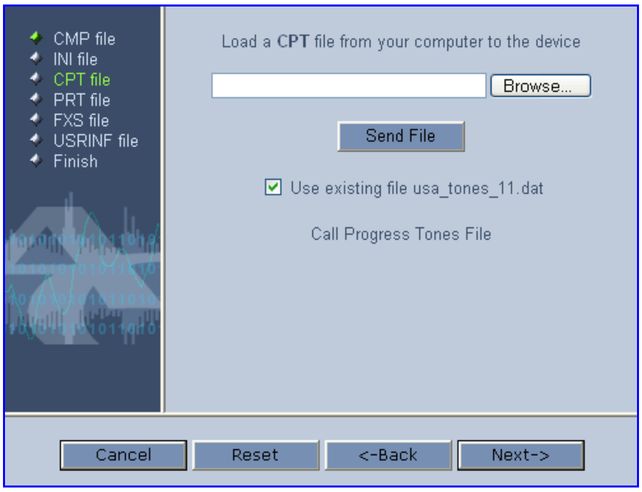
   Note that as you progress, the file type list on the left indicates which file type loading is in process by illuminating green (until 'FINISH').

Figure 5-66: Load an *ini* File Screen



7. In the 'Load an *ini* File' screen, you can now choose to either:
   - Click **Browse** and navigate to the *ini* file; the check box 'Use existing configuration', by default checked, becomes unchecked. Click **Send File**; the *ini* file is loaded to the MediaPack and you're notified as to a successful loading.
   - Ignore the **Browse** button (its field remains undefined and the check box 'Use existing configuration' remains checked by default).
   - Ignore the **Browse** button and uncheck the 'Use existing configuration' check box; no *ini* file is loaded, the MediaPack uses its factory-preconfigured values.
8. You can now choose to either:
   - Click **Cancel**; the MediaPack resets utilizing the *cmp, ini* and all other configuration files that were previously stored in flash memory. Note that these are NOT the files you loaded in the previous Wizard steps.
   - Click **Reset**; the MediaPack resets, utilizing the new *cmp* and *ini* file you loaded up to now as well as utilizing the other configuration files.
   - Click **Back**; the 'Load a *cmp* file' screen is reverted to; refer to Figure 5-64.
   - Click **Next**; the 'Load a CPT File' screen opens, refer to Figure 5-67; Loading a new CPT file or any other auxiliary file listed in the Wizard is optional.

Figure 5-67: Load a CPT File Screen



9. Follow the same procedure you followed when loading the *ini* file (refer to Step 7). The same procedure applies to the 'Load a coefficient file' screen.
10. In the 'Finish' screen (refer to Figure 5-68), the **Next** button is disabled. Complete the upgrade process by clicking **Reset** or **Cancel**.
    - Click **Reset**, the MediaPack 'burns' the newly loaded files to flash memory. The 'Burning files to flash memory' screen appears. Wait for the 'burn' to finish. When it finishes, the 'End Process' screen appears displaying the burned configuration files (refer to Figure 5-69).
    - Click **Cancel**, the MediaPack resets, utilizing the files previously stored in flash memory. (Note that these are NOT the files you loaded in the previous Wizard steps).

Figure 5-68: Finish Screen

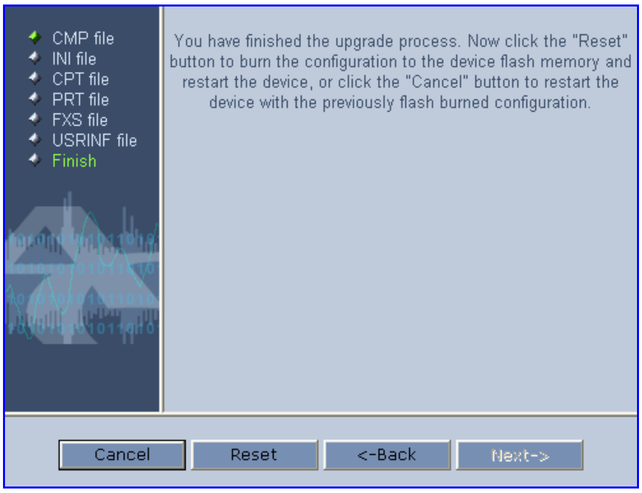
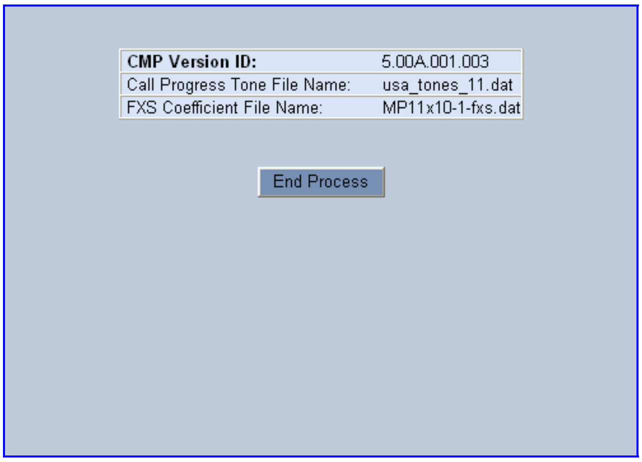


Figure 5-69: 'End Process' Screen



11. Click the **End Process** button; the 'Quick Setup' screen appears and the full Web application is reactivated.

5.9.2 Auxiliary Files

The ‘Auxiliary Files’ screen enables you to load to the gateway the following files: Call Progress Tones, coefficient, Prerecorded Tones (PRT) and User Information. The Voice Prompts file is currently not applicable to the MediaPack. For detailed information on these files, refer to Section 6 on page 209. For information on deleting these files from the MediaPack, refer to Section 5.7.3 on page 193. Table 5-63 presents a brief description of each auxiliary file.

Table 5-63: Auxiliary Files Descriptions

File Type	Description
FXS Coefficient	This file contains the telephony interface configuration information for the VoIP gateway. This information includes telephony interface characteristics, such as DC and AC impedance, feeding current and ringing voltage. This file is specific to the type of telephony interface that the VoIP gateway supports. In most cases you have to load this type of file. **Note:** Use the parameter ‘CountryCoefficients‘ (described in Table 5-35 on page 132) to configure the FXO coefficients.
Call Progress Tones	This is a region-specific, telephone exchange-dependent file that contains the Call Progress Tones levels and frequencies that the VoIP gateway uses. The default CPT file is: U.S.A.
Prerecorded Tones	The *dat* PRT file enhances the gateway’s capabilities of playing a wide range of telephone exchange tones that cannot be defined in the Call Progress Tones file.
User Information	The User Information file maps PBX extensions to IP numbers. This file can be used to represent PBX extensions as IP phones in the global ‘IP world’.

➢ To load an auxiliary file to the gateway, take these 8 steps:

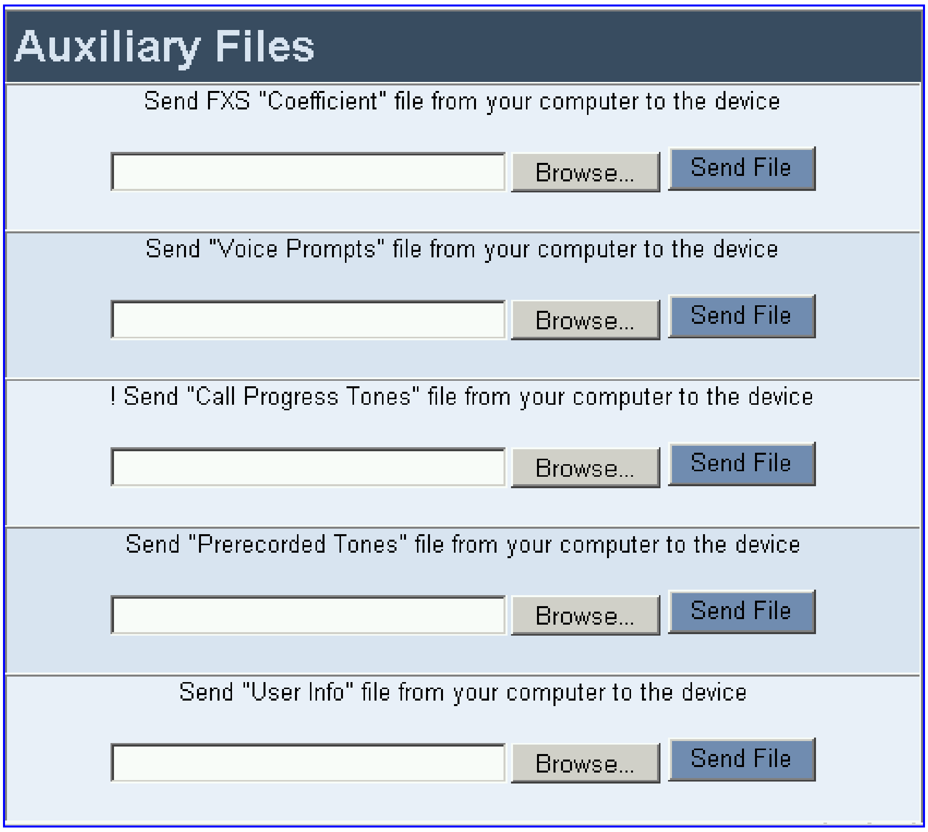
1. Open the ‘Auxiliary Files’ screen (**Software Upgrade** menu > **Load Auxiliary Files**); the ‘Auxiliary Files’ screen is displayed.
2. Click the **Browse** button that is in the field for the type of file you want to load.
3. Navigate to the folder that contains the file you want to load.
4. Select the file and click the **Open** button; the name and path of the file appear in the field beside the **Browse** button.
5. Click the **Send File** button that is next to the field that contains the name of the file you want to load. An exclamation mark in the screen section indicates that the file’s loading doesn’t take effect on-the-fly (e.g., CPT file).
6. Repeat steps 2 to 5 for each file you want to load.

Notes:

- Saving an auxiliary file to flash memory may disrupt traffic on the MediaPack. To avoid this, disable all traffic on the device before saving to flash memory by performing a graceful lock (refer to Section 5.10.1 on page 204).
- A MediaPack reset is required to activate a loaded CPT file, and may be required for the activation of certain *ini* file parameters.

7. To save the loaded auxiliary files so they are available after a power fail, refer to Section 5.10.2 on page 205.
8. To reset the MediaPack, refer to Section 5.10.3 on page 206.

Figure 5-70: Auxiliary Files Screen



5.9.2.1 Loading the Auxiliary Files via the *ini* File

➢ To load the auxiliary files via the *ini* file, take these 3 steps:

1. In the *ini* file, define the auxiliary files to be loaded to the MediaPack. You can also define in the *ini* file whether the loaded files should be stored in the non-volatile memory so that the TFTP process is not required every time the MediaPack boots up.
2. Locate the auxiliary files you want to load and the *ini* file in the same directory.
3. Invoke a BootP/TFTP session; the *ini* and auxiliary files are loaded onto the MediaPack.

Table 5-64 below describes the *ini* file parameters that are associated with the configuration files.

Table 5-64: Configuration Files *ini* File Parameters

ini File Parameter Name	Description
CallProgressTonesFileName	The name (and path) of the file containing the Call Progress Tones definition.
FXSLoopCharacteristicsFileName	The name (and path) of the file providing the FXS line characteristic parameters.
PrerecordedTonesFileName	The name (and path) of the file containing the Prerecorded Tones.
UserInfoFileName	The name (and path) of the file containing the User Information data.
SaveConfiguration	Determines if the gateway's configuration (parameters and files) is saved to flash (non-volatile memory). 0 = Configuration isn't saved to flash memory. 1 = Configuration is saved to flash memory (default).

5.10 Maintenance

The Maintenance menu is used for the following operations:

- Locking and unlocking the gateway (refer to Section 5.10.1 on page 204)
- Saving the gateway's configuration (refer to Section 5.10.2 on page 205)
- Resetting the gateway (refer to Section 5.10.3 on page 206)

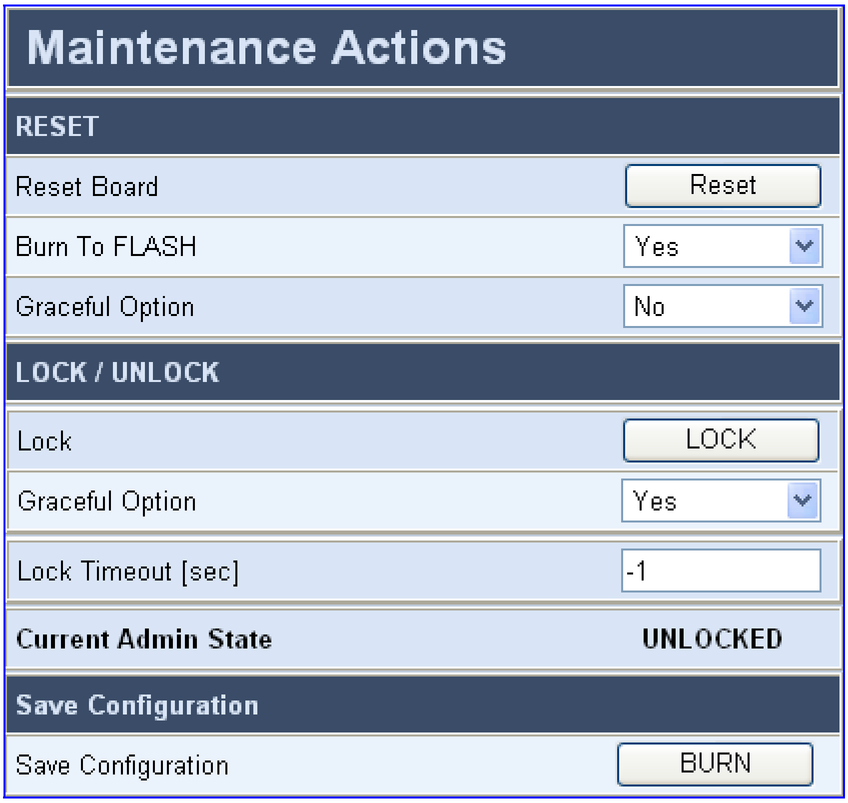
5.10.1 Locking and Unlocking the Gateway

The Lock and Unlock options allow you to lock the gateway so that it does not accept any new incoming calls. This is beneficial when, for example, you are uploading new software files to the gateway and you don't want any traffic to interfere with the process.

➢ To lock the gateway, take these 4 steps:

1. Open the 'Maintenance Actions' screen (**Maintenance** menu); the 'Maintenance Actions' screen is displayed.

Figure 5-71: Maintenance Actions Screen



2. Under the LOCK / UNLOCK group, from the 'Graceful Option' drop-down list, select one of the following options:
   - 'Yes': The gateway is 'locked' only after the user-defined time in the 'Lock Timeout' field (refer to Step 3) expires or no more active traffic exists (the earliest thereof). In addition, no new traffic is accepted.
   - 'No': The gateway is "locked" regardless of traffic. Any existing traffic is terminated immediately.
3. In the 'Lock Timeout' field (relevant only if the 'Graceful Option' in the previous step is set to 'Yes'), enter the time (in seconds) after which the gateway locks. Note that if no traffic exists and the time has not expired, the gateway locks.

4. Click the **LOCK** button. If 'Graceful Option' is set to 'Yes', the lock is delayed and a screen displaying the number of remaining calls and time is displayed. Otherwise, the lock process begins immediately. The Current Admin State displays the current state: LOCKED or UNLOCKED.

➢ To unlock the gateway, take these 2 steps:

1. Access the 'Maintenance Actions' screen as described above in the previous procedure.
2. Click the **UNLOCK** button. Unlock starts immediately and the gateway is ready for new incoming calls.

5.10.2 Saving Configuration

The 'Maintenance Actions' screen enables you to save the current parameter configuration and the loaded auxiliary files to the *non-volatile* memory (i.e., flash) so they are available after a hardware reset (or power fail). Parameters that are only saved to the *volatile* memory (RAM) revert to their previous settings after hardware reset.

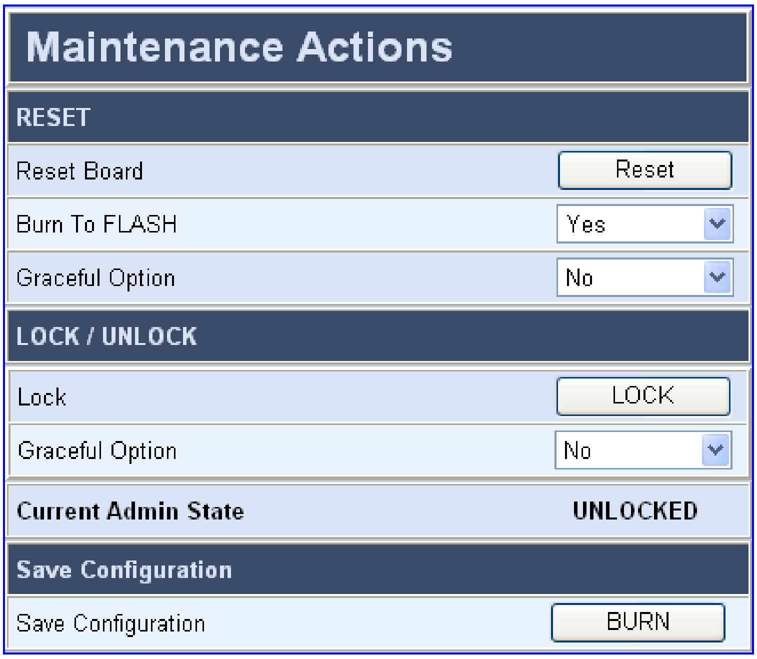
Notes:

- Saving changes to the *non-volatile* memory may disrupt traffic on the gateway. To avoid this, disable all new traffic before saving by performing a graceful lock (refer to Section 5.10.1 on page 204).
- In the Web interface, parameters prefixed with an exclamation mark ('!') are saved to the non-volatile memory only after a device reset.

➢ To save the changes to the *non-volatile*, take these 2 steps:

1. Open the 'Maintenance Actions' screen (**Maintenance** menu); the 'Maintenance Actions' screen is displayed.

Figure 5-72: Maintenance Actions Screen



2. Click the **BURN** button; a confirmation message appears when the save is completed successfully.

5.10.3 Resetting the MediaPack

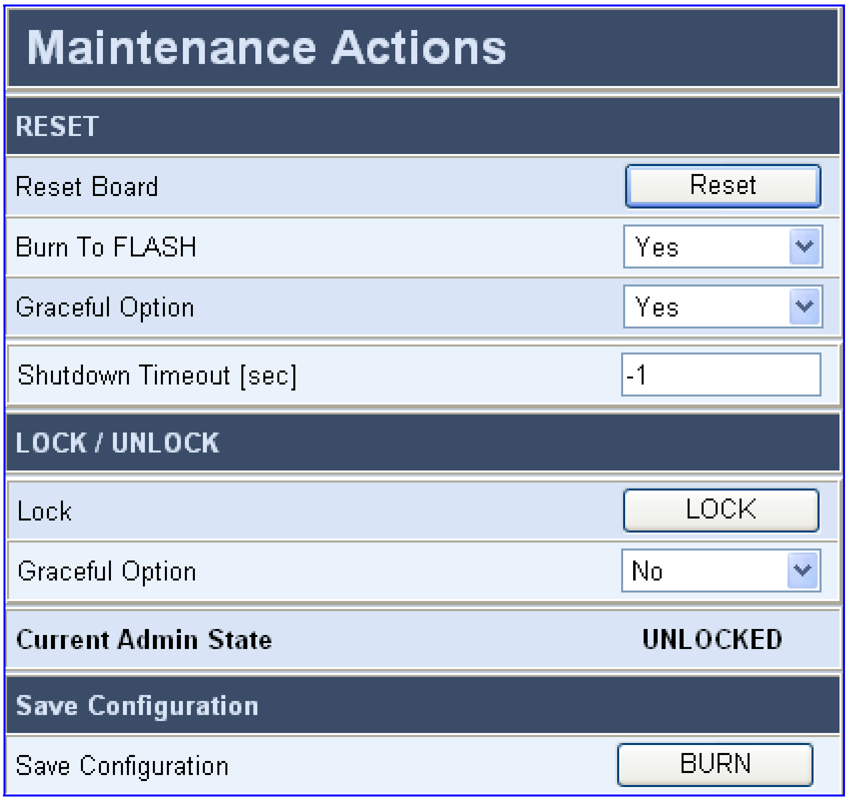
The 'Maintenance Actions' screen enables you to remotely reset the gateway. Before you reset the gateway, you can choose the following options:

- Save the gateway's current configuration to the flash memory (non-volatile).
- Perform a graceful shutdown. Reset starts only after a user-defined time expires or no more active traffic exists (the earliest thereof).

➢ To reset the gateway, take these 5 steps:

1. Open the 'Maintenance Actions' screen (**Maintenance** menu); the 'Maintenance Actions' screen is displayed.

Figure 5-73: Maintenance Actions Screen



2. Under the RESET group, from the 'Burn To FLASH' drop-down list, select one of the following options:
   - 'Yes': The gateway's current configuration is burned (i.e., saved) to the flash memory prior to reset (default).
   - 'No': Resets the device without burning (i.e., saving) the current configuration to flash (discards all unsaved modifications to the configuration).
3. Under the RESET group, from the 'Graceful Option' drop-down list, select one of the following options:
   - 'Yes': Reset starts only after the user-defined time in the 'Shutdown Timeout' field (refer to Step 4) expires or no more active traffic exists (the earliest thereof). In addition, no new traffic is accepted.
   - 'No': Reset starts regardless of traffic and any existing traffic is terminated at once.

4. In the 'Shutdown Timeout' field (relevant only if the 'Graceful Option' in the previous step is set to 'Yes'), enter the time after which the gateway resets. Note that if no traffic exists and the time has not expired, the gateway resets.
8. Click the **Reset** button. If 'Graceful Option' is set to 'Yes', the reset is delayed and a screen displaying the number of remaining calls and time is displayed. When the device resets, a message is displayed informing of the waiting period.

5.11 Logging Off the Embedded Web Server

The **Log Off** button enables you to log off the Embedded Web Server and to re-access it with a different account. For detailed information on the Web User Accounts, refer to Section 5.2.1 on page 49.

➢ To log off the Embedded Web Server, take these 2 steps:

1. Click the **Log Off** button on the main menu bar; the Log Off prompt screen is displayed.

Figure 5-74: Log off Prompt

2. Click **OK**; the Web session is logged off.

Reader's Notes

6 *ini* File Configuration of the MediaPack

As an alternative to configuring the VoIP gateway using the Web Interface (refer to Chapter 5 on page 49), it can be configured by loading the *ini* file containing Customer-configured parameters.

The *ini* file is loaded via the BootP/TFTP utility (refer to Appendix C on page 349) or via any standard TFTP server. It can also be loaded through the Web Interface (refer to Section 5.6.3 on page 165).

The *ini* file configuration parameters are stored in the MediaPack non-volatile memory after the file is loaded. When a parameter is missing from the *ini* file, a default value is assigned to that parameter (according to the *cmp* file loaded on the MediaPack) and stored in the non-volatile memory (thereby overriding the value previously defined for that parameter). Therefore, to restore the default configuration parameters, use the *ini* file without any valid parameters or with a semicolon (;) preceding all lines in the file.

Some of the MediaPack parameters are configurable through the *ini* file only (and not via the Web). These parameters usually determine a low-level functionality and are seldom changed for a specific application.

Note: For detailed explanation of each parameter, refer to Chapter 5 on page 49.

6.1 Secured *ini* File

The *ini* file contains sensitive information that is required for the functioning of the MediaPack. It is loaded to, or retrieved from, the device via TFTP or HTTP. These protocols are unsecured and vulnerable to potential hackers. Therefore an encoded *ini* file significantly reduces these threats.

You can choose to load an encoded *ini* file to the MediaPack. When you load an encoded *ini* file, the retrieved *ini* file is also encoded. Use the 'TrunkPack Downloadable Conversion Utility' to encode or decode the *ini* file before you load it to, or retrieve it from the device. Note that the encoded *ini* file's loading procedure is identical to the regular *ini* file's loading procedure. For information on encoding / decoding an *ini* file, refer to Section E.1.2 on page 365.

6.2 Modifying an *ini* File

➢ To modify the *ini* file, take these 3 steps:

1. Get the *ini* file from the gateway using the Embedded Web Server (refer to Section 5.6.3 on page 165).
2. Open the file (the file is open in Notepad or a Customer-defined text file editor) and modify the *ini* file parameters according to your requirements; save and close the file.
3. Load the modified *ini* file to the gateway (using either BootP/TFTP utility or the Embedded Web Server).

This method preserves the programming that already exists in the device, including special default values that were preconfigured when the unit was manufactured.

Tip: Before loading the *ini* file to the gateway, verify that the extension of the *ini* file saved on your PC is correct: Verify that the check box 'Hide file extension for known file types' (My computer>Tools>Folder Options>View) is unchecked. Then, confirm that the *ini* file name extension is xxx.ini and NOT erroneously xxx.ini.ini or xxx~.ini.

6.3 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 in Figure 6-1 below.

Figure 6-1: *ini* File Structure

```
[Sub Section Name]

Parameter_Name = Parameter_Value
Parameter_Name = Parameter_Value

; REMARK

[Sub Section Name]
```

6.3.1 The *ini* File Structure Rules

- The *ini* file name mustn't include hyphens or spaces, use underscore instead.
- Lines beginning with a semi-colon ';' (as the first character) are ignored.
- A Carriage Return must be the final character of each line.
- The 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.
- String parameters, representing file names, for example CallProgressTonesFileName, must be placed between two inverted commas ('…').
- The parameter name is NOT case-sensitive; the parameter value is not case-sensitive *except for coder names*.
- The *ini* file should be ended with one or more carriage returns.

6.3.2 The *ini* File Example

Figure 6-2 shows an example of an *ini* file for the VoIP gateway.

Figure 6-2: SIP *ini* File Example

```
[Channel Params]
DJBufMinDelay = 75
RTPRedundancyDepth = 1
DefaultNumber = 101
MaxDigits = 3
CoderName = g7231,90
; Phone of each endpoint
EnableSyslog = 0
[Files]
CallProgressTonesFilename = 'CPUSA.dat'
FXSLoopCharacteristicsFileName = 'coeff.dat'
SaveConfiguration = 1
```

7 Using BootP / DHCP

The MediaPack uses the Bootstrap Protocol (BootP) and the Dynamic Host Configuration Protocol (DHCP) to obtain its networking parameters and configuration automatically after it is reset. BootP and DHCP are also used to provide the IP address of a TFTP server on the network, and files (*cmp* and *ini*) to be loaded into memory.

DHCP is a communication protocol that automatically assigns IP addresses from a central point. BootP is a protocol that enables a device to discover its own IP address. Both protocols have been extended to enable the configuration of additional parameters specific to the MediaPack.

Note: BootP is normally used to initially configure the MediaPack. Thereafter, BootP is no longer required as all parameters can be stored in the gateway's non-volatile memory and used when BootP is inaccessible. BootP can be used again to change the IP address of the MediaPack (for example).

7.1 BootP/DHCP Server Parameters

BootP/DHCP can be used to provision the following parameters (included in the BootP/DHCP reply). Note that only the IP address and subnet mask are mandatory:

- IP address, subnet mask - These mandatory parameters are sent to the MediaPack every time a BootP/DHCP process occurs.
- Default gateway IP address - An optional parameter that is sent to the MediaPack only if configured in the BootP/DHCP server.
- TFTP server IP address - An optional parameter that contains the address of the TFTP server from which the firmware (*cmp*) and *ini* files are loaded.
- DNS server IP address (primary and secondary) - Optional parameters that contain the IP addresses of the primary and secondary DNS servers. These parameters are available only in DHCP and from Boot version 1.92.
- Syslog server IP address - An optional parameter that is sent to the MediaPack only if configured. This parameter is available only in DHCP.
- SIP server IP address – Two optional parameters that are sent to the MediaPack only if configured. These parameters are available only in DHCP.
- Firmware file name – An optional parameter that contains the name of the firmware file to be loaded to the gateway via TFTP.
- *ini* file name - An optional parameter that contains the name of the *ini* file to be loaded to the gateway via TFTP.

7.2 Using DHCP

When the gateway is configured to use DHCP (DHCPEnable = 1), it attempts to contact the local DHCP server to obtain the networking parameters (IP address, subnet mask, default gateway, primary/secondary DNS server and two SIP server addresses). These network parameters have a 'time limit'. After the time limit expires, the gateway must 'renew' its lease from the DHCP server.

Note that if the DHCP server denies the use of the gateway's current IP address and specifies a different IP address (according to RFC 1541), the gateway must change its networking parameters. If this happens while calls are in progress, they are not automatically rerouted to the new network address (since this function is beyond the scope of a VoIP gateway). Therefore, administrators are advised to configure DHCP servers to allow renewal of IP addresses.

> **Note:** If the gateway's network cable is disconnected and reconnected, a DHCP renewal is performed (to verify that the gateway is still connected to the same network).

When DHCP is enabled, the gateway also includes its product name (e.g., 'MP-118 FXS') in the DHCP 'option 60' Vendor Class Identifier. The DHCP server can use this product name to assign an IP address accordingly.

> **Note:** After power-up, the gateway performs two distinct DHCP sequences. Only in the second sequence, DHCP 'option 60' is contained. If the gateway is reset from the Web/SNMP, only a single DHCP sequence containing 'option 60' is sent.

If DHCP procedure is used, the new gateway IP address, allocated by the DHCP server, must be detected.

> **Note:** If, during operation, the IP address of the gateway is changed as a result of a DHCP renewal, the gateway is automatically reset.

➢ To detect the gateway's IP address, follow one of the procedures below:

- Starting with Boot version 1.92, the gateway can use a host name in the DHCP request. The host name is set to acl_nnnnn, where nnnnn stands for the gateway's serial number (the serial number is equal to the last 6 digits of the MAC address converted from Hex to decimal). If the DHCP server registers this host name to a DNS server, the user can access the gateway (through a Web browser) using a URL of http://acl_<serial number> (instead of using the gateway's IP address). For example, if the gateway's MAC address is 00908f010280, the DNS name is acl_66176.
- After physically resetting the gateway its IP address is displayed in the 'Client Info' column in the BootP/TFTP configuration utility (refer to Figure C-1 on page 351).
- Use a serial communication software (refer to Section 4.2.4 on page 44).
- Contact your System Administrator.

7.3 Using BootP

7.3.1 Upgrading the MediaPack

When upgrading the MediaPack (loading new software onto the gateway) using the BootP/TFTP configuration utility:

- From version 4.4 to version 4.4 or to any higher version, the device retains its configuration (*ini* file). However, the auxiliary files (CPT, logo, etc.) may be erased.
- From version 4.6 to version 4.6 or to any higher version, the device retains its configuration (*ini* file) and auxiliary files (CPT, logo, etc.).

You can also use the Software Upgrade wizard, available through the Web Interface (refer to Section 5.9.1 on page 197).

Note: To save the *cmp* file to non-volatile memory, use the **-fb** command line switches. If the file is not saved, the gateway reverts to the old version of software after the next reset. For information on using command line switches, refer to Section C.11.6 on page 359.

7.3.2 Vendor Specific Information Field

The MediaPack uses the vendor specific information field in the BootP request to provide device-related initial startup information. The BootP/TFTP configuration utility displays this information in the 'Client Info' column (refer to Figure C-1).

Note: This option is not available on DHCP servers.

The Vendor Specific Information field is disabled by default. To enable / disable this feature: set the *ini* file parameter 'ExtBootPReqEnable' (Table 5-55 on page 182) or use the '-be' command line switch (refer to Table C-1 on page 359).

Table 7-1 details the vendor specific information field according to device types:

Table 7-1: Vendor Specific Information Field

Tag #	Description	Value	Length
220	Gateway Type	#13 = MP-124 #14 = MP-118 #15 = MP-114 #16 = MP-112	1
221	Current IP Address	XXX.XXX.XXX.XXX	4
222	Burned Boot Software Version	X.XX	4
223	Burned *cmp* Software Version	XXXXXXXXXXXX	12
224	Geographical Address	0 – 31	1
225	Chassis Geographical Address	0 – 31	1
228	Indoor / Outdoor (Indoor is valid only for FXS. FXO is always Outdoor.)	#0 = Indoor #1 = Outdoor	1
229	E&M	N/A	1
230	Analog Channels	2 / 4 / 8 / 24	1

Table 7-2 exemplifies the structure of the vendor specific information field for a TP-1610 slave module with IP address 10.2.70.1.

Table 7-2: Structure of the Vendor Specific Information Field

Vendor-Specific Information Code	Total Length	Tag Num	Length	Value	Tab Num	Length	Value	Tag Num	Length	Value (1)	Value (2)	Value (3)	Value (4)	Tag End
42	12	220	1	2	225	1	1	221	4	10	2	70	1	255

8 Telephony Capabilities

8.1 Working with Supplementary Services

The MediaPack SIP FXS and FXO gateways support the following supplementary services:

- Call Hold / Retrieve; refer to Section 8.1.1 on page 215.
- Consultation / Alternate; refer to Section 8.1.2 on page 216.
- Transfer (Refer + Replaces); refer to Section 8.1.3 on page 216.
- Call Forward (3xx Redirect Responses); refer to Section 8.1.4 on page 217.
- Call Waiting (182 Queued Response); refer to Section 8.1.5 on page 217.
- Message Waiting Indication (MWI); refer to Section 8.1.6 on page 218.

To activate these supplementary services (Hold, Transfer, Forward, Waiting and MWI) on the MediaPack gateway, enable each service's corresponding parameter either from the Web Interface or via the *ini* file. Note that all call participants must support the specific used method.

Note: When working with application servers (such as BroadSoft's BroadWorks) in client server mode (the application server controls all supplementary services and keypad features by itself), the gateway's supplementary services must be disabled.

8.1.1 Call Hold and Retrieve

8.1.1.1 Initiating Hold/Retrieve

- Active calls can be put on-hold by pressing the phone's hook-flash button.
- The party that initiates the hold is called the holding party; the other party is called the held party.
- After a successful Hold, the holding party hears a Dial Tone.
- Call retrieve can be performed only by the holding party while the call is held and active.
- The holding party performs the retrieve by pressing the hook-flash.
- After a successful retrieve, voice is connected again.
- Hold is performed by sending a REINVITE with the IP address 0.0.0.0 or 'a=sendonly' in the SDP according to the parameter 'HoldFormat'.

8.1.1.2 Receiving Hold / Retrieve

- When an active call receives REINVITE message with either the IP address 0.0.0.0 or the 'inactive' string in SDP, the gateway stops sending RTP and plays a local Held Tone.
- When an active call receives REINVITE message with 'sendonly' string in SDP, the gateway stops sending RTP and listens to the remote party. In this mode, it is expected that on-hold music (or any other hold tone) is to be played (over IP) by the remote party.

8.1.2 Consultation / Alternate

- The Consultation feature is relevant only for the holding party (applicable only to the MediaPack/FXS gateway).
- After holding a call (by pressing hook-flash), the holding party hears dial tone and can now initiate a new call, which is called a consultation call.
- While hearing a dial tone or when dialing to the new destination (before dialing is complete) the user can retrieve the held call by pressing hook-flash.
- The held call can’t be retrieved while Ringback tone is heard.
- After the consultation call is connected, the user can switch between the held and active call by pressing hook-flash.

8.1.3 Call Transfer

There are two types of call transfers:

- Consultation Transfer (REFER + REPLACES)
- Blind Transfer (REFER)

The common way to perform a consultation transfer is as follows:

In the transfer scenario there are three parties:

Party A = transferring, Party B = transferred, Party C = transferred to.

- A Calls B.
- B answers.
- A presses the hook-flash and puts B on-hold (party B hears a hold tone)
- A dials C.
- After A completed dialing C, he can perform the transfer by onhook the A phone.
- After the transfer is completed B and C parties are engaged in a call.

The transfer can be initiated at any of the following stages of the call between A to C:

- Just after completing dialing C phone number - Transfer from setup.
- While hearing Ringback - Transfer from alert.
- While speaking to C - Transfer from active.

Blind transfer is performed after we have a call between A and B, and party A decides to transfer the call to C immediately without speaking with C.

The result of the transfer is a call between B and C (just like consultation transfer only skipping the consultation stage).

Note the following SIP issues:

- Transfer is initiated by sending REFER with REPLACES.
- The gateway can receive and act upon receiving REFER with or without REPLACES.
- The gateway can receive and act upon receiving INVITE with REPLACES, in which case the old call is replaced by the new one.
- The INVITE with REPLACES can be used to implement Directed Call Pickup.

8.1.4 Call Forward

Five forms of call forward are supported:

- **Immediate:** Any incoming call is forwarded immediately and unconditionally.
- **Busy:** Incoming call is forwarded if the endpoint is busy.
- **No Reply:** The incoming call is forwarded if it isn't answered for a specified time.
- **On Busy or No Reply:** Forward incoming calls when the port is busy or when calls are not answered after a specified time.
- **Do Not Disturb:** Immediately reject incoming calls. Upon receipt of a Do Not Disturb call, the gateway responds with a 603 Decline SIP code.

Three forms of forwarding parties are available:

- **Served party:** the party that is configured to forward the call (MediaPack/FXS)
- **Originating party:** the party that initiated the first call (MediaPack/FXS or FXO)
- **Diverted party:** the new destination of the forwarded call (MediaPack/FXS or FXO)

The served party (MediaPack/FXS) can be configured through the Web browser (refer to Section 5.5.9.4 on page 122) or via *ini* file to activate one of the call forward modes. These modes are configurable per gateway's endpoint.

Note the following SIP issues:

- Initiating forward – When forward is initiated, the gateway sends a 302 response with a contact that contains the phone number from the forward table and its corresponding IP address from the routing table (or, when Proxy is used, the proxy's IP address).
- Receiving forward – The gateway handles 3xx responses for redirecting calls with a new contact.

8.1.5 Call Waiting

The Call Waiting feature enables FXS gateways to accept an additional (second) call on busy endpoints. If an incoming IP call is designated to a busy port, the called party hears call waiting tone (several configurable short beeps) and (for Bellcore and ETSI Caller IDs) can view the Caller ID string of the incoming call. The calling party hears a Call Waiting Ringback Tone. Called party can accept the new call, using hook-flash, and can toggle between the two calls.

To enable Call Waiting:

- Set 'EnableCallWaiting = 1'.
- Set 'EnableHold = 1'.
- Define the Call Waiting indication and Call Waiting Ringback tones in the Call Progress Tones file. You can define up to four Call Waiting indication tones (refer to the parameter 'FirstCallWaitingToneID' in Table 5-35).
- To configure the Call Waiting indication tone cadence, modify the following parameters: 'NumberOfWaitingIndications', 'WaitingBeepDuration' and 'TimeBetweenWaitingIndications'.
- To configure a delay interval before a Call Waiting Indication is played to the currently busy port use the parameter 'TimeBeforeWaitingIndication'. This enables the caller to hang up before disturbing the called party with Call Waiting Indications. Applicable only to FXS gateways.

Both the calling and the called sides are supported by FXS gateways; the FXO gateways support only the calling side.

To indicate Call Waiting, the gateway sends a 182 - call queued response.

The gateway identifies a Waiting Call when a 182 (call queued response) is received.

8.1.6 Message Waiting Indication

Support for Message Waiting Indication (MWI) according to IETF <draft-ietf-sipping-mwi-04.txt>, including SUBSCRIBE (to MWI server). MediaPack/FXS gateways can accept an MWI NOTIFY message that indicates waiting messages or that the MWI is cleared. Users are informed of these messages by a stutter dial tone. The stutter and confirmation tones are defined in the CPT file (refer to Section 15.1 on page 325). If the MWI display is configured, the number of waiting messages is also displayed. If the MWI lamp is configured, the phone's lamp (on a phone that is equipped with an MWI lamp) is lit. The gateway can subscribe to the MWI server per port (usually used on FXS) or per gateway (used on FXO).

To configure MWI set the following parameters:

- EnableMWI
- MWIServerIP
- MWIAnalogLamp
- MWIDisplay
- StutterToneDuration
- EnableMWISubscription
- MWIExpirationTime
- SubscribeRetryTime
- SubscriptionMode
- CallerIDType (determines the standard for detection of MWI signals)
- ETSIVMWITypeOneStandard
- BellcoreVMWITypeOneStandard

8.2 Configuring the DTMF Transport Types

You can control the way DTMF digits are transported over the IP network to the remote endpoint. The following five modes are supported:

1. Using INFO message according to the Nortel IETF draft: In this mode DTMF digits are carried to the remote side within INFO messages. To enable this mode set:
   - 'RxDTMFOption = 0' (**Protocol Management** > **Protocol Definition** > **DTMF & Dialing** > 'Declare RFC 2833 in SDP' = 'No')
   - 'TxDTMFOption = 1' (1st to 5th DTMF Option = INFO (Nortel))
     Note that in this mode DTMF digits are erased from the audio stream (DTMFTransportType is automatically set to 0 (DTMF Mute)).
2. Using INFO message according to Cisco's mode:

   In this mode DTMF digits are carried to the remote side within INFO messages. To enable this mode set:
   - 'RxDTMFOption = 0' (Declare RFC 2833 in SDP = No)
   - 'TxDTMFOption = 3' (1st to 5th DTMF Option = INFO (Cisco))
     Note that in this mode DTMF digits are erased from the audio stream (DTMFTransportType is automatically set to 0 (DTMF Mute)).

3. Using NOTIFY messages according to <draft-mahy-sipping-signaled-digits-01.txt>: In this mode DTMF digits are carried to the remote side using NOTIFY messages. To enable this mode set:
   - 'RxDTMFOption = 0' (Declare RFC 2833 in SDP = No)
   - 'TxDTMFOption = 2' (1^{st} to 5^{th} DTMF Option = NOTIFY)
     Note that in this mode DTMF digits are erased from the audio stream (DTMFTransportType is automatically set to 0 (DTMF Mute)).
4. Using RFC 2833 relay with Payload type negotiation: In this mode, DTMF digits are carried to the remote side as part of the RTP stream in accordance with RFC 2833 standard. To enable this mode set:
   - 'TxDTMFOption = 4' (1^{st} to 5^{th} DTMF Option = RFC 2833)
   - 'RxDTMFOption = 3' (Declare RFC 2833 in SDP = Yes)
     Note that to set the RFC 2833 payload type with a different value (other than its default, 96) configure the 'RFC2833PayloadType' (RFC 2833 Payload Type) parameter. The gateway negotiates the RFC 2833 payload type using local and remote SDP and sends packets using the PT from the received SDP. The gateway expects to receive RFC 2833 packets with the same PT as configured by the 'RFC2833PayloadType' parameter. If the remote side doesn't include 'telephony-event' in its SDP, the gateway sends DTMF digits in transparent mode (as part of the voice stream).
5. Sending DTMF digits (in RTP packets) as part of the audio stream (DTMF Relay is disabled):
   Note that this method is normally used with G.711 coders; with other LBR coders the quality of the DTMF digits is reduced. To ser this mode:
   - 'TxDTMFOption = 0' (1^{st} to 5^{th} DTMF Option = Disable)
   - 'RxDTMFOption = 0' (Declare RFC 2833 in SDP = No)
   - 'DTMFTransportType = 2' (DTMF Transport Type = Transparent DTMF)

Notes:

- The gateway is always ready to receive DTMF packets over IP, in all possible transport modes: INFO messages, NOTIFY and RFC 2833 (in proper payload type) or as part of the audio stream.
- To exclude RFC 2833 Telephony event parameter from the gateway's SDP, set 'RxDTMFOption = 0' in the *ini* file.

The following parameters affect the way the MediaPack SIP handles the DTMF digits:

- TxDTMFOption, RxDTMFOption and RFC2833PayloadType (described in Table 5-7).
- MGCPDTMFDetectionPoint, DTMFDigitLength and DTMFInterDigitInterval (Table 5-47).
- DTMFVolume and DTMFTransportType (Table 5-42).

8.3 Fax & Modem Transport Modes

8.3.1 Fax/Modem Settings

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

- **Fax relay:** demodulation / modulation
- **Bypass:** using a high bit rate coder to pass the signal
- **Transparent:** passing the signal in the current voice coder

When the fax relay mode is 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 distinction is determined. The packets being sent to the network at this stage are T.38-compliant fax relay packets.

8.3.2 Configuring Fax Relay Mode

When FaxTransportMode = 1 (relay mode), then on detection of fax the channel automatically switches from the current voice coder to answer tone mode, and then to T.38-compliant fax relay mode.

When fax transmission has ended, the reverse switching from fax relay to voice is performed. This mode switching automatically occurs at both 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 'FaxRelayEnhancedRedundancyDepth' parameters. Although this is a proprietary redundancy scheme, it should not create problems when working with other T.38 decoders.

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

8.3.3 Configuring Fax/Modem Bypass Mode

When VxxTransportType= 2 (FaxModemBypass, Vxx can be one of the following: V32/V22/Bell/V34/Fax), then on detection of fax/modem, the channel automatically switches from the current voice coder to a high bit-rate coder (G.711 or G.726), as defined by the user, with the FaxModemBypassCoderType configuration parameter.

During the bypass period, the coder uses 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 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 switching, from bypass coder to regular voice coder, is carried out.

8.3.4 Supporting V.34 Faxes

V.34 faxes don't comply with the T.38 relay standard. We therefore provide the optional modes described under Sections 8.3.4.1 and 8.3.4.2:

Note that the CNG detector is disabled (CNGDetectorMode=0) in all the following examples.

8.3.4.1 Using Bypass Mechanism for V.34 Fax Transmission

In this proprietary scenario, the media gateway uses a high bit-rate coder to transmit V.34 faxes, enabling the full utilization of its speed.

Refer to the following configurations:

```
FaxTransportMode = 2 (Bypass)
V34ModemTransportType = 2 (Modem bypass)
V32ModemTransportType = 2
V23ModemTransportType = 2
V22ModemTransportType = 2
```

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

Or

```
FaxTransportMode = 1 (Relay)
V34ModemTransportType = 2 (Modem bypass)
V32ModemTransportType = 2
V23ModemTransportType = 2
V22ModemTransportType = 2
```

In this configuration, T.30 fax uses T.38 Relay mode while V.34 fax uses Bypass mode.

8.3.4.2 Using Relay mode for both T.30 and V.34 faxes

In this scenario, V.34 fax machines are compelled to use their backward compatibility with T.30 faxes; as a T.30 machine, the V.34 fax can use T.38 Relay mode.

Refer to the following configuration:

```
FaxTransportMode = 1 (Relay)
V34ModemTransportType = 0 (Transparent)
V32ModemTransportType = 0
V23ModemTransportType = 0
V22ModemTransportType = 0
```

Both T.30 and V.34 faxes use T.38 Relay mode. This configuration forces the V.34 fax machine to operate in the slower T.30 mode.

8.3.5 Supporting V.152 Implementation

The MediaPack gateway supports the ITU-T recommendation V.152 (Procedures for Supporting Voice-Band Data over IP Networks). Voice-band data (VBD) is the transport of modem, facsimile, and text telephony signals over a voice channel of a packet network with a codec appropriate for such signals.

For V.152 capability, the gateway supports T.38 as well as VBD codecs (i.e., G.711 A-law and G.711 µ-law). The selection of capabilities is performed using the coders table.

When in VBD mode for V.152 implementation, support is negotiated between the gateway and the remote endpoint at the establishment of the call. During this time, initial exchange of call capabilities is exchanged in the outgoing SDP. These capabilities include whether VBD is supported and associated RTP payload types ('gpmd' SDP attribute), supported codecs, and packetization periods for all codec payload types ('ptime' SDP attribute). After this initial negotiation, no Re-INVITE messages are necessary as both endpoints are synchronized in terms of the other side's capabilities. If negotiation fails (i.e., no match was achieved for any of the transport capabilities), fallback to existing logic occurs (according to the parameter IsFaxUsed).

Below is an example of media descriptions of an SDP indicating support for V.152.

```
v=0
o=-  0 0 IN IPV4 <IPAdressA>
s=-
t=0 0
p=+1
c=IN IP4  <IPAddressA
m=audio <udpPort A> RTP/AVP 18 0
a=ptime:10
a=rtpmap:96 PCMU/8000
a=gpmd: 96 vbd=yes
```

In the example above, V.152 implementation is supported (using the dynamic payload type 96 and G.711 mu-law as the VBD codec) as well as the voice codecs G.711 mu-law and G.729.

Instead of using VBD transport mode, the V.152 implementation can use alternative relay fax transport methods (e.g., fax relay over IP using T.38). The preferred V.152 transport method is indicated by the SDP ‘pmft’ attribute. Omission of this attribute in the SDP content means that VBD mode is the preferred transport mechanism for voice-band data.

8.4 FXO Operating Modes

This section provides a description of the FXO operating modes and gateway configurations for Tel-to-IP and IP-to-Tel calls.

8.4.1 IP-to-Telephone Calls

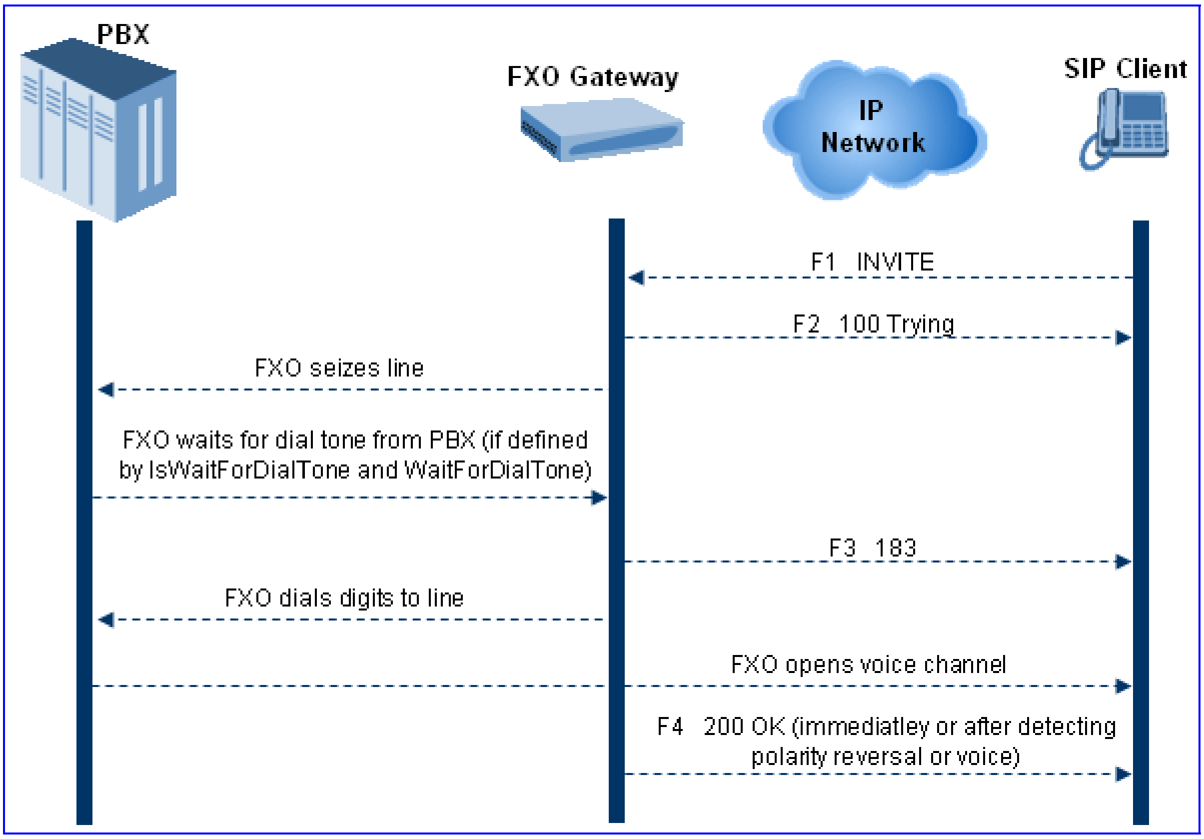
The FXO gateway provides the following FXO operating modes for IP-to-Tel calls:

- One-stage dialing
  - Waiting for dial tone
  - Time to wait before dialing
  - Answer supervision
- Two-stage dialing
- Dialing time
  - Disconnect supervision
  - DID wink

8.4.1.1 One-Stage Dialing

One-stage dialing is when the FXO gateway receives an IP-to-Tel call, off-hooks the PBX line connected to the telephone, and then immediately dials the destination telephone number. In other words, the IP caller doesn't dial the PSTN number upon hearing a dial tone.

Figure 8-1: Call Flow for One-Stage Dialing



One -stage dialing incorporates the following FXO functionality:

- **Waiting for Dial Tone**

  The Waiting for Dial Tone feature enables the gateway to dial the digits to the Tel side only after detecting a dial tone from the PBX line. The *ini* file parameter IsWaitForDialTone is used to configure this operation.

- **Time to Wait Before Dialing**

  The Time to Wait Before Waiting feature defines the time (in msec) between seizing the FXO line and starting to dial the digits. The *ini* file parameter WaitForDialTime is used to configure this operation.

Note: The *ini* file parameter IsWaitForDialTone must be disabled for this mode.

- **Answer Supervision**

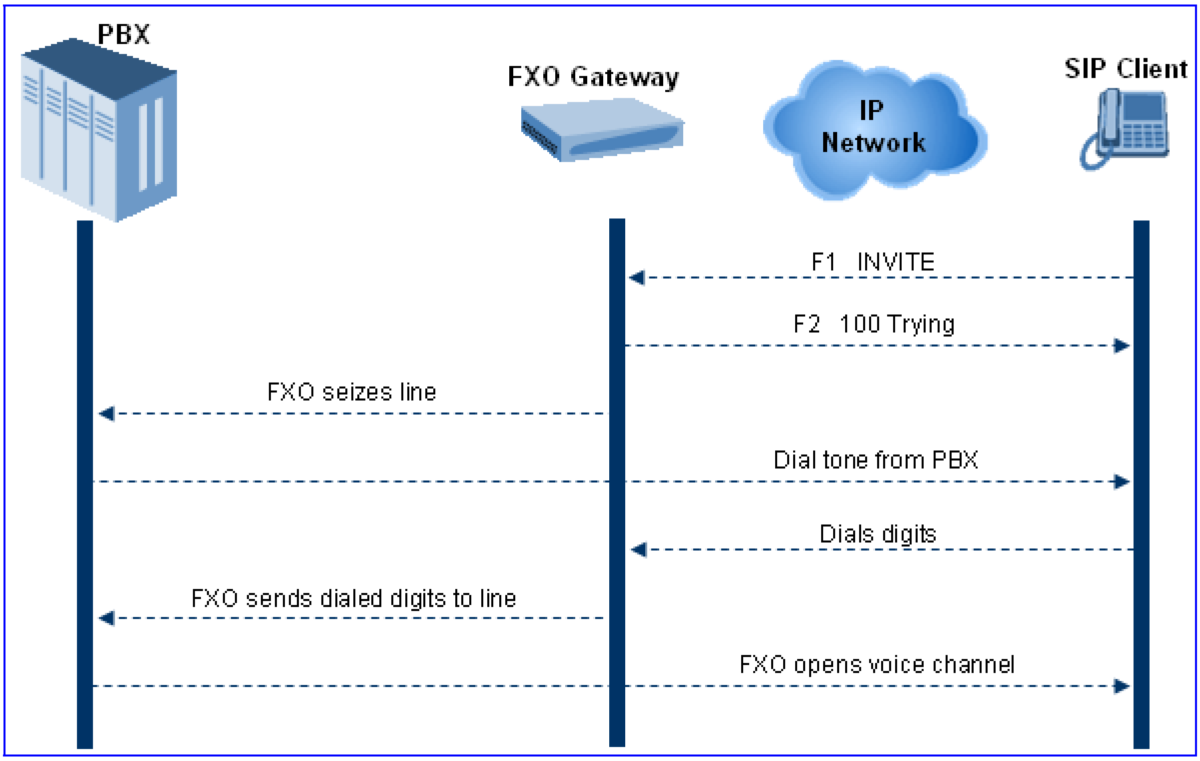
  The Answer Supervision feature enables the FXO gateway to determine when a call is connected, by using one of the following methods:

  - Polarity Reversal: the gateway sends a 200 OK in response to an INVITE only when it detects a polarity reversal.
  - Voice Detection: the gateway sends a 200 OK in response to an INVITE only when it detects the start of speech (or ringback tone) from the Tel side.

8.4.1.2 Two-Stage Dialing

Two-stage dialing is when the IP caller is required to dial twice. The caller initially dials to the FXO gateway, and only after receiving a dial tone from the PBX (via the FXO gateway), dials the destination telephone number.

Figure 8-2: Call Flow for Two-Stage Dialing



Two-stage dialing implements the Dialing Time feature. Dialing Time allows you to define the time that each digit can be separately dialed. By default, the overall dialing time per digit is 200 msec. The longer the telephone number, the greater the dialing time will be.

The relevant parameters for configuring Dialing Time include the following:

- DTMFDigitLength (100 msec): time for generating DTMF tones to the PSTN (PBX) side
- DTMFInterDigitInterval (100 msec): time between generated DTMF digits to PSTN (PBX) side

8.4.1.3 Call Termination (Disconnect Supervision) on the FXO Gateway

The FXO Disconnect Supervision enables the gateway's FXO ports to monitor call-progress tones from a PBX or from the PSTN. This allows the FXO to determine when the call has terminated on the PBX side, and thereby, preventing analog trunks (i.e., lines to the PBX) from getting "stuck" when the called phone hangs up.

The PBX doesn't disconnect the call, but instead signals to the gateway that the call is disconnected using one of the following methods:

- **Detection of polarity reversal / current disconnect**

  This is the recommended method. The call is immediately disconnected after polarity reversal or current disconnect is detected on the Tel side (assuming the PBX / CO produces this signal).

  Relevant parameters: EnableReversalPolarity, EnableCurrentDisconnect, CurrentDisconnectDuration, CurrentDisconnectDefaultThreshold, and TimeToSampleAnalogLineVoltage.

- **Detection of Reorder / Busy / Dial tones**

  The call is immediately disconnected after Reorder / Busy / Dial tone is detected on the Tel side (assuming the PBX / CO generates this tone). This method requires the correct tone frequencies and cadence to be defined in the Call Progress Tones file. If these frequencies are unknown, define them in the CPT file (the tone produced by the PBX / CO must be recorded and its frequencies analyzed -- refer to Section E.2.7 on page 374). This method is slightly less reliable than the previous one. You can use the CPTWizard (described in Section E.1.3 on page 366) to analyze Call Progress Tones generated by any PBX or telephone network.

  Relevant parameters: DisconnectOnBusyTone and DisconnectOnDialTone.

- **Detection of silence**

  The call is disconnected after silence is detected on both call directions for a specific (configurable) amount of time. The call isn't disconnected immediately; therefore, this method should only be used as a backup mode.

  Relevant parameters: EnableSilenceDisconnect and FarEndDisconnectSilencePeriod (with DSP templates number 2 or 3).

- **A special DTMF code**

  A digit pattern that when received from the Tel side, indicates to the gateway to disconnect the call.

  Relevant *ini* file parameter: TelDisconnectCode.

- **Interruption of RTP stream**

  Relevant parameters: BrokenConnectionEventTimeout and DisconnectOnBrokenConnection.

Note: This method operates correctly only if silence suppression is not used.

- **Protocol-based termination of the call from the IP side**

Note: The implemented disconnect method must be supported by the CO or PBX.

8.4.1.4 DID Wink

The gateway's FXO ports support Direct Inward Dialing (DID). DID is a service offered by telephone companies that enables callers to dial directly to an extension on a PBX without the assistance of an operator or automated call attendant. This service makes use of DID trunks, which forward only the last three to five digits of a phone number to the PBX. If, for example, a company has a PBX with extensions 555-1000 to 555-1999, and a caller dials 555-1234, the local central office (CO) would forward, for example, only 234 to the PBX. The PBX would then ring extension 234.

DID wink enables the originating end to seize the line by going off-hook. It waits for acknowledgement from the other end before sending digits. This serves as an integrity check that identifies a malfunctioning trunk and allows the network to send a re-order tone to the calling party.

The "start dial" signal is a wink from the PBX to the FXO gateway. The FXO then sends the last four to five DTMF digits of the called number. The PBX uses these digits to complete the routing directly to an internal station (telephone or equivalent)

- DID Wink can be used for connection to EIA/TIA-464B DID Loop Start lines
- Both FXO (detection) and FXS (generation) are supported

8.4.2 Telephone-to-IP Calls

The FXO gateway provides the following FXO operating modes for Tel-to-IP calls:

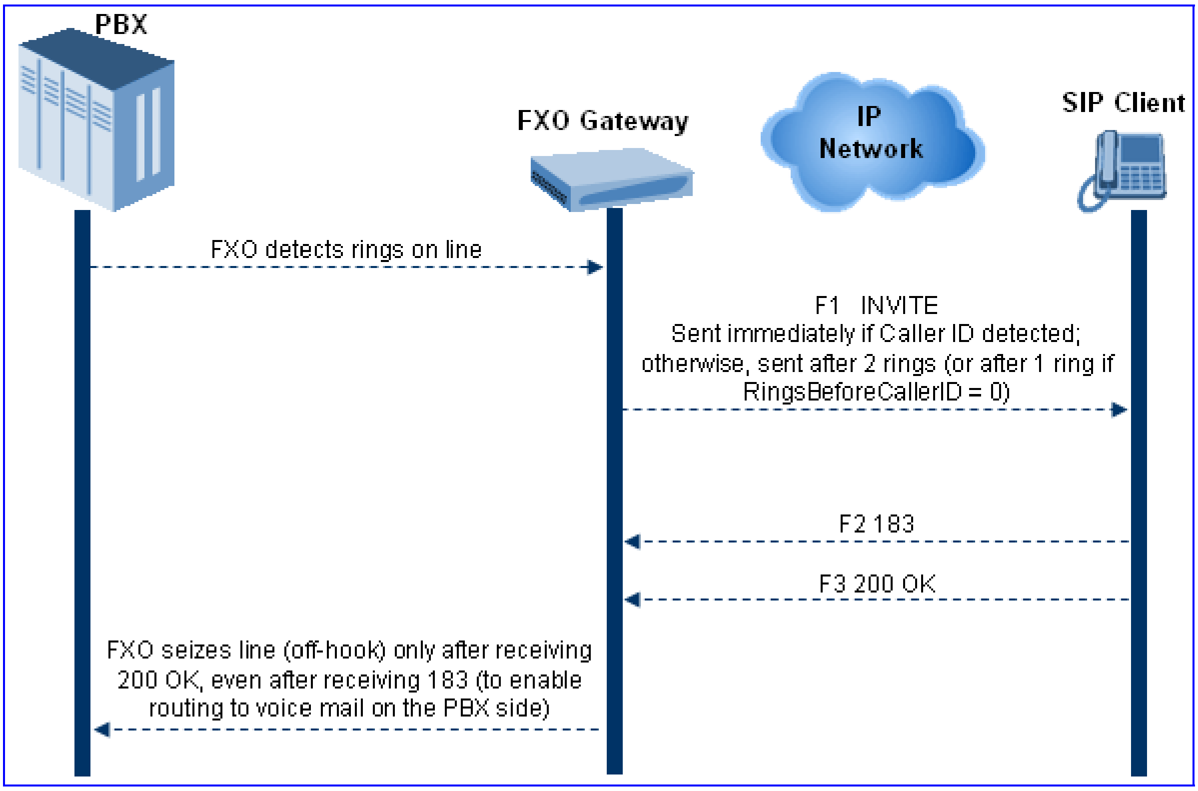
- Automatic Dialing
- Collecting Digits Mode
- Ring Detection Timeout
- FXO Supplementary Services
  - Hold/Transfer Toward the Tel side
  - Blind Transfer to the Tel side
  - Hold/Transfer Toward the IP side

8.4.2.1 Automatic Dialing

Automatic dialing is defined using the ini file parameter TargetOfChannelX (where 'X' is the channel number) or the embedded Web server's 'Automatic Dialing' screen (refer to Section 5.5.9.2 on page 120).

The SIP call flow diagram below illustrates Automatic Dialing.

Figure 8-3: Call Flow for Automatic Dialing

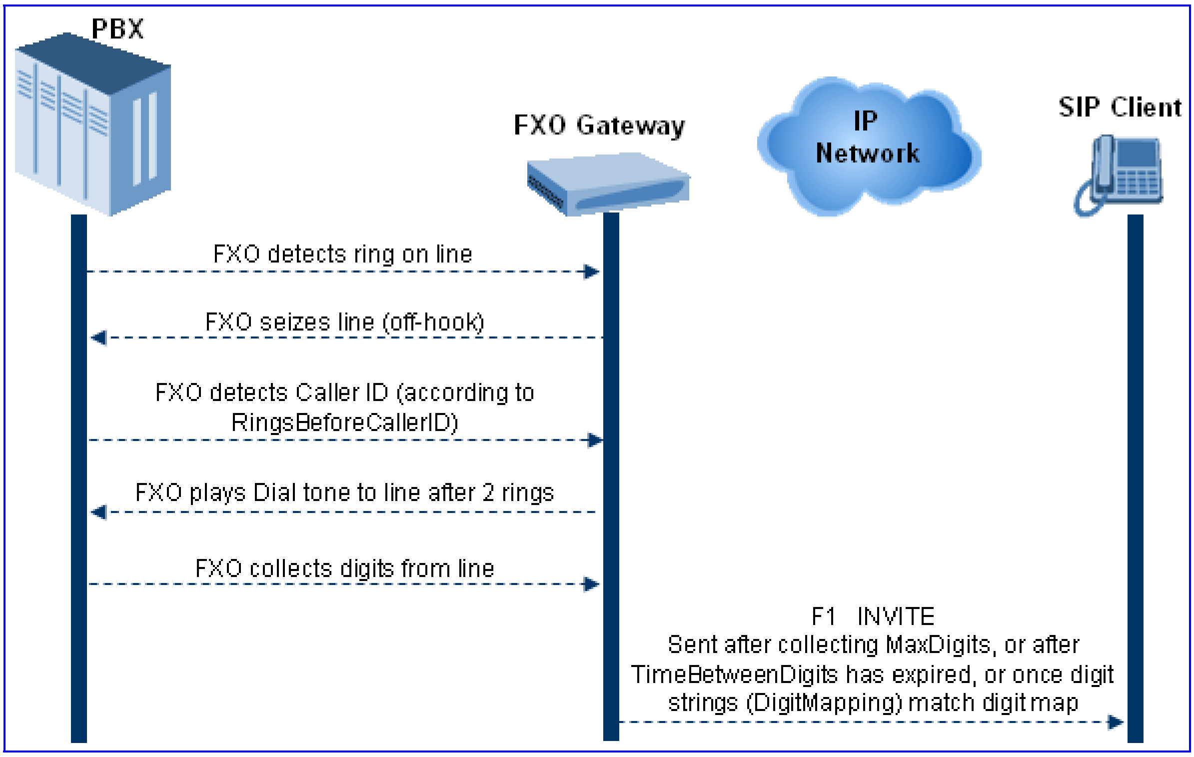


8.4.2.2 Collecting Digits Mode

When automatic dialing is not defined, the gateway collects the digits.

The SIP call flow diagram below illustrates the Collecting Digits Mode.

Figure 8-4: Call Flow for Collecting Digits Mode



8.4.2.3 Ring Detection Timeout

The *ini* file parameters IsWaitForDialTone and WaitForDialTone apply to Ring Detection Timeout. The operation of Ring Detection Timeout depends on the following:

- No automatic dialing and Caller ID is enabled: if the second ring signal doesn't arrive for Ring Detection Timeout, the gateway doesn't initiate a call to the IP.
- Automatic dialing is enabled: if the remote party doesn't answer the call, and the ringing signal stops for Ring Detection Timeout, the FXO releases the IP call.

Ring Detection Timeout supports full ring cycle of ring on and ring off (from ring start to ring start).

8.4.2.4 FXO Supplementary Services

- **Hold / Transfer toward the Tel side**

  The *ini* file parameter LineTransferMode must be set to 0 (default).

  If the FXO receives a hook-flash from the IP side (using out-of-band or RFC 2833), the gateway sends the hook-flash to the Tel side by one of the following:

  - Performing a hook flash (i.e., on-hook and off-hook)
  - Sending a hook-flash code (defined by the *ini* file parameter HookFlashCode)

  The PBX may generate a dial tone that is sent to the IP, and the IP side may dial digits of a new destination.

- **Blind Transfer to the Tel side**

  A blind transfer is one in which the transferring phone connects the caller to a destination line before ringback begins. The *ini* file parameter LineTransferMode must be set to 1.

  The blind transfer call process is as follows:

  - FXO receives a REFER request from the IP side
  - FXO sends a hook-flash to the PBX, dials the digits (that are received in the Refer-To header), and then drops the line (on-hook). Note that the time between flash to dial is according to the WaitForDialTime parameter.
  - PBX performs the transfer internally

- **Hold / Transfer toward the IP side**

  The FXO gateway doesn't initiate hold / transfer as a response to input from the Tel side. If the FXO receives a REFER request (with or without replaces), it generates a new INVITE according to the Refer-To header.

8.5 ThroughPacket™

The gateway supports a proprietary method to aggregate RTP streams from several channels to reduce the bandwidth overhead caused by the attached Ethernet, IP, UDP and RTP headers, and to reduce the packet / data transmission rate. This option reduces the load on network routers and can typically save 50% (e.g., for G.723) on IP bandwidth.

ThroughPacket™ is accomplished by aggregating payloads from several channels that are sent to the same destination IP address into a single IP packet.

ThroughPacket™ can be applied to the entire gateway or, using IP Profile, to specific IP destinations (refer to Section 5.5.6.3 on page 113). Note that ThroughPacket™ must be enabled on both gateways.

To enable ThroughPacket™ set the parameter 'RemoteBaseUDPPort' to a nonzero value. Note that the value of 'RemoteBaseUDPPort' on the local gateway must equal the value of 'BaseUDPPort' of the remote gateway. The gateway uses these parameters to identify and distribute the payloads from the received multiplexed IP packet to the relevant channels.

In ThroughPacket™ mode, the gateway uses a single UDP port for all incoming multiplexed packets and a different port for outgoing packets. These ports are configured using the parameters 'L1L1ComplexTxUDPPort' and 'L1L1ComplexRxUDPPort'.

When ThroughPacket™ is used, Call statistics aren't available (since there is no RTCP flow).

8.6 Dynamic Jitter Buffer Operation

Voice frames are transmitted at a fixed rate. If the frames arrive at the other end at the same rate, voice quality is perceived as good. In many cases, however, some frames can arrive slightly faster or slower than the other frames. This is called jitter (delay variation), and degrades the perceived voice quality. To minimize this problem, the gateway uses a jitter buffer. The jitter buffer collects voice packets, stores them and sends them to the voice processor in evenly spaced intervals.

The MediaPack uses a dynamic jitter buffer that can be configured using two parameters:

- Minimum delay, 'DJBufMinDelay' (0 msec to 150 msec):
  Defines the starting jitter capacity of the buffer. For example, at 0 msec, there is no buffering at the start. At the default level of 10 msec, the gateway always buffers incoming packets by at least 10 msec worth of voice frames.
- Optimization Factor, 'DJBufOptFactor' (0 to 12, 13):
  Defines how the jitter buffer tracks to changing network conditions. When set at its maximum value of 12, the dynamic buffer aggressively tracks changes in delay (based on packet loss statistics) to increase the size of the buffer and doesn't decays back down. This results in the best packet error performance, but at the cost of extra delay. At the minimum value of 0, the buffer tracks delays only to compensate for clock drift and quickly decays back to the minimum level. This optimizes the delay performance but at the expense of a higher error rate.

The default settings of 10 msec Minimum delay and 10 Optimization Factor should provide a good compromise between delay and error rate. The jitter buffer 'holds' incoming packets for 10 msec before making them available for decoding into voice. The coder polls frames from the buffer at regular intervals in order to produce continuous speech. As long as delays in the network do not change (jitter) by more than 10 msec from one packet to the next, there is always a sample in the buffer for the coder to use. If there is more than 10 msec of delay at any time during the call, the packet arrives too late. The coder tries to access a frame and is not able to find one. The coder must produce a voice sample even if a frame is not available. It therefore compensates for the missing packet by adding a Bad-Frame-Interpolation (BFI) packet. This loss is then flagged as the buffer being too small. The dynamic algorithm then causes the size of the buffer to increase for the next voice session. The size of the buffer may decrease again if the gateway notices that the buffer is not filling up as much as expected. At no time does the buffer decrease to less than the minimum size configured by the Minimum delay parameter.

Special Optimization Factor Value: 13

One of the purposes of the Jitter Buffer mechanism is to compensate for clock drift. If the two sides of the VoIP call are not synchronized to the same clock source, one RTP source generates packets at a lower rate, causing under-runs at the remote Jitter Buffer. In normal operation (optimization factor 0 to 12), the Jitter Buffer mechanism detects and compensates for the clock drift by occasionally dropping a voice packet or by adding a BFI packet.

Fax and modem devices are sensitive to small packet losses or to added BFI packets. Therefore to achieve better performance during modem and fax calls, the Optimization Factor should be set to 13. In this special mode the clock drift correction is performed less frequently - only when the Jitter Buffer is completely empty or completely full. When such condition occurs, the correction is performed by dropping several voice packets simultaneously or by adding several BFI packets simultaneously, so that the Jitter Buffer returns to its normal condition.

8.7 Configuring the Gateway's Alternative Routing (based on Connectivity and QoS)

The Alternative Routing feature enables reliable routing of Tel to IP calls when a Proxy isn't used. The MediaPack gateway periodically checks the availability of connectivity and suitable Quality of Service (QoS) before routing. If the expected quality cannot be achieved, an alternative IP route for the prefix (phone number) is selected.

8.7.1 Alternative Routing Mechanism

When a Tel→IP call is routed through the MediaPack gateway, the call's destination number is compared to the list of prefixes defined in the Tel to IP Routing table (described in Section 5.5.5.2 on page 100). The Tel to IP Routing table is scanned for the destination number's prefix starting at the top of the table. When an appropriate entry (destination number matches one of the prefixes) is found; the prefix's corresponding destination IP address is checked. If the destination IP address is disallowed, an alternative route is searched for in the following table entries.

Destination IP address is disallowed if no ping to the destination is available (ping is continuously initiated every 7 seconds), when an inappropriate level of QoS was detected, or when DNS host name is not resolved. The QoS level is calculated according to delay or packet loss of previously ended calls. If no call statistics are received for two minutes, the QoS information is reset.

The MediaPack gateway matches the rules starting at the top of the table. For this reason, enter the main IP route above any alternative route.

8.7.2 Determining the Availability of Destination IP Addresses

To determine the availability of each destination IP address (or host name) in the routing table, one (or all) of the following (configurable) methods are applied:

- **Connectivity:** The destination IP address is queried periodically (currently only by ping).
- **QoS:** The QoS of an IP connection is determined according to RTCP statistics of previous calls. Network delay (in msec) and network packet loss (in percentage) are separately quantified and compared to a certain (configurable) threshold. If the calculated amounts (of delay or packet loss) exceed these thresholds the IP connection is disallowed.
- **DNS resolution:** When host name is used (instead of IP address) for the destination route, it is resolved to an IP address by a DNS server. Connectivity and QoS are then applied to the resolved IP address.

8.7.3 Relevant Parameters

The following parameters (described in Table 5-16) are used to configure the Alternative Routing mechanism:

- AltRoutingTel2IPEnable
- AltRoutingTel2IPMode
- IPConnQoSMaxAllowedPL
- IPConnQoSMaxAllowedDelay

8.8 Mapping PSTN Release Cause to SIP Response

The MediaPack FXO gateway is used to interoperate between the SIP network and the PSTN/PBX. This interoperability includes the mapping of PSTN/PBX Call Progress Tones to SIP 4xx or 5xx responses for IP→Tel calls. The converse is also true: For Tel→IP calls, the SIP 4xx or 5xx responses are mapped to tones played to the PSTN/PBX.

When establishing an IP→Tel call the following rules are applied:

If the remote party (PSTN/PBX) is busy and the FXO gateway detects a Busy tone, it sends 486 busy to IP. If it detects a Reorder tone, it sends 404 not found (no route to destination) to IP. In both cases the call is released. Note that if 'DisconnectOnBusyTone = 0' the FXO gateway ignores the detection of Busy/Reorder tones and doesn't release the call.

For all other MediaPack FXS/FXO release types (caused when there are no free channels in the specific hunt group, or when an appropriate rule for routing the call to a hunt group doesn't exist, or if the phone number isn't found), the MediaPack sends SIP response (to IP) according to the parameter 'DefaultReleaseCause'. This parameter defines Q.931 release causes. Its default value is '3', which is mapped to SIP 404 response. By changing its value to '34' SIP 503 response is sent. Other causes can be used as well.

8.9 Call Detail Record

The Call Detail Record (CDR) contains vital statistic information on calls made by the gateway. CDRs are generated at the end and (optionally) at the beginning of each call (determined by the parameter 'CDRReportLevel') and sent to a Syslog server. The destination IP address for CDR logs is determined by the parameter 'CDRSyslogServerIP'.

The table below lists the CDR fields that are supported.

Table 8-1: Supported CDR Fields (continues on pages 232 to 233)

Field Name	Description
Cid	Port Number
CallId	H.323/SIP Call Identifier
Trunk	N/A
BChan	N/A
ConId	H.323/SIP Conference ID
TG	Trunk Group Number
EPTyp	Endpoint Type
Orig	Call Originator (IP, Tel)
SourceIp	Source IP Address
DestIp	Destination IP Address
TON	Source Phone Number Type
NPI	Source Phone Number Plan
SrcPhoneNum	Source Phone Number
SrcNumBeforeMap	Source Number Before Manipulation
TON	Destination Phone Number Type
NPI	Destination Phone Number Plan
DstPhoneNum	Destination Phone Number
DstNumBeforeMap	Destination Number Before Manipulation
Durat	Call Duration
Coder	Selected Coder
Intrv	Packet Interval
RtpIp	RTP IP Address

Table 8-1: Supported CDR Fields (continues on pages 232 to 233)

Field Name	Description
Port	Remote RTP Port
TrmSd	Initiator of Call Release (IP, Tel, Unknown)
TrmReason	Termination Reason
Fax	Fax Transaction during the Call
InPackets	Number of Incoming Packets
OutPackets	Number of Outgoing Packets
PackLoss	Number of Incoming Lost Packets
UniqueId	unique RTP ID
SetupTime	Call Setup Time
ConnectTime	Call Connect Time
ReleaseTime	Call Release Time
RTPdelay	RTP Delay
RTPjitter	RTP Jitter
RTPssrc	Local RTP SSRC
RemoteRTPssrc	Remote RTP SSRC
RedirectReason	Redirect Reason
TON	Redirection Phone Number Type
NPI	Redirection Phone Number Plan
RedirectPhonNum	Redirection Phone Number

8.10 Supported RADIUS Attributes

Use Table 8-2 below for explanations on the RADIUS attributes contained in the communication packets transmitted between the MediaPack and a RADIUS Server.

Table 8-2: Supported RADIUS Attributes (continues on pages 233 to 235)

Attribute Number	Attribute Name	VSA No.	Purpose	Value Format	Sample	AAA[1]
Request Attributes						
1	User-Name		Account number or calling party number or blank	String up to 15 digits long	5421385747	Start Acc Stop Acc
4	NAS-IP-Address		IP address of the requesting MediaPack	Numeric	192.168.14.43	Start Acc Stop Acc
6	Service-Type		Type of service requested	Numeric	1: login	Start Acc Stop Acc
26	h323-incoming-conf-id	1	H.323/SIP call identifier	Up to 32 octets		Start Acc Stop Acc
26	h323-remote-address	23	IP address of the remote gateway	Numeric		Stop Acc
26	h323-conf-id	24	H.323/SIP call identifier	Up to 32		Start Acc

[1] The values in column 'AAA' are as follows:
'Start Acc' - Start Accounting
'Stop Acc' - Stop Accounting

Table 8-2: Supported RADIUS Attributes (continues on pages 233 to 235)

Attribute Number	Attribute Name	VSA No.	Purpose	Value Format	Sample	AAA[1]
				octets		Stop Acc
26	h323-setup-time	25	Setup time in NTP format 1	String		Start Acc Stop Acc
26	h323-call-origin	26	The call's originator: Answering (IP) or Originator (PSTN)	String	Answer, Originate etc	Start Acc Stop Acc
26	h323-call-type	27	Protocol type or family used on this leg of the call	String	VoIP	Start Acc Stop Acc
26	h323-connect-time	28	Connect time in NTP format	String		Stop Acc
26	h323-disconnect-time	29	Disconnect time in NTP format	String		Stop Acc
26	h323-disconnect-cause	30	Q.931 disconnect cause code	Numeric		Stop Acc
26	h323-gw-id	33	Name of the gateway	String	SIPIDString	Start Acc Stop Acc
30	Called-Station-Id			String	8004567145	Start Acc
			Destination phone number	String	2427456425	Stop Acc
31	Calling-Station-Id		Calling Party Number (ANI)	String	5135672127	Start Acc Stop Acc
40	Acct-Status-Type		Account Request Type (start or stop) **Note:** 'start' isn't supported on the Calling Card application.	Numeric	1: start, 2: stop	Start Acc Stop Acc
41	Acct-Delay-Time		No. of seconds tried in sending a particular record	Numeric	5	Start Acc Stop Acc
42	Acct-Input-Octets		Number of octets received for that call duration	Numeric		Stop Acc
43	Acct-Output-Octets		Number of octets sent for that call duration	Numeric		Stop Acc
44	Acct-Session-Id		A unique accounting identifier - match start & stop	String	34832	Start Acc Stop Acc
46	Acct-Session-Time		For how many seconds the user received the service	Numeric		Stop Acc
47	Acct-Input-Packets		Number of packets received during the call	Numeric		Stop Acc
48	Acct-Output-Packets		Number of packets sent during the call	Numeric		Stop Acc
61	NAS-Port-Type		MediaPack physical port type on which the call is active	String	0: Asynchronous	Start Acc Stop Acc
Response Attributes						
26	h323-return-code	103	The reason for failing authentication (0 = ok, other number failed)	Numeric	0 Request accepted	Stop Acc
44	Acct-Session-Id		A unique accounting identifier – match start & stop	String		Stop Acc

8.10.1 RADIUS Server Messages

In Figure 8-5 below, non-standard parameters are preceded with brackets.

Figure 8-5: Accounting Example

```
Accounting-Request (361)
user-name = 111
acct-session-id = 1
nas-ip-address = 212.179.22.213
nas-port-type = 0
acct-status-type = 2
acct-input-octets = 4841
acct-output-octets = 8800
acct-session-time = 1
acct-input-packets = 122
acct-output-packets = 220
called-station-id = 201
calling-station-id = 202
// Accounting non-standard parameters:
(4923 33) h323-gw-id =
(4923 23) h323-remote-address = 212.179.22.214
(4923 1) h323-ivr-out = h323-incoming-conf-id:02102944 600a1899 3fd61009 0e2f3cc5
(4923 30) h323-disconnect-cause = 22 (0x16)
(4923 27) h323-call-type = VOIP
(4923 26) h323-call-origin = Originate
(4923 24) h323-conf-id = 02102944 600a1899 3fd61009 0e2f3cc5
```

8.11 Proxy or Registrar Registration Example

The REGISTER message is sent to the Registrar's IP address (if configured) or to the Proxy's IP address. The message is sent per gateway or per gateway endpoint according to the 'AuthenticationMode' parameter. Usually the FXS gateways are registered per gateway port, while FXO gateways send a single registration message, where Username is used instead of phone number in From/To headers. The registration request is resent according to the parameter 'RegistrartionTimeDivider'. For example, if 'RegistrationTimeDivider = 70' (%) and Registration Expires time = 3600, the gateway resends its registration request after 3600 x 70% = 2520 sec. The default value of 'RegistrartionTimeDivider' is 50%.

```
REGISTER sip:servername 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
```

The 'servername' string is defined according to the following rules:

- The 'servername' is equal to 'RegistrarName' if configured. The 'RegistrarName' can be any string.
- Otherwise, the 'servername' is equal to 'RegistrarIP' (either FQDN or numerical IP address), if configured.
- Otherwise the 'servername' is equal to 'ProxyName' if configured. The 'ProxyName' can be any string.
- Otherwise the 'servername' is equal to 'ProxyIP' (either FQDN or numerical IP address).

The 'sipgatewayname' parameter (defined in the *ini* file or set from the Web browser), can be any string. Some Proxy servers require that the 'sipgatewayname' (in REGISTER messages) is set equal to the Registrar/Proxy IP address or to the Registrar/Proxy domain name.

8.12 Configuration Examples

8.12.1 Establishing a Call between Two Gateways

After you've installed and set up the MediaPack, you can ensure that it functions as expected by establishing a call between it and another gateway. This section exemplifies how to configure two 8-port MediaPack FXS SIP gateways to establish a call. After configuration, you can make calls between telephones connected to a single MediaPack gateway or between the two MediaPack gateways.

In the following example, the IP address of the first gateway is 10.2.37.10 and its endpoint numbers are 101 to 108. The IP address of the second gateway is 10.2.37.20 and its endpoint numbers are 201 to 208.

In this example, a SIP Proxy is not used. Internal call routing is performed using the internal 'Tel to IP Routing' table.

➢ To configure the two gateways, take these 4 steps:

1. For the *first* MediaPack gateway (10.2.37.10), in the 'Endpoint Phone Numbers' screen, assign the phone numbers 101 to 108 for the gateway's endpoints.

Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Profile ID
1 FXS	1-8	101		

2. For the *second* MediaPack gateway (10.2.37.20), in the 'Endpoint Phone Numbers' screen, assign the phone numbers 201 to 208 for the gateway's endpoints.

Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Profile ID
1 FXS	1-8	201		

3. Configure the following settings for *both* gateways:

   In the 'Tel to IP Routing' screen, in the first row, enter 10 in the 'Destination Phone Prefix' field and enter the IP address of the first gateway (10.2.37.10) in the field 'IP Address'. In the second row, enter 20 and the IP address of the second gateway (10.2.37.20) respectively. These settings enable the routing (from both gateways) of outgoing Tel→IP calls that start with 10 to the first gateway and calls that start with 20 to the second gateway.

	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Profile ID	Status
1	10	*	10.2.37.10	0	n/a
2	20	*	10.2.37.20	0	n/a

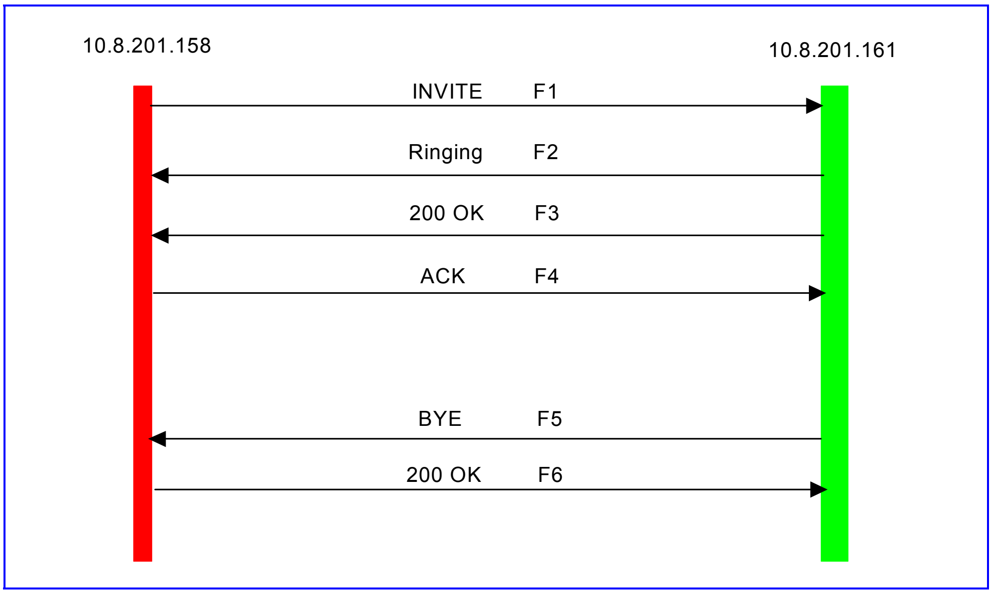
4. Make a call. Pick up the phone connected to port #1 of the first MediaPack and dial 102 (to the phone connected to port #2 of the same gateway). Listen out for progress tones at the calling endpoint and for ringing tone at the called endpoint. Answer the called endpoint, speak into the calling endpoint, and check the voice quality. Dial 201 from the phone connected to port #1 of the first MediaPack gateway; the phone connected to port #1 of the second MediaPack rings. Answer the call and check the voice quality.

8.12.2 SIP Call Flow

The following Call Flow describes SIP messages exchanged between two MediaPack gateways during simple call.

Telephone ‘6000’ dials ‘2000’, sending INVITE message to Gateway 10.8.201.161.

Figure 8-6: SIP Call Flow



F1 10.8.201.158 ==> 10.8.201.161 INVITE

```
INVITE sip:6000@10.8.201.161;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.201.158;branch=z9hG4bKacolwbzYF
From: <sip:2000@10.8.201.158>;tag=1c3535
To: <sip:6000@10.8.201.161>
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
CSeq: 20214 INVITE
Contact: <sip:2000@10.8.201.158;user=phone>
User-Agent: Audiocodes-Sip-Gateway/MP-118 FXS/v.4.20.299.410
Supported: 100rel,em
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO
Content-Type: application/sdp
Content-Length: 208

v=0
s=Phone-Call
t=0 0
o=AudiocodesGW 87943 43401 IN IP4 10.8.201.158
c=IN IP4 10.8.201.158
m=audio 6000 RTP/AVP 8 96
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F2 10.8.201.161 ==> 10.8.201.158 180 RINGING

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.8.201.158;branch=z9hG4bKacolwbzYF
From: <sip:2000@10.8.201.158>;tag=1c3535
To: <sip:6000@10.8.201.161>;tag=1c29715
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
Server: Audiocodes-Sip-Gateway/MP-118 FXS/v.4.20.299.410
CSeq: 20214 INVITE
Supported: 100rel,em
Content-Length: 0
```

Note: Phone ‘2000’ answers the call, and sends 200 OK message to gateway 10.8.201.158.

F3 10.8.201.161 ==> 10.8.201.158 200 OK

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.8.201.158;branch=z9hG4bKacolwbzYF
From: <sip:2000@10.8.201.158>;tag=1c3535
To: <sip:6000@10.8.201.161>;tag=1c29715
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
CSeq: 20214 INVITE
Contact: <sip:6000@10.8.201.161;user=phone>
Server: Audiocodes-Sip-Gateway/MP-118 FXS/v.4.20.299.410
Supported: 100rel,em
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO
Content-Type: application/sdp
Content-Length: 208

v=0
s=Phone-Call
t=0 0
o=AudiocodesGW 30762 37542 IN IP4 10.8.201.161
c=IN IP4 10.8.201.161
m=audio 4040 RTP/AVP 8 96
a=rtpmap:8 pcma/8000
a=ptime:20
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
```

F4 10.8.201.158 ==> 10.8.201.161 ACK

```
ACK sip:6000@10.8.201.161;user=phone;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.201.158;branch=z9hG4bKachoWSQxD
From: <sip:2000@10.8.201.158>;tag=1c3535
To: <sip:6000@10.8.201.161>;tag=1c29715
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
User-Agent: Audiocodes-Sip-Gateway/MP-118 FXS/v.4.20.299.410
CSeq: 20214 ACK
Supported: 100rel,em
Content-Length: 0
```

Note: Phone ‘6000’ goes onhook, gateway 10.8.201.161 sends BYE to gateway 10.8.201.158. Voice path is established.

F5 10.8.201.161 ==> 10.8.201.158 BYE

```
BYE sip:2000@10.8.201.158;user=phone;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.201.161;branch=z9hG4bKacLBzZgmA
From: <sip:6000@10.8.201.161>;tag=1c29715
To: <sip:2000@10.8.201.158>;tag=1c3535
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
User-Agent: Audiocodes-Sip-Gateway/MP-118 FXS/v.4.20.299.410
CSeq: 34541 BYE
Supported: 100rel,em
Content-Length: 0
```

F6 10.8.201.158 ==> 10.8.201.161 200 OK

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.8.201.161;branch=z9hG4bKacLBzZgmA
From: <sip:6000@10.8.201.161>;tag=1c29715
To: <sip:2000@10.8.201.158>;tag=1c3535
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
Server: Audiocodes-Sip-Gateway/MP-118 FXS/v.4.20.299.410
CSeq: 34541 BYE
Supported: 100rel,em
Content-Length: 0
```

8.12.3 SIP Authentication Example

MediaPack gateways support basic and digest (MD5) authentication types, according to SIP RFC 3261 standard. A proxy server might require authentication before forwarding an INVITE message. A Registrar/Proxy 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 sent to Registrar/Proxy 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
User-Agent: Audiocodes-Sip-Gateway/MP-118 FXS/v.4.20.299.410
CSeq: 1 REGISTER
Contact: sip:122@10.1.1.200:
Expires:3600
```

On receiving this request the Registrar/Proxy 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 gateway to be able to 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
Server: Audiocodes-Sip-Gateway/MP-118 FXS/v.4.20.299.410
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 returns a 200 OK 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
```

8.12.4 Remote IP Extension between FXO and FXS

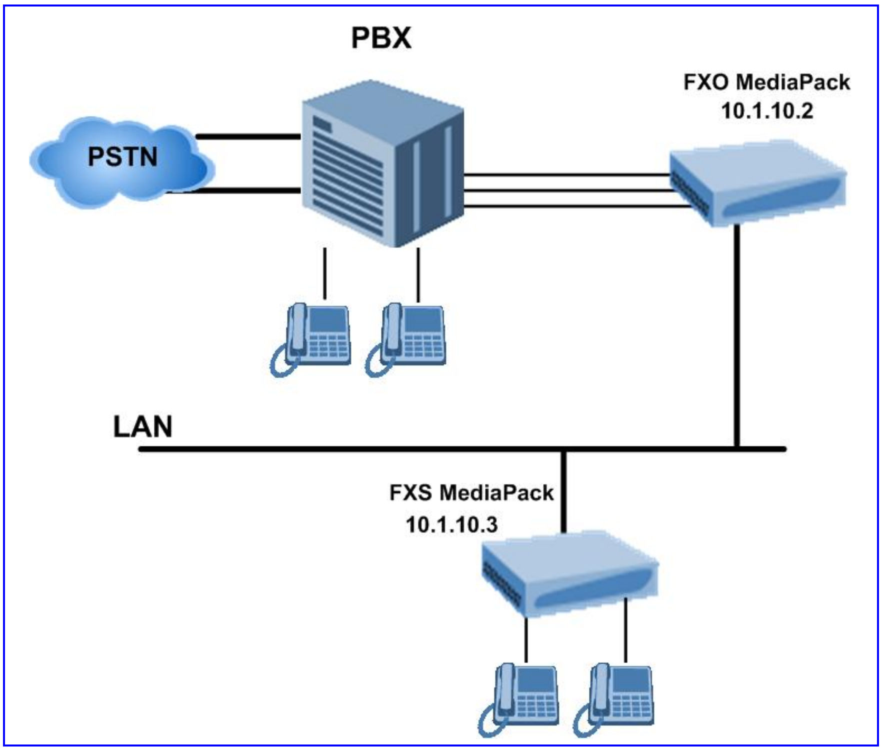
This application explains how to implement remote extension via IP, using 8-port FXO and 8-port FXS MediaPack gateways. In this configuration, PBX incoming calls are routed to the ‘Remote Extension’ via the FXO and FXS gateways.

Requirements:

- One FXO MediaPack gateway
- One FXS MediaPack gateway
- Analog phones (POTS)
- PBX – one or more PBX loop start lines
- LAN

Connect the FXO MediaPack ports directly to the PBX lines as shown in the diagram below:

Figure 8-7: MediaPack FXS & FXO Remote IP Extension



8.12.4.1 Dialing from Remote Extension (Phone connected to FXS)

➢ **To configure the call, take these 6 steps:**

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

8.12.4.2 Dialing from other PBX line, or from PSTN

➢ To configure the call, take these 5 steps:

1. Dial the PBX subscriber number the same way as if the user's phone was connected directly to PBX.
2. When the PBX rings the FXO MediaPack, the ring signal is immediately 'sent' to the phone connected to the FXS MediaPack.
3. Once the phone's handset, connected to the FXS MediaPack is raised, the FXO MediaPack 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 FXS MediaPack ports. Each PBX line is routed to the same phone (connected to the FXS MediaPack).
5. The call is disconnected when the phone connected to FXS MediaPack goes onhook.

8.12.4.3 FXS MediaPack Configuration (using the Embedded Web Server)

➢ To configure the FXS MediaPack, take these 3 steps:

1. In the 'Endpoint Phone Numbers' screen, assign the phone numbers 100 to 107 for the gateway's endpoints.

Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Profile ID
1 FXS	1-8	100		0

2. In the 'Automatic Dialing' screen, enter the phone numbers of the FXO MediaPack gateway in the 'Destination Phone Number' fields. When a phone connected to port #1 goes offhook, the FXS gateway automatically dials the number '200'.

Automatic Dialing

Gateway Port	Destination Phone Number	Auto Dial Status
Port 1 FXS	200	Enable
Port 2 FXS	201	Enable
Port 3 FXS	202	Enable
Port 4 FXS	203	Enable
Port 5 FXS	204	Enable
Port 6 FXS	205	Enable
Port 7 FXS	206	Enable
Port 8 FXS	207	Enable

3. In the 'Tel to IP Routing' screen, enter 20 in the 'Destination Phone Prefix' field, and the IP address of the FXO MediaPack gateway (10.1.10.2) in the field 'IP Address'.

	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Profile ID	Status
1	20	*	10.1.10.2	0	n/a

Note: In remote extensions, for the transfer to function, hold must be disabled on the FXS (i.e., Enable Hold = 0).

8.12.4.4 FXO MediaPack Configuration (using the Embedded Web Server)

➢ To configure the FXO MediaPack, take these 4 steps:

1. In the 'Endpoint Phone Numbers' screen, assign the phone numbers 200 to 207 for the gateway's endpoints.

Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Profile ID
1 FXO	1-8	200		0

2. In the 'Automatic Dialing' screen, enter the phone numbers of the FXS MediaPack gateway in the 'Destination Phone Number' fields. When a ringing signal is detected at port #1, the FXO gateway automatically dials the number '100'.

Automatic Dialing

Gateway Port	Destination Phone Number	Auto Dial Status
Port 1 FXO	100	Enable
Port 2 FXO	101	Enable
Port 3 FXO	102	Enable
Port 4 FXO	103	Enable
Port 5 FXO	104	Enable
Port 6 FXO	105	Enable
Port 7 FXO	106	Enable
Port 8 FXO	107	Enable

3. In the 'Tel to IP Routing' screen, enter 10 in the 'Destination Phone Prefix' field, and the IP address of the FXS MediaPack gateway (10.1.10.3) in the field 'IP Address'.

	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Profile ID	Status
1	10	*	10.1.10.3	0	n/a

4. In the 'Protocol Management' screen, set the parameter 'Dialing Mode' to 'Two Stage' (IsTwoStageDial=1).

Reader's Notes

9 Networking Capabilities

9.1 Ethernet Interface Configuration

Using the parameter ‘EthernetPhyConfiguration‘, users can control the Ethernet connection mode.

Either the manual modes (10 Base-T Half-Duplex, 10 Base-T Full-Duplex, 100 Base-TX Half-Duplex, 100 Base-TX Full-Duplex) or Auto-Negotiate mode can be used.

Auto-Negotiation falls back to Half-Duplex mode when the opposite port is not Auto-Negotiate, but the speed (10 Base-T, 100 Base-TX) in this mode is always configured correctly. Note that configuring the gateway to Auto-Negotiate mode while the opposite port is set manually to Full-Duplex (either 10 Base-T or 100 Base-TX) is invalid (as it causes the gateway to fall back to Half-Duplex mode while the opposite port is Full-Duplex). It is also invalid to set the gateway to one of the manual modes while the opposite port is either Auto-Negotiate or not exactly matching (both in speed and in duplex mode). Users are encouraged to always prefer Full-Duplex connections to Half-Duplex ones and 100 Base-TX to 10 Base-T (due to the larger bandwidth). It is strongly recommended to use the same mode in both link partners. Any mismatch configuration can yield unexpected functioning of the Ethernet connection.

Note that when remote configuration is performed, the gateway should be in the correct Ethernet setting prior to the time this parameter takes effect. When, for example, the gateway is configured using BootP/TFTP, the gateway must perform many Ethernet-based transactions prior to reading the *ini* file containing this gateway configuration parameter.

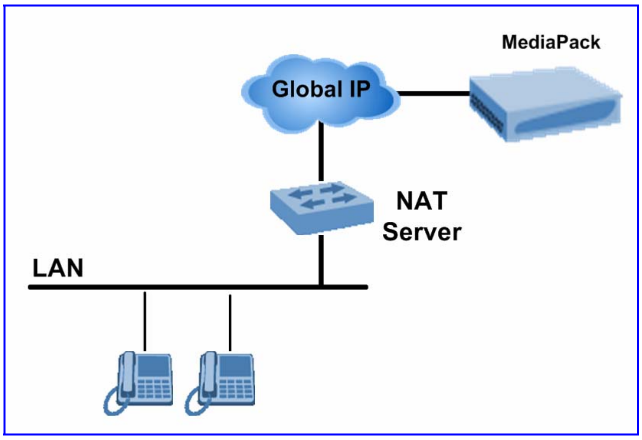
To work around this problem, the gateway always uses the last Ethernet setup mode configured. This way, if users want to configure the gateway to work in a new network environment in which the current Ethernet setting of the gateway is invalid, they should first modify this parameter in the current network so that the new setting holds next time gateway is restarted. After reconfiguration has completed, connect the gateway to the new network and restart it. As a result, the remote configuration process that takes place in the new network uses a valid Ethernet configuration.

9.2 NAT (Network Address Translation) Support

NAT is a mechanism that maps a set of internal IP addresses used within a private network to global IP addresses, providing transparent routing to end hosts. The primary advantages of NAT are (1) Reduction in the number of global IP addresses required in a private network (global IP addresses are only used to connect to the Internet); (2) Better network security by hiding its internal architecture.

Figure 9-1 below illustrates the NAT architecture.

Figure 9-1: NAT Functioning



The way SIP is designed creates a problem for VoIP traffic to pass through NAT. SIP uses IP addresses and port numbers in its message body. The NAT server can't modify SIP messages and therefore, can't change local to global addresses.

Two different streams traverse through NAT: signaling and media. A gateway (located behind a NAT) that initiates a signaling path will have problems in receiving incoming signaling responses (they will be blocked by the NAT). Furthermore, the initiating gateway must notify the receiving gateway where to send the media to.

To solve these problems the following mechanisms are available:

- STUN (refer to Section 9.2.1).
- First Incoming Packet Mechanism (refer to Section 9.2.2 on page 249)
- RTP No-Op packets according to the avt-rtp-noop draft (refer to Section 9.2.3 on page 249).
- For SNMP NAT traversal, refer to Section 14.10 on page 322.

9.2.1 STUN

Simple Traversal of UDP through NATs (STUN) (according to RFC 3489) is a client / server protocol that solves most of the NAT traversal problems. The STUN server operates in the public Internet and the STUN clients are embedded in end-devices (located behind NAT). STUN is used both for the signaling and the media streams. STUN works with many existing NAT types, and does not require any special behavior from them.

STUN enables the gateway to discover the presence (and types) of NATs and firewalls located between it and the public Internet. It provides the gateway with the capability to determine the public IP address and port allocated to it by the NAT. This information is later embedded in outgoing SIP/SDP messages and enables remote SIP user agents to reach the gateway. It also discovers the binding lifetime of the NAT (the refresh rate necessary to keep NAT 'Pinholes' open).

On startup, the gateway sends a STUN Binding Request. The information received in the STUN Binding Response (IP address:port) is used for SIP signaling. This information is updated every NATBindingDefaultTimeout.

At the beginning of each call, if STUN is needed (i.e., not an internal NAT call), the media ports of the call are mapped. The call is delayed until the STUN Binding Response (that includes a global IP:port) for each media (RTP, RTCP and T.38) is received.

To enable STUN:

- Set the parameter EnableSTUN to 1.
- Define the STUN server's address by performing one of the following:
  - Define the STUN server's address by assigning it a domain name. The STUN client can perform the required SRV query to resolve it to an IP address and port, sort the server list, and use the servers according to the sorted list.
  - Determine the IP address of the primary and (optionally) the secondary STUN servers using the parameters STUNServerPrimaryIP and STUNServerSecondaryIP. If the primary STUN server isn't available, the gateway tries to communicate with the secondary server.
- Use the parameter NATBindingDefaultTimeout to determine the default NAT binding lifetime in seconds. STUN is used to refresh the binding information after this time expires.

Notes:

- STUN only applies to UDP (doesn't support TCP and TLS).
- STUN can't be used when the gateway is located behind a symmetric NAT.
- For defining the STUN server's address, either use the stunServerPrimaryIpAddress or the StunServerDomainName method, with priority to the first one.

9.2.2 First Incoming Packet Mechanism

If the remote gateway resides behind a NAT device, it's possible that the MediaPack can activate the RTP/RTCP/T.38 streams to an invalid IP address / UDP port. To avoid such cases, the MediaPack automatically compares the source address of the incoming RTP/RTCP/T.38 stream with the IP address and UDP port of the remote gateway. If the two are not identical, the transmitter modifies the sending address to correspond with the address of the incoming stream. The RTP, RTCP and T.38 can thus have independent destination IP addresses and UDP ports.

Users can choose to disable the NAT mechanism by setting the *ini* file parameter 'DisableNAT' to 1. The two parameters 'EnableIpAddrTranslation' and 'EnableUdpPortTranslation' enable users to specify the type of compare operation that takes place on the first incoming packet. To compare only the IP address, set 'EnableIpAddrTranslation = 1' and 'EnableUdpPortTranslation = 0'. In this case, if the first incoming packet arrives with only a difference in the UDP port, the sending addresses won't change. If both the IP address and UDP port need to be compared, then both parameters need to be set to 1.

9.2.3 No-Op Packets

The gateway's No-Op packet support can be used to verify Real-Time Transport Protocol (RTP) and T.38 connectivity, and to keep NAT bindings and Firewall pinholes open. No-Op packets are available for sending in RTP and T.38 formats.

Users can control the activation of No-Op packets by using the *ini* file parameter NoOperationSendingMode. If No-Op packet transmission is activated, users can control the time interval in which No-Op packets are sent in the case of silence (i.e., no RTP or T.38 traffic). This is performed using the NoOpInterval *ini* parameter.

Note: Receipt of No-Op is always supported.

- **RTP No-OP**

  The RTP No-Op support complies with IETF's draft-wing-avt-rtp-noop-03.txt (titled 'A No-Op Payload Format for RTP'). This IETF document defines a No-Op payload format for RTP.

  The draft defines the RTP payload type as dynamic. Users can control the payload type with which the No-Op packets are sent. This is performed using the RTPNoOpPayloadType *ini* parameter. AudioCodes' default payload type is 120.

- **T.38 No-Op**

  T.38 No-Op packets are sent only while a T.38 session is activated. Sent packets are a duplication of the previously sent frame (including duplication of the sequence number).

9.3 IP Multicasting

The gateway supports IP Multicasting level 1 according to RFC 2236 (i.e. IGMP version 2) for RTP channels. The gateway is capable of transmitting and receiving Multicast packets.

9.4 Point-to-Point Protocol over Ethernet (PPPoE)

PPPoE is a method of sending the Point-to-Point Protocol over Ethernet network.

9.4.1 Point-to-Point Protocol (PPP) Overview

Point-to-Point Protocol (PPP) provides a method of transmitting data over serial point-to-point links. The protocol defines establishing, configuring and testing the data link connection and the network protocol.

The PPP standard describes a state machine used to establish a valid connection between two hosts over a serial connection. There are three major stages described, helping to establish a network layer (such as an IP) connection over the point-to-point link: LCP (Link Configuration Protocol), Authentication, and NCP (Network Control Protocol). Once the network protocol is configured, the two hosts can communicate, sending network layer protocol (such as IP) over the PPP connection (a small PPP header is added at the beginning of each packet).

At the initial phase, the hosts use LCP (link configuration protocol) to negotiate for link characteristic and parameters. Packets sent in this phase have two octets of ‘PPP header’ followed by LCP message with variable length. Various parameters and options are negotiable at this phase, including MRU (maximum receive unit), Authentication Protocol, and others.

Once the link is established (each side sends a ‘configure ack’ message to the other side), the authentication phase may begin. The authentication phase is not mandatory. However, it is negotiated in the link configuration phase. A host may ask other hosts for authentication using Password Authentication Protocol (PAP) or Challenge Handshake Authentication Protocol (CHAP).

The PAP sends the username and password to the remote host unencrypted.

The CHAP is a more sophisticated method of authentication. The two hosts share a ‘secret’. The authenticator sends a ‘challenge’ to the host requesting authentication. The host performs a calculation (one-way hash) using the challenge received from the authenticator and the shared ‘secret’, and sends the result to the authenticator. The authenticator verifies the host if the result of the calculation is correct; otherwise it is rejected.

The last configuration phase, immediately after the authentication phase (or after the Link Configuration) is the Network Control Protocol. There is a family of control protocols for establishing and configuring different network-layer protocols, for example, IPCP (PPP Internet Protocol Control Protocol), IPv6CP (PPP IP v6 Control Protocol), and BCP (PPP Bridging Control Protocol). Each of them handles and manages the specific needs required by their respective network-layer protocol.

When working in an IP network, IPCP is used as the Network Configuration Protocol. The IPCP is used to configure the network layer of the hosts, requesting/declaring on IP Addresses.

Further information on PPP Protocol is available on the IETF website (http://www.ietf.org/rfc/rfc1661.txt). Further information on Password Authentication Protocol is available on the IETF website (http://www.ietf.org/rfc/rfc1334.txt). Further information on Challenge Handshake Authentication Protocol is available on the IETF website (http://www.ietf.org/rfc/rfc1994.txt). Further information on PPP Internet Protocol Control Protocol (IPCP) is available on the IETF website (http://www.ietf.org/rfc/rfc1332.txt).

9.4.2 PPPoE Overview

PPPoE is a method of sending the Point-to-Point Protocol over Ethernet network. PPPoE provides the ability to connect a network of hosts over a simple bridging access device to a remote Access Concentrator. Access control, billing and type of service can be done on a per-user, rather than a per-site, basis.

A common use of the PPPoE is in the ADSL market: The home PC is connected to a modem via Ethernet, and the PC uses the PPPoE to 'simulate' as if it was directly connected to the remote host on a point-to-point connection.

Since PPPoE frames are sent over Ethernet, each PPP session must learn the Ethernet address of the remote peer, as well as establish a unique session identifier. The PPPoE standard describes a discovery protocol that provides this. A PPPoE session begins with a discovery phase. Only after this discovery is completed can the PPP state machine start (with LCP, Authentication etc, as described above).

Each of the Ethernet frames carrying PPP session has a standard Ethernet header followed by PPPoE header, and is sent with the remote host Ethernet MAC address (except for the very first one, in the discovery phase, which is broadcasted to all hosts).

Further information on the transmission of PPPoE is available on the IETF website (http://www.ietf.org/rfc/rfc2516.txt).

9.4.3 PPPoE in AudioCodes Gateways

The AudioCodes gateway contains a PPPoE client embedded in its software. When correctly configured (see *ini* file parameters) the gateway can try to connect to a remote PPPoE Access Concentrator.

When restarting the gateway after several BOOTP attempts, if PPPoE is enabled (see *ini* file parameter EnablePPPoE), the gateway tries to initiate a PPP session.

The gateway initiates a PPPoE discovery phase to discover a PPPoE Access Concentrator. It does this by broadcasting a discovery initialization packet (PADI). If an Access Concentrator exists and replies, the gateway tries to connect to this Access Concentrator. If this initial connection succeeds, then the PPP LCP phase starts - each side of the PPPoE connection sends LCP configuration requests to configure the PPP link.

The gateway PPPoE client supports both PAP and CHAP authentications. The type of authentication protocol used is according to the request from the authentication server. In the LCP configuration phase, the server requires a specific authentication (none, PAP, or CHAP are supported). The *ini* file parameters PPPoEUserName, PPPoEPassword, and PPPoEServerName are used to configure the authentication parameters. If the Access Concentrator is configured to operate in PAP, the PPPoEUserName and PPPoEPassword are used as Username and Password (in this case, the PPPoEServerName parameter is not used). If the Access Concentrator is configured to operate in CHAP, the PPPoEUserName parameter functions as Client Name (sent in the CHAP response packet), while the PPPoEPassword functions as the shared secret (calculated along with the challenge to produce the response). In this case, the PPPoEServerName is the name of the server. Some hosts can be configured to authenticate to multiple servers. In such hosts, the server name is used to identify the "secret" that should be used.

Note: The AudioCodes gateway, being a PPPoE client, requests no authentication.

After the gateway has been authenticated, it needs to configure a network layer protocol. The gateway uses the IP protocol. Therefore, the used NCP will be IPCP (IP Configuration Protocol). In this phase, if the *ini* file parameter PPPoEStaticIPAddress is defined, the gateway requests the remote host to assign this address for its use.

When working in a PPPoE environment, the gateway negotiates for its IP address (as described above). However, if the user desires to disable the PPPoE client, the gateway can be configured to use default values for IP address, subnet mask and default gateway. This can be done using *ini* file parameters PPPoERecoverIPAddress, PPPoERecoverSubnetMask and PPPoERecoverDfgwAddress. These parameters indicate to the gateway that if the PPPoE is disabled and no BOOTP server is activated, as required in the gateway to use a PPPoE environment, then the gateway should use these defaults for its IP configuration.

For a detailed description of the *ini* file parameters for PPPoE, refer to Section 5.6.1.6 on page 149.

Note: When working with a PPPoE server (Access Concentrator) that does not reply to LCP Echo messages (which by default, the gateway periodically sends) you may want to disable the LCP Echo messages by using the *ini* file parameter PPPoELcpEchoEnable. (For a description of this parameter, refer to Section 5.6.1.6 on page 149.)

9.5 Robust Reception of RTP Streams

This mechanism filters out unwanted RTP streams that are sent to the same port number on the gateway. These multiple RTP streams can result from traces of previous calls, call control errors and deliberate attacks.

When more than one RTP stream reaches the gateway on the same port number, the gateway accepts only one of the RTP streams and rejects the rest of the streams. The RTP stream is selected according to the following procedure:

The first packet arriving on a newly opened channel sets the source IP address and UDP port from which further packets are received. Thus, the source IP address and UDP port identify the currently accepted stream. If a new packet arrives whose source IP address or UDP port are different to the currently accepted RTP stream, there are two options:

- The new packet has a source IP address and UDP port which are the same as the remote IP address and UDP port that were stated during the opening of the channel. In this case, the gateway reverts to this new RTP stream.
- The new packet has any other source IP address and UDP port, in which case the packet is dropped.

9.6 Multiple Routers Support

Multiple routers support is designed to assist the media gateway when it operates in a multiple routers network. The gateway learns the network topology by responding to ICMP redirections and caches them as routing rules (with expiration time).

When a set of routers operating within the same subnet serve as gateways to that network and intercommunicate using a dynamic routing protocol, the routers can determine the shortest path to a certain destination and signal the remote host the existence of the better route. Using multiple router support the media gateway can utilize these router messages to change its next hop and establish the best path.

Note: Multiple Routers support is an integral feature that doesn't require configuration.

9.7 Simple Network Time Protocol Support

Simple Network Time Protocol (SNTP) client functionality generates requests and reacts to the resulting responses using the NTP version 3 protocol definitions (according to RFC 1305). Through these requests and responses, the NTP client is able to synchronize the system time to a time source within the network, thereby eliminating any potential issues should the local system clock 'drift' during operation. By synchronizing time to a network time source, traffic handling, maintenance, and debugging actions become simplified for the network administrator.

The NTP client follows a simple process in managing system time; the NTP client requests an NTP update, receives an NTP response, and updates the local system clock based on a configured NTP server within the network.

The client requests a time update from a specified NTP server at a specified update interval. In most situations this update interval should be every 24 hours based on when the system was restarted. The NTP server identity (as an IP address) and the update interval are configurable parameters that can be specified either in the *ini* file (NTPServerIP, NTPUpdateInterval respectively) or via an SNMP MIB object.

When the client receives a response to its request from the identified NTP server it must be interpreted based on time zone, or location, offset that the system is to a standard point of reference called the Universal Time Coordinate (UTC). The time offset that the NTP client should use is a configurable parameter that can be specified either in the *ini* file (NTPServerUTCOffset) or via an SNMP MIB object.

If required, the clock update is performed by the client as the final step of the update process. The update is done in such a way as to be transparent to the end users. For instance, the response of the server may indicate that the clock is running too fast on the client. The client slowly robs bits from the clock counter in order to update the clock to the correct time. If the clock is running too slow, then in an effort to catch the clock up, bits are added to the counter, causing the clock to update quicker and catch up to the correct time. The advantage of this method is that it does not introduce any disparity in the system time, that is noticeable to an end user, or that could corrupt call timeouts and timestamps.

9.8 IP QoS via Differentiated Services (DiffServ)

DiffServ is architecture providing different types or levels of service for IP traffic. DiffServ (according to RFC 2474) offers the capability to prioritize certain traffic types, depending on their priority, thereby accomplishing a higher-level QoS at the expense of other traffic types. By prioritizing packets, DiffServ routers can minimize transmission delays for time-sensitive packets such as VoIP packets.

The MediaPack can be configured to set a different DiffServ value to IP packets according to their class-of-service (Network, Premium Media, Premium Control, Gold and Bronze).

For the mapping of an application to its class-of-service, refer to Table 9-1 on page 255.

The DiffServ parameters are described in Table 5-36.

9.9 VLANS and Multiple IPs

9.9.1 Multiple IPs

Media, Control, and Management (OAM) traffic in the gateway can be assigned one of the following IP addressing schemes:

- Single IP address for all traffic (i.e., Media, Control, and OAM).
- Separate IP address for each traffic type.

  For separate IP addresses, the different traffic types are separated into three dedicated networks. Instead of a single IP address, the gateway is assigned three IP addresses and subnet masks, each relating to a different traffic type. This architecture enables users to integrate the gateway into a three-network environment that is focused on security and segregation. Each entity in the gateway (e.g., Web and RTP) is mapped to a single traffic type (according to Table 9-1 on page 255) in which it operates.
- Two separate IP addresses (Dual IP mode)--one for a specific traffic type and the other for a combination of two traffic types.

  In Dual IP mode, the gateway is assigned two IP addresses for the different traffic types. One IP address is assigned to a combination of two traffic types (Media and Control, OAM and Control, or OAM and Media), while the other IP address is assigned to whichever traffic type that is not included in this combination. For example, a typical scenario using this mode would include one IP address assigned for Control and OAM, and another IP address assigned for Media.

For detailed information on integrating the MediaPack into a VLAN and multiple IPs network, refer to Section 9.9.3 on page 256. For detailed information on configuring the multiple IP parameters, refer to Section 5.6.1.1 on page 138.

Notes:

- A default gateway is supported only for the Media traffic type; for the other two, use the IP Routing table.
- The IP address and subnet mask used in the Single IP Network mode are carried over to the OAM traffic type in the Multiple IP Network mode.

9.9.2 IEEE 802.1p/Q (VLANs and Priority)

The Virtual Local Area Network (VLAN) mechanism enables the MediaPack to be integrated into a VLAN-aware environment that includes switches, routers and endpoints.

When in VLAN-enabled mode, each packet is tagged with values that specify its priority (class-of-service) (IEEE 802.1p) and the identifier (traffic type) of the VLAN to which it belongs (media, control or management) (IEEE 802.1Q).

The class-of-service mechanism can be utilized to accomplish Ethernet QoS. Packets sent by the MediaPack to the Ethernet network are divided into five, different-priority classes (Network, Premium media, Premium control, Gold and Bronze). The priority of each class is determined by a corresponding *ini* file parameter.

Traffic type tagging can be used to implement Layer 2 VLAN security. By discriminating traffic into separate and independent domains, the information is preserved within the VLAN. Incoming packets received from an incorrect VLAN are discarded.

For the mapping of an application to its class-of-service and traffic type, refer to Table 9-1 below.

Media traffic type is assigned ‘Premium media’ class of service, Management traffic type is assigned ‘Bronze’ class of service, and Control traffic type is assigned ‘Premium control’ class of service.

For example, RTP/RTCP traffic is assigned the Media VLAN ID and ‘Premium media’ class of service, whereas Web traffic is assigned the Management VLAN ID and ‘Bronze’ class of service. Each of these parameters can be configured with a 802.1p/q value: traffic type to VLAN ID, and class of service to 802.1p priority.

Notes:

- As a safety measure, the VLAN mechanism is activated only when the gateway is loaded from the flash memory. Therefore, when using BootP:
  Load an *ini* file with ‘VlanMode = 1’ and ‘SaveConfiguration = 1’. Then (after the gateway is active) reset the gateway with TFTP disabled, or by using any method except for BootP.
- The gateway must be connected to a VLAN-aware switch, and the switch’s PVID must be equal to the gateway’s native VLAN ID.

For information on how to configure VLAN parameters, refer to Section 5.6.1.5 on page 147.

Table 9-1: Traffic / Network Types and Priority (continues on pages 255 to 255)

Application	Traffic / Network Types	Class-of-Service (Priority)
Debugging interface	Management	Bronze
Telnet	Management	Bronze
DHCP	Management	Network
Web server (HTTP)	Management	Bronze
SNMP GET/SET	Management	Bronze
Web server (HTTPS)	Management	Bronze
IPSec IKE	Determined by the service	Determined by the service
RTP traffic	Media	Premium media
RTCP traffic	Media	Premium media
T.38 traffic	Media	Premium media
SIP	Control	Premium control
SIP over TLS (SIPS)	Control	Premium control
Syslog	Management	Bronze
ICMP	Management	Determined by the initiator of the request
ARP listener	Determined by the initiator of the request	Network
SNMP Traps	Management	Bronze
DNS client	EnableDNSasOAM	Network
NTP	EnableNTPasOAM	Depends on the traffic type: Control: Premium control Management: Bronze
NFS	NFSServers_VlanType in the NFSServers table	Gold

9.9.2.1 Operation

- **Outgoing packets (from the gateway to the switch):**
All outgoing packets are tagged, each according to its interface (control, media or OAM). If the gateway's native ID is identical to one of the other IDs (usually to the OAM ID), this ID (e.g., OAM) is set to zero on outgoing packets (VlanSendNonTaggedOnNative = 0). This method is called Priority Tagging (p tag without Q tag). If the parameter VlanSendNonTaggedOnNative is set to 1, the gateway sends regular packets (with no VLAN tag).
- **Incoming packets (from the switch to the gateway):**
The switch sends all packets intended for the gateway (according to the switch's configuration) to the gateway without altering them. For packets whose VLAN ID is identical to the switch's PVID, the switch removes the tag and sends a packet.

The gateway only accepts packets that have a VLAN ID identical to one of its interfaces (control, media or OAM). Packets with a VLAN ID that is 0 or packets without a tag are accepted only if the gateway's native VLAN ID is identical to the VLAN ID of one of its interfaces. In this case, the packets are sent to the relevant interface. All other packets are rejected.

9.9.3 Getting Started with VLANS and Multiple IPs

By default the MediaPack operates without VLANs and multiple IPs, using a single IP address, subnet mask and default gateway IP address. This section provides an example of the configuration required to integrate the MediaPack into a VLAN and multiple IPs network using the Embedded Web Server (refer to Section 9.9.3.1 on page 257) and *ini* file (refer to Section 9.9.3.2 on page 259). Table 9-2 below shows an example configuration that is implemented in the following sections.

Table 9-2: Example of VLAN and Multiple IPs Configuration

Network Type	IP Address	Subnet Mask	Default Gateway IP Address	VLAN ID	External Routing Rule
OAM	10.31.174.50	255.255.0.0	0.0.0.0	4	83.4.87.X
Control	10.32.174.50	255.255.0.0	0.0.0.0	5	130.33.4.6
Media	10.33.174.50	255.255.0.0	10.33.0.1	6	--

Note that since a default gateway is available only for the Media network, for the MediaPack to be able to communicate with an external device / network on its OAM and Control networks, IP routing rules must be used.

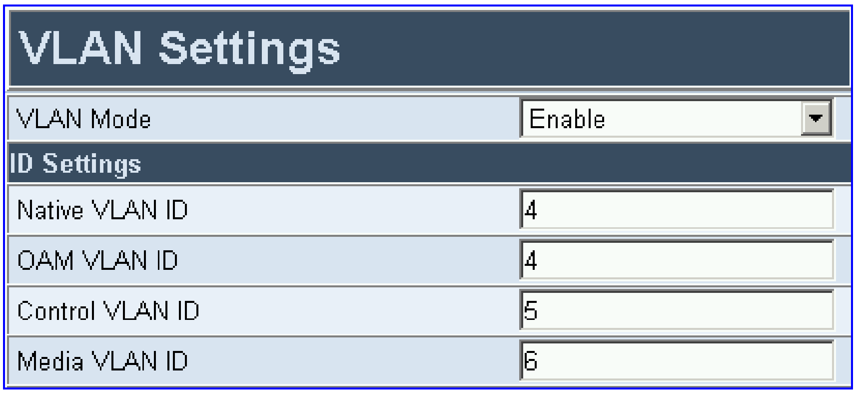
Note: The values provided in Sections 9.9.3.1 and 9.9.3.2 are sample parameter values only and are to be replaced with actual values appropriate to your system.

9.9.3.1 Integrating Using the Embedded Web Server

➢ **To integrate the MediaPack into a VLAN and multiple IPs network using the Embedded Web Server, take these 7 steps:**

1. Access the Embedded Web Server (Section 5.3 on page 51).
2. Use the Software Upgrade Wizard (Section 5.9.1 on page 197) to load and *burn* the firmware version to the MediaPack (VLANs and multiple IPs support is available only when the firmware is burned to flash).
3. Configure the VLAN parameters by completing the following steps:
   - Open the 'VLAN Settings' screen (**Advanced Configuration** menu > **Network Settings** > **VLAN Settings** option); the 'VLAN Settings' screen is displayed.
   - Modify the VLAN parameters to correspond to the values shown in Figure 9-2 below.

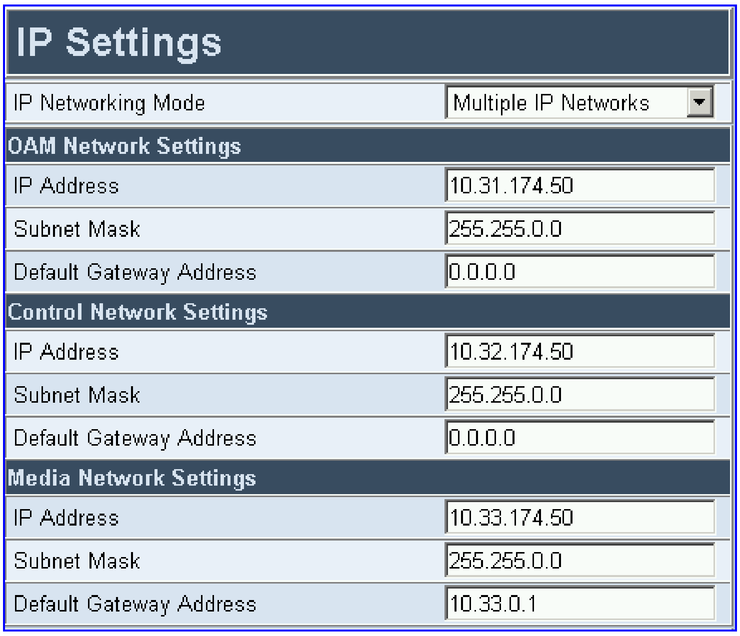
Figure 9-2: Example of the VLAN Settings Screen



   - Click the **Submit** button to save your changes.
4. Configure the multiple IP parameters by completing the following steps:
   - Open the 'IP Settings' screen (**Advanced Configuration** menu > **Network Settings** > **IP Settings** option); the 'IP Settings' screen is displayed.
   - Modify the IP parameters to correspond to the values shown in Figure 9-3. Note that the OAM, Control and Media Network Settings parameters appear only after you select the options 'Multiple IP Networks' or 'Dual IP' in the field 'IP Networking Mode'.

Note: Configure the OAM parameters only if the OAM networking parameters are different from the networking parameters used in the Single IP Network mode.

Figure 9-3: Example of the IP Settings Screen

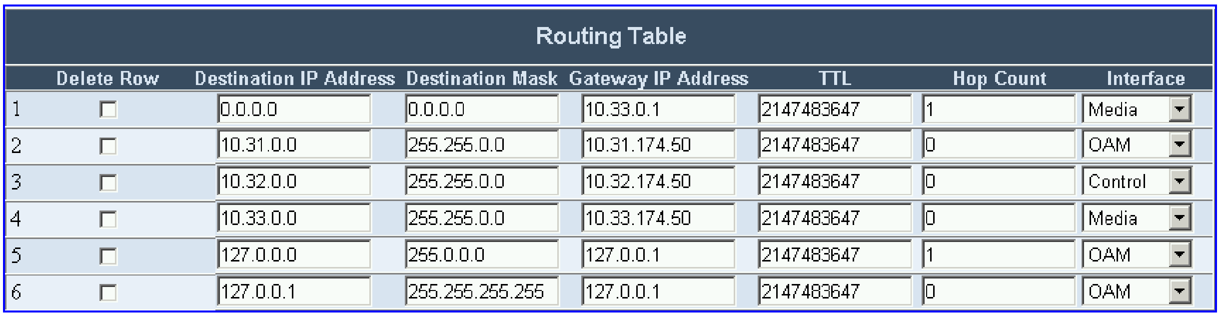


IP Settings	
IP Networking Mode	Multiple IP Networks
OAM Network Settings	
IP Address	10.31.174.50
Subnet Mask	255.255.0.0
Default Gateway Address	0.0.0.0
Control Network Settings	
IP Address	10.32.174.50
Subnet Mask	255.255.0.0
Default Gateway Address	0.0.0.0
Media Network Settings	
IP Address	10.33.174.50
Subnet Mask	255.255.0.0
Default Gateway Address	10.33.0.1

- Click the **Submit** button to save your changes.

5. Configure the IP Routing table by completing the following steps (the IP Routing table is required to define static routing rules for the OAM and Control networks since a default gateway isn't supported for these networks):
   - Open the 'IP Routing Table' screen (**Advanced Configuration** menu > **Network Settings** > **IP Routing Table** option); the 'IP Routing Table' screen is displayed.

Figure 9-4: Example of the IP Routing Table Screen



Routing Table

	Delete Row	Destination IP Address	Destination Mask	Gateway IP Address	TTL	Hop Count	Interface
1		0.0.0.0	0.0.0.0	10.33.0.1	2147483647	1	Media
2		10.31.0.0	255.255.0.0	10.31.174.50	2147483647	0	OAM
3		10.32.0.0	255.255.0.0	10.32.174.50	2147483647	0	Control
4		10.33.0.0	255.255.0.0	10.33.174.50	2147483647	0	Media
5		127.0.0.0	255.0.0.0	127.0.0.1	2147483647	1	OAM
6		127.0.0.1	255.255.255.255	127.0.0.1	2147483647	0	OAM

   - Use the 'Add a new table entry' pane to add the routing rules shown in Table 9-3 below.

Table 9-3: Example of IP Routing Table Configuration

Destination IP Address	Destination Mask	Gateway IP Address	Hop Count	Network Type
130.33.4.6	255.255.255.255	10.32.0.1	20	Control
83.4.87.6	255.255.255.0	10.31.0.1	20	OAM

   - Click the **Submit** button to save your changes.

6. Save your changes to flash so they are available after a power fail, refer to Section 5.10.2 on page 205.
7. Reset the gateway (refer to Section 5.10.3 on page 206).

9.9.3.2 Integrating Using the *ini* File

➢ **To integrate the MediaPack into a VLAN and multiple IPs network using the *ini* file, take these 3 steps:**

1. Prepare an *ini* file with parameters shown in Figure 6-1 (refer to the following notes):
   - If the BootP/TFTP utility and the OAM interface are located in the same network, the Native VLAN ID (VlanNativeVlanId) must be equal to the OAM VLAN ID (VlanOamVlanId), which in turn must be equal to the PVID of the switch port the gateway is connected to. Therefore, set the PVID of the switch port to 4 (in this example).
   - Configure the OAM parameters (LocalOAMPAddress, LocalOAMSubnetMask and LocalOAMDefaultGW) only if the OAM networking parameters are different from the networking parameters used in the Single IP Network mode.
   - The IP Routing table is required to define static routing rules for the OAM and Control networks since a default gateway isn't supported for these networks.

Figure 9-5: Example of VLAN and Multiple IPs *ini* File Parameters

```
; VLAN Configuration
VlanMode=1
VlanOamVlanId=4
VlanNativeVlanId=4
VlanControlVlanId=5
VlanMediaVlanID=6

; Multiple IPs Configuration
EnableMultipleIPs=1
LocalMediaIPAddress=10.33.174.50
LocalMediaSubnetMask=255.255.0.0
LocalMediaDefaultGW=10.33.0.1
LocalControlIPAddress=10.32.174.50
LocalControlSubnetMask=255.255.0.0
LocalControlDefaultGW=0.0.0.0
LocalOAMPAddress=10.31.174.50
LocalOAMSubnetMask=255.255.0.0
LocalOAMDefaultGW=0.0.0.0

; IP Routing table parameters
RoutingTableDestinationsColumn = 130.33.4.6, 83.4.87.6
RoutingTableDestinationMasksColumn = 255.255.255.255 , 255.255.255.0
RoutingTableGatewaysColumn = 10.32.0.1 , 10.31.0.1
RoutingTableInterfacesColumn = 1 , 0
RoutingTableHopsCountColumn = 20,20
```

2. Use the BootP/TFTP utility (Section C.6 on page 350) to load and *burn* (-fb option) the firmware version and the *ini* file you prepared in the previous step to the MediaPack (VLANs and multiple IPs support is available only when the firmware is burned to flash).
3. Reset the MediaPack after disabling it on the BootP/TFTP utility.

Reader's Notes

10 Advanced System Capabilities

10.1 Restoring Networking Parameters to their Initial State

You can use the 'Reset' button to restore the MediaPack networking parameters to their factory default values (described in Table 4-1) and to reset the username and password.

Note that the MediaPack returns to the software version burned in flash. This process also restores the MediaPack parameters to their factory settings. Therefore, you must load your previously backed-up *ini* file, or the default *ini* file (received with the software kit) to set them to their correct values.

➢ To restore the networking parameters of the MP-11x to their initial state, take these 4 steps:

1. Press in the 'Reset' button uninterruptedly for a duration of more than six seconds; the gateway is restored to its factory settings (username: 'Admin', password: 'Admin').
2. Assign the MP-11x IP address (refer to Section 4.2 on page 41).
3. Load your previously backed-up *ini* file, or the default *ini* file (received with the software kit). To load the *ini* file via the Embedded Web Server, refer to Section 5.6.3 on page 165.
4. Press again on the 'Reset' button (this time for a short period).

➢ To restore the networking parameters of the MP-124 to their initial state, take these 6 steps:

1. Disconnect the MP-124 from the power and network cables.
2. Reconnect the power cable; the gateway is powered up. After approximately 45 seconds the **Ready** LED turns to green and the **Control** LED blinks for about 3 seconds.
3. While the **Control** LED is blinking, press shortly on the reset button (located on the left side of the front panel); the gateway resets a second time and is restored with factory default parameters (username: 'Admin', password: 'Admin').
4. Reconnect the network cable.
5. Assign the MP-124 IP address (refer to Section 4.2 on page 41).
6. Load your previously backed-up *ini* file, or the default *ini* file (received with the software kit). To load the *ini* file via the Embedded Web Server, refer to Section 5.6.3 on page 165.

10.2 Establishing a Serial Communications Link with the MediaPack

Use serial communication software (e.g., HyperTerminal™) to establish a serial communications link with the MediaPack via the RS-232 connection. You can use this link to change the networking parameters (Section 4.2.4 on page 44) and to receive error / notification messages.

➢ To establish a serial communications link with the MediaPack via the RS-232 port, take these 2 steps:

1. Connect the RS-232 port to your PC (For the MP-124, refer to Section 3.2.4.1 on page 40. For the MP-11x, refer to Section 3.1.5.1 on page 34).
2. Use a serial communication software (e.g., HyperTerminal™) with the following communications port settings:
   - Baud Rate: 115,200 bps (MP-124), 9,600 bps (MP-11x)
   - Data bits: 8
   - Parity: None
   - Stop bits: 1
   - Flow control: None

Note that after resetting the gateway, the information, shown in Figure 11-1 below, appears on the terminal screen. This information can be used to determine possible MediaPack initialization problems, such as incorrectly defined (or undefined) local IP address, subnet mask, etc.

Figure 10-1: RS-232 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
```

10.3 Automatic Update Mechanism

The MediaPack is capable of automatically updating its *cmp*, *ini* and configuration files. These files can be stored on any standard Web, FTP or NFS server/s and can be loaded periodically to the gateway via HTTP, HTTPS, FTP or NFS. This mechanism can be used even for Customer Premise(s) Equipment (CPE) devices that are installed behind NAT and firewalls.

The Automatic Update mechanism is applied separately to each file. For the detailed list of available files and their corresponding parameters, refer to Table 5-56 on page 184.

> **Note:** The Automatic Update mechanism assumes the external Web server conforms to the HTTP standard. If the Web server ignores the If-Modified-Since header, or doesn't provide the current date and time during the HTTP 200 OK response, the gateway may reset itself repeatedly. To overcome this problem, adjust the update frequency (AutoUpdateFrequency).

Three methods are used to activate the Automatic Update mechanism:

- After the MediaPack starts-up (refer to the Startup process described in Figure 10-3 on page 266).
- At a configurable time of the day (e.g., 18:00). This option is disabled by default.
- At fixed intervals (e.g., every 60 minutes). This option is disabled by default.

The following *ini* file example can be used to activate the Automatic Update mechanism.

Figure 10-2: Example of an *ini* File Activating the Automatic Update Mechanism

```
# DNS is required for specifying domain names in URLs
DnsPriServerIP = 10.1.1.11

# Load an extra configuration ini file using HTTP
IniFileURL = 'http://webserver.corp.com/AudioCodes/inifile.ini'
# Load Call Progress Tones file using HTTPS
CptFileUrl = 'https://10.31.2.17/usa_tones.dat'
# Load Voice Prompts file using FTPS with user ‘root’ and password ‘wheel’
VPFileUrl = 'ftps://root:wheel@ftpserver.corp.com/vp.dat'

# Update every day at 03:00 AM
AutoUpdatePredefinedTime = '03:00'
# Note: The cmp file isn’t updated since it is disabled by default
(AutoUpdateCmpFile).
```

Refer to the following notes:

- When HTTP or HTTPS are used, the gateway contacts the Web server/s and queries for the requested files. The *ini* file is loaded only if it was modified since the last automatic update. The *cmp* file is loaded only if its version is different from the version stored on the gateway’s non-volatile memory. All other auxiliary files (e.g., CPT) are updated only once. To update a previously-loaded auxiliary file, you must update the parameter containing its URL.
- To load different configurations (*ini* files) for specific gateways, add the string ‘<MAC>’ to the URL. This mnemonic is replaced with the MediaPack hardware MAC address. Resulting in an *ini* file name request that contains the gateway’s MAC address.
- To automatically update the *cmp* file, use the parameter ‘CmpFileURL’ to specify its name and location. As a precaution (to protect the MediaPack from an accidental update) the Automatic Update mechanism doesn’t apply to the *cmp* file by default. Therefore, (to enable it), set the parameter ‘AutoUpdateCmpFile’ to 1.

The following example illustrates how to utilize Automatic Updates for deploying devices with minimum manual configuration.

➢ To utilize Automatic Updates for deploying the MediaPack with minimum manual configuration, take these 5 steps:

1. Set up a Web server (in the following example it is http://www.corp.com/) where all configuration files are to be stored.
2. To each device, pre-configure the following parameter (DHCP / DNS are assumed): IniFileURL = 'http://www.corp.com/master_configuration.ini'
3. Create a file named master_configuration.ini, with the following text:

```
# Common configuration for all devices
# ------------------------------------
CptFileURL = 'http://www.corp.com/call_progress.dat'
# Check for updates every 60 minutes
AutoUpdateFrequency = 60

# Additional configuration per device
# -----------------------------------
# Each device loads a file named after its MAC address,
# (e.g., config_00908F033512.ini)
IniFileURL = 'http://www.corp.com/config_<MAC>.ini'

# Reset the device after configuration is updated.
# The device resets after all of the files are processed.
ResetNow = 1
```

   You can modify the master_configuration.ini file (or any of the config_<MAC>.ini files) at any time. The MediaPack queries for the latest version every 60 minutes and applies the new settings immediately.

4. For additional security, use HTTPS or FTPS. The MediaPack supports HTTPS (RFC 2818) and FTPS using the AUTH TLS method <draft-murray-auth-ftp-ssl-16> for the Automatic Update mechanism.
5. To load configuration files from an NFS server, the NFS file system parameters should be defined in the configuration *ini* file. The following is an example of an *ini* file for loading files from NFS servers using NFS version 2.

```
# Define NFS servers for Automatic Update
[ NFSServers ]
FORMAT NFSServers_Index = NFSServers_HostOrIP, NFSServers_RootPath,
NFSServers_NfsVersion;
NFSServers 1 = 10.31.2.10, /usr/share, 2 ;
NFSServers 2 = 192.168.100.7, /d/shared, 2 ;
[ \NFSServers ]

CptFileUrl = 'file://10.31.2.10/usr/share/public/usa_tones.dat'
VpFileUrl = 'file://192.168.100.7/d/shared/audiocodes/voiceprompt.dat'
```

10.4 Startup Process

The startup process (illustrated in Figure 10-3 on page 266) begins when the gateway is reset (physically or from the Web / SNMP) and ends when the operational software is running. In the startup process, the network parameters, software and configuration files are obtained.

After the gateway powers up or after it is physically reset, it broadcasts a BootRequest message to the network. If it receives a reply (from a BootP server), it changes its network parameters (IP address, subnet mask and default gateway address) to the values provided. If there is no reply from a BootP server and if DHCP is enabled (DHCPEnable = 1), the gateway initiates a standard DHCP procedure to configure its network parameters.

After changing the network parameters, the gateway attempts to load the *cmp* and various configuration files from the TFTP server's IP address, received from the BootP/DHCP servers. If a TFTP server's IP address isn't received, the gateway attempts to load the software (*cmp*) file and / or configuration files from a preconfigured TFTP server (refer to Section 10.3 on page 263). Thus, the gateway can obtain its network parameters from BootP or DHCP servers and its software and configuration files from a different TFTP server (preconfigured in *ini* file).

If BootP/DHCP servers are not found or when the gateway is reset from the Web / SNMP, it retains its network parameters and attempts to load the software (*cmp*) file and / or configuration files from a preconfigured TFTP server.

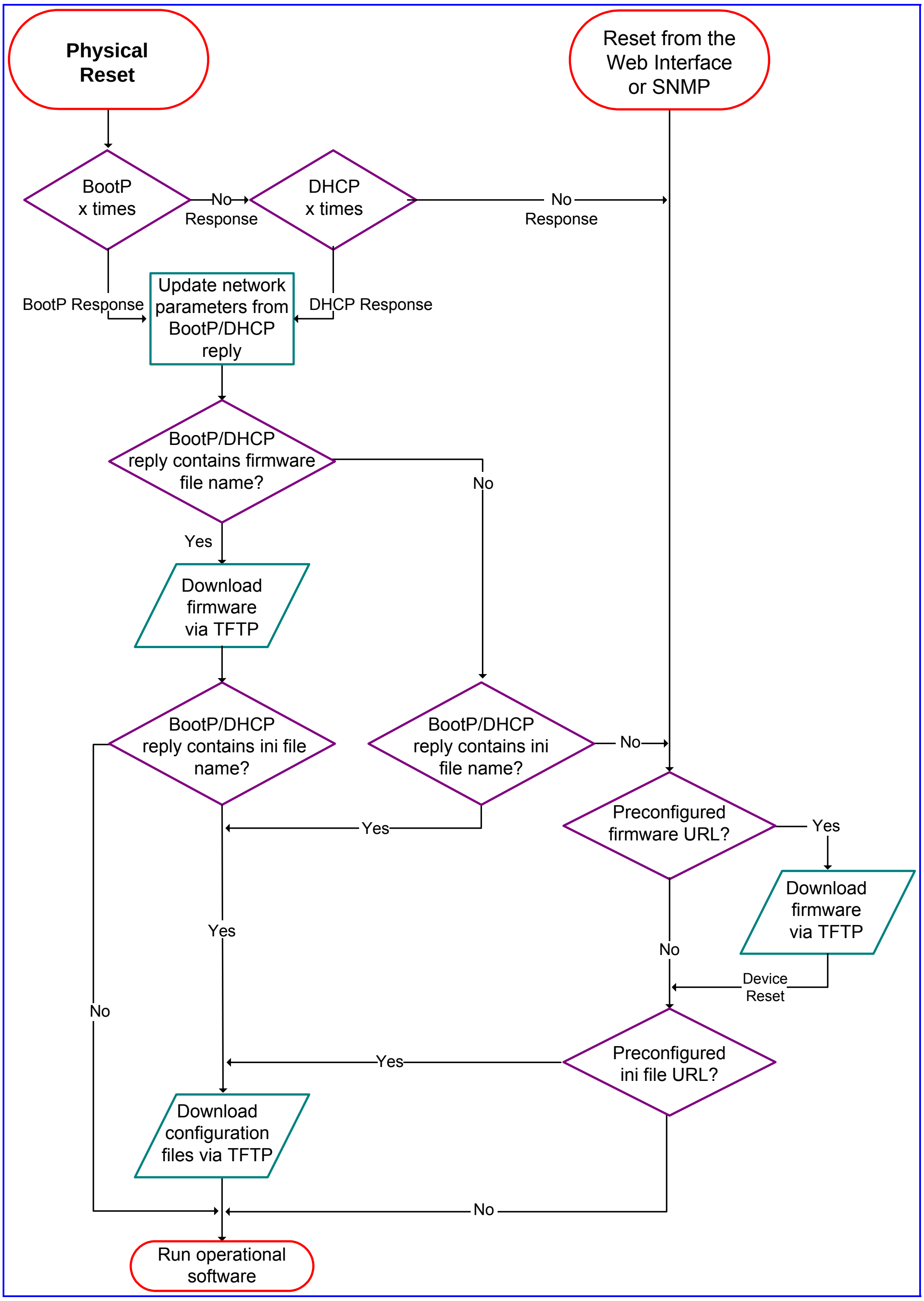
If a preconfigured TFTP server doesn't exist, the gateway operates using the existing software and configuration files loaded on its non-volatile memory.

Note that after the operational software runs, if DHCP is configured, the gateway attempts to renew its lease with the DHCP server.

Notes:

- Though DHCP and BootP servers are very similar in operation, the DHCP server includes some differences that could 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 gateway configuration.
- The time duration between BootP/DHCP requests is set to 1 second by default. This can be changed by the BootPDelay *ini* file parameter. Also, the number of requests is 3 by default and can be changed by BootPRetries *ini* file parameter (both parameters can also be set using the BootP command line switches).

Figure 10-3: MediaPack Startup Process



10.5 Using Parameter Tables

The MediaPack uses parameter tables to group related parameters of specific entities and manage them together. These tables, similar to regular parameters, can be configured via the *ini* file, Embedded Web Server, SNMP, etc.

Tables are composed of lines and columns. Columns represent parameters' types. Lines represent specific entities. The instances in each line are called line attributes. Lines in table may represent (for example) a trunk, an NFS file system, list of timers for a given application, etc.

Table 10-1 and Table 10-2 below provide useful examples for reference.

Table 10-1: Example of Parameter Table - Remote Management Connections

Index Fields: 1. Connection Number				
Connection Number	**User Name**	**User Password**	**Time Connected (msec)**	**Permissions**
0	Admin	Yellow9	0	All
1	Gillian	Red5	1266656	Read Only
2	David	Orange6	0	Read Write

Table 10-2: Example of Parameter Table - Port-to-Port Connections

Index Fields: 1. Source Ports 2. Destination IP 3. Destination Port				
Source Port	**Destination IP**	**Destination Port**	**Connection Name**	**Application Type**
2020	10.4.1.50	2020	ATM_TEST_EQ	LAB_EQ
2314	212.199.201.20	4050	ATM_ITROP_LOOP	LAB_EQ
6010	10.3.3.41	6010	REMOTE_MGMT	MGMT

Note: Table 10-1 and Table 10-2 are provided as examples for the purpose of illustration only and are not actually implemented in the MediaPack.

10.5.1 Table Indices

Each line in a table must be unique. Therefore, each table defines one or more Index fields. The combination of the Index fields determines the 'line-tag'. Each line-tag appears only once.

In the example provided in Table 10-1 there is only one Index field. This is the simplest way to mark lines.

In the example provided in Table 10-2 there are three Index fields. This more complicated method is a result of the application it represents.

10.5.2 Table Permissions

Each column has a 'permission' attribute that is applied to all instances in the column. This permission determines if and when a field can be modified. Several permissions can be applied to each column.

The following permissions are available:

- **Read:** The value of the field can be read.
- **Write:** The value of the field can be modified.
- **Create:** A value for the field must be provided at creation time (the default values, set to all fields, determine the initial values).
- **Maintenance Write:** The value of the field can only be modified when the entity represented by the line is in maintenance state (each table includes rules that determine when it is in maintenance state).

In the example in Table 10-1 it is assumed that the columns 'User Name' and 'User Password' have Read-Create permissions. The column 'Time Connected' has a Read permission, and the column 'Permissions' has Read-Create-Maintenance Write permissions.

10.5.3 Dynamic Tables vs. Static Tables

- **Static Tables:** Static tables don't support adding new lines or removing (deleting) existing lines. All lines in a Static table are pre-configured with default values. Users can only modify the values of the existing lines. After reset, all lines in a Static table are available.
- **Dynamic Tables:** Dynamic tables support adding and removing lines. They are always initialized as empty tables with no lines. Users should add lines to a Dynamic table via the *ini* file or at run-time via the Embedded Web Server for example.

Note: Certain dynamic tables may initialize a line (or more) at start-up. If so, it is explained in the specific table's documentation.

10.5.4 Secret Tables

A table is defined as a secret table if it contains at least a single secret data field or if it depends on another secret table. A secret data field is a field that mustn't be revealed to the user. For example, in the IPSec application, IPSec tables are defined as secret tables as the IKE table contains a pre-shared key that must be concealed. Therefore, the SPD table that depends on the IKE table is defined as a secret table as well.

There are two major differences between tables and secret tables:

- The secret field itself cannot be viewed via SNMP, Web or any other application.
- *ini* file behavior: Secret tables are never displayed in an uploaded *ini* file (e.g., when performing a 'Get *ini* File from Web' operation). Instead, there is a commented title that states that the secret table is present at the gateway and is not to be revealed. Secret tables are always kept in the gateway's non-volatile memory and can be over-written by new tables that are provided in a new *ini* file. If a secret table appears in an *ini* file, it replaces the current table regardless of its content. To delete a secret table from the gateway, provide an empty table of the same type (with no data lines) as part of a new *ini* file; the empty table replaces the previous table in the gateway.

10.5.5 Using the *ini* File to Configure Parameter Tables

You can use the *ini* file to add / modify parameter tables. When using tables, Read-Only parameters are not loaded, as they cause an error when trying to reload the loaded file. Therefore, Read-Only parameters mustn't be included in tables in the *ini* file. Consequently, tables are loaded with all parameters having at least one of the following permissions: Write, Create or Maintenance Write.

Parameter tables (in an uploaded *ini* file) are grouped according to the applications they configure (e.g., NFS, IPSec). When loading an *ini* file to the gateway, the recommended policy is to include only tables that belong to applications that are to be configured (Dynamic tables of other applications are empty, but static tables are not).

The *ini* file includes a Format line that defines the columns of the table to be modified (this may vary from *ini* file to *ini* file for the same table). The Format line must only include columns that can be modified (parameters that are not specified as Read-Only).

An exception is Index-fields that are always mandatory. In the example provided in Table 10-1, all fields except for the 'Time Connected' field are loaded.

10.5.5.1 Structure of Parameter Tables in the *ini* File

Tables are composed of four elements:

- **Title of the table:** The name of the table in square brackets (e.g., [MY_TABLE_NAME]).
- **A Format line:** Specifies the columns of the table (by their string names) that are to be configured.
    - The first word of the Format line must be 'FORMAT', followed by the names of the Indices fields, and an equal sign '='. After the equal sign the names of the columns are listed.
    - Items must be separated by a comma ','.
    - The Format line must end with a semicolon ';'.
- **Data line(s):** Contain the actual values of the parameters. The values are interpreted according to the Format line. The first word of the Data line must be the table's string name followed by the Index fields.
    - Items must be separated by a comma ','.
    - A Data line must end with a semicolon ';'.
- **End-of-Table-Mark:** Indicates the end of the table. The same string used for the table's title, preceded by a backslash '\' (e.g., [\MY_TABLE_NAME]).

Figure 10-4 displays an example of the structure of a parameter table in the *ini* file.

Figure 10-4: Structure of a Parameter Table in the *ini* File

```
; Table: Items Table.
; Fields: Item_Name, Item_Serial_Number, Item_Color, Item_weight.
; NOTE: Item_Color is not specified. It will be given default value.
[Items_Table]
; Fields declaration
Format Item_Index =  Item_Name, Item_Serial_Number, Item_weight;
Items_Table 0 = Computer, 678678, 6;
Items_Table 6 = Computer-screen, 127979, 9;
Items_Table 2 = Computer-pad, 111111, $$;
[\Items_Table]
```

Refer to the following notes:

- Indices (in both the Format and the Data lines) must appear in the same order determined by the specific table's documentation. The Index field must never be omitted.
- The Format line can include a sub-set of the configurable fields in a table. In this case, all other fields are assigned with the pre-defined default values for each configured line.
- The order of the fields in the Format line isn't significant (as opposed to the Index-fields). The fields in the Data lines are interpreted according to the order specified in the Format line.
- The sign '$$' in a Data line indicates that the user wants to assign the pre-defined default value to it.
- The order of the Data lines is insignificant.
- Data lines must match the Format line, i.e., it must contain exactly the same number of Indices and Data fields and must be in exactly the same order.
- A line in a table is identified by its table-name and Index fields. Each such line may appear only once in the *ini* file.
- Table dependencies:
  Certain tables may depend on other tables. For example, one table may include a field that specifies an entry in another table. This method is used to specify additional attributes of an entity, or to specify that a given entity is part of a larger entity. The tables must appear in the order of their dependency (i.e., if Table X is referred to by Table Y, Table X must appear in the *ini* file before Table Y).

10.6 Customizing the MediaPack Web Interface

Customers incorporating the MediaPack into their portfolios can customize the Web Interface to suit their specific corporate logo and product naming conventions.

Customers can customize the Web Interface's title bar (AudioCodes' title bar is shown in Figure 10-5; a customized title bar is shown in Figure 10-7).

Figure 10-5: User-Customizable Web Interface Title Bar

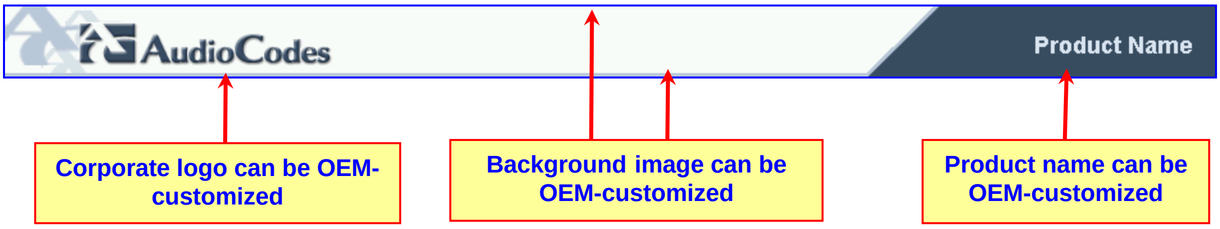


Figure 10-6: Customized Web Interface Title Bar

➢ To customize the title bar via the Web Interface, take these 3 steps:

1. Replace the main corporate logo (refer to Section 10.6.1 below).
2. Replace the title bar's background image file (refer to Section 10.6.2 on page 274).
3. Customize the product's name (refer to Section 10.6.3 on page 275).

10.6.1 Replacing the Main Corporate Logo

The main corporate logo can be replaced either with a different logo image file (refer to Section 10.6.1.1 below) **or** with a text string (refer to Section 10.6.1.2 on page 273). Note that when the main corporation logo is replaced, AudioCodes' logo on the left bar (refer to Figure 5-2) and in the Software Upgrade Wizard (Section 5.9.1 on page 197) disappear.

Also note that the browser's title bar is automatically updated with the string assigned to the WebLogoText parameter when AudioCodes' default logo is not used.

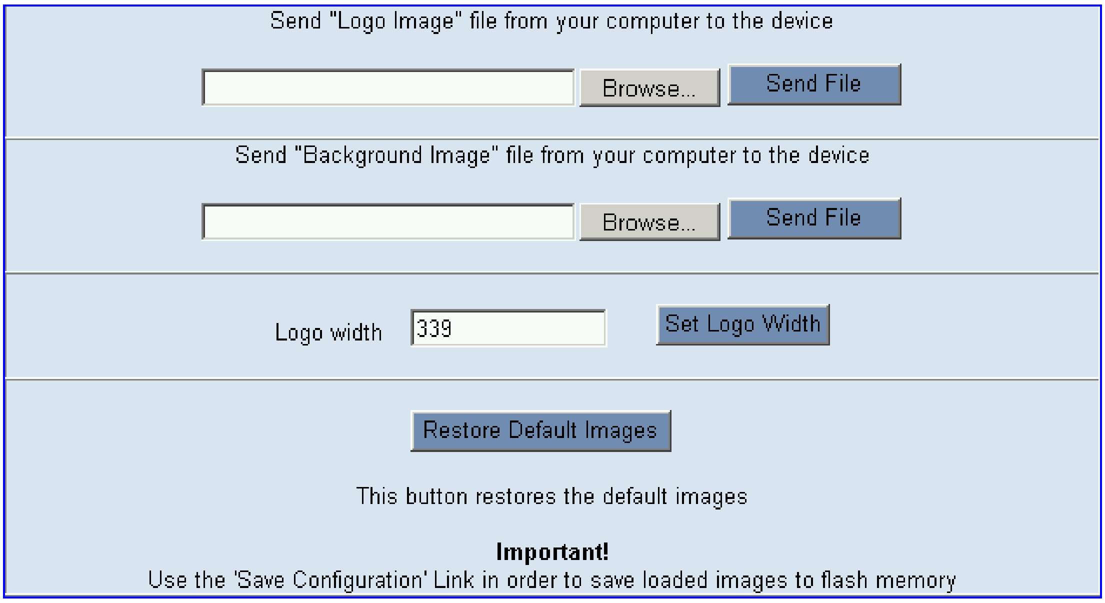
10.6.1.1 Replacing the Main Corporate Logo with an Image File

Note:	Use a gif, jpg or jpeg file for the logo image. It is important that the image file has a fixed height of 59 pixels (the width can be configured up to a maximum of 339 pixels). The size of the image files (logo and background) is limited each to 64 kbytes.

➢ **To replace the default logo with your own corporate image via the Web Interface, take these 7 steps:**

1. Access the MediaPack Embedded Web Server (refer to Section 5.3 on page 51).
2. In the URL field, append the suffix 'AdminPage' (note that it's case-sensitive) to the IP address, e.g., http://10.1.229.17/AdminPage.
3. Click **Image Load to Device**; the Image Download screen is displayed (shown in Figure 10-7).

Figure 10-7: Image Download Screen



4. Click the **Browse** button in the 'Send Logo Image File from your computer to the device' box. Navigate to the folder that contains the logo image file you want to load.
5. Click the **Send File** button; the file is sent to the device. When loading is complete, the screen is automatically refreshed and the new logo image is displayed.
6. Note the appearance of the logo. If you want to modify the width of the logo (the default width is 339 pixels), in the 'Logo Width' field, enter the new width (in pixels) and click the **Set Logo Width** button.
7. To save the image to flash memory so it is available after a power fail, refer to Section 5.10.2 on page 205.

The new logo appears on all Web Interface screens.

Tip:	If you encounter any problem during the loading of the files, or you want to restore the default images, click the **Restore Default Images** button.

➢ To replace the default logo with your own corporate image via the *ini* file, take these 2 steps:

1. Place your corporate logo image file in the same folder as where the device's *ini* file is located (i.e., the same location defined in the BootP/TFTP configuration utility). For detailed information on the BootP/TFTP, refer to Appendix C on page 349.
2. Add/modify the two *ini* file parameters in Table 10-3 according to the procedure described in Section 6.2 on page 209.

Note: Loading the device's *ini* file via the 'Configuration File' screen in the Web Interface doesn't load the corporate logo image files as well.

Table 10-3: Customizable Logo *ini* File Parameters

Parameter	Description
LogoFileName	The name of the image file containing your corporate logo. Use a gif, jpg or jpeg image file. The default is AudioCodes' logo file. **Note:** The length of the name of the image file is limited to 47 characters.
LogoWidth	Width (in pixels) of the logo image. **Note:** The optimal setting depends on the resolution settings. The default value is 339, which is the width of AudioCodes' displayed logo.

10.6.1.2 Replacing the Main Corporate Logo with a Text String

The main corporate logo can be replaced with a text string.

- To replace AudioCodes' default logo with a text string *via the Web Interface*, modify the two *ini* file parameters in Table 10-4 according to the procedure described in Section 10.6.4 on page 276.
- To replace AudioCodes' default logo with a text string *via the ini* file, add/modify the two *ini* file parameters in Table 10-4 according to the procedure described in Section 6.2 on page 209.

Table 10-4: Web Appearance Customizable *ini* File Parameters

Parameter	Description
UseWebLogo	0 = Logo image is used (default). 1 = Text string is used instead of a logo image.
WebLogoText	Text string that replaces the logo image. The string can be up to 15 characters.

10.6.2 Replacing the Background Image File

The background image file is duplicated across the width of the screen. The number of times the image is duplicated depends on the width of the background image and screen resolution. When choosing your background image, keep this in mind.

> **Note:** Use a gif, jpg or jpeg file for the background image. It is important that the image file has a fixed height of 59 pixels. The size of the image files (logo and background) is limited each to 64 kbytes.

➢ To replace the background image via the Web, take these 6 steps:

1. Access the MediaPack Embedded Web Server (refer to Section 5.3 on page 51).
2. In the URL field, append the suffix 'AdminPage' (note that it's case-sensitive) to the IP address, e.g., http://10.1.229.17/AdminPage.
3. Click the **Image Load to Device**, the Image load screen is displayed (shown in Figure 10-7).
4. Click the **Browse** button in the 'Send Background Image File from your computer to gateway' box. Navigate to the folder that contains the background image file you want to load.
5. Click the **Send File** button; the file is sent to the device. When loading is complete, the screen is automatically refreshed and the new background image is displayed.
6. To save the image to flash memory so it is available after a power fail, refer to Section 5.10.2 on page 205.

The new background appears on all Web Interface screens.

> **Tip 1:** If you encounter any problem during the loading of the files, or you want to restore the default images, click the **Restore Default Images** button.
>
> **Tip 2:** When replacing both the background image and the logo image, first load the logo image followed by the background image.

➢ To replace the background image via the *ini* file, take these 2 steps:

1. Place your background image file in the same folder as where the device's *ini* file is located (i.e., the same location defined in the BootP/TFTP configuration utility). For detailed information on the BootP/TFTP, refer to Appendix C on page 349.
2. Add/modify the *ini* file parameters in Table 10-5 according to the procedure described in Section 6.2 on page 209.

Note that loading the device's *ini* file via the 'Configuration File' screen in the Web Interface doesn't load the logo image file as well.

Table 10-5: Customizable Logo *ini* File Parameters

Parameter	Description
BkgImageFileName	The name (and path) of the file containing the new background. Use a gif, jpg or jpeg image file. The default is AudioCodes background file. **Note:** The length of the name of the image file is limited to 47 characters.

10.6.3 Customizing the Product Name

The Product Name text string can be modified according to OEMs specific requirements.

- To replace AudioCodes' default product name with a text string *via the Web Interface*, modify the two *ini* file parameters in Table 10-6 according to the procedure described in Section 10.6.4 on page 276.
- To replace AudioCodes' default product name with a text string *via the ini* file, add/modify the two *ini* file parameters in Table 10-6 according to the procedure described in Section 6.2 on page 209.

Table 10-6: Web Appearance Customizable *ini* File Parameters

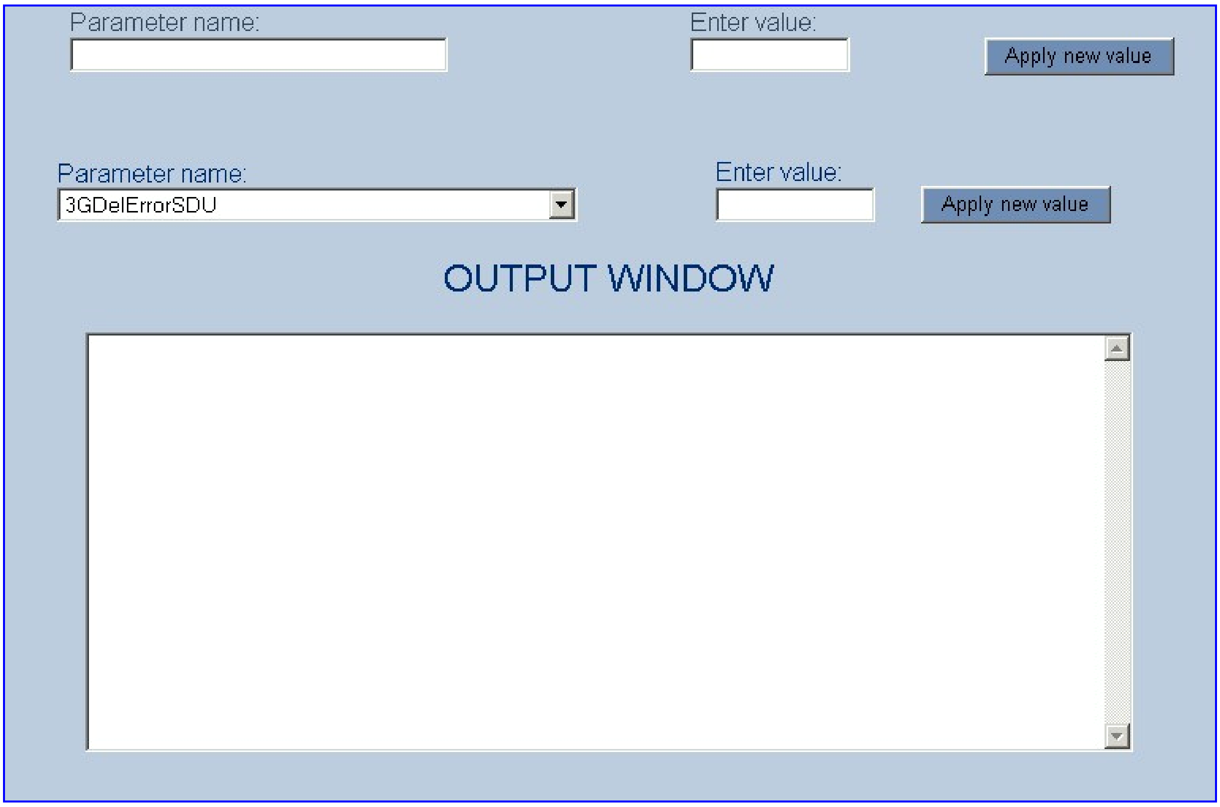
Parameter	Description
UseProductName	0 = Don't change the product name (default). 1 = Enable product name change.
UserProductName	Text string that replaces the product name. The default is 'MediaPack'. The string can be up to 29 characters.

10.6.4 Modifying *ini* File Parameters via the Web AdminPage

➢ **To modify *ini* file parameters via the AdminPage, take these 6 steps:**

1. Access the MediaPack Embedded Web Server (refer to Section 5.3 on page 51).
2. In the URL field, append the suffix 'AdminPage' (note that it's case-sensitive) to the IP address, e.g., http://10.1.229.17/AdminPage.
3. Click the **INI Parameters** option, the INI Parameters screen is displayed (shown in Figure 10-8).

Figure 10-8: INI Parameters Screen



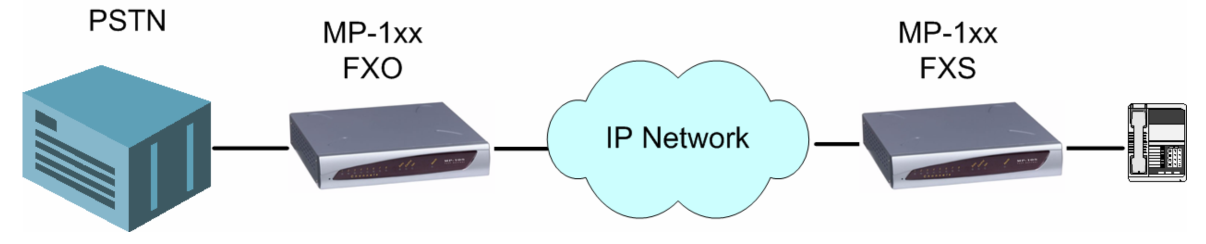
4. From the 'Parameter Name' drop-down list, select the required *ini* file parameter.
5. In the 'Enter Value' field, enter the parameter's new value.
6. Click the **Apply New Value** button; the INI Parameters screen is refreshed, the parameter name with the new value appears in the fields at the top of the screen and the 'Output Window' pane displays a log displaying information on the operation.

Note: You cannot load the image files (e.g., logo/background image files) to the device by choosing a file name parameter in this screen.

11 Special Applications - Metering Tones Relay

The MediaPack FXS and FXO gateways can be used to relay standard 12 or 16 kHz metering tones over the IP network as illustrated in Figure 11-1 below.

Figure 11-1: Metering Tone Relay Architecture



After a call is established between the FXS and FXO gateways, the PSTN generates 12 or 16 kHz metering tones towards the FXO gateway. The FXO gateway detects these pulses and relays them, over IP, to the FXS gateway using a proprietary INFO messages (shown in Figure 11-2). The FXS gateway generates the same pulses to the connected phone.

The parameter ‘MeteringType’ (described in Table 5-35) is used to determine the frequency of the metering tone (12 kHz (default) or 16 kHz). In addition, the correct (12 or 16 kHz) coefficient must be used for both FXS and FXO gateways.

To enable this feature configure ‘SendMetering2IP = 1’.

The proprietary INFO message used to relay the metering tone pulse contains a ‘Content-Type: message/Metering’:

Figure 11-2: Proprietary INFO Message for Relaying Metering Tones

```
INFO sip:108@10.13.83.1 SIP/2.0
Via: SIP/2.0/UDP 10.13.83.2;branch=z9hG4bKacEizRjAa
Max-Forwards: 70
From: "aviad" <sip:201@10.13.83.2>;tag=1c1638621413
To: <sip:108@10.13.83.1;user=phone>;tag=1c1412617336
Call-ID: 2031013892fcCd@10.13.83.2
CSeq: 3 INFO
Contact: <sip:201@10.13.83.2>
Supported: em,timer,replaces,path
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE
User-Agent: Audiocodes-Sip-Gateway-MP-114 FXS/v.4.40.0.18700
Content-Type: message/Metering
Content-Length: 0
```

Reader's Notes

12 Security

This section describes the security mechanisms and protocols implemented on the MediaPack. The following list specifies the available security protocols and their objectives:

- IPSec and IKE protocols are part of the IETF standards for establishing a secured IP connection between two applications. IPSec and IKE are used in conjunction to provide security for control and management protocols but not for media (refer to Section 12.1 below).
- SSL (Secure Socket Layer) / TLS (Transport Layer Security) – The SSL / TLS protocols are used to provide privacy and data integrity between two communicating applications over TCP/IP. They are used to secure the following applications: SIP Signaling (SIPS), Web access (HTTPS) and Telnet access (refer to Section 12.2 on page 288).
- Secured RTP (SRTP) according to RFC 3711, used to encrypt RTP and RTCP transport (refer to Section 12.3 on page 293).
- RADIUS (Remote Authentication Dial-In User Service) - RADIUS server is used to enable multiple-user management on a centralized platform (refer to Section 12.4 on page 294).
- Internal Firewall allows filtering unwanted inbound traffic (refer to Section 12.5 on page 297).

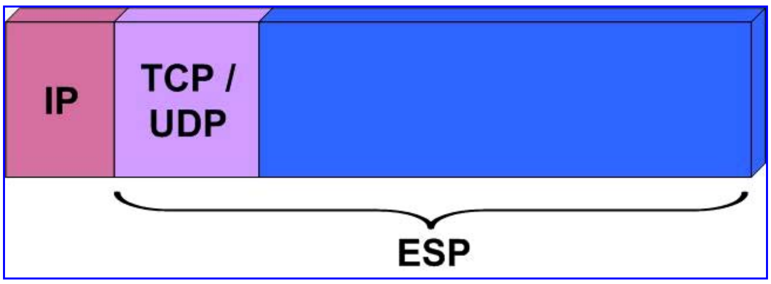
12.1 IPSec and IKE

IP Security (IPSec) and Internet Key Exchange (IKE) protocols are part of the IETF standards for establishing a secured IP connection between two applications (also referred to as peers). Providing security services at the IP layer, IPSec and IKE are transparent to IP applications.

IPSec and IKE are used in conjunction to provide security for control and management (e.g., SNMP and Web) protocols, but not for media (i.e., RTP, RTCP and T.38).

IPSec is responsible for securing the IP traffic. This is accomplished by using the Encapsulation Security Payload (ESP) protocol to encrypt the IP payload (illustrated in Figure 12-1 below). The IKE protocol is responsible for obtaining the IPSec encryption keys and encryption profile known as IPSec Security Association (SA).

Figure 12-1: IPSec Encryption



Note: IPSec doesn't function properly if the gateway's IP address is changed on-the-fly due to the fact that the crypto hardware can only be configured on reset. Therefore, reset the gateway after you change its IP address.

12.1.1 IKE

IKE is used to obtain the Security Associations (SA) between peers (the gateway and the application it's trying to contact). The SA contains the encryption keys and profile used by the IPSec to encrypt the IP stream. The IKE table lists the IKE peers with which the gateway performs the IKE negotiation (up to 20 peers are available).

The IKE negotiation is separated into two phases: main mode and quick mode. The main mode employs the Diffie-Hellman (DH) protocol to obtain an encryption key (without any prior keys), and uses a pre-shared key to authenticate the peers. The created channel secures the messages of the following phase (quick mode) in which the IPSec SA properties are negotiated.

The IKE negotiation is as follows:

- Main mode (the main mode creates a secured channel for the quick mode)
    - **SA negotiation:** The peers negotiate their capabilities using two proposals. Each proposal includes three parameters: Encryption method, Authentication protocol and the length of the key created by the DH protocol. The key's lifetime is also negotiated in this stage. For detailed information on configuring the main mode proposals, refer to Section 12.1.3.1 on page 281.
    - **Key exchange (DH):** The DH protocol is used to create a phase-1 key.
    - **Authentication:** The two peers authenticate one another using the pre-shared key (configured by the parameter 'IKEPolicySharedKey').
- Quick mode (quick mode negotiation is secured by the phase-1 SA)
    - **SA negotiation:** The peers negotiate their capabilities using a single proposal. The proposal includes two parameters: Encryption method and Authentication protocol. The lifetime is also negotiated in this stage. For detailed information on configuring the quick mode proposal, refer to the SPD table under Section 12.1.3.2 on page 284.
    - **Key exchange:** a symmetrical key is created using the negotiated SA.

IKE specifications include the following:

- Authentication mode - pre-shared key only
- Main mode is supported for IKE Phase 1
- Supported IKE SA encryption algorithms - Data Encryption Standard (DES), 3DES, and Advanced Encryption Standard (AES)
- Hash types for IKE SA - SHA1 and MD5

12.1.2 IPSec

IPSec is responsible for encrypting and decrypting the IP streams.

The IPSec Security Policy Database (SPD) table defines up to 20 IP peers to which the IPSec security is applied. IPSec can be applied to all packets designated to a specific IP address or to a specific IP address, port (source or destination) and protocol type.

Each outgoing packet is analyzed and compared to the SPD table. The packet's destination IP address (and optionally, destination port, source port and protocol type) are compared to each entry in the table. If a match is found, the gateway checks if an SA already exists for this entry. If it doesn't, the IKE protocol is invoked (refer to Section 12.1.1 above) and an IPSec SA is established. The packet is encrypted and transmitted. If a match isn't found, the packet is transmitted un-encrypted.

Note: An incoming packet whose parameters match one of the entries of the SPD table, but received un-encrypted is dropped.

IPSec specifications include the following:

- Transport mode only
- Encapsulation Security Payload (ESP) only
- Support for Cipher Block Chaining (CBC)
- Supported IPSec SA encryption algorithms - DES, 3DES, and AES
- Hash types for IPSec SA are SHA1 and MD5

12.1.3 Configuring the IPSec and IKE

To enable IPSec and IKE on the gateway set the *ini* file parameter 'EnableIPSec' to 1.

12.1.3.1 IKE Configuration

The parameters described in Table 12-1 below are used to configure the first phase (main mode) of the IKE negotiation for a specific peer. A different set of parameters can be configured for each of the 20 available peers.

Up to two IKE main mode proposals (Encryption / Authentication / DH group combinations) can be defined. The same proposals must be configured for all peers.

Table 12-1: IKE Table Configuration Parameters (continues on pages 281 on 282)

Parameter Name	Description
Shared Key **[IKEPolicySharedKey]**	Determines the pre-shared key (in textual format). Both peers must register the same pre-shared key for the authentication process to succeed. **Note 1:** The pre-shared key forms the basis of IPSec security and should therefore be handled cautiously (in the same way as sensitive passwords). It is not recommended to use the same pre-shared key for several connections. **Note 2:** Since the *ini* file is in plain text format, loading it to the gateway over a secure network connection is recommended, preferably over a direct crossed-cable connection from a management PC. For added confidentiality, use the encoded *ini* file option (described in Section 6.1 on page 209). **Note 3:** After it is configured, the value of the pre-shared key cannot be obtained via Web, *ini* file or SNMP (refer to Section 12.1.3.3 on page 287).
First to Fourth Proposal Encryption Type **[IKEPolicyProposalEncryption_X]**	Determines the encryption type used in the main mode negotiation for up to four proposals. X stands for the proposal number (0 to 3). The valid encryption values are: Not Defined (default) DES-CBC **[1]** Triple DES-CBC **[2]** AES **[3]**
First to Fourth Proposal Authentication Type **[IKEPolicyProposalAuthentication_X]**	Determines the authentication protocol used in the main mode negotiation for up to four proposals. X stands for the proposal number (0 to 3). The valid authentication values are: Not Defined (default) HMAC-SHA1-96) **[2]** HMAC-MD5-96 **[4]**
First to Fourth Proposal DH Group **[IKEPolicyProposalDHGroup_X]**	Determines the length of the key created by the DH protocol for up to four proposals. X stands for the proposal number (0 to 3). The valid DH Group values are: Not Defined (default) DH-786-Bit **[0]** DH-1024-Bit **[1]**

Table 12-1: IKE Table Configuration Parameters (continues on pages 281 on 282)

Parameter Name	Description
Authentication Method **[IkePolicyAuthenticationMethod]**	Determines the authentication method for IKE. The valid authentication method values include: 0 = Pre-shared Key (default) 1 = RSA Signiture **Note 1:** For pre-shared key based authentication, peers participating in an IKE exchange must have a prior (out-of-band) knowledge of the common key (see IKEPolicySharedKey parameter). **Note 2:** For RSA signature based authentication, peers must be loaded with a certificate signed by a common CA. For additional information on certificates, refer to Section 12.2.4 on page 290.
IKE SA LifeTime (sec) **[IKEPolicyLifeInSec]**	Determines the time (in seconds) the SA negotiated in the first IKE session (main mode) is valid. After the time expires, the SA is re-negotiated. The default value is 28800 (8 hours).
IKE SA LifeTime (KB) **[IKEPolicyLifeInKB]**	Determines the lifetime (in kilobytes) the SA negotiated in the first IKE session (main mode) is valid. After this size is reached, the SA is re-negotiated. The default value is 0 (this parameter is ignored).
The lifetime parameters (IKEPolicyLifeInSec and IKEPolicyLifeInKB) determine the duration the SA created in the main mode phase is valid. When the lifetime of the SA expires, it is automatically renewed by performing the IKE first phase negotiations. To refrain from a situation where the SA expires, a new SA is being negotiated while the old one is still valid. As soon as the new SA is created, it replaces the old one. This procedure occurs whenever an SA is about to expire.	

If no IKE methods are defined (Encryption / Authentication / DH Group), the default settings (shown in Table 12-2 below) are applied.

Table 12-2: Default IKE First Phase Proposals

	Encryption	Authentication	DH Group
Proposal 0	3DES	SHA1	1024
Proposal 1	3DES	MD5	1024
Proposal 2	3DES	SHA1	786
Proposal 3	3DES	MD5	786

➢ To configure the IKE table using the *ini* file:

The IKE parameters are configured using *ini* file tables (described in Section 10.5 on page 267). Each line in the table refers to a different IKE peer.

The Format line (IKE_DB_INDEX in the example below) specifies the order in which the actual data lines are written. The order of the parameters is irrelevant. Parameters are not mandatory unless stated otherwise. To support more than one Encryption / Authentication / DH Group proposals, for *each* proposal specify the relevant parameters in the Format line. Note that the proposal list must be contiguous.

Figure 12-2: Example of an IKE Table

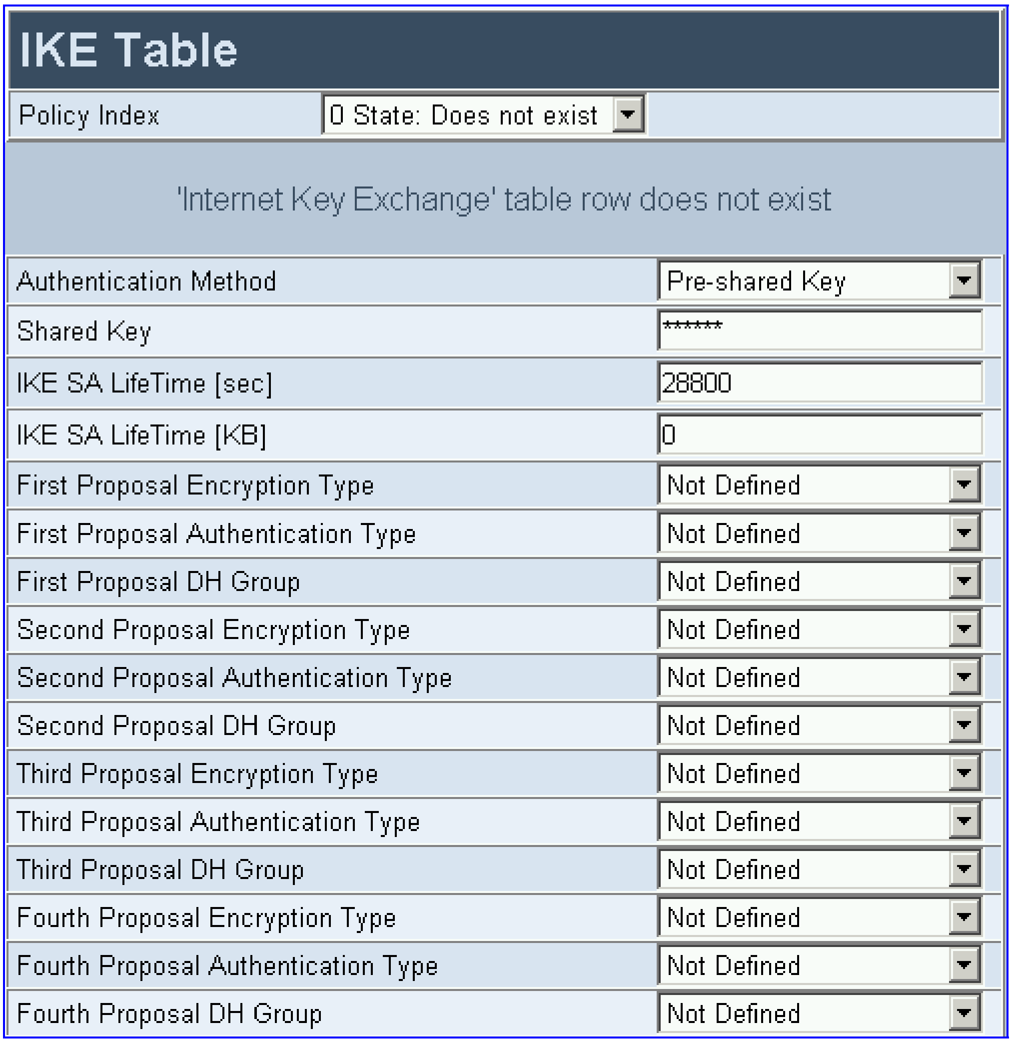
```
[IPSec_IKEDB_Table]

Format IKE_DB_INDEX = IKEPolicySharedKey, IKEPolicyProposalEncryption_0,
IKEPolicypRoposalAuthentication_0, IKEPolicyProposalDHGroup_0,
IKEPolicyProposalEncryption_1, IKEPolicypRoposalAuthentication_1,
IKEPolicyProposalDHGroup_1, IKEPolicyLifeInSec, IkePolicyAuthenticationMethod;

IPSEC_IKEDB_TABLE 0 = 123456789, 1, 2, 0, 2, 2, 1, 28800, 0;

[\IPSEC_IKEDB_TABLE]
```

In the example above, a single IKE peer is configured and a Pre-shared key authentication is selected. Its pre-shared key is 123456789. Two security proposals are configured: DES/SHA1/786DH and 3DES/SHA1/1024DH.

➢ To configure the IKE table using the Embedded Web Server, take these 6 steps:

1. Access the Embedded Web Server (refer to Section 5.3 on page 51).
2. Open the ‘IKE Table’ screen (**Advanced Configuration** menu > **Security Settings** > **IKE Table** option); the ‘IKE Table’ screen is displayed.

Figure 12-3: IKE Table Screen



3. In the ‘Policy Index’ drop-down list, select the peer you want to edit (up to 20 peers can be configured).
4. Configure the IKE parameters according to Table 12-1 on page 281.
5. Click the button **Create**; a row is create in the IKE table
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

To delete a peer from the IKE table, select it in the ‘Policy Index’ drop-down list, click the button **Delete** and click **OK** in the prompt.

12.1.3.2 IPSec Configuration

The parameters described in Table 12-3 below are used to configure the SPD table. A different set of parameters can be configured for each of the 20 available IP destinations.

Table 12-3: SPD Table Configuration Parameters (continues on pages 284 to 285)

<table>
<tr><th>Parameter Name</th><th colspan="2">Description</th></tr>
<tr><td>Remote IP Address
[IPSecPolicyRemoteIPAddress]</td><td>Defines the destination IP address (or a FQDN) the IPSec mechanism is applied to.
This parameter is mandatory.
Note: When a FQDN is used, a DNS server must be configured (DNSPriServerIP).</td><td rowspan="5">IPSec is applied to outgoing packets whose IP address, destination port, source port and protocol type match the values defined for these four parameters.</td></tr>
<tr><td>Local IP Address Type
[IPSecPolicyLocalIPAddressType]</td><td>Determines the local interface to which the encryption is applied (applicable to multiple IPs and VLANs).
0 = OAM interface (default).
1 = Control interface.</td></tr>
<tr><td>Source Port
[IPSecPolicySrcPort]</td><td>Defines the source port the IPSec mechanism is applied to.
The default value is 0 (any port).</td></tr>
<tr><td>Destination Port
[IPSecPolicyDstPort]</td><td>Defines the destination port the IPSec mechanism is applied to.
The default value is 0 (any port).</td></tr>
<tr><td>Protocol
[IPSecPolicyProtocol]</td><td>Defines the protocol type the IPSec mechanism is applied to.
0 = Any protocol (default).
17 = UDP.
6 = TCP.
Or any other protocol type defined by IANA (Internet Assigned Numbers Authority).</td></tr>
<tr><td>Related Key Exchange Method Index
[IPsecPolicyKeyExchangeMethodIndex]</td><td colspan="2">Determines the index for the corresponding IKE entry. Note that several policies can be associated with a single IKE entry.
The valid range is 0 to 19. The default value is 0.</td></tr>
<tr><td colspan="3">**IKE Second Phase Parameters (Quick Mode)**</td></tr>
<tr><td>SA Lifetime (sec)
I[PsecPolicyLifeInSec]</td><td colspan="2">Determines the time (in seconds) the SA negotiated in the second IKE session (quick mode) is valid. After the time expires, the SA is re-negotiated.
The default value is 28800 (8 hours).</td></tr>
<tr><td>SA Lifetime (KB)
[IPSecPolicyLifeInKB]</td><td colspan="2">Determines the lifetime (in kilobytes) the SA negotiated in the second IKE session (quick mode) is valid. After this size is reached, the SA is re-negotiated.
The default value is 0 (this parameter is ignored).</td></tr>
<tr><td colspan="3">The lifetime parameters (IPsecPolicyLifeInSec and IPSecPolicyLifeInKB) determine the duration an SA is valid. When the lifetime of the SA expires, it is automatically renewed by performing the IKE second phase negotiations. To refrain from a situation where the SA expires, a new SA is being negotiated while the old one is still valid. As soon as the new SA is created, it replaces the old one. This procedure occurs whenever an SA is about to expire.</td></tr>
</table>

Table 12-3: SPD Table Configuration Parameters (continues on pages 284 to 285)

Parameter Name	Description
First to Fourth Proposal Encryption Type **[IPSecPolicyProposalEncryption_X]**	Determines the encryption type used in the quick mode negotiation for up to four proposals. X stands for the proposal number (0 to 3). The valid encryption values are: Not Defined (default) None **[0]** = No encryption DES-CBC **[1]** Triple DES-CBC **[2]** AES **[3]**
First to Fourth Proposal Authentication Type **[IPSecPolicyProposalAuthentication_X]**	Determines the authentication protocol used in the quick mode negotiation for up to four proposals. X stands for the proposal number (0 to 3). The valid authentication values are: Not Defined (default) HMAC-SHA-1-96 **[2]** HMAC-MD5-96 **[4]**

If no IPSec methods are defined (Encryption / Authentication), the default settings (shown in Table 12-4 below) are applied.

Table 12-4: Default IKE Second Phase Proposals

	Encryption	Authentication
Proposal 0	3DES	SHA1
Proposal 1	3DES	MD5
Proposal 2	DES	SHA1
Proposal 3	DES	MD5

➢ To configure the SPD table using the *ini* file:

SPD table is configured using *ini* file tables (described in Section 10.5 on page 267). Each line in the table refers to a different IP destination.

The Format line (SPD_INDEX in the example below) specifies the order in which the actual data lines are written. The order of the parameters is irrelevant. Parameters are not mandatory unless stated otherwise. To support more than one Encryption / Authentication proposals, for *each* proposal specify the relevant parameters in the Format line. Note that the proposal list must be contiguous.

Figure 12-4: Example of an SPD Table

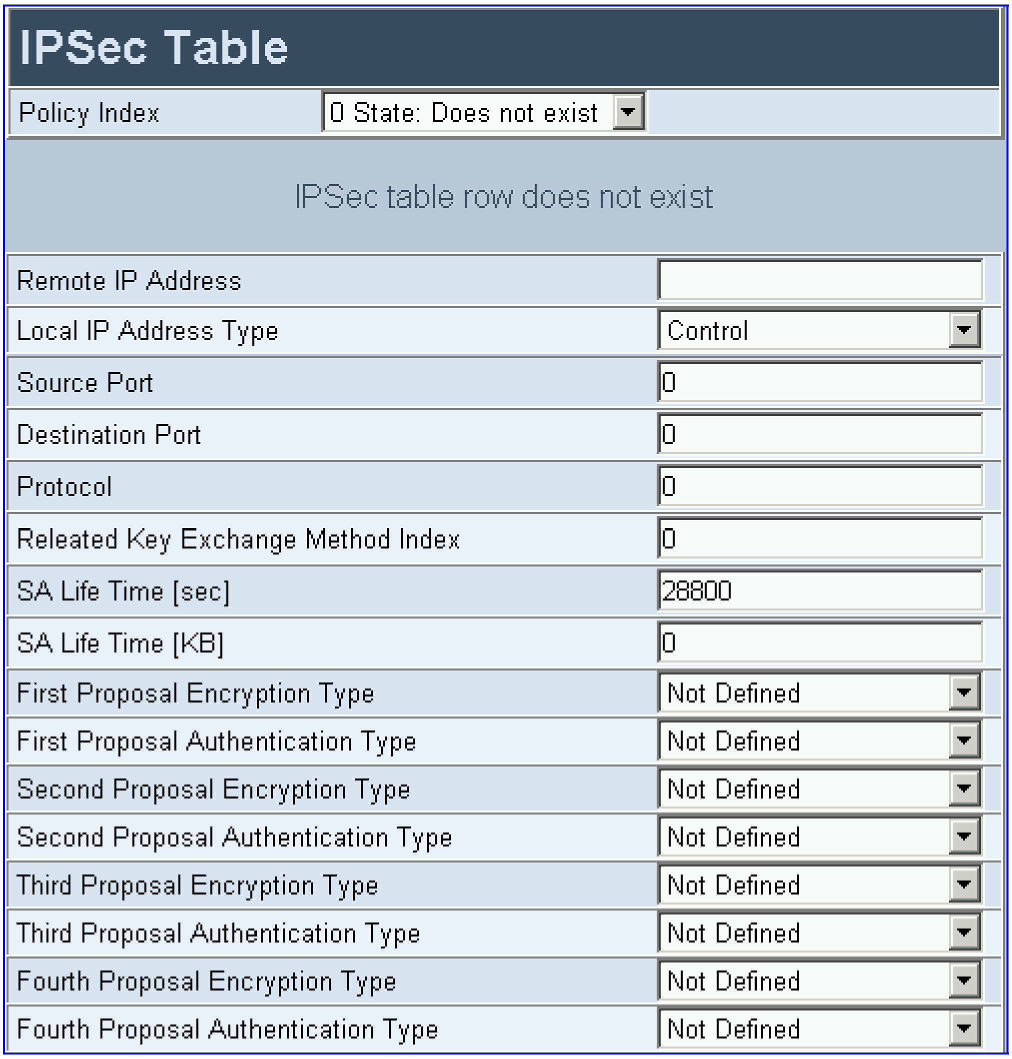
```
[ IPSEC_SPD_TABLE ]

Format SPD_INDEX = IPSecPolicyRemoteIPAddress, IpsecPolicySrcPort,
IPSecPolicyDStPort,IPSecPolicyProtocol, IPSecPolicyLifeInSec,
IPSecPolicyProposalEncryption_0, IPSecPolicyProposalAuthentication_0,
IPSecPolicyProposalEncryption_1, IPSecPolicyProposalAuthentication_1,
IPSecPolicyKeyExchangeMethodIndex, IPSecPolicyLocalIPAddressType;

IPSEC_SPD_TABLE 0 = 10.11.2.21, 0, 0, 17, 900, 1,2, 2,2 ,1, 0;

[ \IPSEC_SPD_TABLE ]
```

In the SPD example, all packets designated to IP address 10.11.2.21 that originates from the OAM interface (regardless to their destination and source ports) and whose protocol is UDP are encrypted, the SPD also defines an SA lifetime of 900 seconds and two security proposals: DES/SHA1 and 3DES/SHA1.

➢ To configure the SPD table using the Embedded Web Server, take these 6 steps:

1. Access the Embedded Web Server (refer to Section 5.3 on page 51).
2. Open the 'IPSec Table' screen (**Advanced Configuration** menu > **Security Settings** > **IPSec Table** option); the 'IPSec Table' screen is displayed.

Figure 12-5: IPSec Table Screen



3. In the 'Policy Index' drop-down list, select the rule you want to edit (up to 20 rules can be configured).
4. Configure the SPD parameters according to Table 12-3 on page 284.
5. Click the button **Create**; a row is create in the SPD table
6. To save the changes so they are available after a power fail, refer to Section 5.10.2 on page 205.

To delete a peer from the SPD table, select it in the 'Policy Index' drop-down list, click the button **Delete** and click **OK** in the prompt.

12.1.3.3 IPSec and IKE Configuration Table's Confidentiality

Since the pre-shared key parameter of the IKE table must remain undisclosed, measures are taken by the *ini* file, Embedded Web Server and SNMP agent to maintain this parameter's confidentiality. On the Embedded Web Server a list of asterisks is displayed instead of the pre-shared key. On SNMP, the pre-shared key parameter is a write-only parameter and cannot be read. In the *ini* file, the following measures to assure the secrecy of the IPSec and IKE tables are taken:

- Hidden IPSec and IKE tables **-** When uploading the *ini* file from the gateway the IPSec and IKE tables are not available. Instead, the notifications (shown in Figure 12-6) are displayed.

Figure 12-6: Example of an *ini* File Notification of Missing Tables

```
;
;  *** TABLE IPSEC_IKEDB_TABLE ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;

;
;  *** TABLE IPSEC_SPD_TABLE ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;
```

- Preserving the values of the parameters in the IPSec and IKE tables from one *ini* file loading to the next – The values configured for the parameters in the IPSec tables in the *ini* file are preserved from one loading to another. If a newly loaded *ini* file doesn't define IPSec tables, the previously loaded tables remain valid. To invalidate a previously loaded *ini* file's IPSec tables, load a new *ini* file with an empty IPSec table (shown below).

Figure 12-7: Empty IPSec / IKE Tables

```
[IPSec_IKEDB_Table]
[\IPSec_IKEDB_Table]

[IPSEC_SPD_TABLE]
[\IPSEC_SPD_TABLE]
```

12.2 SSL/TLS

SSL, also known as TLS, is the method used to secure the MediaPack SIP Signaling connections, Embedded Web Server and Telnet server. The SSL protocol provides confidentiality, integrity and authenticity between two communicating applications over TCP/IP.

Specifications for the SSL/TLS implementation:

- Supports transports: SSL 2.0, SSL 3.0, TLS 1.0
- Supports ciphers: DES, RC4 compatible
- Authentication: X.509 certificates; CRLs are not supported

12.2.1 SIP Over TLS (SIPS)

The MediaPack uses TLS over TCP to encrypt SIP transport and (optionally) to authenticate it. To enable TLS on the MediaPack, set the selected transport type to TLS (SIPTransportType = 2). In this mode the gateway initiates a TLS connection only for the next network hop. To enable TLS all the way to the destination (over multiple hops) set EnableSIPS to 1. When a TLS connection with the gateway is initiated, the gateway also responds using TLS regardless of the configured SIP transport type (in this case, the parameter EnableSIPS is also ignored).

TLS and SIPS use the Certificate Exchange process described in Sections 12.2.4 and 12.2.5. To change the port number used for SIPS transport (by default 5061), use the parameter TLSLocalSIPPort.

When SIPS is used, it is sometimes required to use two-way authentication. When acting as the TLS server (in a specific connection) it is possible to demand the authentication of the client's certificate. To enable two-way authentication on the MediaPack, set the *ini* file parameter, SIPSRequireClientCertificate = 1. For information on installing a client certificate, refer to Section 12.2.5 on page 292.

12.2.2 Embedded Web Server Configuration

For additional security, you can configure the Embedded Web Server to accept only secured (HTTPS) connections by changing the parameter HTTPSOnly to 1 (described in Table 5-50 on page 174).

You can also change the port number used for the secured Web server (by default 443) by changing the *ini* file parameter, HTTPSPort (described in Table 5-55 on page 182).

12.2.2.1 Using the Secured Embedded Web Server

➢ To use the secured Embedded Web Server, take these 3 steps:

1. Access the MediaPack using the following URL:

   https://[host name] or [IP address]

   Depending on the browser's configuration, a security warning dialog may be displayed. The reason for the warning is that the MediaPack initial certificate is not trusted by your PC. The browser may allow you to install the certificate, thus skipping the warning dialog the next time you connect to the MediaPack.

2. If you are using Internet Explorer, click **View Certificate** and then **Install Certificate**.
3. The browser also warns you if the host name used in the URL is not identical to the one listed in the certificate. To solve this, add the IP address and host name (ACL_nnnnnn where nnnnnn is the serial number of the MediaPack) to your hosts file, located at /etc/hosts on UNIX or C:\Windows\System32\Drivers\ETC\hosts on Windows; then use the host name in the URL (e.g., https://ACL_280152).The figure below is an example of a host file:

Figure 12-8: Example of a Host File

```
# This is a sample HOSTS file used by Microsoft TCP/IP for Windows.
# Location: C:\WINDOWS\SYSTEM32\DRIVERS\ETC\hosts
#
127.0.0.1     localhost
10.31.4.47    ACL_280152
```

12.2.3 Secured Telnet

To enable the embedded Telnet server on the MediaPack, set the parameter TelnetServerEnable (described in Table 5-37 on page 142) to 1 (standard mode) or 2 (SSL mode); no information is transmitted in the clear when SSL mode is used.

If the Telnet server is set to SSL mode, a special Telnet client is required on your PC to connect to the Telnet interface over a secured connection; examples include C-Kermit for UNIX, Kermit-95 for Windows, and AudioCodes' acSSLTelnet utility for Windows (that requires prior installation of the free OpenSSL toolkit). Contact AudioCodes to obtain the acSSLTelnet utility.

12.2.4 Server Certificate Replacement

The MediaPack is supplied with a working SSL configuration consisting of a unique self-signed server certificate. If an organizational Public Key Infrastructure (PKI) is used, you may wish to replace this certificate with one provided by your security administrator.

➢ **To replace the MediaPack self-signed certificate, take these 9 steps:**

1. Your network administrator should allocate a unique DNS name for the MediaPack (e.g., dns_name.corp.customer.com). This name is used to access the device, and should therefore be listed in the server certificate.
2. Open the 'Certificates' screen (**Advanced Configuration** menu > **Security Settings** submenu > **Certificates** option); the 'Certificates' screen is displayed (Figure 12-9).

Figure 12-9: Certificate Signing Request Screen

3. In the Subject Name field, enter the DNS name and click **Generate CSR**. A textual certificate signing request, that contains the SSL device identifier, is displayed.
4. Copy this text and send it to your security provider; the security provider (also known as Certification Authority or CA) signs this request and send you a server certificate for the device.
5. Save the certificate in a file (e.g., cert.txt). Ensure the file is a plain-text file with the 'BEGIN CERTIFICATE' header. The figure below is an example of a Base64-Encoded X.509 Certificate.

Figure 12-10: Example of a Base64-Encoded X.509 Certificate


```
-----BEGIN CERTIFICATE-----
MIIDkzCCAnugAwIBAgIEAgAAADANBgkqhkiG9w0BAQQFADA/MQswCQYDVQQGEwJG
UjETMBEGA1UEChMKQ2VydGlwb3N0ZTEbMBkGA1UEAxMSQ2VydGlwb3N0ZSBTZXJ2
ZXVyMB4XDTk4MDYyNDA4MDAwMFoXDTE4MDYyNDA4MDAwMFowPzELMAkGA1UEBhMC
RlIxEzARBgNVBAoTCkNlcnRpcG9zdGUxGzAZBgNVBAMTEkNlcnRpcG9zdGUgU2Vy
dmV1cjCCASEwDQYJKoZIhvcNAQEBBQADggEOADCCAQkCggEAPqd4MziR4spWldGR
x8bQrhZkonWnNm`+Yhb7+4Q67ecf1janH7GcN/SXsfx7jJpreWULf7v7Cvpr4R7qI
JcmdHIntmf7JPM5n6cDBv17uSW63er7NkVnMFHwK1QaGFLMybFkzaeGrvFm4k3lR
efiXDmuOe+FhJgHYezYHf44LvPRPwhSrzi9+Aq3o8pWDguJuZDIUP1F1jMa+LPwv
REXfFcUW+w==
-----END CERTIFICATE-----
```


6. Before continuing, set the parameter HTTPSOnly = 0 to ensure you have a method of accessing the device in case the new certificate doesn't work. Restore the previous setting after testing the configuration.
7. In the Certificates screen (Figure 12-9) locate the server certificate loading section.
8. Click **Browse** and navigate to the *cert.txt* file, click **Send File**.
9. When the operation is completed, save the configuration (Section 5.10.2 on page 205) and restart the MediaPack; the Embedded Web Server uses the provided certificate.

Notes:

- The certificate replacement process can be repeated when necessary (e.g., the new certificate expires).
- It is possible to use the IP address of the MediaPack (e.g., 10.3.3.1) instead of a qualified DNS name in the Subject Name. This practice is not recommended since the IP address is subject to changes and may not uniquely identify the device.
- The server certificate can also be loaded via *ini* file using the parameter 'HTTPSCertFileName'.

12.2.5 Client Certificates

By default, Web servers using SSL provide one-way authentication. The client is certain that the information provided by the Web server is authentic. When an organizational PKI is used, two-way authentication may be desired: both client and server should be authenticated using X.509 certificates. This is achieved by installing a client certificate on the managing PC, and loading the same certificate (in base64-encoded X.509 format) to the MediaPack Trusted Root Certificate Store. The Trusted Root Certificate file should contain both the certificate of the authorized user and the certificate of the CA.

Since X.509 certificates have an expiration date and time, the MediaPack must be configured to use NTP (Section 9.7 on page 253) to obtain the current date and time. Without a correct date and time, client certificates cannot work.

➢ To install a client certificate, take these 6 steps:

1. Before continuing, set HTTPSOnly = 0 to ensure you have a method of accessing the device in case the client certificate doesn't work. Restore the previous setting after testing the configuration.
2. Open the 'Certificates' screen (**Advanced Configuration** menu > **Security Settings** submenu > **Certificates** option); the 'Certificates' screen is displayed (Figure 12-9).
3. To load the Trusted Root Certificate file locate the trusted root certificate loading section.
4. Click **Browse**, and navigate to the file, and then click **Send File**.
5. When the operation is completed, set the *ini* file parameter, HTTPSRequireClientCertificates = 1.
6. Save the configuration (Section 5.10.2 on page 205) and restart the MediaPack.

When a user connects to the secure Web server:

- If the user has a client certificate from a CA listed in the Trusted Root Certificate file, the connection is accepted and the user is prompted for the system password.
- If both the CA certificate and the client certificate appear in the Trusted Root Certificate file, the user is not prompted for a password (thus providing a single-sign-on experience - the authentication is performed using the X.509 digital signature).
- If the user doesn't have a client certificate from a listed CA, or doesn't have a client certificate at all, the connection is rejected.

Notes:

- The process of installing a client certificate on your PC is beyond the scope of this document. For more information, refer to your Web browser or operating system documentation, and/or consult your security administrator.
- The root certificate can also be loaded via *ini* file using the parameter 'HTTPSRootFileName'.

12.3 SRTP

The gateway supports Secured RTP (SRTP) according to RFC 3711. SRTP is used to encrypt RTP and RTCP transport since it is best-suited for protecting VoIP traffic.

SRTP requires a Key Exchange mechanism that is performed according to <draft-ietf-mmusic-sdescriptions-12>. The Key Exchange is executed by adding a 'Crypto' attribute to the SDP. This attribute is used (by both sides) to declare the various supported cipher suites and to attach the encryption key to use. If negotiation of the encryption data is successful, the call is established.

Use the parameter MediaSecurityBehaviour (described in Table 5-50) to select the gateway's mode of operation: Must or Prefer. These modes determine the behavior of the gateway if negotiation of the cipher suite fails.

- Mandatory = the call is terminated. Incoming calls that don't include encryption information are rejected.
- Preferable = an unencrypted call is established. Incoming calls that don't include encryption information are accepted.

To enable SRTP set the parameter EnableMediaSecurity to 1 (described in Table 5-50).

Notes:

- When SRTP is used the channel capacity is reduced (refer to the parameter EnableMediaSecurity.
- The gateway only supports the AES 128 in CM mode cipher suite.

Figure 12-11: Example of crypto Attributes Usage

```
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:PsKoMpHlCg+b5X0YLuSvNrImEh/dAe
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline:IsPtLoGkBf9a+c6XVzRuMqHlDnEiAd
```

12.4 RADIUS Login Authentication

Users can enhance the security and capabilities of logging to the gateway's Web and Telnet embedded servers by using a Remote Authentication Dial-In User Service (RADIUS) to store numerous usernames, passwords and access level attributes (Web only), allowing multiple user management on a centralized platform. RADIUS (RFC 2865) is a standard authentication protocol that defines a method for contacting a predefined server and verifying a given name and password pair against a remote database, in a secure manner.

When accessing the Web and Telnet servers, users must provide a valid username and password. When RADIUS authentication isn't used, the username and password are authenticated with the Embedded Web Server's usernames and passwords of the primary or secondary accounts (refer to Section 5.2.1 on page 49) or with the Telnet server's username and password stored internally in the gateway's memory. When RADIUS authentication is used, the gateway doesn't store the username and password but simply forwards them to the pre-configured RADIUS server for authentication (acceptance or rejection). The internal Web / Telnet passwords can be used as a fallback mechanism in case the RADIUS server doesn't respond (configured by the parameter BehaviorUponRadiusTimeout). Note that when RADIUS authentication is performed, the Web / Telnet servers are blocked until a response is received (with a timeout of 5 seconds).

RADIUS authentication requires HTTP basic authentication, meaning the username and password are transmitted in clear text over the network. Therefore, users are recommended to set the parameter 'HttpsOnly = 1' to force the use of HTTPS, since the transport is encrypted.

12.4.1 Setting Up a RADIUS Server

The following examples refer to FreeRADIUS, a free RADIUS server that can be downloaded from www.freeradius.org. Follow the directions on that site for information on installing and configuring the server. If you use a RADIUS server from a different vendor, refer to its appropriate documentation.

➢ **To set up a RADIUS server, take these 5 steps:**

1. Define the gateway as an authorized client of the RADIUS server, with a predefined 'shared secret' (a password used to secure communication) and a vendor ID. The figure below displays an example of the file clients.conf (FreeRADIUS client configuration).

Figure 12-12: Example of the File clients.conf (FreeRADIUS Client Configuration)

```
#
# clients.conf - client configuration directives
#
client 10.31.4.47 {
        secret          = FutureRADIUS
        shortname       = tp1610_master_tpm
}
```

2. If access levels are required, set up a VSA dictionary for the RADIUS server and select an attribute ID that represents each user's access level. The following example shows a dictionary file for FreeRADIUS that defines the attribute 'ACL-Auth-Level' with ID=35.

Figure 12-13: Example of a Dictionary File for FreeRADIUS (FreeRADIUS Client Configuration)

```
#
# AudioCodes VSA dictionary
#
VENDOR AudioCodes 5003
ATTRIBUTE ACL-Auth-Level 35 integer AudioCodes
VALUE ACL-Auth-Level ACL-Auth-UserLevel 50
VALUE ACL-Auth-Level ACL-Auth-AdminLevel 100
VALUE ACL-Auth-Level ACL-Auth-SecurityAdminLevel 200
```

3. In the RADIUS server, define the list of users authorized to use the gateway, using one of the password authentication methods supported by the server implementation. The following example shows a user configuration file for FreeRADIUS using a plain-text password.

Figure 12-14: Example of a User Configuration File for FreeRADIUS Using a Plain-Text Password

```
# users - local user configuration database

john    Auth-Type := Local, User-Password == "qwerty"
        Service-Type = Login-User,
        ACL-Auth-Level = ACL-Auth-SecurityAdminLevel

larry   Auth-Type := Local, User-Password == "123456"
        Service-Type = Login-User,
        ACL-Auth-Level = ACL-Auth-UserLevel
```

4. Record and retain the IP address, port number, 'shared secret', vendor ID and VSA access level identifier (if access levels are used) used by the RADIUS server.
5. Configure the gateway's relevant parameters according to Section 12.4.2 below.

12.4.2 Configuring RADIUS Support

For information on the RADIUS parameters, refer to Table 5-50 on page 174.

- **To configure RADIUS support on the gateway via the Embedded Web Server, take these 12 steps:**

1. Access the Embedded Web Server (refer to Section 5.3 on page 51).
2. Open the 'General Security Settings' screen (**Advanced Configuration** menu > **Security Settings** > **General Security Settings** option); the 'General Security Settings' screen is displayed.
3. Under section 'General RADIUS Settings', in the field 'Enable RADIUS Access Control', select 'Enable'; the RADIUS application is enabled.
4. In the field 'Use RADIUS for Web / Telnet Login', select 'Enable'; RADIUS authentication is enabled for Web and Telnet login.
5. Enter the RADIUS server IP address, port number and shared secret in the relevant fields.

6. Under section 'RADIUS Authentication Settings', in the field 'Device Behavior Upon RADIUS Timeout', select the gateway's operation if a response isn't received from the RADIUS server after the 5 seconds timeout expires:
   - Deny Access – the gateway denies access to the Web and Telnet embedded servers.
   - Verify Access Locally – the gateway checks the local username and password.
7. In the field 'Local RADIUS Password Cache Timeout', enter a time (in seconds); when this time expires, the username and password verified by the RADIUS server becomes invalid and a username and password must be re-validated with the RADIUS server.
8. In the field 'Local RADIUS Password Cache Mode', select the gateway's mode of operation regarding the above-mentioned 'Local RADIUS Password Cache Timer' option:
   - Reset Timer Upon Access – upon each access to a Web screen, the timer resets (reverts to the initial value configured in the previous step).
   - Absolute Expiry Timer - when you access a Web screen, the timer doesn't reset but rather continues decreasing.
9. In the field 'RADIUS VSA Vendor ID', enter the vendor ID you configured in the RADIUS server:
10. When using the Web access-level mechanism, perform one of the following options:
    - When RADIUS responses include the access level attribute:
      In the field 'RADIUS VSA Access Level Attribute', enter the code that indicates the access level attribute in the Vendor Specific Attributes (VSA) section of the received RADIUS packet.
    - When RADIUS responses don't include the access level attribute:
      In the field 'Default Access Level', enter the default access level that is applied to all users authenticated by the RADIUS server.
11. In the field 'Require Secured Web Connection (HTTPS)', select 'HTTPS only'. It is important you use HTTPS (secure Web server) when connecting to the gateway over an open network, since the password is transmitted in clear text. Similarly, for Telnet, use SSL 'TelnetServerEnable = 2 (refer to Section 12.2.3 on page 289).
12. Save the changes and reset the gateway (refer to Section 5.10.3 on page 206).

After reset, when accessing the Web or Telnet servers, use the username and password you configured in the RADIUS database. The local system password is still active and can be used when the RADIUS server is down.

➢ To configure RADIUS support on the gateway using the *ini* file:

- Add the following parameters to the *ini* file. For information on modifying the *ini* file, refer to Section 6.2 on page 209.
  - EnableRADIUS = 1
  - WebRADIUSLogin = 1
  - RADIUSAuthServerIP = *IP address of RADIUS server*
  - RADIUSAuthPort = *port number of RADIUS server, usually 1812*
  - SharedSecret = *your shared secret'*
  - HTTPSOnly = 1
  - BehaviorUponRadiusTimeout = 1
  - RadiusLocalCacheMode = 1
  - RadiusLocalCacheTimeout = 300
  - RadiusVSAVendorID = *your vendor's ID*
  - RadiusVSAAccessAttribute = *code that indicates the access level attribute*
  - DefaultAccessLevel = *default access level (0 to 200)*

12.5 Internal Firewall

The MediaPack accommodates an internal access list facility, allowing the security administrator to define network traffic filtering rules. The access list provides the following features:

- Block traffic from known malicious sources
- Only allow traffic from known friendly sources, and block all others
- Mix allowed and blocked network sources
- Limit traffic to a predefined rate (blocking the excess)
- Limit traffic to specific protocols, and specific port ranges on the device

The access list consists of a table with up to 50 ordered lines. For each packet received on the network interface, the table is scanned from the top until a matching rule is found (or the table end is reached). This rule can either block the packet or allow it; however it is important to note that subsequent rules aren't scanned. If the table end is reached without a match, the packet is accepted.

Each rule is composed of the following fields (described in Table 5-49 on page 172):

- IP address (or DNS name) of source network
- IP network mask
- Destination UDP/TCP ports (on this device)
- Protocol type
- Maximum packet size, byte rate per second, and allowed data burst
- Action upon match (allow or block)

Figure 12-15 shows an example of an access list definition via *ini* file:

Figure 12-15: Example of an Access List Definition via *ini* File

```
[ ACCESSLIST ]
FORMAT AccessList_Index = AccessList_Source_IP, AccessList_Net_Mask,
AccessList_Start_Port, AccessList_End_Port, AccessList_Protocol,
AccessList_Packet_Size, AccessList_Byte_Rate, AccessList_Byte_Burst,
AccessList_Allow_Type;

AccessList 10 = mgmt.customer.com, 255.255.255.255, 0, 80, tcp, 0, 0, 0, allow ;
AccessList 15 = 192.0.0.0, 255.0.0.0, 0, 65535, any, 0, 40000, 50000, block ;
AccessList 20 = 10.31.4.0, 255.255.255.0, 4000, 9000, any, 0, 0, 0, block ;
AccessList 22 = 10.4.0.0, 255.255.0.0, 4000, 9000, any, 0, 0, 0, block ;
[ \ACCESSLIST ]
```

Explanation of the example access list:

- Rule #10: traffic from the host 'mgmt.customer.com' destined to TCP ports 0 to 80, is always allowed.
- Rule #15: traffic from the 192.xxx.yyy.zzz subnet, is limited to a rate of 40 Kbytes per second (with an allowed burst of 50 Kbytes). Note that the rate is specified in bytes, not bits, per second; a rate of 40000 bytes per second, nominally corresponds to 320 kbps.
- Rule #20: traffic from the subnet 10.31.4.xxx destined to ports 4000 to 9000 is always blocked, regardless of protocol.
- Rule #22: traffic from the subnet 10.4.xxx.yyy destined to ports 4000 to 9000 is always blocked, regardless of protocol.
- All other traffic is allowed.

More complex rules may be defined, relying on the 'single-match' process described above:

Figure 12-16 shows an advanced example of an access list definition via *ini* file:

Figure 12-16: Advanced Example of an Access List Definition via *ini* File

```
[ ACCESSLIST ]
FORMAT AccessList_Index = AccessList_Source_IP, AccessList_Net_Mask,
AccessList_Start_Port, AccessList_End_Port, AccessList_Protocol,
AccessList_Packet_Size, AccessList_Byte_Rate, AccessList_Byte_Burst,
AccessList_Allow_Type;

AccessList 10 = 10.0.0.0, 255.0.0.0, 0, 65535, any, 0, 40000, 50000, allow ;
AccessList 15 = 10.31.4.0, 255.255.255.0, 4000, 9000, any, 0, 0, 0, allow ;
AccessList 20 = 0.0.0.0, 0.0.0.0, 0, 65535, any, 0, 0, 0, block;
 [ \ACCESSLIST ]
```

Explanation of the example access list:

This access list consists of three rules:

- Rule #10: traffic from the subnet 10.xxx.yyy.zzz is allowed if the traffic rate does not exceed 40 KB/s.
- Rule #15: if a packet didn't match rule #10, that is, the excess traffic is over 40 KB/s, and coming from the subnet 10.31.4.xxx to ports 4000 to 9000, then it is allowed.
- Rule #20: all other traffic (which didn't match the previous rules), is blocked.

The internal firewall can also be configured via the Embedded Web Server (refer to Section 5.6.5.3 on page 171).

12.6 Network Port Usage

The following table lists the default TCP/UDP network port numbers used by the MediaPack. Where relevant, the table lists the *ini* file parameters that control the port usage and provide source IP address filtering capabilities.

Table 12-5: Default TCP/UDP Network Port Numbers

Port Number	Peer Port	Application	Notes
2	2	Debugging interface	Always ignored
23	-	Telnet	Disabled by default (TelnetServerEnable). Configurable (TelnetServerPort), access controlled by WebAccessList
68	67	DHCP	Active only if DHCPEnable = 1
80	-	Web server (HTTP)	Configurable (HTTPPort), can be disabled (DisableWebTask or HTTPSOnly). Access controlled by WebAccessList
161	-	SNMP GET/SET	Configurable (SNMPPort), can be disabled (DisableSNMP). Access controlled by SNMPTrustedMGR
443	-	Web server (HTTPS)	Configurable (HTTPSPort), can be disabled (DisableWebTask). Access controlled by WebAccessList
500	-	IPSec IKE	Can be disabled (EnableIPSec)
6000, 6010 and up	-	RTP traffic	Base port number configurable (BaseUDPPort), fixed increments of 10. The number of ports used depends on the channel capacity of the device.
6001, 6011 and up	-	RTCP traffic	Always adjacent to the RTP port number
6002, 6012 and up	-	T.38 traffic	Always adjacent to the RTCP port number
5060	5060	SIP	Configurable (LocalSIPPort [UDP], TCPLocalSIPPort [TCP]).
5061	5061	SIP over TLS (SIPS)	Configurable (TLSLocalSIPPort)
(random) > 32767	514	Syslog	Disabled by default (EnableSyslog).
(random) > 32767	-	Syslog ICMP	Disabled by default (EnableSyslog).
(random) > 32767	-	ARP listener	
(random) > 32767	162	SNMP Traps	Can be disabled (DisableSNMP)
(random) > 32767	-	DNS client	

12.7 Recommended Practices

To improve network security, the following guidelines are recommended when configuring the MediaPack:

- Set the password of the primary web user account (refer to 5.6.5.1 on page 168) to a unique, hard-to-hack string. Do not use the same password for several devices as a single compromise may lead to others. Keep this password safe at all times and change it frequently.
- If possible, use a RADIUS server for authentication. RADIUS allows you to set different passwords for different users of the MediaPack, with centralized management of the password database. Both Web and Telnet interfaces support RADIUS authentication (refer to Section 12.3 on page 293).
- If the number of users that access the Web and Telnet interfaces is limited, you can use the 'Web and Telnet Access List' to define up to ten IP addresses that are permitted to access these interfaces. Access from an undefined IP address is denied (refer to Section 5.6.5.2 on page 170).
- Use IPSec to secure traffic to all management and control hosts. Since IPSec encrypts all traffic, hackers cannot capture sensitive data transmitted on the network, and malicious intrusions are severely limited.
- Use HTTPS when accessing the Web interface. Set HTTPSOnly to 1 to allow only HTTPS traffic (and block port 80). If you don't need the Web interface, disable the Web server (DisableWebTask).
- If you use Telnet, do not use the default port (23). Use SSL mode to protect Telnet traffic from network sniffing.
- If you use SNMP, do not leave the community strings at their default values as they can be easily guessed by hackers (refer to Section 14.8.1 on page 313).
- Use a firewall to protect your VoIP network from external attacks. Network robustness may be compromised if the network is exposed to Denial of Service (DoS) attacks. DoS attacks are mitigated by Stateful firewalls. Do not allow unauthorized traffic to reach the MediaPack.

12.8 Legal Notice

By default, the MediaPack supports export-grade (40-bit and 56-bit) encryption due to US government restrictions on the export of security technologies. To enable 128-bit and 256-bit encryption on your device, contact your AudioCodes representative.

This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (www.openssl.org)

This product includes cryptographic software written by Eric Young' (eay@cryptsoft.com).

13 Diagnostics

Several diagnostic tools are provided, enabling you to identify correct functioning of the MediaPack, or an error condition with a probable cause and a solution or workaround.

- Front and rear-panel LEDs on the MediaPack. The MP-11x front-panel LEDs are described in Table 2-1 on page 26. The MP-124 front-panel LEDs are described in Table 2-4 on page 27, the rear-panel LEDs are described in Table 2-6 on page 28.
- Self-Testing on hardware initialization (refer to Section 13.1 below).
- Error / notification messages via the following interfaces:
    - Syslog: Log messages can be viewed using an external Syslog server, refer to Section 13.2 on page 301, or on the 'Message Log' screen in the Embedded Web Server, refer to Section 5.7.2 on page 192. Note that the 'Message Log' screen is not recommended for prolong debugging.
    - RS-232 terminal: For information on establishing a serial communications link with the MediaPack, refer to Section 10.2 on page 262.

13.1 Self-Testing

The MediaPack features two self-testing modes used to identify faulty hardware components:

- **Rapid:** The Rapid test is performed every time the MediaPack starts up. It is executed each time the MediaPack completes its initialization process. This is a short test phase in which the only error detected and reported is failure in initializing hardware components. If an error is detected, an error message is sent to the Syslog.
- **Detailed:** Used in addition to the Rapid and Enhanced test modes. The test is performed on startup, when initialization of the MediaPack is completed and if the parameter EnableDiagnostics is set to 1 or 2. In this mode, the MediaPack tests its DSPs, RAM and flash memory. When EnableDiagnostics is set to 1, flash is tested thoroughly, when EnableDiagnostics is set to 2, flash is only partially tested. While the Detailed test is running, the **Ready** and **Fail** LEDs are lit.
  If an error is detected, an error message is sent to the Syslog.

Warning: To continue regular operation the Detailed test must be disabled. Set the parameter EnableDiagnostics to 0 and reset the MediaPack.

13.2 MediaPack Line Testing

The MediaPack features a mechanism that performs tests on the telephone lines connected to FXS and FXO ports. These tests provide various line measurements.

Note: The line testing mechanism is performed on channel 1 and therefore, disconnects any call that is in progress on this channel. Therefore, it's recommended to perform the testing only when there are no calls in progress.

The following line tests are available on FXS gateways:

- Hardware revision number.
- Temperature (above or below limit, only if a thermometer is installed).
- Hook state.

- Coefficients checksum.
- Message waiting indication status.
- Ring state.
- Reversal polarity state.
- Line current (only on port 0).
- Line voltage between TIP and RING (only on port 0).
- 3.3 V reading (only on port 0).
- Ring voltage (only on port 0).
- Long line current (only on port 0).

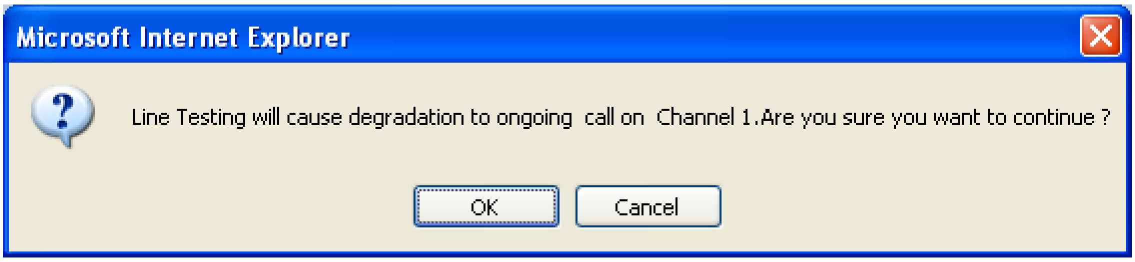
The following line tests are available on FXO gateways:

- Hardware revision number.
- Hook state.
- Reversal polarity state.

➢ To perform analog line testing using the embedded Web server, take these 2 steps:

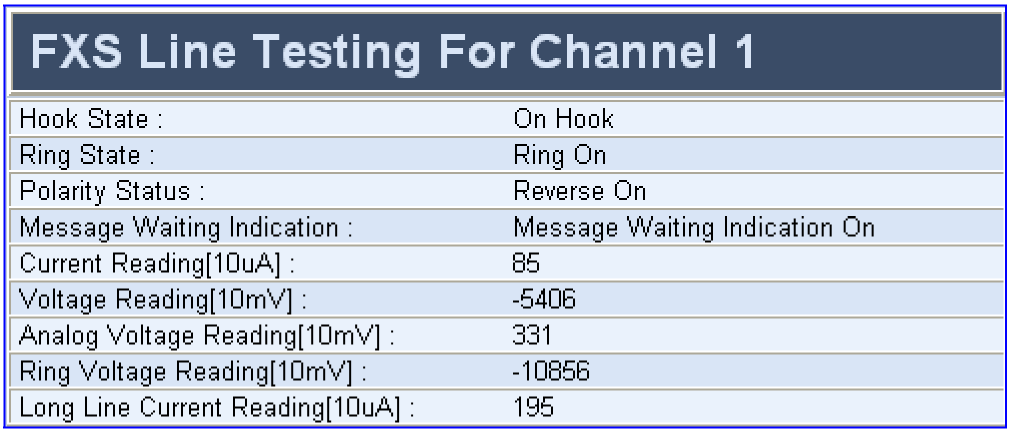
1. Click the **Analog Line Testing** submenu (**Status & Diagnostics** menu > **Analog Line Testing**); the Analog Line Testing confirmation box is displayed.

Figure 13-1: Analog Line Testing Confirmation Box



2. Click **OK** to confirm that you want to continue start the test; the 'FXS Line Testing For Channel 1' screen appears.

Figure 13-2: FXS Line Testing For Channel 1 Screen



FXS Line Testing For Channel 1

Hook State :	On Hook
Ring State :	Ring On
Polarity Status :	Reverse On
Message Waiting Indication :	Message Waiting Indication On
Current Reading[10uA] :	85
Voltage Reading[10mV] :	-5406
Analog Voltage Reading[10mV] :	331
Ring Voltage Reading[10mV] :	-10856
Long Line Current Reading[10uA] :	195

To perform the test again, click the **ReTest** button.

13.3 Syslog Support

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

Since each process, application and operating system was written 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 can be defined using SyslogServerPort parameter (default port is 514).

The Syslog message is transmitted as an ASCII (American Standard Code for Information Interchange) 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.

For example:

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

13.3.1 Syslog Servers

Users can use the provided AudioCodes Syslog server (ACSyslog) or any other third-party Syslog servers.

Examples of Syslog servers available as shareware on the Internet:

- Kiwi Enterprises: www.kiwisyslog.com/
- The US CMS Server: uscms.fnal.gov/hanlon/uscms_server/
- TriAction Software: www.triaction.nl/Products/SyslogDaemon.asp
- Netal SL4NT 2.1 Syslog Daemon: www.netal.com

A typical Syslog server application enables filtering of the messages according to priority, IP sender address, time, date, etc.

13.3.2 Operation

The Syslog client, embedded in the MediaPack, sends error reports/events generated by the MediaPack unit application to a Syslog server using IP/UDP protocol.

➢ To activate the Syslog client on the MediaPack, take these 5 steps:

1. Set the parameter 'EnableSyslog' to 1 (refer to Table 5-51 on page 177).
2. Use the parameter 'SyslogServerIP' to define the IP address of the Syslog server you use (refer to Table 5-51 on page 177).
3. Use the parameter 'SyslogServerPort' to define the UDP port number of the Syslog server (refer to Table 5-51 on page 177).
4. To determine the Syslog logging level use the parameter 'GWDebugLevel' (refer to Table 5-8 on page 78).
5. To enable the gateway to send log messages that report certain types of Web actions according to a pre-defined filter use the parameter 'ActivityListToLog' (described in Table 5-51 on page 177).

14 SNMP-Based Management

Simple Network Management Protocol (SNMP) is a standard-based network control protocol used to manage elements in a network. The SNMP Manager (usually implemented by a Network Manager (NM) or an Element Manager (EM)) connects to an SNMP Agent (embedded on a remote Network Element (NE)) to perform network element Operation, Administration and Maintenance (OAM).

Both the SNMP Manager and the NE refer to the same database to retrieve information or configure parameters. This database is referred to as the Management Information Base (MIB), and is a set of statistical and control values. Apart from the standard MIBs documented in IETF RFCs, SNMP additionally enables the use of private MIBs, containing a non-standard information set (specific functionality provided by the NE).

Directives, issued by the SNMP Manager 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 that identifier, or set the identifier to a new value (configuration). The SNMP Agent can also send unsolicited events towards the EM, called SNMP traps.

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 EM client programs so that they can become aware of MIB variables and their use.

The device contains an embedded SNMP Agent supporting both general network MIBs (such as the IP MIB), VoP-specific MIBs (such as RTP) and our proprietary MIBs (acBoard, acGateway, acAlarm and other MIBs), enabling a deeper probe into the inter-working of the device. All supported MIB files are supplied to customers as part of the release.

14.1 About SNMP

14.1.1 SNMP Message Standard

Four types of SNMP messages are defined:

- **Get:** A request that returns the value of a named object.
- **Get-Next:** A request that returns the next name (and value) of the 'next' object supported by a network device given a valid SNMP name.
- **Set:** A request that sets a named object to a specific value.
- **Trap:** A message generated asynchronously by network devices. It is an unsolicited message from an agent to the manager.

Each of these 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 an operant 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.

- **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 a ‘trap’ message) to notify or alert 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 above message types are all encoded into messages referred to as Protocol Data Units (PDUs) that are interchanged between SNMP devices.

14.1.2 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 the following four main branches:

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

The ‘tree’ structure described above is an integral part of the SNMP standard, though 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 sections 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. For "Discrete" objects, this instance number is zero. For "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, so 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 can 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).

14.1.3 SNMP Extensibility Feature

One of the principal components of an 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.

14.2 Carrier Grade Alarm System

The basic alarm system has been extended to a carrier-grade alarm system. A carrier-grade alarm system provides a reliable alarm reporting mechanism that takes into account EMS outages, network outages, and transport mechanism such as SNMP over UDP.

A carrier-grade alarm system is characterized by the following:

- The device has a mechanism that allows a manager to determine which alarms are currently active in the device. That is, the device maintains an active alarm table.
- The device has a mechanism to allow a manager to detect lost alarm raise and clear notifications [sequence number in trap, current sequence number MIB object].
- The device has a mechanism to allow a manager to recover lost alarm raise and clear notifications [maintains a log history].
- The device sends a cold start trap to indicate that it is starting. This allows the EMS to synchronize its view of the device's active alarms.

When the SNMP alarm traps are sent, the carrier-grade alarm system does not add or delete alarm traps as part of the feature. This system provides the mechanism for viewing of history and current active alarm information.

14.2.1 Active Alarm Table

The device maintains an active alarm table to allow a manager to determine which alarms are currently active in the device. Two views of the active alarm table are supported by the agent:

- acActiveAlarmTable in the enterprise acAlarm
- alarmActiveTable and alarmActiveVariableTable in the IETF standard ALARM-MIB (rooted in the AC tree)

The acActiveAlarmTable is a simple, one-row per alarm table that is easy to view with a MIB browser.

The ALARM-MIB is currently a draft standard and therefore has no OID assigned to it. In the current software release, the MIB is rooted in the experimental MIB subtree. In a future release, after the MIB has been ratified and an OID assigned, it is to move to the official OID

14.2.2 Alarm History

The device maintains a history of alarms that have been raised and traps that have been cleared to allow a manager to recover any lost, raised or cleared traps. Two views of the alarm history table are supported by the agent:

- acAlarmHistoryTable in the enterprise acAlarm
- nlmLogTable and nlmLogVariableTable in the standard NOTIFICATION-LOG-MIB

As with the acActiveAlarmTable, the acAlarmHistoryTable is a simple, one-row-per-alarm table that is easy to view with a MIB browser.

14.3 Cold Start Trap

MediaPack technology supports a cold start trap to indicate that the device is starting. This allows the manager to synchronize its view of the device's active alarms. Two different traps are sent at start-up:

- The standard coldStart trap - iso(1).org(3).dod(6).internet(1). snmpV2(6). snmpModules(3). snmpMIB(1). snmpMIBObjects(1). snmpTraps(5). coldStart(1) - sent at system initialization.
- The enterprise acBoardEvBoardStarted which is generated at the end of system initialization. This is more of an 'application-level' cold start sent after the entire initializing process is complete and all the modules are ready.

14.4 Third-Party Performance Monitoring Measurements

Performance measurements are available for a third-party performance monitoring system through an SNMP interface. These measurements can be polled at scheduled intervals by an external poller or utility in a media server or other off-device system.

The device provides two types of performance measurements:

- **Gauges:** Gauges represent the current state of activities on the device. Gauges, unlike counters, can decrease in value, and like counters, can increase. The value of a gauge is the current value or a snapshot of the current activity on the device.
- **Counters:** Counters always increase in value and are cumulative. Counters, unlike gauges, never decrease in value unless the off-device system is reset, the counters are then zeroed.

Performance measurements are provided by several proprietary MIBs that are located under the 'performance' sub tree:

iso(1).org(3).dod(6).internet(1).private(4).enterprises(1).audioCodes(5003).acPerformance(10).

Two formats of performance monitoring MIBs are available:

- Old format (obsolete as of version 4.6):
  Each MIB is composed of a list of single MIB objects, each relates to a separate attribute within a gauge or a counter. All counters and gauges provide the current time value only.
  - acPerfMediaGateway - a generic-type of PM MIB that covers:
    - Control protocol
    - RTP stream
    - System packets statistics
  - acPerfMediaServices - Media services devices specific performance MIB.
  - acPerfH323SIPGateway – holds statistics on Tel to IP and vice versa.
- New format:
  The following MIBs feature an identical structure. Each includes two major sub-trees.
  - Configuration sub tree – enables configuration of general attributes of the MIB and specific attributes of the monitored objects.
  - Data sub tree

  The monitoring results are presented in tables. Each table includes one or two indices. When there are two indices, the first index is a sub-set in the table (e.g., trunk number) and the second (or a single where there is only one) index represents the interval number (present - 0, previous - 1 and the one before - 2).

The MIBs include:

- acPMMedia: for media (voice) related monitoring (e.g., RTP, DSP's).
- acPMControl: for Control-Protocol related monitoring (e.g., connections, commands).
- acPMAnalog: for analog channels in offhook state.
- acPMSystem: for general (system related) monitoring.

The log trap, acPerformanceMonitoringThresholdCrossing (non-alarm), is sent out every time the threshold of a Performance Monitored object is crossed. The severity field is 'indeterminate' when the crossing is above the threshold and 'cleared' when it falls bellow the threshold. The 'source' varbind in the trap indicates the object for which the threshold is being crossed.

14.5 Total Counters

The counter's attribute 'total' accumulates counter values since the board's most recent restart. The user can reset the total's value by setting the Reset-Total object.

Each MIB module has its own Reset Total object, as follows:

- **PM-Analog:** acPMAnalogConfigurationResetTotalCounters
- **PM-Control:** acPMControlConfigurationResetTotalCounters
- **PM-Media:** acPMMediaConfigurationResetTotalCounters
- **PM-PSTN:** acPMPSTNConfigurationResetTotalCounters
- **PM-System:** acPMSystemConfigurationResetTotalCounters

14.6 Supported MIBs

The MediaPack contains an embedded SNMP Agent supporting the following MIBs:

- **Standard MIB (MIB-2):** 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 indicators.
- **RTP MIB:** The RTP MIB is supported in conformance with the IETF RFC 2959. It contains objects relevant to the RTP streams generated and terminated by the device and to RTCP information related to these streams.
- **NOTIFICATION-LOG-MIB:** This standard MIB (RFC 3014 - iso.org.dod.internet.mgmt.mib-2) is supported as part of our implementation of carrier grade alarms.
- **ALARM-MIB:** This is an IETF MIB (RFC 3877) also supported as part of our implementation of carrier grade alarms. This MIB is a new standard and is therefore under the audioCodes.acExperimental branch.
- **SNMP-TARGET-MIB:** According to RFC 2273. It allows for the configuration of trap destinations and trusted managers.
- **SNMP MIB:** This MIB (RFC 3418) allows support of the coldStart and authenticationFailure traps.
- **SNMP Framework MIB:** (RFC 3411).
- **SNMP Usm MIB:** this MIB (RFC 3414) implements the user-based Security Model.
- **SNMP Vacm MIB:** This MIB (RFC 3415) implements the view-based Access Control Model.
- **SNMP Community MIB:** This MIB (RFC 3584). implements community string management.
- **ipForward MIB:** (RFC 2096) fully supported

- **RTCP-XR:** This MIB (RFC) implements the following partial support:
  - The rtcpXrCallQualityTable is fully supported.
  - In the rtcpXrHistoryTable, support of the RCQ objects is provided only with no more than 3 intervals, 15 minutes long each.
  - Supports the rtcpXrVoipThresholdViolation trap.

In addition to the standard MIBs, the complete product series contains proprietary MIBs:

- **AC-TYPES MIB:** lists the known types defined by the complete product series. This is referred to by the sysObjectID object in the MIB-II.

Note: The acBoard MIB is still supported, but is being replaced by five newer proprietary MIBs.

As noted above, five new MIBs cover the device's general parameters. Each contains a Configuration subtree for configuring related parameters. In some, there also are Status and Action subtrees.

The 5 MIBs are:

- AC-ANALOG-MIB
- AC-CONTROL-MIB
- AC-MEDIA-MIB
- AC-PSTN-MIB
- AC-SYSTEM-MIB

Other proprietary MIBs are:

- **acGateway MIB:** This proprietary MIB contains objects related to configuration of the device when applied as a SIP or H.323 media gateway only. This MIB complements the other proprietary MIBs.

  The acGateway MIB has the following groups:
  - Common - for parameters common to both SIP and H.323
  - SIP - for SIP parameters only
  - H.323 - for H.323 parameters only
- **acAlarm:** This is a proprietary carrier-grade alarm MIB. It is a simpler implementation of the notificationLogMIB and the IETF suggested alarmMIB (both also supported in all MediaPack and related devices).

  The acAlarm MIB has the following groups:
  - ActiveAlarm - straightforward (single-indexed) table, listing all currently active alarms, together with their bindings (the alarm bindings are defined in acAlarm. acAlarmVarbinds and also in acBoard.acTrap. acBoardTrapDefinitions. oid_1_3_6_1_4_1_5003_9_10_1_21_2_0).
  - acAlarmHistory - straightforward (single-indexed) table, listing all recently raised alarms together with their bindings (the alarm bindings are defined in acAlarm. acAlarmVarbinds and also in acBoard.acTrap. acBoardTrapDefinitions. oid_1_3_6_1_4_1_5003_9_10_1_21_2_0).

  The table size can be altered via notificationLogMIB.notificationLogMIBObjects.nlmConfig.nlmConfigGlobalEntryLimit or notificationLogMIB.notificationLogMIBObjects.nlmConfig.nlmConfigLogTable.nlm ConfigLogEntry.nlmConfigLogEntryLimit.

  The table size can be any value from 10 to 100 and is 100 by default.

Note 1: The following are special notes pertaining to MIBs:

- A detailed explanation of each parameter can be viewed in an SNMP browser in the 'MIB Description' field.
- Not all groups in the MIB are functional. Refer to version release notes.
- Certain parameters are non-functional. Their MIB status is marked 'obsolete'.
- When a parameter is set to a new value via SNMP, the change may affect device functionality immediately or may require that the device be soft reset for the change to take effect. This depends on the parameter type.

Note 2: The current (updated) device configuration parameters are programmed into the device provided that the user does not load an *ini* file to the device after reset. Loading an *ini* file after reset overrides the updated parameters.

Additional MIBs are to be supported in future releases.

14.7 Traps

Note: As of this version all traps are sent from the SNMP port (default 161). This is part of the NAT traversal solution.

Full proprietary trap definitions and trap Varbinds are found in the acBoard and acAlarm MIBs. The following proprietary traps are supported. For detailed information on these traps, refer to Appendix F on page 375:

Table 14-1: Proprietary Traps Description

Trap	Description
acBoardFatalError	Sent whenever a fatal device error occurs.
acBoardConfigurationError	Sent when a device's settings are illegal. The trap contains a message describing the illegality of the setting. Note: Not applicable to IPM-260).
acBoardTemperatureAlarm	Sent when a board exceeds its temperature limits.
acBoardEvResettingBoard	Sent after the device is reset.
acBoardEvBoardStarted	Sent after the device is successfully restored and initialized following reset.
acgwAdminStateChange	Sent when Graceful Shutdown commences and ends.
acOperationalStateChange	Sent if the operational state of the node changes to disabled. Cleared when the operational state of the node changes to enabled.
acBoardCallResourcesAlarm	Sent when no free channels are available.
acBoardControllerFailureAlarm	Sent when the Gatekeeper/Proxy is not found or registration failed. Internal routing table can be used for routing.
acFeatureKeyError	Intended to relay Feature Key errors etc. (will be supported in the next applicable release).
acBoardOverloadAlarm	Sent when an overload in one or more of the system's components occurs.
acActiveAlarmTableOverflow	Sent to indicate that an active alarm could not be entered into the Active Alarm table because the table was full.
acKeepAlive	Part of the NAT traversal mechanism. If the STUN application detects a NAT, this trap is sent on regular time laps - 9/10 of the acSysSTUNBindingLifeTime object. The AdditionalInfo1 varbind has the MAC address of the gateway.
acNATTraversalAlarm	Sent when the NAT is placed in front of a gateway that is identified as a symmetric NAT. It is cleared when a non-symmetric NAT or no NAT replaces the symmetric one.
acEnhancedBITStatus	This trap is used to indicate the status of the Built In Test (BIT). The information in the trap contains board hardware elements being tested and their status. The information is presented in the additional info fields.
acBoardEthernetLinkAlarm	Sent when an Ethernet link(s) is down.
acPerformanceMonitoringThresholdCrossing	Sent every time the threshold of a Performance Monitored object is crossed. The severity field is 'indeterminate' when the crossing is above the threshold, and 'cleared' when it goes back under the threshold. The 'source' varbind in the trap indicates the object for which the threshold is being crossed.
acHTTPDownloadResult	Sent at the success or failure of HTTP download.

In addition to the listed traps, the device also supports the following standard traps:

- coldStart
- authenticationFailure
- linkDown
- linkup
- entConfigChange

14.8 SNMP Interface Details

This section describes details of the SNMP interface that is required when developing an Element Manager (EM) for any of the TrunkPack-VoP Series products, or to manage a device with a MIB browser.

The gateway offers the following SNMP security features:

- SNMPv2c community strings
- SNMPv3 User-based Security Model (USM) users
- SNMP encoded over IPSec (refer to Section 12.1 on page 279)
- Combinations of the above

Currently, both SNMP and *ini* file commands and downloads are not encrypted. For *ini* file encoding, refer to Section E.1.2 on page 365.

14.8.1 SNMP Community Names

By default, the device uses a single, read-only community string of ‘public’ and a single read-write community string of ‘private’.

The following community strings can be defined:

- Up to five read-only community strings
- Up to five read-write community strings
- A single trap community string

Each community string must be associated with one of the following predefined SNMP groups:

Table 14-2: SNMP Predefined Groups

Group	Gets Access	Sets Access	Sends Traps
ReadGroup	Yes	No	Yes
ReadWriteGroup	Yes	Yes	Yes
TrapGroup	No	No	Yes

14.8.1.1 Configuration of Community Strings via the Web

For detailed information on configuration the community strings via the Embedded Web Server, refer to Section 5.6.6.2 on page 180.

14.8.1.2 Configuration of Community Strings via the *ini* File

The following *ini* file parameters are used to configure community strings:

- SNMPReadOnlyCommunityString_<x> = '#######'
- SNMPReadWriteCommunityString_<x> = '#######'

Where <x> is a number from 0 to 4. The '#' character represents any alphanumeric character. The maximum length of the string is 20 characters.

14.8.1.3 Configuration of Community Strings via SNMP

To configure read-only and read-write community strings, the EM must use the SNMP-COMMUNITY-MIB. To configure the trap community string, the EM must also use the snmpVacmMIB and the snmpTargetMIB.

➢ **To add a read-only community string (v2user), take this step:**

- Add a new row to the srCommunityTable with CommunityName v2user and GroupName ReadGroup.

➢ **To delete the read-only community string (v2user), take these 2 steps:**

1. If v2user is being used as the trap community string, follow the procedure for changing the trap community string (see below).
2. Delete the srCommunityTable row with CommunityName v2user.

➢ **To add a read-write community string (v2admin), take this step:**

- Add a new row to the srCommunityTable with CommunityName of v2admin and GroupName ReadWriteGroup.

➢ **To delete the read-write community string (v2admin), take these 2 steps:**

1. If v2admin is being used as the trap community string, follow the procedure for changing the trap community string. (See below.)
2. Delete the srCommunityTable row with a CommunityName of v2admin and GroupName of ReadWriteGroup.

➢ **To change the only read-write community string from v2admin to v2mgr, take these 4 steps:**

1. Follow the procedure above to add a read-write community string to a row for v2mgr.
2. Set up the EM so that subsequent ‘set’ requests use the new community string, v2mgr.
3. If v2admin is being used as the trap community string, follow the procedure to change the trap community string (see below).
4. Follow the procedure above to delete a read-write community name in the row for v2admin.

The following procedure assumes that a row already exists in the srCommunityTable for the new trap community string. The trap community string can be part of the TrapGroup, ReadGroup or ReadWriteGroup. If the trap community string is used solely for sending traps (recommended), it should be made part of the TrapGroup.

➢ **To change the trap community string, take these 2 steps:**

1. Add a row to the vacmSecurityToGroupTable with these values: SecurityModel=2, SecurityName=the new trap community string, GroupName=TrapGroup, ReadGroup or ReadWriteGroup. The SecurityModel and SecurityName objects are row indices.
2. Modify the SecurityName field in the sole row of the snmpTargetParamsTable.

Note: You must add GroupName and RowStatus on the same set.

14.8.2 SNMP v3 USM Users

You can define up to 10 User-based Security Model (USM) users (USM users are referred to as “v3 users”). Each v3 user can be associated with an authentication type (none, MD5, or SHA-1) and a privacy type (none, DES, 3DES, or AES).

Table 14-3: SNMP v3 Security Levels

Security Level	Authentication	Privacy
noAuthNoPriv(1)	None	None
authNoPriv(2)	MD5 or SHA-1	None
authPriv(3)	MD5 or SHA-1	DES, 3DES, AES128, AES192, or AES256

Each SNMP v3 user must be associated with one of the predefined groups listed in the following table:

Table 14-4: SNMP v3 Predefined Groups

Group	Get Access	Set Access	Send Traps	Security Level
ReadGroup1	Yes	No	Yes	noAuthNoPriv(1)
ReadWriteGroup1	Yes	Yes	Yes	noAuthNoPriv(1)
TrapGroup1	No	No	Yes	noAuthNoPriv(1)
ReadGroup2	Yes	No	Yes	authNoPriv(2)
ReadWriteGroup2	Yes	Yes	Yes	authNoPriv(2)
TrapGroup2	No	No	Yes	authNoPriv(2)
ReadGroup3	Yes	No	Yes	authPriv(3)
ReadWriteGroup3	Yes	Yes	Yes	authPriv(3)
TrapGroup3	No	No	Yes	authPriv(3)

14.8.2.1 Configuring SNMP v3 Users via the *ini* File

Use the SNMPUsers *ini* table to add, modify, and delete SNMPv3 users. For a description of the SNMPUsers table *ini* file parameters, refer to Section 5.6.6.3 on page 181.

> **Note:** The SNMPUsers *ini* table is a hidden parameter. Therefore, when you perform a "Get ini File" operation using the Web interface, the table will not be included in the generated file.

You can enter keys in the form of a text password or in the form of a localized key in hex format. If using a text password, then it should be at least eight characters in length. Below is an example of a localized key format:

```
26:60:d8:7d:0d:4a:d6:8c:02:73:dd:22:96:a2:69:df
```

The following example configuration creates three SNMPv3 USM users:

```
[ SNMPUsers ]
FORMAT SNMPUsers_Index = SNMPUsers_Username, SNMPUsers_AuthProtocol,
SNMPUsers_PrivProtocol, SNMPUsers_AuthKey, SNMPUsers_PrivKey, SNMPUsers_Group;
SNMPUsers 0 = v3user, 0, 0, -, -, 0;
SNMPUsers 1 = v3admin1, 1, 0, myauthkey, -, 1;
SNMPUsers 2 = v3admin2, 2, 1, myauthkey, myprivkey, 1;
[ \SNMPUsers ]
```

The example above creates the following three v3 users:

- The user "v3user" is defined for a security level of noAuthNoPriv(1) and is associated with ReadGroup1.
- The user "v3admin1" is defined for a security level of authNoPriv(2) with authentication protocol MD5. The authentication text password is "myauthkey" and the user will be associated with ReadWriteGroup2.
- The user "v3admin2" is defined for a security level of authPriv(3) with authentication protocol SHA-1 and privacy protocol DES. The authentication text password is "myauthkey", the privacy text password is "myprivkey", and the user will be associated with ReadWriteGroup3.

14.8.2.2 Configuring SNMP v3 Users via SNMP

To configure SNMP v3 users, the EM must use the standard snmpUsmMIB and the snmpVacmMIB.

➢ To add a read-only, noAuthNoPriv SNMPv3 user (v3user), take these 3 steps:

1. Clone the row with the same security level. After the clone step, the status of the row is notReady(3).
2. Activate the row (i.e., set the row status to active(1)).
9. Add a row to the vacmSecurityToGroupTable for SecurityName v3user, GroupName ReadGroup1, and SecurityModel usm(3).

> **Note:** A row with the same security level (noAuthNoPriv) must already exist in the usmUserTable. (See the usmUserTable for details).

➢ To delete the read-only, noAuthNoPriv SNMPv3 user (v3user), take these 3 steps:

1. If v3 user is associated with a trap destination, follow the procedure for associating a different user to that trap destination. (See below.)
2. Delete the vacmSecurityToGroupTable row for SecurityName v3user, GroupName ReadGroup1, and SecurityModel usm.
3. Delete the row in the usmUserTable for v3user.

➢ To add a read-write, authPriv SNMPv3 user (v3user), take these 4 steps:

1. Clone the row with the same security level.
2. Change the authentication key and privacy key.
3. Activate the row. That is, set the row status to active(1).
4. Add a row to the vacmSecurityToGroupTable for SecurityName v3admin1, GroupName ReadWriteGroup3, and SecurityModel usm(3).

Note: A row with the same security level (authPriv) must already exist in the usmUserTable (see the usmUserTable for details).

➢ To delete the read-write, authPriv SNMPv3 user (v3admin1), take these 3 steps:

1. If v3admin1 is associated with a trap destination, follow the procedure for associating a different user to that trap destination. (See below.)
2. Delete the vacmSecurityToGroupTable row for SecurityName v3admin1, GroupName ReadWriteGroup1, and SecurityModel usm.
3. Delete the row in the usmUserTable for v3admin1.

14.8.3 Trusted Managers

By default, the agent accepts ‘get’ and ‘set’ requests from any IP address, as long as the correct community string is used in the request. Security can be enhanced via the use of Trusted Managers. A Trusted Manager is an IP address from which the SNMP Agent accepts and processes ‘get’ and ‘set’ requests. An EM can be used to configure up to five Trusted Managers.

Note: If Trusted Managers are defined, all community strings work from all Trusted Managers. That is, there is no way to associate a community string with particular trusted managers.

The concept of trusted managers is considered to be a weak form of security and is therefore, not a required part of SNMPv3 security, which uses authentication and privacy. However, the board’s SNMP agent applies the trusted manager concept as follows:

- There is no way to configure trusted managers for only a SNMPv3 user. An SNMPv2c community string must be defined.
- If specific IPs are configured as trusted managers (via the community table), then only SNMPv3 users on those trusted managers are given access to the agent’s MIB objects.

14.8.3.1 Configuration of Trusted Managers via *ini* File

To set the Trusted Mangers table from start-up, write the following in the *ini* file:

SNMPTRUSTEDMGR_X = D.D.D.D

where X is any integer between 0 and 4 (0 sets the first table entry, 1 sets the second, and so on), and D is an integer between 0 and 255.

14.8.3.2 Configuration of Trusted Managers via SNMP

To configure Trusted Managers, the EM must use the SNMP-COMMUNITY-MIB, the snmpTargetMIB and the TGT-ADDRESS-MASK-MIB.

- **To add the first Trusted Manager, take these 3 steps:**

(The following procedure assumes that there is at least one configured read-write community. There are currently no Trusted Managers. The taglist for columns for all srCommunityTable rows are currently empty).

1. Add a row to the snmpTargetAddrTable with these values: Name=mgr0, TagList=MGR, Params=v2cparams.
2. Add a row to the tgtAddressMaskTable table with these values: Name=mgr0, tgtAddressMask=255.255.255.255:0. The agent doesn't allow creation of a row in this table unless a corresponding row exists in the snmpTargetAddrTable.
3. Set the value of the TransportLabel field on each non-TrapGroup row in the srCommunityTable to MGR.

The following procedure assumes that there is at least one configured read-write community. There are currently one or more Trusted Managers. The taglist for columns for all rows in the srCommunityTable are currently set to MGR. This procedure must be performed from one of the existing Trusted Managers.

- **To add a subsequent Trusted Manager, take these 2 steps:**

1. Add a row to the snmpTargetAddrTable with these values: Name=mgrN, TagList=MGR, Params=v2cparams, where N is an unused number between 0 and 4.
2. Add a row to the tgtAddressMaskTable table with these values: Name=mgrN, tgtAddressMask=255.255.255.255:0.

   An alternative to the above procedure is to set the tgtAddressMask column while you are creating other rows in the table.

The following procedure assumes that there is at least one configured read-write community. There are currently two or more Trusted Managers. The taglist for columns for all rows in the srCommunityTable are currently set to MGR. This procedure must be performed from one of the existing trusted managers, but not the one that is being deleted.

- **To delete a Trusted Manager (not the final one), take this step:**

- Remove the appropriate row from the snmpTargetAddrTable.

The change takes effect immediately. The deleted trusted manager cannot access the device. The agent automatically removes the row in the tgtAddressMaskTable.

The following procedure assumes that there is at least one configured read-write community. There is currently only one Trusted Manager. The taglist for columns for all rows in the srCommunityTable are currently set to MGR. This procedure must be performed from the final Trusted Manager.

➢ **To delete the final Trusted Manager, take these 2 steps:**

1. Set the value of the TransportLabel field on each row in the srCommunityTable to the empty string.
2. Remove the appropriate row from the snmpTargetAddrTable

The change takes effect immediately. All managers can now access the device.

14.8.4 SNMP Ports

The SNMP Request Port is 161 and the Trap Port is 162. These ports can be changed by setting parameters in the device *ini* file. The parameter name is:

SNMPPort = <port_number>
Valid UDP port number; default = 161

This parameter specifies the port number for SNMP requests and responses. Usually, it should not be specified. Use the default.

14.8.5 Multiple SNMP Trap Destinations

An agent can send traps to up to five managers. For each manager, set the manager's IP address, receiving port number, and enable sending traps to that manager.

The user also has the option of associating a trap destination with a specific SNMPv3 USM user. Traps are then sent to that trap destination using the SNMPv3 format and the authentication and privacy protocol configured for that user.

To configure the trap managers table use:

- The Embedded Web Server, refer to Section 5.6.6.1 on page 178.
- The *ini* file, refer to Section 14.8.1.1 below.
- SNMP, refer to Section 14.8.1.3 on page 314.

14.8.5.1 Configuring Trap Manager via Host Name

One of the five available SNMP managers can be defined using a FQDN. In the current version, this option can only be configured via the *ini* file (SNMPTrapManagerHostName).

The gateway tries to resolve the host name at start up. Once the name is resolved (IP is found), the resolved IP address replaces the last entry in the trap manager table (defined by the parameter 'SNMPManagerTableIP_x') and the last trap manager entry of snmpTargetAddrTable in the snmpTargetMIB. The port is 162 (unless specified otherwise), the row is marked as 'used' and the sending is 'enabled'.

When using 'host name' resolution, any changes made by the user to this row in either MIBs are overwritten by the gateway when a resolving is redone (once an hour).

Note that several traps may be lost until the resolving is complete.

14.8.5.2 Configuring Trap Managers via the *ini* File

In the MediaPack *ini* file, the parameters below can be set to enable or disable the sending of SNMP traps. Multiple trap destinations can be supported on the device by setting multiple trap destinations in the *ini* file.

- **SNMPManagerTrapSendingEnable_<x>:** indicates whether or not traps are to be sent to the specified SNMP trap manager. A value of '1' means that it is enabled, while a value of '0' means disabled. <x> represents a number 0, 1, 2 which is the array element index. Currently, up to five SNMP trap managers can be supported.
- **SNMPManagerTrapUser_<x>**: indicates to send an SNMPv2 trap using the trap user community string configured with the SNMPTrapCommunityString parameter. The user may instead specify an SNMPv3 user name

Figure 14-1 presents an example of entries in a device *ini* file regarding SNMP. The device can be configured to send to multiple trap destinations. The lines in the file below are commented out with the ';' at the beginning of the line. All of the lines below are commented out since the first line character is a semi-colon.

Figure 14-1: Example of Entries in a Device *ini* file Regarding SNMP

```
; SNMP trap destinations
; The board maintains a table of trap destinations containing 5 ;rows. The rows
are numbered 0..4. Each block of 4 items below ;apply to a row in the table.

; To configure one of the rows, uncomment all 4 lines in that ;block. Supply an
IP address and if necessary, change the port ;number.
; To delete a trap destination, set ISUSED to 0.
; -change these entries as needed
;SNMPManagerTableIP_0=
;SNMPManagerTrapPort_0=162
;SNMPManagerIsUsed_0=1
;SNMPManagerTrapSendingEnable_0=1
;
;SNMPManagerTableIP_1=
;SNMPManagerTrapPort_1=162
;SNMPManagerIsUsed_1=1
;SNMPManagerTrapSendingEnable_1=1
;
;SNMPManagerTableIP_2=
;SNMPManagerTrapPort_2=162
;SNMPManagerIsUsed_2=1
;SNMPManagerTrapSendingEnable_2=1
;
;SNMPManagerTableIP_3=
;SNMPManagerTrapPort_3=162
;SNMPManagerIsUsed_3=1
;SNMPManagerTrapSendingEnable_3=1
;
;SNMPManagerTableIP_4=
;SNMPManagerTrapPort_4=162
;SNMPManagerIsUsed_4=1
;SNMPManagerTrapSendingEnable_4=1
```

To configure the trap manger host name use the parameter SNMPTrapManagerHostName. For example: SNMPTrapManagerHostName = 'myMananger.corp.MyCompany.com'.

Note: The same information configurable in the *ini* file can also be configured via the acBoardMIB.

14.8.5.3 Configuring Trap Managers via SNMP

The standard snmpTargetMIB interface is available for configuring trap managers.

Note: The acBoard MIB is planned to become obsolete. The only relevant section in this MIB is the trap sub tree acTrap.

➢ To add an SNMPv2 trap destination, take the following step:

- Add a row to the snmpTargetAddrTable with these values:
  Name=trapN, TagList=AC_TRAP, Params=v2cparams, where N is an unused number between 0 and 4.

  All changes to the trap destination configuration take effect immediately.

➢ To add an SNMPv3 trap destination, take these 2 steps:

1. Add a row to the snmpTargetAddrTable with these values:
   - Name=trapN, where N is an unused number between 0 and 4
   - TagList=AC_TRAP
   - Params=usm<user>, where <user> is the name of the SNMPv3 with which this user is associated
2. If a row does not already exist for this combination of user and SecurityLevel, add a row to the snmpTargetParamsTable with these values:
   - Name=usm<user>
   - MPModel=3(SNMPv3)
   - SecurityModel=3 (usm)
   - SecurityName=<user>
   - SecurityLevel=M, where M is either 1(noAuthNoPriv), 2(authNoPriv), or 3(authPriv)

All changes to the trap destination configuration take effect immediately.

➢ To delete a trap destination, take the following step:

- Remove the appropriate row from the snmpTargetAddrTable.

You can change the IP address and/or port number for an existing trap destination. The same effect can be achieved by removing a row and adding a new row

➢ To modify a trap destination, take the following step:

- Modify the IP address and/or port number for the appropriate row in the snmpTargetAddrTable.

➢ To disable a trap destination, take the following step:

- Change TagList on the appropriate row in the snmpTargetAddrTable to the empty string.

➢ To enable a trap destination, take the following step:

- Change TagList on the appropriate row in the snmpTargetAddrTable to ‘AC_TRAP’.

14.9 SNMP Manager Backward Compatibility

With support for the Multi Manager Trapping feature, the older acSNMPManagerIP MIB object, synchronized with the first index in the snmpManagers MIB table, is also supported. This is translated in two features:

- SET/GET to either of the two MIB objects is identical.
  i.e., as far as the SET/GET are concerned OID 1.3.6.1.4.1.5003.9.10.1.1.2.7 is identical to OID 1.3.6.1.4.1.5003.9.10.1.1.2.21.1.1.3.
- When setting ANY IP to the acSNMPManagerIP (this is the older parameter, not the table parameter), two more parameters are SET to ENABLE. snmpManagerIsUsed.0 and snmpManagerTrapSendingEnable.0 are both set to 1.

14.10 SNMP NAT Traversal

A NAT placed between the gateway and the element manager calls for traversal solutions:

- **Trap source port:** all traps are sent out from the SNMP port (default 161). A manager receiving these traps can use the binding information (in the UDP layer) to traverse the NAT back to the device.
  The trap destination address (port and IP) are as configured in the snmpTargetMIB.
- **acKeepAliveTrap:** this trap is designed to be a constant life signal from the device to the manager allowing the manager NAT traversal at all times. The acBoardTrapGlobalsAdditionalInfo1 varbind has the device's serial number.
  The destination port (i.e., the manager port for this trap) can be set to be different than the port to which all other traps are sent. To do this, use the acSysSNMPKeepAliveTrapPort object in the acSystem MIB or the *ini* file parameter KeepAliveTrapPort.
  The trap is instigated in three ways:
  - Via an *ini* file parameter ('SendKeepAliveTrap = 1'). This ensures that the trap is continuously sent. The frequency is set via the 9/10 of the acSysSTUNBindingLifeTime object.
  - After the STUN client has discovered a NAT (any NAT).
  - If the STUN client cannot contact a STUN server.

Note: The two latter options require the STUN client be enabled (EnableSTUN). Also, once the acKeepAlive trap is instigated it does not stop.

- The manager can see the NAT type in the MIB:
  audioCodes(5003).acProducts(9).acBoardMibs(10).acSystem(10).acSystemStatus(2).acSysNetwork(6).acSysNAT(2).acSysNATType(1)
- The manger also has access to the STUN client configuration:
  audioCodes(5003).acProducts(9).acBoardMibs(10).acSystem(10).acSystemConfiguration(1).acSysNetworkConfig(3).acSysNATTraversal(6).acSysSTUN(21)
- **acNATTraversalAlarm:** When the NAT is placed in front a device that is identified as a symmetric NAT, this alarm is raised. It is cleared when a non-symmetric NAT or no NAT replace the symmetric one.

14.11 SNMP Administrative State Control

14.11.1 Node Maintenance

Node maintenance for the MediaPack is provided by an SNMP interface. The acBoardMIB provides two parameters for graceful and forced shutdowns of the MediaPack:

- acgwAdminState
- acgwAdminStateLockControl

The acgwAdminState is used either to request (set) a shutdown (0), undo shutdown (2), or to view (get) the gateway condition (0 = locked; 1 = shutting down; 2 = unlocked).

The acgwAdminStateLockControl is used to set a time limit (in seconds) for the shutdown where 0 means shutdown immediately (forced), -1 means no time limit (graceful), and x where x>0 indicates a time limit in seconds (timed limit is considered a graceful shutdown).

Note: The acgwAdminStateLockControl must be set first followed by the acgwAdminState.

14.11.2 Graceful Shutdown

acgwAdminState is a read-write MIB object. When a get request is sent for this object, the agent returns the board's current administrative state.

The possible values received on a **get** request include the following:

- locked(0): the board is locked
- shuttingDown(1): the board is currently performing a graceful lock
- unlocked(2): the board is unlocked

On a **set** request, the manager supplies one of the following desired administrative states:

- locked(0)
- unlocked(2)

When the board changes to either shuttingDown or locked state, an adminStateChange alarm is raised. When the board changes to an unlocked state, the adminStateChange alarm is cleared.

Before setting acgwAdminState to perform a lock, acgwAdminStateLockControl should be set first to control the type of lock that is performed. The possible values for the acgwAdminStateLockControl include the following:

- 1 = Perform a graceful lock. Calls are allowed to complete. No new calls are allowed to be originated on this device.
- 0 = Perform a forced lock. Calls are immediately terminated.
- Any number greater than 0 = Time in seconds before the graceful lock turns into a forced lock.

14.12 AudioCodes' Element Management System

Using AudioCodes' Element Management System (EMS) is recommended to Customers requiring large deployments (multiple media gateways in globally distributed enterprise offices, for example), that need to be managed by central personnel.

The EMS is not included in the device's supplied package. Contact AudioCodes for detailed information on AudioCodes' EMS and on AudioCodes' EVN - Enterprise VoIP Network – solution for large VoIP deployments.

15 Configuration Files

This section describes the configuration *dat* files that are loaded (in addition to the *ini* file) to the gateway. The configuration files are:

- Call Progress Tones file (refer to Section 15.1 on page 325).
- Prerecorded Tones file (refer to Section 15.2 on page 330).
- FXS Coefficient file (refer to Section 15.3 on page 331).
- User Information file (refer to Section 15.4 on page 332).

To load either of the configuration files to the MediaPack use the Embedded Web Server (refer to Section 5.9.2 on page 202) or alternatively specify the name of the relevant configuration file in the gateway's *ini* file and load it (the *ini* file) to the gateway (refer to Section 5.9.2.1 on page 203).

15.1 Configuring the Call Progress Tones and Distinctive Ringing File

The Call Progress Tones and Distinctive Ringing, configuration file used by the MediaPack is a binary file (with the extension *dat*) that is comprised of two sections. The first section contains the definitions of the Call Progress Tones (levels and frequencies) that are detected / generated by the MediaPack. The second section contains the characteristics of the distinctive ringing signals that are generated by the MediaPack.

Users can either use, one of the supplied MediaPack configuration (*dat*) files, or construct their own file. To construct their own configuration file, users are recommended, to modify the supplied *usa_tone.ini* file (in any standard text editor) to suit their specific requirements, and to convert it (the modified *ini* file) into binary format using the TrunkPack Downloadable Conversion Utility. For the description of the procedure on how to convert CPT *ini* file to a binary *dat* file, refer to Section E.1.1 on page 364.

Note that only the *dat* file can be loaded to the MediaPack gateway.

To load the Call Progress Tones *(dat)* file to the MediaPack, use the Embedded Web Server (refer to Section 5.6.4 on page 166) or the *ini* file (refer to Section 5.9.2.1 on page 203).

15.1.1 Format of the Call Progress Tones Section in the *ini* File

Users can create up to 32 different Call Progress Tones, each with frequency and format attributes.

The frequency attribute can be single or dual-frequency (in the range of 300 Hz to 1980 Hz), or an Amplitude Modulated (AM). In total, up to 64 different frequencies are supported. Only eight AM tones, in the range of 1 to 128 kHz, can be configured (the detection range is limited to 1 to 50 kHz). Note that when a tone is composed of a single frequency, the second frequency field must be set to zero.

The format attribute can be one of the following:

- **Continuous:** (e.g., dial tone) a steady non-interrupted sound. Only the 'First Signal On time' should be specified. All other on and off periods must be set to zero. In this case, the parameter specifies the detection period. For example, if it equals 300, the tone is detected after 3 seconds (300 x 10 msec). The minimum detection time is 100 msec.
- **Cadence:** A repeating sequence of on and off sounds. Up to four different sets of on / off periods can be specified.
- **Burst:** A single sound followed by silence. Only the 'First Signal On time' and 'First Signal Off time' should be specified. All other on and off periods must be set to zero. The burst tone is detected after the off time is completed.

Users can specify several tones of the same type. These additional tones are used only for tone detection. Generation of a specific tone conforms to the first definition of the specific tone. For example, users can define an additional dial tone by appending the second dial tone's definition lines to the first tone definition in the *ini* file. The MediaPack reports dial tone detection if either of the two tones is detected.

The Call Progress Tones section of the *ini* file comprises the following segments:

- **[NUMBER OF CALL PROGRESS TONES]:** Contains the following key: 'Number of Call Progress Tones' defining the number of Call Progress Tones that are 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

Figure 15-1: Call Progress Tone Types

```
1 - Dial Tone
2 - Ringback Tone
3 - Busy Tone
7 - Reorder Tone
8 - Confirmation Tone
9 - Call Waiting Tone
15 - Stutter Dial Tone
16 - Off Hook Warning Tone
17 - Call Waiting Ringback Tone
23 - Hold Tone
```

  - **Tone Modulation Type**: Either Amplitude Modulated (1) or regular (0).
  - **Tone Form:** The tone's format, can be one of the following:
    - Continuous
    - Cadence
    - Burst
  - **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 (not relevant to AM tones).
  - **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 (not relevant to AM tones).
  - **Low Freq Level [-dBm]:** Generation level 0 dBm to –31 dBm in [dBm] (not relevant to AM tones).
  - **High Freq Level:** Generation level. 0 to –31 dBm. The value should be set to '32' in the case of a single tone (not relevant to AM tones).
  - **First Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the first cadence on-off cycle. For be continuous tones, this parameter defines the detection period. For burst tones, it defines the tone's duration.
  - **First Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the first cadence on-off cycle (for cadence tones). For burst tones, this parameter defines the off time required after the burst tone ends and the tone detection is reported. For continuous tones, this parameter is ignored.
  - **Second Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the second cadence on-off cycle. Can be omitted if there isn't a second cadence.
  - **Second Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the second cadence on-off cycle. Can be omitted if there isn't a second cadence.
  - **Third Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the third cadence ON-OFF cycle. Can be omitted if there isn't a third cadence.

- **Third Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the third cadence ON-OFF cycle. Can be omitted if there isn't a third cadence.
- **Forth Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the fourth cadence ON-OFF cycle. Can be omitted if there isn't a fourth cadence.
- **Forth Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the fourth cadence ON-OFF cycle. Can be omitted if there isn't a fourth cadence.
- **Carrier Freq [Hz]:** the frequency of the carrier signal for AM tones.
- **Modulation Freq [Hz]:** the frequency of the modulated signal for AM tones (valid range from 1 Hz to 128 Hz).
- **Signal Level [-dBm]:** the level of the tone for AM tones.
- **AM Factor [steps of 0.02]:** the amplitude modulation factor (valid range from 1 to 50. Recommended values from 10 to 25).

Notes:

- When the same frequency is used for a continuous tone and a cadence tone, the 'Signal On Time' parameter of the continuous tone must have a value that is greater than the 'Signal On Time' parameter of the cadence tone. Otherwise the continuous tone is detected instead of the cadence tone.
- The tones frequency should differ by at least 40 Hz from one tone to other defined tones.

For example: to configure the dial tone to 440 Hz only, define the following text:

Figure 15-2: Defining a Dial Tone Example

```
#Dial tone
[CALL PROGRESS TONE #1]
Tone Type=1
Tone Form =1 (continuous)
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
```

Figure 15-3: Example of Ringing Burst

```
#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]=300
```

15.1.2 Format of the Distinctive Ringing Section in the *ini* File

Distinctive Ringing is only applicable to MediaPack/FXS gateways. Using the distinctive ringing section of this configuration file, the user can create up to 16 distinctive ringing patterns.

Each ringing pattern configures the ringing tone frequency and up to 4 ringing cadences. The same ringing frequency is used for all the ringing pattern cadences. The ringing frequency can be configured in the range of 10 Hz to 200 Hz with a 5 Hz resolution. Each of the ringing pattern cadences is specified by the following parameters:

- **Burst Ring On Time:** Configures the cadence to be a burst cadence in the entire ringing pattern. The burst relates to On time and the Off time of the same cadence. It must appear between 'First/Second/Third/Fourth' string and the 'Ring On/Off Time' This cadence rings once during the ringing pattern. Otherwise, the cadence is interpreted as cyclic: it repeats for every ringing cycle.
- **Ring On Time:** specifies the duration of the ringing signal.
- **Ring Off Time:** specifies the silence period of the cadence.

Note: In SIP the distinctive ringing pattern is selected according to Alert-Info header that is included in INVITE message. For example: Alert-Info <Bellcore-dr2>, or Alert-Info<http://…/Bellcore-dr2>. 'dr2' defines ringing pattern # 2. If the Alert-Info header is missing, the default ringing tone (0) is played.

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

- **[NUMBER OF DISTINCTIVE RINGING PATTERNS]:** Contains the following key:
  - 'Number of Distinctive Ringing Patterns' defining the number of Distinctive Ringing signals that are defined in the file.
- **[Ringing Pattern #X]:** Contains the Xth ringing pattern definition (starting from 0 and not exceeding the number of Distinctive Ringing patterns defined in the first section minus 1) using the following keys:
  - **Ring Type:** Must be equal to the Ringing Pattern number.
  - **Freq [Hz]:** Frequency in hertz of the ringing tone.
  - **First (Burst) Ring On Time [10 msec]:** 'Ring On' period (in 10 msec units) for the first cadence on-off cycle.
  - **First (Burst) Ring Off Time [10 msec]:** 'Ring Off' period (in 10 msec units) for the first cadence on-off cycle.
  - **Second (Burst) Ring On Time [10 msec]:** 'Ring On' period (in 10 msec units) for the second cadence on-off cycle.
  - **Second (Burst) Ring Off Time [10 msec]:** 'Ring Off' period (in 10 msec units) for the second cadence on-off cycle.
  - **Third (Burst) Ring On Time [10 msec]:** 'Ring On' period (in 10 msec units) for the third cadence on-off cycle.
  - **Third (Burst) Ring Off Time [10 msec]:** 'Ring Off' period (in 10 msec units) for the third cadence on-off cycle.
  - **Fourth (Burst) Ring On Time [10 msec]:** 'Ring Off' period (in 10 msec units) for the fourth cadence on-off cycle.
  - **Fourth (Burst) Ring Off Time [10 msec]:** 'Ring Off' period (in 10 msec units) for the fourth cadence on-off cycle.

15.1.2.1 Examples of Various Ringing Signals

Figure 15-4: Examples of Various Ringing Signals

```
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=3

#Regular North American Ringing Pattern
[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 1
[Ringing Pattern #1]
Ring Type=1
Freq [Hz]=20
First Ring On Time [10msec]=200
First Ring Off Time [10msec]=400

#GR-506-CORE Ringing Pattern 2
[Ringing Pattern #2]
Ring Type=2
Freq [Hz]=20
First Ring On Time [10msec]=80
First Ring Off Time [10msec]=40
Second Ring On Time [10msec]=80
Second Ring Off Time [10msec]=400
```

15.2 Prerecorded Tones (PRT) File

The Call Progress Tones mechanism has several limitations, such as a limited number of predefined tones and a limited number of frequency integrations in one tone. To work around these limitations and provide tone generation capability that is more flexible, the PRT file can be used. If a specific prerecorded tone exists in the PRT file, it takes precedence over the same tone that exists in the CPT file and is played instead of it.

Note that the prerecorded tones are used only for generation of tones. Detection of tones is performed according to the CPT file.

15.2.1 PRT File Format

The PRT *dat* file contains a set of prerecorded tones to be played by the MediaPack during operation. Up to 40 tones (totaling approximately one minute) can be stored in a single file in flash memory. The prerecorded tones (raw data PCM or L8 files) are prepared offline using standard recording utilities (such as CoolEdit™) and combined into a single file using the TrunkPack Downloadable Conversion utility (refer to Section E.1.3 on page 366).

The raw data files must be recorded with the following characteristics:

- Coders: G.711 A-law, G.711 µ-law or Linear PCM
- Rate: 8 kHz
- Resolution: 8-bit
- Channels: mono

The generated PRT file can then be loaded to the MediaPack using the BootP/TFTP utility (refer to Section 5.9.2.1 on page 203) or via the Embedded Web Server (Section 5.9.2 on page 202).

The prerecorded tones are played repeatedly. This enables you to record only part of the tone and play it for the full duration. For example, if a tone has a cadence of 2 seconds on and 4 seconds off, the recorded file should contain only these 6 seconds. The PRT module repeatedly plays this cadence for the configured duration. Similarly, a continuous tone can be played by repeating only part of it.

Note: The maximum size of a PRT file that can be loaded to the gateway is 100 KB.

15.3 The Coefficient Configuration File

The Coeff_FXS.dat file is used to provide best termination and transmission quality adaptation for different line types for FXS gateways. This adaptation is performed by modifying the telephony interface characteristics (such as DC and AC impedance, feeding current and ringing voltage).

The *coeff.dat* configuration file is produced specifically 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 and40 V RMS ringing voltage for REN = 2.

To load the coeff.dat file to the MediaPack use the Embedded Web Server (refer to Section 5.6.4 on page 166) or alternatively specify the FXS coeff.dat file name in the gateway's *ini* file (refer to Section 5.9.2.1 on page 203).

The Coeff.dat 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 doesn't require hardware modifications, so that a single device is able to meet requirements for different markets. The digital design of the filters and gain stages also ensures high reliability, no drifts (over temperature or time) and simple variations between different line types.

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

Note: Use the parameter 'CountryCoefficients' (described in Table 5-35 on page 132) to configure the FXO coefficients.

15.4 User Information File

The User Information file maps PBX extensions (connected to the MediaPack gateway) to global IP phone numbers (alphanumerical). In this context, a global IP number serves as a routing identifier for calls in the 'IP World'. The PBX extension uses this mapping to emulate the behavior of an IP phone. Note that the mapping mechanism is disabled by default and must be activated using the parameter 'EnableUserInfoUsage' (described in Section 5.5.2.1).

Each line in the file represents a mapping rule of a single PBX extension (up to 100 rules can be configured). Each line includes five items separated with commas. The items are described in Table 15-1 below. An example of a User Information file is shown in Figure 15-5 below.

Each PBX extension registers separately (a REGISTER message is sent for each entry, only if AuthenticationMode is set to 'Per Endpoint') using the IP number in the From / To headers. The REGISTER messages are sent gradually (i.e., initially, the gateway sends requests according to the maximum number of allowed SIP dialogs (configured by the parameter NumberOfActiveDialogs), after each received response, the subsequent request is sent). Therefore, no more than 'NumberOfActiveDialogs' dialogs are active simultaneously. The username and password are used for SIP Authentication when required.

The calling number of outgoing Tel to IP calls is first translated to an IP number and then (if defined) the manipulation rules are performed. The Display Name is used in the From header in addition to the IP number.

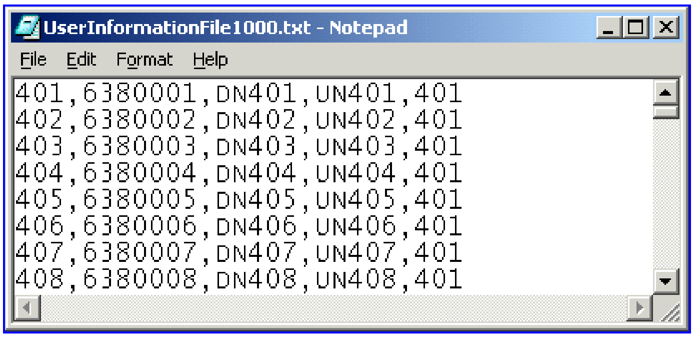
The called number of incoming IP to Tel calls is translated to a PBX extension only after manipulation rules (if defined) are performed.

The User Information file is a text file (the file size mustn't exceed 10,800 bytes) that can be loaded via the *ini* file (UserInfoFileName, described in Table 5-64), the Embedded Web Server (refer to Section 5.9.2 on page 202) or by using the automatic update mechanism (UserInfoFileURL, refer to Section 10.3 on page 263).

Table 15-1: User Information Items

Item	Description	Maximum Size
PBX extension #	The relevant PBX extension number	10
Global IP #	The relevant IP phone number	20
Display name	A string that represents the PBX extensions for the Caller ID.	30
Username	A string that represents the username for SIP registration.	20
Password	A string that represents the password for SIP registration.	20

Figure 15-5: Example of a User Information File



16 Selected Technical Specifications

16.1 MP-11x Specifications

Table 16-1: MP-11x Functional Specifications (continues on pages 333 to 335)

Channel Capacity	
Available Ports	MP-112 2 ports* MP-114 4 ports MP-118 8 ports * The MP-112 differs from the MP-114 and MP-118. Its configuration excludes the RS-232 connector, the Lifeline option and outdoor protection.
MP-11x/FXS Functionality	
FXS Capabilities	Short or Long Haul (Automatic Detection): REN2: Up to 10 km (32,800 feet) using 24 AWG line. REN5: Up to 3.5 km (11,400 feet) using 24 AWG line. **Note:** The lines were tested under the following conditions: ring voltage greater than 30 Vrms, offhook loop current greater than 20 mA (all lines ring simultaneously).
	MP-11x includes lightning and high voltage protection for outdoor operation.
	Caller ID generation: Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation, ETSI Type 1, NTT, Denmark, India, Brazil, British and DTMF ETSI CID (ETS 300-659-1).
	Programmable Line Characteristics: Battery feed, line current, hook thresholds, AC impedance matching, hybrid balance, Tx & Rx frequency response, Tx & Rx Gains. **Note:** For a specific coefficient file contact AudioCodes.
	Programmable ringing signal. Up to three cadences and frequency 15 to 200 Hz.
	Over-temperature protection for abnormal situations as shorted lines.
	Loop-backs for testing and maintenance.
MP-11x/FXO Functionality	
FXO Capabilities (Note: doesn't apply to the MP-112)	Short or Long Haul.
	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. **Note:** For country-specific coefficients use the parameter CountryCoefficients.
	Caller ID detection: Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation, ETSI Type 1, NTT, Denmark, India, Brazil, British and DTMF ETSI CID (ETS 300-659-1).
Additional Features	
Polarity Reversal / Wink	Immediate or smooth to prevent erroneous ringing
Metering Tones	12/16 KHz sinusoidal bursts
Distinctive Ringing	By frequency (15-100 Hz) and cadence patterns
Message Waiting Indication	DC voltage generation (TIA/EIA-464-B), V23 FSK data, Stutter dial tone and DTMF based.

Table 16-1: MP-11x Functional Specifications **(continues on pages 333 to 335)**

Voice & Tone Characteristics	
Voice Compression	G.711 PCM at 64 kbps µ-law/A-law (10, 20, 30, 40, 50, 60, 80, 100, 120 msec) G.723.1 MP-MLQ at 5.3 or 6.3 kbps (30, 60, 90 msec) G.726 at 32 kbps ADPCM (10, 20, 30, 40, 50, 60, 80, 100, 120 msec) G.729 CS-ACELP 8 Kbps Annex A / B (10, 20, 30, 40, 50, 60 msec)
Silence Suppression	G.723.1 Annex A G.729 Annex B PCM and ADPCM - Standard Silence Descriptor (SID) with Proprietary Voice Activity Detection (VAD) and Comfort Noise Generation (CNG).
Packet Loss Concealment	G.711 appendix 1 G.723.1 G.729 a/b
Echo Canceler	G.165 and G.168 2000, 64 msec
Gain Control	Programmable
DTMF Transport (in-band)	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	32 tones: single tone, dual tones or AM tones, programmable frequency & amplitude; 64 frequencies in the range 300 to 1980 Hz, 1 to 4 cadences per tone, up to 4 sets of ON/OFF periods.
Output Gain Control	-32 dB to +31 dB in steps of 1 dB
Input Gain Control	-32 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 seconds round trip)
Modem Transparency	Auto switch to PCM or ADPCM on V.34 or V.90 modem detection
Protocols	
VoIP Signaling Protocol	SIP RFC 3261
Communication Protocols	RTP/RTCP packetization. IP stack (UDP, TCP, RTP). Remote Software load (TFTP, HTTP and HTTPS).
Line Signaling Protocols	Loop start
Processor	
Control Processor	Motorola PowerQUICC 870
Control Processor Memory	SDRAM - 32 MB
Signal Processors	AudioCodes AC482 VoIP DSP
Interfaces	
FXS Telephony Interface	2, 4 or 8 Analog FXS phone or fax ports, loop start (RJ-11)
FXO Telephony Interface	4 or 8 Analog FXO PSTN/PBX loop start ports
Combined FXS / FXO	MP-118: 4 FXS + 4 FXO ports; MP-114: 2 FXS + 2 FXO ports
Network Interface	10/100 Base-TX
RS-232 Interface	RS-232 Terminal Interface (requires a DB-9 to PS/2 adaptor).
Indicators	Channel status and activity LEDs
Lifeline	The Lifeline provides a wired analog POTS phone connection to any PSTN or PBX FXS port when there is no power, or the network fails. Combined FXS/FXO gateways provide a Lifeline connection available on all FXS ports. **Note:** The Lifeline splitter (for FXS gateways) is a special order option.

Table 16-1: MP-11x Functional Specifications (continues on pages 333 to 335)

Connectors & Switches	
Rear Panel	
8 Analog Lines (MP-118)	8 RJ-11 connectors
4 Analog Lines (MP-114)	4 RJ-11 connectors
2 Analog Lines (MP-112)	2 RJ-11 connectors
AC power supply socket	100-240~0.3A max.
Ethernet	10/100 Base-TX, RJ-45
RS-232	Console PS/2 port
Reset Button	Resets the MP-11x
Physical	
Dimensions (HxWxD)	42 x 172 x 220 mm
Weight	0.5 kg (Approx.)
Environmental	Operational: 5° to 40° C 41° to 104° F Storage: -25° to 70° C -77° to 158° F Humidity: 10 to 90% non-condensing
Mounting	Rack mount, Desktop, Wall mount. **Note:** The rack mount is a special order option.
Electrical	100-240 VAC Nominal 50/60 Hz
Type Approvals	
Safety and EMC	UL 60950-1, FCC part 15 Class B CE Mark EN 60950-1, EN 55022, EN 55024, EN61000-3-2, EN61000-3-3, EN55024.
Management	
Configuration	Gateway configuration using Web browser or *ini* files
Management and Maintenance	SNMP v2c; SNMP v3
	Syslog, per RFC 3164
	Local RS-232 terminal
	Web Management via HTTP or HTTPS
	Telnet

16.2 MP-124 Specifications

Table 16-2: MP-124 Functional Specifications (continues on pages 336 to 338)

Channel Capacity	
Available Ports	MP-124 24 ports
FXS Functionality	
FXS Capabilities	Short or Long Haul (Automatic Detection): REN2: Up to 15.5 km (50,800 feet) using 24 AWG line. REN3: Up to 9 km (30,000 feet) using 24 AWG line. REN5: Up to 5.5 km (18,000 feet) using 24 AWG line. **Note:** The lines were tested under the following conditions: ring voltage greater than 32 Vrms, offhook loop current greater than 20 mA (all lines ring simultaneously).
	Includes lightning and high voltage protection for outdoor operation.
	Caller ID generation: Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation, ETSI Type 1, NTT, Denmark, India, Brazil, British and DTMF ETSI CID (ETS 300-659-1).
	Programmable Line Characteristics: Battery feed, line current, hook thresholds, AC impedance matching, hybrid balance, Tx & Rx frequency response, Tx & Rx Gains. **Note:** For a specific coefficient file contact AudioCodes.
	Programmable ringing signal. Up to three cadences and frequency 15 to 200 Hz.
	Over-temperature protection for abnormal situations as shorted lines.
	Loop-backs for testing and maintenance.
Additional Features	
Polarity Reversal / Wink	Immediate or smooth to prevent erroneous ringing
Metering Tones	12/16 KHz sinusoidal bursts
Distinctive Ringing	By frequency (15-100 Hz) and cadence patterns
Message Waiting Indication	DC voltage generation (TIA/EIA-464-B), V23 FSK data, Stutter dial tone and DTMF based.
Voice & Tone Characteristics	
Voice Compression	G.711 PCM at 64 kbps µ-law/A-law (10, 20, 30, 40, 50, 60, 80, 100, 120 msec) G.723.1 MP-MLQ at 5.3 or 6.3 kbps (30, 60, 90 msec) G.726 at 32 kbps ADPCM (10, 20, 30, 40, 50, 60, 80, 100, 120 msec) G.729 CS-ACELP 8 Kbps Annex A / B (10, 20, 30, 40, 50, 60 msec)
Silence Suppression	G.723.1 Annex A G.729 Annex B PCM and ADPCM - Standard Silence Descriptor (SID) with Proprietary Voice Activity Detection (VAD) and Comfort Noise Generation (CNG).
Packet Loss Concealment	G.711 appendix 1 G.723.1 G.729 a/b
Echo Canceler	G.165 and G.168 2000, 64 msec
Gain Control	Programmable
DTMF Transport (in-band)	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.

Table 16-2: MP-124 Functional Specifications (continues on pages 336 to 338)

Call Progress Tone Detection and Generation	32 tones: single tone, dual tones or AM tones, programmable frequency & amplitude; 64 frequencies in the range 300 to 1980 Hz, 1 to 4 cadences per tone, up to 4 sets of ON/OFF periods.
Output Gain Control	-32 dB to +31 dB in steps of 1 dB
Input Gain Control	-32 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 seconds round trip)
Modem Transparency	Auto switch to PCM or ADPCM on V.34 or V.90 modem detection
Protocols	
VoIP Signaling Protocol	SIP RFC 3261
Communication Protocols	RTP/RTCP packetization. IP stack (UDP, TCP, RTP). Remote Software load (TFTP, HTTP and HTTPS).
Line Signaling Protocols	Loop start
Processor	
Control Processor	Motorola PowerQUICC 860
Control Processor Memory	SDRAM – 64 MB
Signal Processors	AudioCodes AC482 VoIP DSP
Interfaces	
FXS Telephony Interface	24 Analog FXS phone or fax ports, loop start (RJ-11)
Network Interface	10/100 Base-TX
RS-232 Interface	RS-232 Terminal Interface (DB-9).
Indicators	Channel status and activity LEDs
Connectors & Switches	
Rear Panel:	
24 Analog Lines	50-pin Telco shielded connector
Ethernet	10/100 Base-TX, RJ-45 shielded connector
RS-232	Console port - DB-9
AC power supply socket	100-240~0.8A max
Front Panel:	
Reset Button	Resets the MP-124
Physical	
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: 1.8 kg 4 lb
Environmental	Operational: 5° to 40° C 41° to 104° F Storage: -25° to 70° C -77° to 158° F Humidity: 10 to 90% non-condensing
Mounting	Rack mount, Desktop
Electrical	100-240 VAC Nominal 50/60 Hz

Table 16-2: MP-124 Functional Specifications (continues on pages 336 to 338)

Type Approvals	
Safety and EMC	UL 60950-1, FCC part 15 Class B CE Mark EN 60950-1, EN 55022, EN 55024, EN61000-3-2, EN61000-3-3, EN55024.
Management	
Configuration	Gateway configuration using Web browser or *ini* files
Management and Maintenance	SNMP v2c; SNMP v3
	Syslog, per RFC 3164
	Local RS-232 terminal
	Web Management via HTTP or HTTPS
	Telnet

All specifications in this document are subject to change without prior notice.

A MediaPack SIP Software Kit

Table A-1 describes the standard supplied software kit for MediaPack FXS/FXO SIP gateways. The supplied documentation includes this User's Manual, the MP-11x & MP-124 MGCP-H.323-SIP Fast Track Guide, and the MP-11x & MP-124 SIP Release Notes.

Table A-1: MediaPack SIP Supplied Software Kit

File Name	Description
Ram.cmp files	
MP124_SIP_xxx.cmp	Image file containing the software for the MP-124/FXS gateway.
MP118_SIP_xxx.cmp	Common Image file Image file containing the software for MP-11x/FXS gateways.
***ini* files and utilities**	
SIPgw_MP124.ini	Sample *Ini* file for MP-124/FXS gateway.
SIPgw_fxs_MP118.ini	Sample *ini* file for MP-118/FXS gateways.
SIPgw_fxs_MP114.ini	Sample *ini* file for MP-114/FXS gateways.
SIPgw_fxs_MP112.ini	Sample *ini* file for MP-112/FXS gateways.
Usa_tones_xx.dat	Default loadable Call Progress Tones *dat* file.
Usa_tones_xx.ini	Call progress Tones *ini* file (used to create *dat* file).
MP1xx_Coeff_FXS.dat	Telephony interface configuration file for MediaPack/FXS gateways.
DConvert.exe	TrunkPack Downloadable Conversion Utility
ACSyslog08.exe	Syslog server.
bootp.exe	BootP/TFTP configuration utility
CPTWizard.exe	Call Progress Tones Wizard
MIBs Files	MIB library for SNMP browser

Reader's Notes

B SIP Compliance Tables

The MediaPack gateways comply with RFC 3261, as shown in the following sections.

B.1 SIP Functions

Table B-1: SIP Functions

Function	Supported
User Agent Client (UAC)	Yes
User Agent Server (UAS)	Yes
Proxy Server	Third-party only (tested with, amongst others, Ubiquity, Delta3, Microsoft, 3Com, BroadSoft, Snom, and Cisco Proxies)
Redirect Server	Third-party
Registrar Server	Third-party
Event Publication Agent (EPA)	Yes
Event State Compositor (ESC)	Third-party

B.2 SIP Methods

Table B-2: SIP Methods

Method	Supported	Comments
INVITE	Yes	
ACK	Yes	
BYE	Yes	
CANCEL	Yes	
REGISTER	Yes	Send only
REFER	Yes	
NOTIFY	Yes	
INFO	Yes	
OPTIONS	Yes	
PRACK	Yes	
UPDATE	Yes	
PUBLISH	Yes	Send only
SUBSCRIBE	Yes	

B.3 SIP Headers

The following SIP Headers are supported by the gateway:

Table B-3: SIP Headers (continues on pages 342 to 343)

Header Field	Supported
Accept	Yes
Accept–Encoding	Yes
Alert-Info	Yes
Allow	Yes
Also	Yes
Asserted-Identity	Yes
Authorization	Yes
Call-ID	Yes
Call-Info	Yes
Contact	Yes
Content-Disposition	Yes
Content-Encoding	Yes
Content-Length	Yes
Content-Type	Yes
Cseq	Yes
Date	Yes
Diversion	Yes
Encryption	No
Expires	Yes
Fax	Yes
From	Yes
History-Info	Yes
Join	Yes
Max-Forwards	Yes
Messages-Waiting	Yes
MIN-SE	Yes
Organization	No
P-Asserted-Identity	Yes
P-Preferred-Identity	Yes
Priority	No
Proxy- Authenticate	Yes
Proxy- Authorization	Yes
Proxy- Require	Yes
Prack	Yes
Reason	Yes
Record- Route	Yes
Refer-To	Yes
Referred-By	Yes
Replaces	Yes
Require	Yes
Remote-Party-ID	Yes

Table B-3: SIP Headers (continues on pages 342 to 343)

Header Field	Supported
Response- Key	Yes
Retry- After	Yes
Route	Yes
Rseq	Yes
Session-Expires	Yes
Server	Yes
SIP-If-Match	Yes
Subject	Yes
Supported	Yes
Timestamp	Yes
To	Yes
Unsupported	Yes
User- Agent	Yes
Via	Yes
Voicemail	Yes
Warning	Yes
WWW- Authenticate	Yes

B.4 SDP Headers

The following SDP Headers are supported by the gateway:

Table B-4: SDP Headers

SDP Header Element	Supported
v - Protocol version	Yes
o - Owner/ creator and session identifier	Yes
a - Attribute information	Yes
c - Connection information	Yes
d - Digit	Yes
m - Media name and transport address	Yes
s - Session information	Yes
t - Time alive header	Yes
b - Bandwidth header	Yes
u - Uri Description Header	Yes
e - Email Address header	Yes
i - Session Info Header	Yes
p - Phone number header	Yes
y - Year	Yes

B.5 SIP Responses

The following SIP responses are supported by the gateway:

- 1xx Response - Information Responses
- 2xx Response - Successful Responses
- 3xx Response - Redirection Responses
- 4xx Response - Client Failure Responses
- 5xx Response - Server Failure Responses
- 6xx Response - Global Responses

B.5.1 1xx Response – Information Responses

Table B-5: 1xx SIP Responses

1xx Response		Supported	Comments
100	Trying	Yes	The SIP gateway generates this response upon receiving of Proceeding message from ISDN or immediately after placing a call for CAS signaling.
180	Ringing	Yes	The SIP gateway generates this response for an incoming INVITE message. On receiving this response, the gateway waits for a 200 OK response.
181	Call is being forwarded	Yes	The SIP gateway does not generate these responses. However, the gateway does receive them. The gateway processes these responses the same way that it processes the 100 Trying response.
182	Queued	Yes	The SIP gateway generates this response in Call Waiting service. When SIP gateway receives a 182 response, it plays a special waiting Ringback tone to TEL side.
183	Session Progress	Yes	The SIP gateway generates this response if Early Media feature is enabled and if the gateway plays a Ringback tone to IP

B.5.2 2xx Response – Successful Responses

Table B-6: 2xx SIP Responses

2xx Response		Supported	Comments
200	OK	Yes	
202	Accepted	Yes	

B.5.3 3xx Response – Redirection Responses

Table B-7: 3xx SIP Responses

3xx Response		Supported	Comments
300	Multiple Choice	Yes	The gateway responds with an ACK and resends the request to first in the contact list, new address.
301	Moved Permanently	Yes	The gateway responds with an ACK and resends the request to new address.
302	Moved Temporarily	Yes	The SIP gateway generates this response when call forward is used, to redirect the call to another destination. If such response is received, the calling gateway initiates an INVITE message to the new destination.
305	Use Proxy	Yes	The gateway responds with an ACK and resends the request to new address.
380	Alternate Service	Yes	"

B.5.4 4xx Response – Client Failure Responses

Table B-8: 4xx SIP Responses (continues on pages 345 to 346)

4xx Response		Supported	Comments
400	Bad Request	Yes	The gateway does not generate this response. On reception of this message, before a 200 OK has been received, the gateway responds with an ACK and disconnects the call.
401	Unauthorized	Yes	Authentication support for Basic and Digest. On receiving this message the GW issues a new request according to the scheme received on this response
402	Payment Required	Yes	The gateway does not generate this response. On reception of this message, before a 200 OK has been received, the gateway responds with an ACK and disconnects the call.
403	Forbidden	Yes	The gateway does not generate this response. On reception of this message, before a 200 OK has been received, the gateway responds with an ACK and disconnects the call.
404	Not Found	Yes	The SIP gateway generates this response if it is unable to locate the callee. On receiving this response, the gateway notifies the User with a Reorder Tone.
405	Method Not Allowed	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
406	Not Acceptable	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
407	Proxy Authentication Required	Yes	Authentication support for Basic and Digest. On receiving this message the GW issues a new request according to the scheme received on this response.
408	Request Timeout	Yes	The gateway generates this response if the no-answer timer expires. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.

Table B-8: 4xx SIP Responses (continues on pages 345 to 346)

4xx Response		Supported	Comments
409	Conflict	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
410	Gone	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
411	Length Required	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
413	Request Entity Too Large	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
414	Request-URL Too Long	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
415	Unsupported Media	Yes	If the gateway receives a 415 Unsupported Media response, it notifies the User with a Reorder Tone. The gateway generates this response in case of SDP mismatch.
420	Bad Extension	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
480	Temporarily Unavailable	Yes	If the gateway receives a 480 Temporarily Unavailable response, it notifies the User with a Reorder Tone. This response is issued if there is no response from remote.
481	Call Leg/Transaction Does Not Exist	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
482	Loop Detected	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
483	Too Many Hops	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
484	Address Incomplete	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
485	Ambiguous	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
486	Busy Here	Yes	The SIP gateway generates this response if the called party is off hook and the call cannot be presented as a call waiting call. On receiving this response, the gateway notifies the User and generates a busy tone.
487	Request Canceled	Yes	This response indicates that the initial request is terminated with a BYE or CANCEL request.
488	Not Acceptable	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.

B.5.5 5xx Response – Server Failure Responses

Table B-9: 5xx SIP Responses

5xx Response		Comments
500	Internal Server Error	On reception of any of these Responses, the GW releases the call, sending appropriate release cause to PSTN side. The GW generates 5xx response according to PSTN release cause coming from PSTN.
501	Not Implemented	
502	Bad gateway	
503	Service Unavailable	
504	Gateway Timeout	
505	Version Not Supported	

B.5.6 6xx Response – Global Responses

Table B-10: 6xx SIP Responses

6xx Response		Comments
600	Busy Everywhere	On reception of any of these Responses, the GW releases the call, sending appropriate release cause to PSTN side.
603	Decline	
604	Does Not Exist Anywhere	
606	Not Acceptable	

Reader's Notes

C BootP/TFTP Configuration Utility

The BootP/TFTP utility enables you to easily configure and provision our boards and media gateways. Similar to third-party BootP/TFTP utilities (which are also supported) but with added functionality; our BootP/TFTP utility can be installed on Windows™ 98 or Windows™ NT/2000/XP. The BootP/TFTP utility enables remote reset of the device to trigger the initialization procedure (BootP and TFTP). It contains BootP and TFTP utilities with specific adaptations to our requirements.

C.1 When to Use the BootP/TFTP

The BootP/TFTP utility can be used with the device as an alternative means of initializing the gateways. Initialization provides a gateway with an IP address, subnet mask, and the default gateway IP address. The tool also loads default software, *ini* and other configuration files. BootP Tool can also be used to restore a gateway to its initial configuration, such as in the following instances:

- The IP address of the gateway is not known.
- The Web browser has been inadvertently turned off.
- The Web browser password has been forgotten.
- The gateway has encountered a fault that cannot be recovered using the Web browser.

Tip: The BootP is normally used to configure the device's initial parameters. Once this information has been provided, the BootP is no longer needed. All parameters are stored in non-volatile memory and used when the BootP is not accessible.

C.2 An Overview of BootP

BootP is a protocol defined in RFC 951 and RFC 1542 that enables an internet device to discover its own IP address and the IP address of a BootP on the network, and to obtain the files from that utility that need to be loaded into the device to function.

A device that uses BootP when it powers up broadcasts a BootRequest message on the network. A BootP on the network receives this message and generates a BootReply. The BootReply indicates the IP address that should be used by the device and specifies an IP address from which the unit may load configuration files using Trivial File Transfer Protocol (TFTP) described in RFC 906 and RFC 1350.

C.3 Key Features

- Internal BootP supporting hundreds of entities
- Internal TFTP
- Contains all required data for our products in predefined format
- Provides a TFTP address, enabling network separation of TFTP and BootP utilities
- Tools to backup and restore the local database
- Templates
- User-defined names for each entity
- Option for changing MAC address
- Protection against entering faulty information
- Remote reset

- Unicast BootP response
- User-initiated BootP respond, for remote provisioning over WAN
- Filtered display of BootP requests
- Location of other BootP utilities that contain the same MAC entity
- Common log window for both BootP and TFTP sessions
- Works with Windows™ 98, Windows™ NT, Windows™ 2000 and Windows™ XP

C.4 Specifications

- BootP standards: RFC 951 and RFC 1542
- TFTP standards: RFC 1350 and RFC 906
- Operating System: Windows™ 98, Windows™ NT, Windows™ 2000 and Windows™ XP
- Max number of MAC entries: 200

C.5 Installation

➢ To install the BootP/TFTP on your computer, take these 2 steps:

1. Locate the BootP folder on the VoIP gateway supplied CD ROM and open the file Setup.exe.
2. Follow the prompts from the installation wizard to complete the installation.

➢ To open the BootP/TFTP, take these 2 steps:

1. From the **Start** menu on your computer, navigate to **Programs**, and then click **BootP**.
2. The first time that you run the BootP/TFTP, the program prompts you to set the user preferences. Refer to the Section C.10 on page 353 for information on setting the preferences.

C.6 Loading the *cmp* File, Booting the Device

Once the application is running, and the preferences were set (refer to Section C.10), for each unit that is to be supported, enter parameters into the tool to set up the network configuration information and initialization file names. Each unit is identified by a MAC address. For information on how to configure (add, delete and edit) units, refer to Section C.11 on page 355.

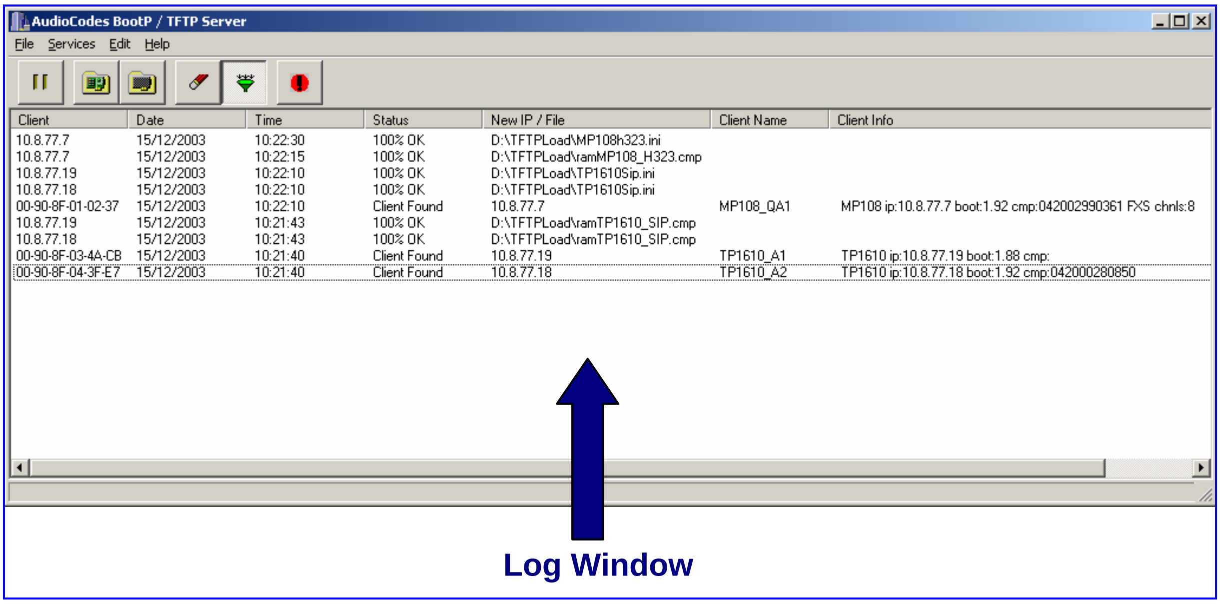
➢ To load the software and configuration files, take these 4 steps:

1. Create a folder on your computer that contains all software and configuration files that are needed as part of the TFTP process.
2. Set the BootP and TFTP preferences (refer to Section C.10).
3. Add client configuration for the VoIP gateway that you want to initialize by the BootP, refer to Section C.11.1.
4. Reset the VoIP gateway, either physically or remotely, causing the device to use BootP to access the network and configuration information.

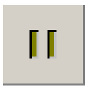
C.7 BootP/TFTP Application User Interface

Figure C-1 shows the main application screen for the BootP/TFTP utility.

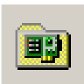
Figure C-1: Main Screen



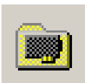
C.8 Function Buttons on the Main Screen



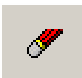
Pause: Click this button to pause the BootP Tool so that no replies are sent to BootP requests. Click the button again to restart the BootP Tool so that it responds to all BootP requests. The **Pause** button provides a depressed graphic when the feature is active.



Edit Clients: Click this button to open a new window that enables you to enter configuration information for each supported VoIP gateway. Details on the Clients window are provided in Section C.11 on page 355.

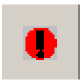


Edit Templates: Click this button to open a new window that enables you to create or edit standard templates. These templates can be used when configuring new clients that share most of the same settings. Details on the **Templates** window are provided in Section C.12 on page 360.



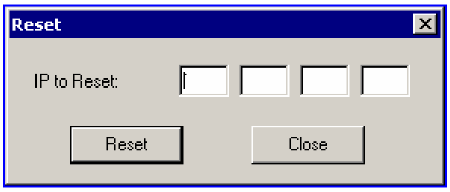
Clear Log: Click this button to clear all entries from the Log Window portion of the main application screen. Details on the log window are provided in Section C.9 on page 352.

Filter Clients: Click this button to prevent the BootP Tool from logging BootP requests received from disabled clients or from clients which do not have entries in the Clients table.



Reset: Click this button to open a new window where you enter an IP address requests for a gateway that you want to reset.

Figure C-2: Reset Screen



When a gateway resets, it first sends a BootRequest. Therefore, Reset can be used to force a BootP session with a gateway without needing to power cycle the gateway. As with any BootP session, the computer running the BootP Tool must be located on the same subnet as the controlled VoIP gateway.

C.9 Log Window

The log window (refer to Figure C-1 on the previous page) records all BootP request and BootP reply transactions, as well as TFTP transactions. For each transaction, the log window displays the following information:

- **Client:** shows the Client address of the VoIP gateway, which is the MAC address of the client for BootP transactions or the IP address of the client for TFTP transactions.
- **Date:** shows the date of the transaction, based on the internal calendar of the computer.
- **Time:** shows the time of day of the transaction, based on the internal clock of the computer.
- **Status:** indicates the status of the transaction.
    - *Client Not Found:* A BootRequest was received but there is no matching client entry in the BootP Tool.
    - *Client Found:* A BootRequest was received and there is a matching client entry in the BootP Tool. A BootReply is sent.
    - *Client's MAC Changed:* There is a client entered for this IP address but with a different MAC address.
    - *Client Disabled:* A BootRequest was received and there is a matching client entry in the BootP tool but this entry is disabled.
    - *Listed At:* Another BootP utility is listed as supporting a particular client when the Test Selected Client button is clicked (for details on Testing a client, refer to Section C.11.4 on page 357).
    - *Download Status:* Progress of a TFTP load to a client, shown in %.
- **New IP / File:** shows the IP address applied to the client as a result of the BootP transaction, as well as the file name and path of a file transfer for a TFTP transaction.
- **Client Name:** shows the client name, as configured for that client in the Client Configuration screen.

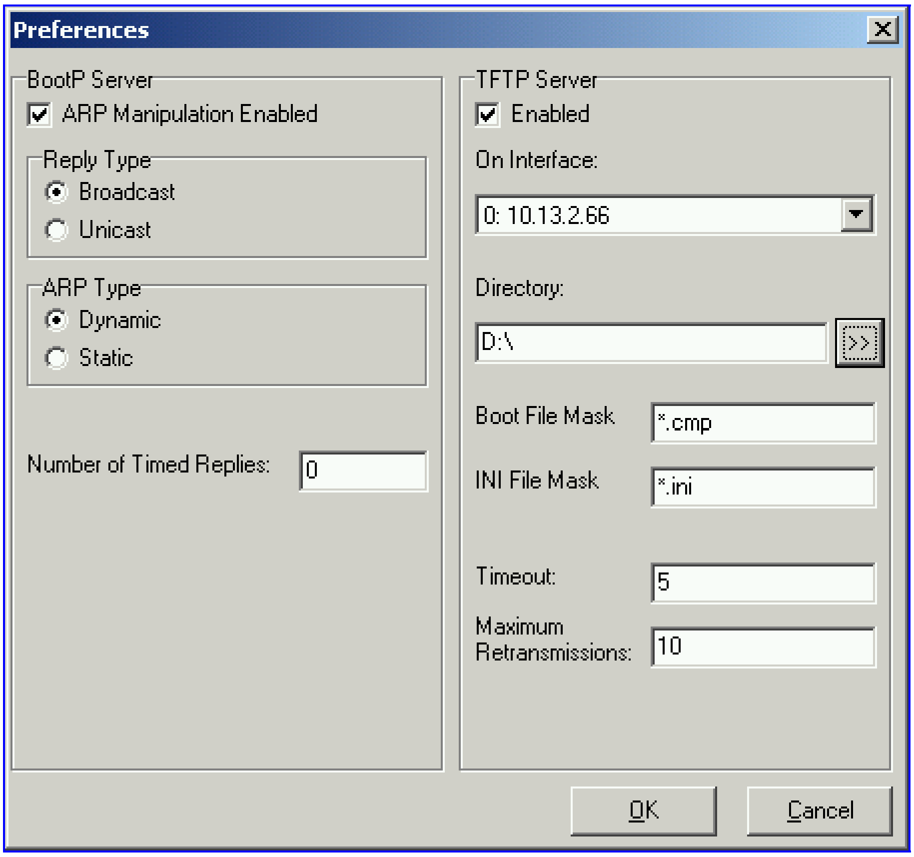
Use right-click on a line in the Log Window to open a pop-up window with the following options:

- **Reset:** Selecting this option results in a reset command being sent to the client VoIP gateway. The program searches its database for the MAC address indicated in the line. If the client is found in that database, the program adds the client MAC address to the Address Resolution Protocol (ARP) table for the computer. The program then sends a reset command to the client. This enables a reset to be sent without knowing the current IP address of the client, as long as the computer sending the reset is on the same subnet.
  Note: To use reset as described above, the user must have administrator privileges on the computer. Attempting to perform this type of reset without administrator privileges on the computer results in an error message. **ARP Manipulation Enable** must also be turned on in the **Preferences** window.
- **View Client:** Selecting this option, or double clicking on the line in the log window, opens the **Client Configuration** window. If the MAC address indicated on the line exists in the client database, it is highlighted. If the address is not in the client database, a new client is added with the MAC address filled out. You can enter data in the remaining fields to create a new client entry for that client.

C.10 Setting the Preferences

The Preferences window, Figure C-3, is used to configure the BootP Tool parameters.

Figure C-3: Preferences Screen



C.10.1 BootP Preferences

ARP is a common acronym for Address Resolution Protocol, and is the method used by all Internet devices to determine the link layer address, such as the Ethernet MAC address, in order to route Datagrams to devices that are on the same subnet.

When ARP Manipulation is enabled on this screen, the BootP Tool creates an ARP cache entry on your computer when it receives a BootP BootRequest from the VoIP gateway. Your computer uses this information to send messages to the VoIP gateway without using ARP again. This is particularly useful when the gateway does not yet have an IP address and, therefore, cannot respond to an ARP.

Because this feature creates an entry in the computer ARP cache, Administrator Privileges are required. If the computer is not set to allow administrator privileges, ARP Manipulation cannot be enabled.

- **ARP Manipulation Enabled:** Enable ARP Manipulation to remotely reset a gateway that does not yet have a valid IP address.

  If ARP Manipulation is enabled, the following two commands are available.

  - **Reply Type:** Reply to a BootRequest can be either **Broadcast** or **Unicast**. The default for the BootP Tool is **Broadcast**. In order for the reply to be set to **Unicast**, ARP Manipulation must first be enabled. This then enables the BootP Tool to find the MAC address for the client in the ARP cache so that it can send a message directly to the requesting device. Normally, this setting can be left at **Broadcast**.
  - **ARP Type:** The type of entry made into the ARP cache on the computer, once **ARP Manipulation** is enabled, can be either **Dynamic** or **Static**. Dynamic entries expire after a period of time, keeping the cache clean so that stale entries do not consume computer resources. The Dynamic setting is the default setting and the setting most often used. Static entries do not expire.
- **Number of Timed Replies:** This feature is useful for communicating to VoIP gateways that are located behind a firewall that would block their BootRequest messages from getting through to the computer that is running the BootP Tool. You can set this value to any whole digit. Once set, the BootP Tool can send that number of BootReply messages to the destination immediately after you send a remote reset to a VoIP gateway at a valid IP address. This enables the replies to get through to the VoIP gateway even if the BootRequest is blocked by the firewall. To turn off this feature, set the **Number of Timed Replies** = 0.

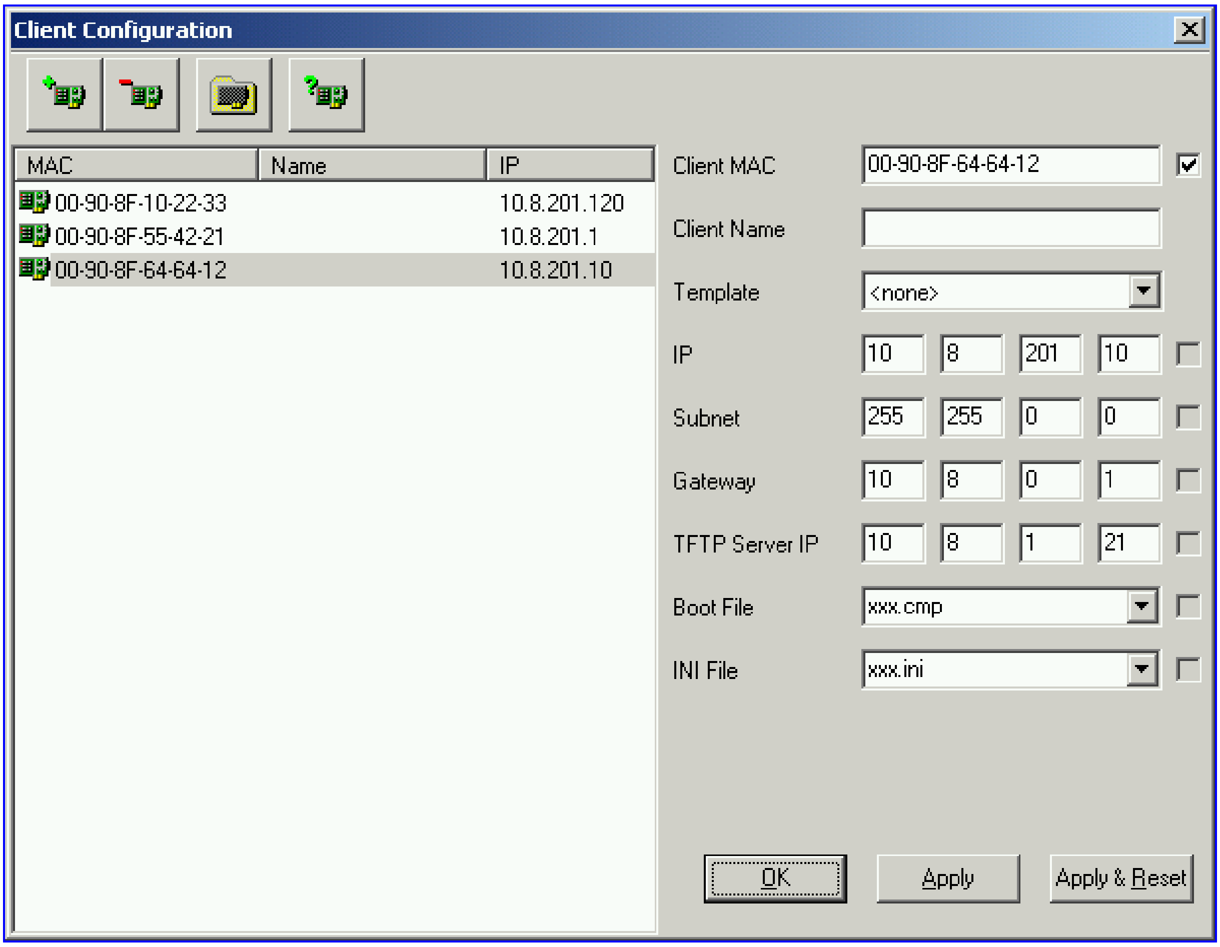
C.10.2 TFTP Preferences

- **Enabled:** To enable the TFTP functionality of the BootP Tool, check the box beside this heading. If you want to use another TFTP application, other than the one included with the BootP Tool, unselect the box.
- **On Interface:** This pull down menu displays all network interfaces currently available on the computer. Select the interface that you want to use for the TFTP. Normally, there is only one choice.
- **Directory:** This option is enabled only when the TFTP is enabled. Use this parameter to specify the folder that contains the files for the TFTP utility to manage (*cmp*, *ini*, Call Progress Tones, etc.).
- **Boot File Mask:** Boot File Mask specifies the file extension used by the TFTP utility for the boot file that is included in the BootReply message. This is the file that contains VoIP gateway software and normally appears as *cmp*.
- ***ini* File Mask:** *ini* File mask specifies the file extension used by the TFTP utility for the configuration file that is included in the BootReply message. This is the file that contains VoIP gateway configuration parameters and normally appears as *ini*.
- **Timeout:** This specifies the number of seconds that the TFTP utility waits before retransmitting TFTP messages. This can be left at the default value of 5 (the more congested your network, the higher the value you should define in these fields).
- **Maximum Retransmissions:** Specifies the number of times that the TFTP utility tries to resend messages after timing out. This can be left at the default value of 10 (the more congested your network, the higher the value should be defined in these fields).

C.11 Configuring the BootP Clients

The Clients window, shown in Figure C-4, is used to set up the parameters for each specific VoIP gateway.

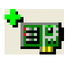
Figure C-4: Client Configuration Screen



C.11.1 Adding Clients

Adding a client creates an entry in the BootP Tool for a specific gateway.

➢ **To add a client to the list without using a template, take these 3 steps:**

1. Click **Add New Client**  ; a client with blank parameters is displayed.
2. Enter values in the fields on the right side of the window, using the guidelines for the fields in Section C.11.5 on page 357.
3. Click **Apply** to save this entry to the list of clients, or click **Apply & Reset** to save this entry to the list of clients and send a reset message to that gateway to immediately implement the settings.

   Note: To use **Apply & Reset** you must enable **ARP Manipulation** in the **Preferences** window. Also, you must have administrator privileges for the computer you are using.

An easy way to create several clients that use similar settings is to create a template. For information on how to create a template, refer to Section C.12 on page 360.

➢ To add a client to the list using a template, take these 5 steps:

1. Click **Add New Client** ; a client with blank parameters is displayed.
2. In the field 'Template', located on the right side of the 'Client Configuration Window', click on the down arrow to the right of the entry field and select the template that you want to use.
3. The values provided by the template are automatically entered into the parameter fields on the right side of the 'Client Configuration Window'. To use the template parameters, leave the check box next to that parameter selected. The parameter values appear in gray text.
4. To change a parameter to a different value, unselect the check box to the right of that parameter. This clears the parameter provided by the template and enables you to edit the entry. Clicking the check box again restores the template settings.
5. Click **Apply** to save this entry to the list of clients or click **Apply & Reset** to save this entry to the list of clients and send a reset message to that gateway to immediately implement the settings.

   Note: To use **Apply & Reset** you must enable **ARP Manipulation** in the **Preferences** window. Also, you must have administrator privileges for the computer you are using.

C.11.2 Deleting Clients

➢ To delete a client from the BootP Tool, take these 3 steps:

1. Select the client that you wish to delete by clicking on the line in the window for that client.
2. Click the **Delete Current Client** button ; a warning pops up.
3. To delete the client, click **Yes**.

C.11.3 Editing Client Parameters

➢ To edit the parameters for an existing client, take these 4 steps:

1. Select the client that you wish to edit by clicking on the line in the window for that client.
2. Parameters for that client display in the parameter fields on the right side of the window.
3. Make the changes required for each parameter.
4. Click **Apply** to save the changes, or click **Apply & Reset** to save the changes and send a reset message to that gateway to immediately implement the settings. **Note:** To use **Apply & Reset** you must enable **ARP Manipulation** in the **Preferences** window. Also, you must have administrator privileges for the computer you are using.

C.11.4 Testing the Client

There should only be one BootP utility supporting any particular client MAC active on the network at any time.

➢ To check if other BootP utilities support this client, take these 4 steps:

1. Select the client that you wish to test by clicking the client name in the main area of the **Client Configuration Window**.
2. Click the **Test Selected Client** button .
3. Examine the Log Window on the Main Application Screen. If there is another BootP utility that supports this client MAC, there is a response indicated from that utility showing the status Listed At along with the IP address of that utility.
4. If there is another utility responding to this client, you must remove that client from either this utility or the other one.

C.11.5 Setting Client Parameters

Client parameters are listed on the right side of the **Client Configuration Window**.

- **Client MAC:** The Client MAC is used by BootP to identify the VoIP gateway. The MAC address for the VoIP gateway is printed on a label located on the VoIP gateway hardware. Enter the Ethernet MAC address for the VoIP gateway in this field. Click the box to the right of this field to enable this particular client in the BootP tool (if the client is disabled, no replies are sent to BootP requests).
  Note: When the MAC address of an existing client is edited, a new client is added, with the same parameters as the previous client.
- **Client Name:** Enter a descriptive name for this client so that it is easier to remember which VoIP gateway the record refers to. For example, this name could refer to the location of the gateway.
- **Template:** Click the pull down arrow if you wish to use one of the templates that you configured. This applies the parameters from that template to the remaining fields. Parameter values that are applied by the template are indicated by a check mark in the box to the right of that parameter. Uncheck this box if you want to enter a different value. If templates are not used, the box to the right of the parameters is colored gray and is not selectable.
- **IP:** Enter the IP address you want to apply to the VoIP gateway. Use the normal dotted decimal format.
- **Subnet:** Enter the subnet mask you want to apply to the VoIP gateway. Use the normal dotted decimal format. Ensure that the subnet mask is correct. If the address is incorrect, the VoIP gateway may not function until the entry is corrected and a BootP reset is applied.
- **Gateway:** Enter the IP address for the data network gateway used on this subnet that you want to apply to the VoIP gateway. The data network gateway is a device, such as a router, that is used in the data network to interface this subnet to the rest of the enterprise network.
- **TFTP Server IP:** This field contains the IP address of the TFTP utility that is used for file transfer of software and initialization files to the gateway. When creating a new client, this field is populated with the IP address used by the BootP Tool. If a different TFTP utility is to be used, change the IP address in this field to the IP address used by the other utility.

- **Boot File:** This field specifies the file name for the software (*cmp*) file that is loaded by the TFTP utility to the VoIP gateway after the VoIP gateway receives the BootReply message. The actual software file is located in the TFTP utility directory that is specified in the BootP **Preferences** window. The software file can be followed by command line switches. For information on available command line switches, refer to Section C.11.6 on page 359.

Notes:

- Once the software file loads into the gateway, the gateway begins functioning from that software. In order to save this software to non-volatile memory, (only the *cmp* file, i.e., the compressed firmware file, can be burned to your device's flash memory), the -fb flag must be added to the end of the file name. If the file is not saved, the gateway reverts to the old version of software after the next reset.
- The **Boot file** field can contain up to two file names: *cmp* file name to be used for load of application image and *ini* file name to be used for gateway provisioning. Either one, two or no file names can appear in the **Boot file** field. To use both file names use the ';' separator (without blank spaces) between the xxx.*cmp* and the yyy.*ini* files (e.g., *ram.cmp;SIPgw.ini*).

- ***ini* File:** This field specifies the configuration *ini* file that the gateway uses to program its various settings. Enter the name of the file that is loaded by the TFTP utility to the VoIP gateway after it receives the BootReply message. The actual *ini* file is located in the TFTP utility directory that is specified in the BootP Preferences window.

C.11.6 Using Command Line Switches

You can add command line switches in the field **Boot File**.

➢ To use a Command Line Switch, take these 4 steps:

1. In the field **Boot File**, leave the file name defined in the field as it is (e.g., ramxxx.*cmp*).
2. Place your cursor after *cmp.*
3. Press the space bar.
4. Type in the switch you require.

Example: ‘ramxxx.*cmp* –fb’ to burn flash memory.

‘ramxxx.*cmp* -fb -em 4’ to burn flash memory and for Ethernet Mode 4 (auto-negotiate).

Table C-1 lists and describes the switches that are available:

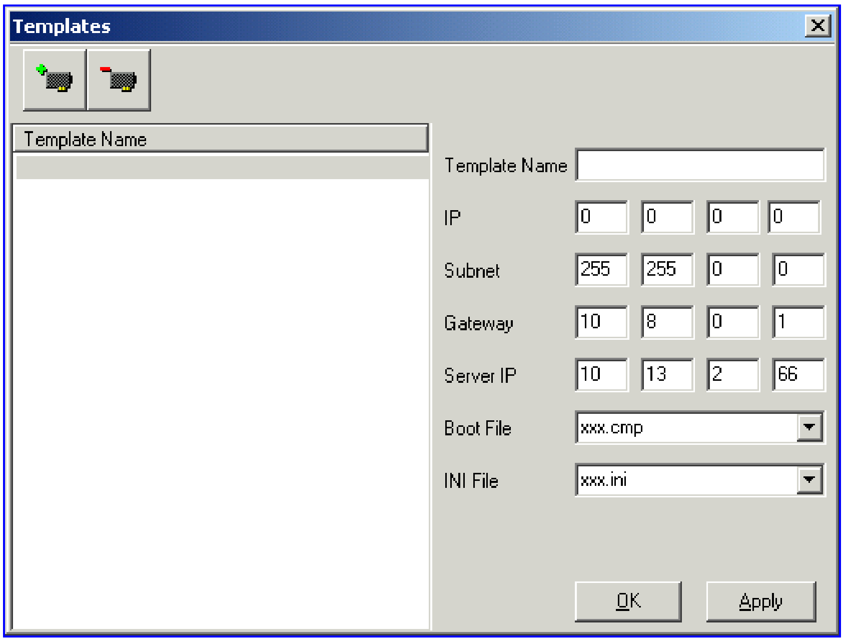
Table C-1: Command Line Switch Descriptions

<table>
<tr><th>Switch</th><th colspan="2">Description</th></tr>
<tr><td>-fb</td><td colspan="2">Burn ram.cmp in flash (only for cmp files)</td></tr>
<tr><td>-em #</td><td colspan="2">Use this switch to set Ethernet mode.
0 = 10 Base-T half-duplex
1 = 10 Base-T full-duplex
2 = 100 Base-TX half-duplex
3 = 100 Base-TX full-duplex
4 = auto-negotiate (default)
For detailed information on Ethernet interface configuration, refer to Section 9.1 on page 247.</td></tr>
<tr><td rowspan="2">-br</td><td colspan="2">This parameter is used to:
Note: This switch takes effect only from the next gateway reset.</td></tr>
<tr><td>Set the number of BootP requests the gateway sends during start-up. The gateway stops sending BootP requests when either BootP reply is received or number of retries is reached.
1 = 1 BootP retry, 1 second
2 = 2 BootP retries, 3 seconds
3 = 3 BootP retries, 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 indefinitely</td><td>Set the number of DHCP packets the gateway sends.
After all packets were sent, if there's still no reply, the gateway loads from flash.
1 = 4 DHCP packets
2 = 5 DHCP packets
3 = 6 DHCP packets (default)
4 = 7 DHCP packets
5 = 8 DHCP packets
6 = 9 DHCP packets
7 = 10 DHCP packets
15 = 18 DHCP packets</td></tr>
<tr><td>-bs</td><td colspan="2">Use –bs 1 to enable the Selective BootP mechanism.
Use –bs 0 to disable the Selective BootP mechanism.
The Selective BootP mechanism (available from Boot version 1.92) enables the gateway’s integral BootP client to filter unsolicited BootP/DHCP replies (accepts only BootP replies that contain the text ‘AUDC’ in the vendor specific information field). This option is useful in environments where enterprise BootP/DHCP servers provide undesired responses to the gateway’s BootP requests.</td></tr>
<tr><td>-be</td><td colspan="2">Use -be 1 for the device to send device-related initial startup information (such as board type, current IP address, software version) in the vendor specific information field (in the BootP request). This information can be viewed in the main screen of the BootP/TFTP, under column 'Client Info‘ (refer to Figure C-1 showing BootP/TFTP main screen with the column 'Client Info' on the extreme right). For a full list of the vendor specific Information fields, refer to Section 7.3.2 on page 214.
Note: This option is not available on DHCP servers.</td></tr>
</table>

C.12 Managing Client Templates

Templates can be used to simplify configuration of clients when most of the parameters are the same.

Figure C-5: Templates Screen



➢ To create a new template, take these 4 steps:

1. Click the **Add New Template** button .
2. Fill in the default parameter values in the parameter fields.
3. Click **Apply** to save this new template.
4. Click **OK** when you are finished adding templates.

➢ To edit an existing template, take these 4 steps:

1. Select the template by clicking on its name from the list of templates in the window.
2. Make changes to the parameters, as required.
3. Click **Apply** to save this new template.
4. Click **OK** when you are finished editing templates.

➢ To delete an existing template, take these 3 steps:

1. Select the template by clicking its name from the list of templates in the window.
2. Click the **Delete Current Template** button ; a warning pop up message appears.
3. To delete the template, click **Yes**. Note that if this template is currently in use, the template cannot be deleted.

D RTP/RTCP Payload Types and Port Allocation

RTP Payload Types are defined in RFC 3550 and RFC 3551. We have added new payload types to enable advanced use of other coder types. These types are reportedly not used by other applications.

D.1 Packet Types Defined in RFC 3551

Table D-1: Packet Types Defined in RFC 3551

Payload Type	Description	Basic Packet Rate [msec]
0	G.711 µ-Law	10,20
2	G.726-32	10,20
4	G.723 (6.3/5.3 kbps)	30
8	G.711 A-Law	10,20
18	G.729A/B	20
200	RTCP Sender Report	Randomly, approximately every 5 seconds (when packets are sent by channel)
201	RTCP Receiver Report	Randomly, approximately every 5 seconds (when channel is only receiving)
202	RTCP SDES packet	
203	RTCP BYE packet	
204	RTCP APP packet	

D.2 Defined Payload Types

Table D-2: Defined Payload Types

Payload Type	Description	Basic Packet Rate [msec]
96	RFC 2833 DTMF relay	20
102	Fax Bypass	20
103	Modem Bypass	20
104	RFC 2198 (Redundancy)	Same as channel's voice coder.
105	NSE Bypass	

D.3 Default RTP/RTCP/T.38 Port Allocation

The following table shows the default RTP/RTCP/T.38 port allocation.

Table D-3: 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
9	6080	6081	6082
10	6090	6091	6092
11	6100	6101	6102
12	6110	6111	6112
13	6120	6121	6122
14	6130	6131	6132
15	6140	6141	6142
16	6150	6151	6152
17	6160	6161	6162
18	6170	6171	6172
19	6180	6181	6182
20	6190	6191	6192
21	6200	6201	6202
22	6210	6211	6212
23	6220	6221	6222
24	6230	6231	6232

Note: To configure the gateway to use the same port for both RTP and T.38 packets, set the parameter 'T38UseRTPPort' to 1.

E Accessory Programs and Tools

The accessory applications and tools shipped with the device provide you with friendly interfaces that enhance device usability and smooth your transition to the new VoIP infrastructure. The following applications are available:

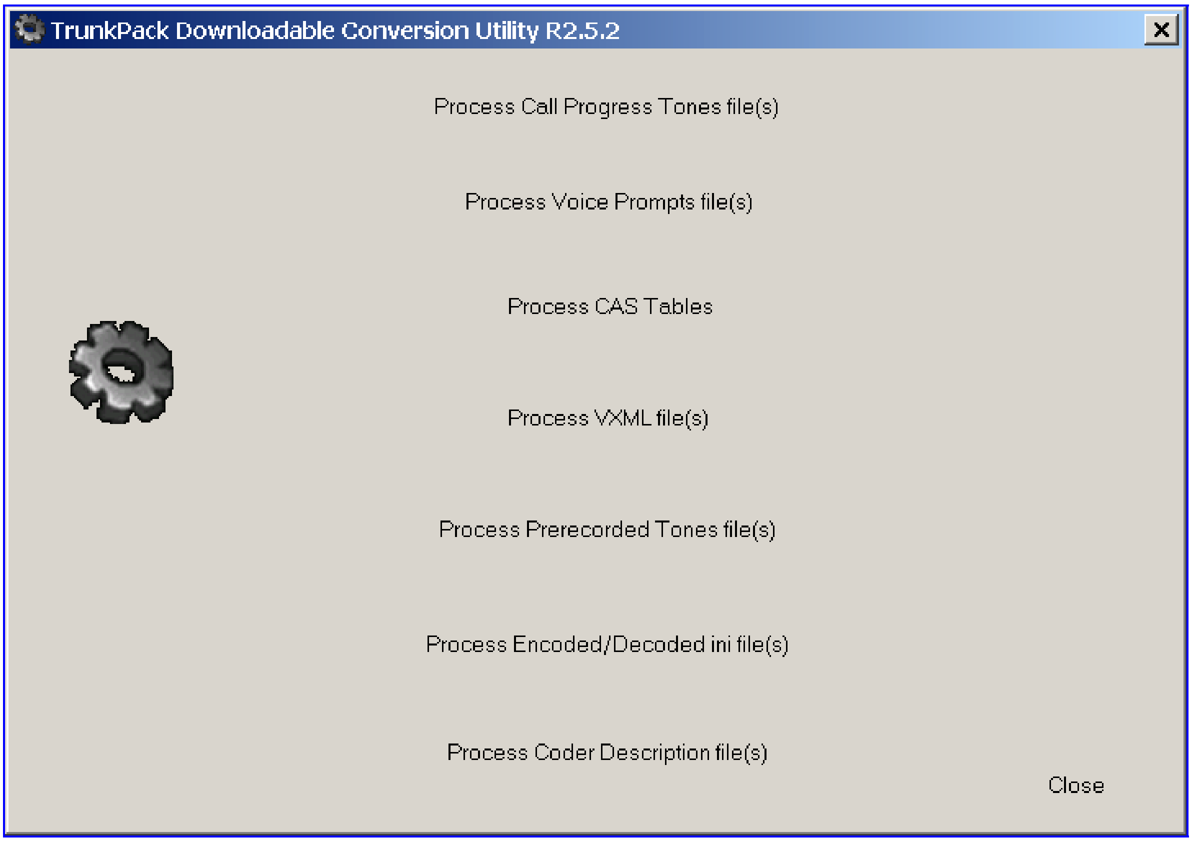
- TrunkPack Downloadable Conversion Utility (refer to Section E.1 below).
- Call Progress Tones Wizard (refer to Section E.1.3 on page 366).

E.1 TrunkPack Downloadable Conversion Utility

Use the TrunkPack Downloadable Conversion Utility to:

- Create a loadable Call Progress Tones file (refer to Section E.1.1 on page 364).
- Encode / decode an *ini* file (refer to Section E.1.2 on page 365).
- Create a loadable Prerecorded Tones file (refer to Section E.1.3 on page 366).

Figure E-1: TrunkPack Downloadable Conversion Utility Opening Screen



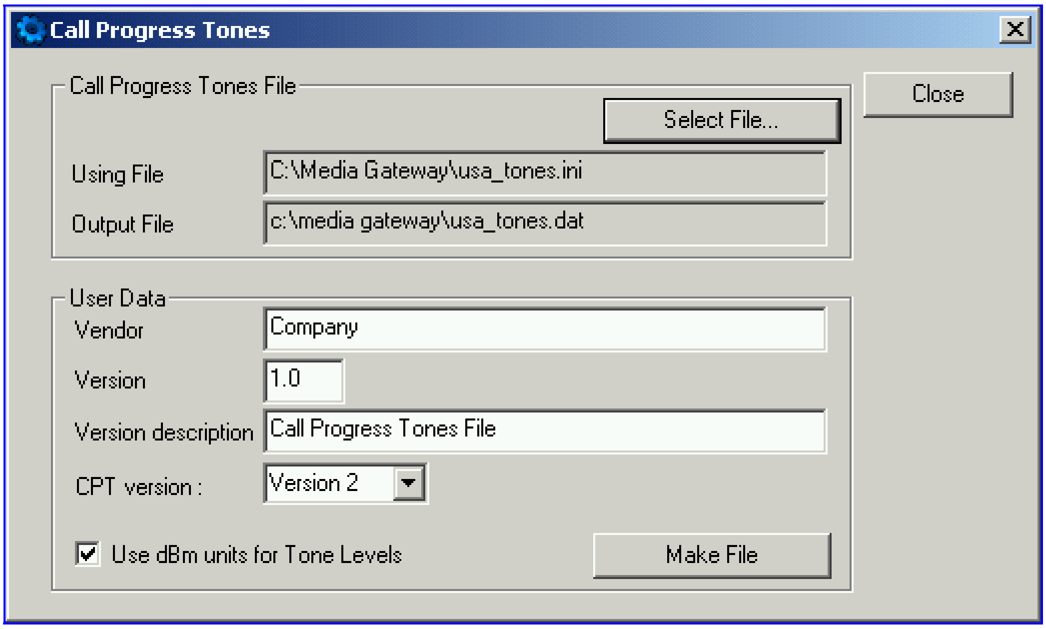
E.1.1 Converting a CPT *ini* File to a Binary *dat* File

For detailed information on creating a CPT *ini* file, refer to Section 15.1 on page 325.

➢ To convert a CPT *ini* file to a binary *dat* file, take these 10 steps:

1. Execute the TrunkPack Downloadable Conversion Utility, DConvert.exe (supplied with the software package); the utility's main screen opens (shown in Figure E-1).
2. Click the **Process Call Progress Tones File(s)** button; the 'Call Progress Tones' screen, shown in Figure E-2, opens.

Figure E-2: Call Progress Tones Conversion Screen



3. Click the **Select File…** button that is in the 'Call Progress Tone File' box.
4. Navigate to the folder that contains the CPT *ini* file you want to convert.
5. Click the *ini* file and click the **Open** button; the name and path of both the *ini* file and the (output) *dat* file appears in the fields below the Select File button.
6. Enter the Vendor Name, Version Number and Version Description in the corresponding required fields under the 'User Data' section.
   - The maximum length of the Vendor field is 256 characters.
   - The format of the Version field is composed of two integers separated by a period '.' (e.g., 1.2, 23.4, 5.22).
   - The maximum length of the Version Description field is 256 characters.
7. The default value of the CPT Version drop-down list is Version 3. Do one of the following:
   - If the software version you are using is prior to version 4.4, select Version 1 (to maintain backward compatibility).
   - If the software version you are using is 4.4, select Version 2.
   - Otherwise, leave the value at its default.
8. Check the 'Use dBm units for Tone Levels' check box. Note that the levels of the Call Progress Tones (in the CPT file) must be in -dBm units.
9. Click the **Make File** button; you're prompted that the operation (conversion) was successful.
10. Close the application.

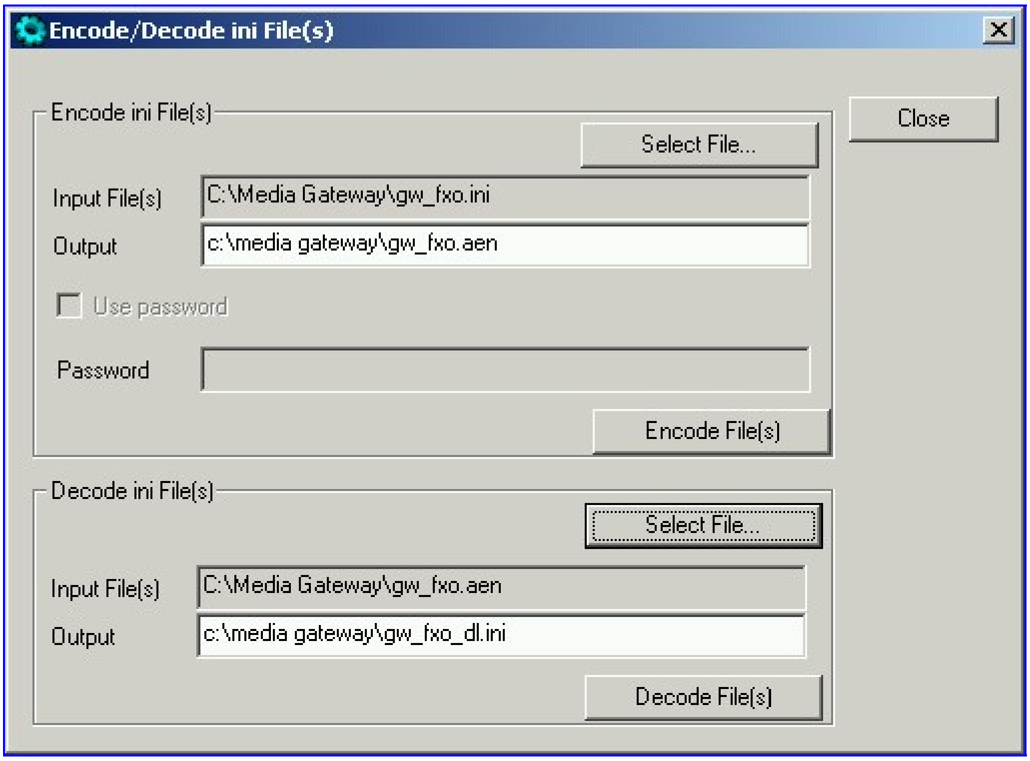
E.1.2 Encoding / Decoding an *ini* File

For detailed information on secured *ini* file, refer to Section 6.1 on page 209.

➢ To encode an *ini* file, take these 6 steps:

1. Execute the TrunkPack Downloadable Conversion Utility, DConvert.exe (supplied with the software package); the utility's main screen opens (shown in Figure E-1).
2. Click the **Process Encoded/Decoded *ini* file(s)** button; the 'Encode/Decode *ini* File(s)' screen, shown in Figure E-3, opens.

Figure E-3: Encode/Decode *ini* File(s) Screen



3. Click the **Select File…** button under the 'Encode *ini* File(s)' section.
4. Navigate to the folder that contains the *ini* file you want to encode.
5. Click the *ini* file and click the **Open** button; the name and path of both the *ini* file and the output encoded file appear in the fields under the **Select File** button. Note that the name and extension of the output file can be modified.
6. Click the **Encode File(s)** button; an encoded *ini* file with the name and extension you specified is created.

➢ To decode an encoded *ini* file, take these 4 steps:

1. Click the **Select File…** button under the 'Decode *ini* File(s)' section.
2. Navigate to the folder that contains the file you want to decode.
3. Click the file and click the **Open** button. the name and path of both the encode *ini* file and the output decoded file appear in the fields under the **Select File** button. Note that the name of the output file can be modified.
4. Click the **Decode File(s)** button; a decoded *ini* file with the name you specified is created.

Note that the decoding process verifies the input file for validity. Any change made to the encoded file causes an error and the decoding process is aborted.

E.1.3 Creating a Loadable Prerecorded Tones File

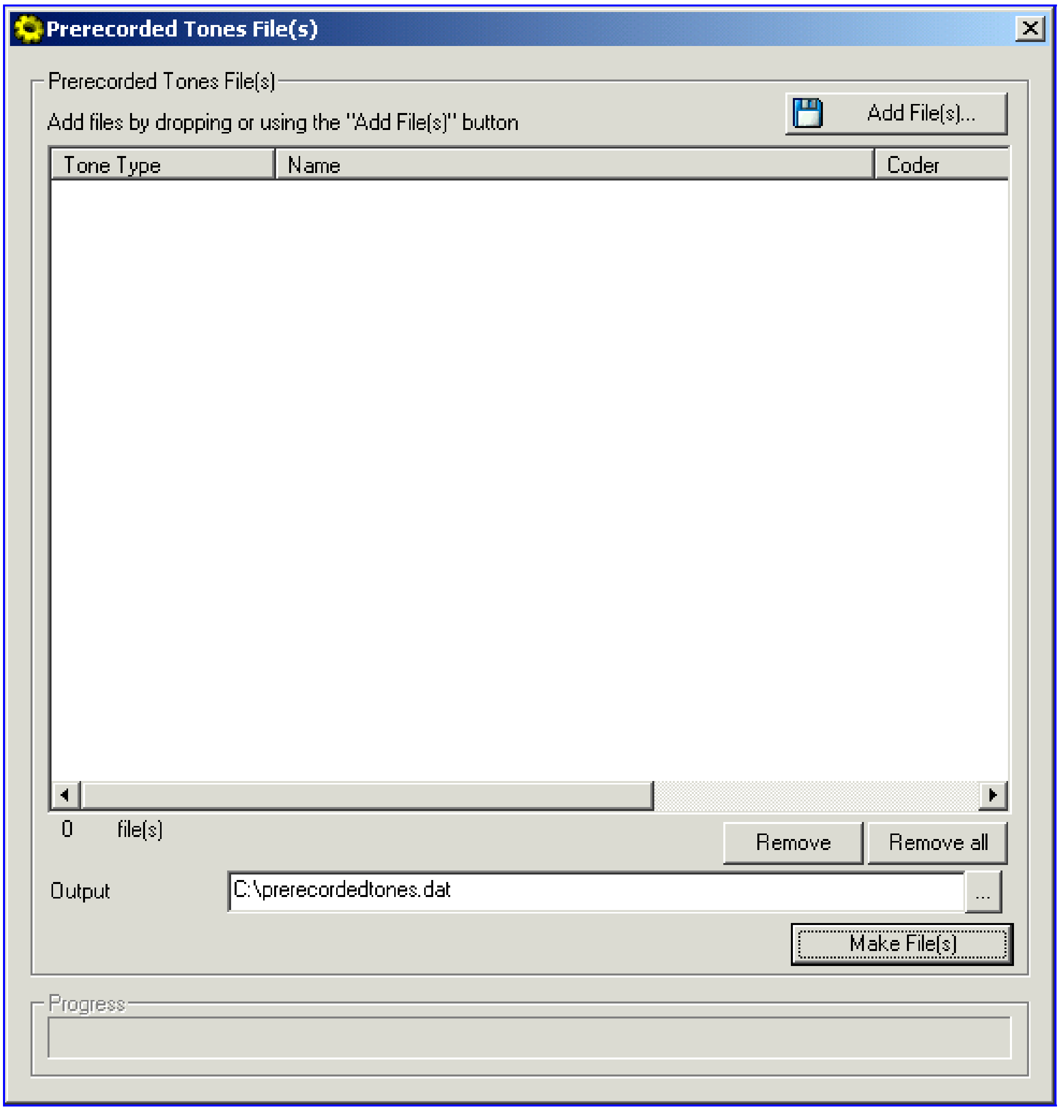
For detailed information on the PRT file, refer to Section 15.2 on page 330.

Note: The maximum size of a PRT file that can be loaded to the gateway is 100 KB.

➢ **To create a loadable PRT *dat* file from your raw data files, take these 7 steps:**

1. Prepare the prerecorded tones (raw data PCM or L8) files you want to combine into a single *dat* file using standard recording utilities.
2. Execute the TrunkPack Downloadable Conversion utility, DConvert.exe (supplied with the software package); the utility's main screen opens (shown in Figure E-1).
3. Click the **Process Prerecorded Tones File(s)** button; the Prerecorded Tones File(s) screen, shown in Figure E-4, opens.

Figure E-4: Prerecorded Tones Screen



4. To add the prerecorded tone files (you created in Step 1) to the ‘Prerecorded Tones’ screen follow one of these procedures:
   - Select the files and drag them to the ‘Prerecorded Tones’ screen.
   - Click the **Add File(s)** button; the ‘Select Files’ screen opens. Select the required Prerecorded Tone files and click the **Add>>** button. Close the ‘Select Files’ screen.
5. For each raw data file, define a Tone Type, a Coder and a Default Duration by completing the following steps:
   - Double-click or right-click the required file; the ‘File Data’ window (shown in Figure E-5) appears.
   - From the ‘Type’ drop-down list, select the tone type this raw data file is associated with.
   - From the ‘Coder’ drop-down list, select the coder that corresponds to the coder this raw data file was *originally* recorded with.
   - In the ‘Description’ field, enter additional identifying information (optional).
   - In the ’Default’ field, enter the default duration this raw data file is repeatedly played.
   - Close the ‘File Data’ window (press the **Esc** key to cancel your changes); you are returned to the Prerecorded Tones File(s) screen.

Figure E-5: File Data Window

6. In the ‘Output’ field, specify the output directory in which the PRT file is generated followed by the name of the PRT file (the default name is *prerecordedtones.dat*). Alternatively, use the Browse button to select a different output file. Navigate to the desired file and select it; the selected file name and its path appear in the ‘Output’ field.
7. Click the **Make File(s)** button; the Progress bar at the bottom of the window is activated. The *dat* file is generated and placed in the directory specified in the ‘Output’ field. A message box informing you that the operation was successful indicates that the process is completed.

E.2 Call Progress Tones Wizard

This section describes the Call Progress Tones Wizard (CPTWizard), an application designed to facilitate the provisioning of a MediaPack/FXO gateway by recording and analyzing Call Progress Tones generated by any PBX or telephone network.

E.2.1 About the Call Progress Tones Wizard

The Call Progress Tones wizard helps detect the Call Progress Tones generated by your PBX (or telephone exchange) and creates a basic Call Progress Tones *ini* file (containing definitions for all relevant Call Progress Tones), providing a good starting point when configuring an MediaPack/FXO gateway. This *ini* file can then be converted to a *dat* file that can be loaded to the gateway using the TrunkPack Downloadable Conversion utility.

To use this wizard, a MediaPack/FXO gateway connected to your PBX with two physical phone lines is required. This gateway must be configured with factory-default settings and shouldn’t be used for phone calls during the operation of the wizard.

Note that firmware version 4.2 and above is required on the gateway.

E.2.2 Installation

The CPTWizard can be installed on any Windows 2000 or Windows XP based PC. Windows-compliant networking and audio peripherals are required for full functionality.

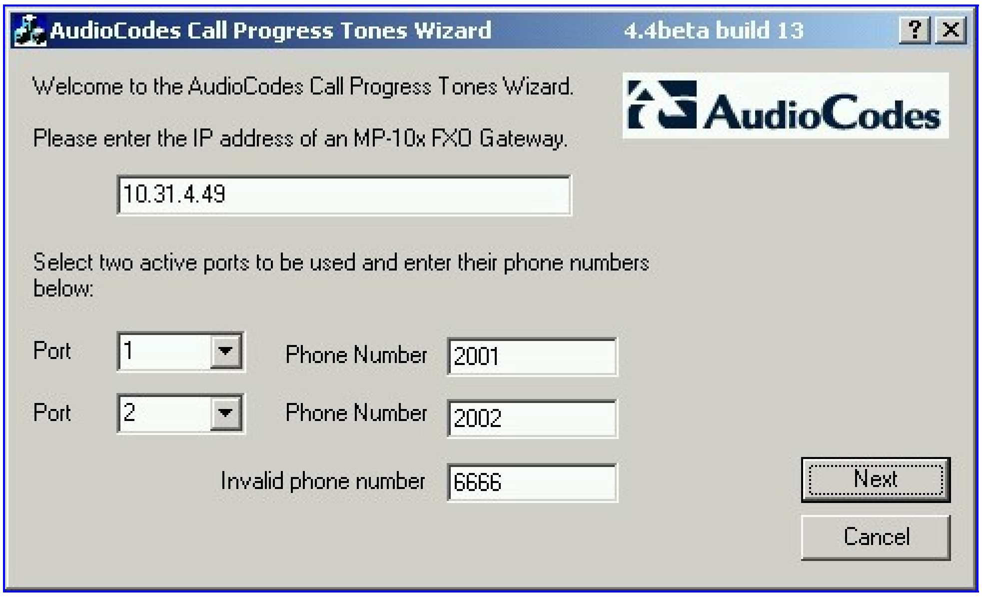
To install the CPTWizard, copy the files from the supplied installation kit to any folder on your PC. No further setup is required (approximately 5 MB of hard disk space are required).

E.2.3 Initial Settings

➢ To start the CPTWizard, take these 5 steps:

1. Execute the CPTWizard.exe file; the wizard's initial settings screen is displayed.

Figure E-6: Initial Settings Screen



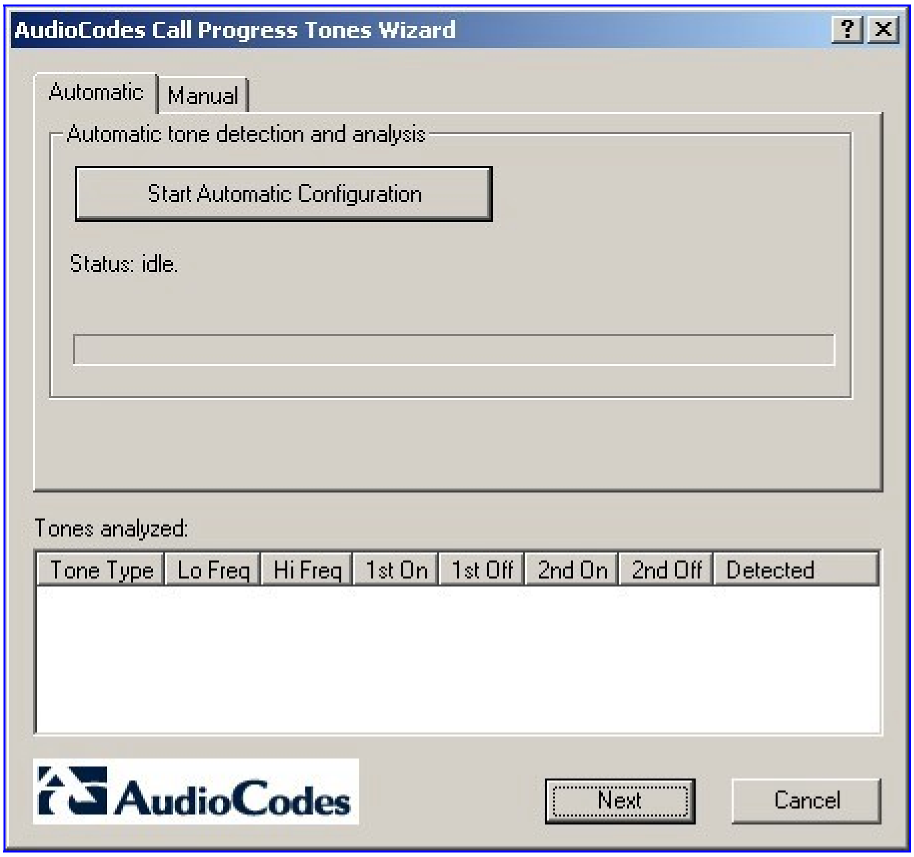
2. Enter the IP address of the MediaPack/FXO gateway you are using.
3. Select the gateway's ports that are connected to your PBX, and specify the phone number of each extension.
4. In the 'Invalid phone number' field, enter a number that generates a 'fast busy' tone when dialed. Usually, any incorrect phone number should cause a 'fast busy' tone.
5. Click **Next**.

Note: The CPTWizard communicates with the FXO gateway via TPNCP (TrunkPack Network Control Protocol). If this protocol has been disabled in the gateway configuration, the CPTWizard doesn't display the next screen and an error is reported.

E.2.4 Recording Screen – Automatic Mode

After the connection to the MediaPack/FXO gateway is established, the recording screen is displayed.

Figure E-7: Recording Screen –Automatic Mode

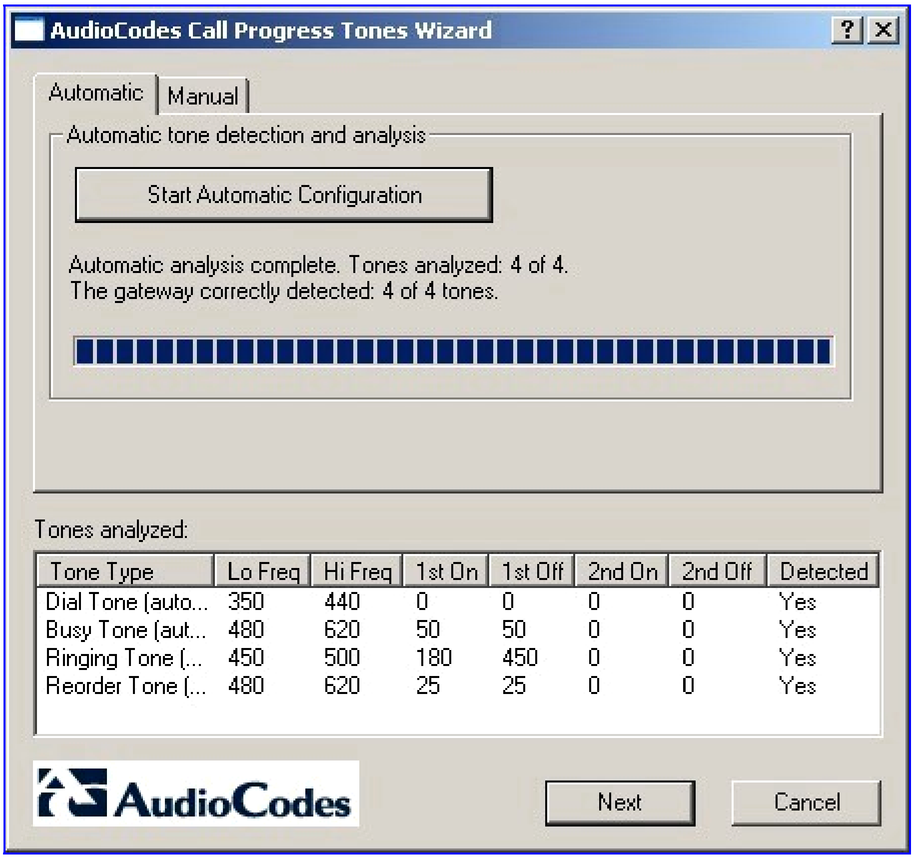


➢ To start recording in automatic mode, take these 3 steps:

1. Click the **Start Automatic Configuration** button; the wizard starts the following Call Progress Tones detection sequence (the operation takes approximately 60 seconds to complete):
   - Sets port 1 offhook, listens to the dial tone.
   - Sets port 1 and port 2 offhook, dials the number of port 2, listens to the busy tone.
   - Sets port 1 offhook, dials the number of port 2, listens to the Ringback tone.
   - Sets port 1 offhook, dials an invalid number, listens to the reorder tone.

2. The wizard then analyzes the recorded Call Progress Tones and displays a message specifying the tones that were detected (by the gateway) and analyzed (by the wizard) correctly. At the end of a successful detection operation, the detected Call Progress Tones are displayed in the **Tones Analyzed** pane (refer to Figure E-8).

Figure E-8: Recording Screen after Automatic Detection



3. All four Call Progress Tones are saved (as standard A-law PCM at 8000 bits per sample) in the same directory as the CPTWizard.exe file is located, with the following names:
   - cpt_recorded_dialtone.pcm
   - cpt_recorded_busytone.pcm
   - cpt_recorded_ringtone.pcm
   - cpt_recorded_invalidtone.pcm

Notes:

- If the gateway is configured correctly (with a Call Progress Tones *dat* file loaded to the gateway), all four Call Progress Tones are detected by the gateway. By noting whether the gateway detects the tones or not, you can determine how well the Call Progress Tones *dat* file matches your PBX. During the first run of the CPTWizard, it is likely that the gateway does not detect any tones.
- Some tones cannot be detected by the MediaPack gateway hardware (such as 3-frequency tones and complex cadences). CPTWizard is therefore limited to detecting only those tones that can be detected on the MediaPack gateway.

At this stage, you can either click **Next** to generate a Call Progress Tones *ini* file and terminate the wizard, or continue to manual recording mode.

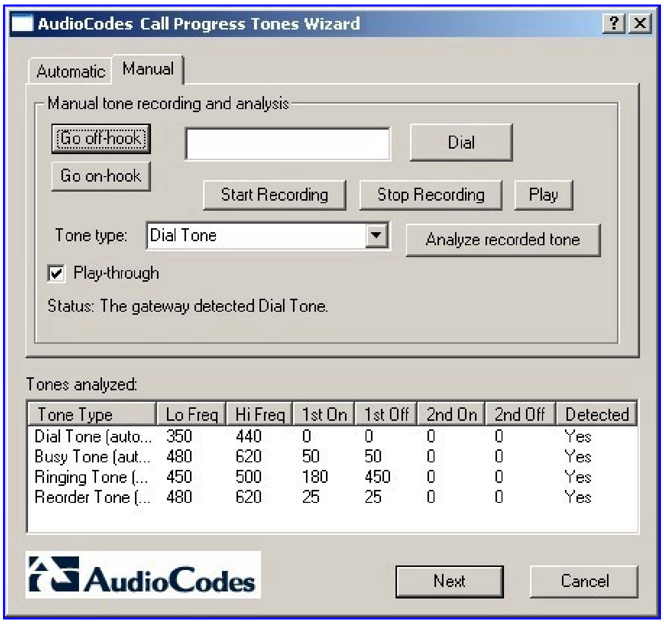
E.2.5 Recording Screen – Manual Mode

In manual mode you can record and analyze tones, included in the Call Progress Tones *ini* file, in addition to those tones analyzed when in automatic mode.

➢ To start recording in manual mode, take these 6 steps:

1. Click the **Manual** tab at the top of the recording screen, the manual recording screen is displayed.

Figure E-9: Recording Screen - Manual Mode



2. Check the **play-through** check box to hear the tones through your PC speakers.
3. Click the **Go offhook** button, enter a number to dial in the **Dial** field, and click the **Dial** button. When you're ready to record, click the **Start Recording** button; when the desired tone is complete, click **Stop Recording**. (The recorded tone is saved as 'cpt_manual_tone.pcm'.)

Note: Due to some PC audio hardware limitations, you may hear 'clicks' in play-through mode. It is safe to ignore these clicks.

4. Select the tone type from the drop-down list and click **Analyze recorded tone**; the analyzed tone is added to the **Tones analyzed** list at the bottom of the screen. It is possible to record and analyze several different tones for the same tone type (e.g., different types of 'busy' signal).
5. Repeat the process for more tones, as necessary.
6. When you're finished adding tones to the list, click **Next** to generate a Call Progress Tones *ini* file and terminate the wizard.

E.2.6 The Call Progress Tones *ini* File

After the Call Progress Tones detection is complete, a text file named call_progress_tones.ini is created in the same directory as the directory in which the CPTWizard.exe is located. This file contains:

- Information about each tone that was recorded and analyzed by the wizard. This information includes frequencies and cadence (on/off) times, and is required for using this file with the TrunkPack Downloadable Conversion utility.

Figure E-10: Call Progress Tone Properties

```
[CALL PROGRESS TONE #1]
Tone Type=1
Low Freq [Hz]=350
High Freq [Hz]=440
Low Freq Level [-dBm]=0
High Freq Level [-dBm]=0
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
```

- Information related to possible matches of *each* tone with the CPTWizard's internal database of well-known tones. This information is specified as comments in the file, and is ignored by the TrunkPack Downloadable Conversion utility.

Figure E-11: Call Progress Tone Database Matches

```
# Recorded tone: Busy Tone (automatic configuration)
## Matches: PBX name=ITU Anguilla, Tone name=Busy tone
## Matches: PBX name=ITU Antigua and Barbuda, Tone name=Busy tone
## Matches: PBX name=ITU Barbados, Tone name=Busy tone
## Matches: PBX name=ITU Bermuda, Tone name=Busy tone
## Matches: PBX name=ITU British Virgin Islan, Tone name=Busy tone
## Matches: PBX name=ITU Canada, Tone name=Busy tone
## Matches: PBX name=ITU Dominica (Commonweal, Tone name=Busy tone
## Matches: PBX name=ITU Hongkong, China, Tone name=Busy tone
## Matches: PBX name=ITU Jamaica, Tone name=Busy tone
## Matches: PBX name=ITU Korea (Republic of), Tone name=Busy tone
## Matches: PBX name=ITU Montserrat, Tone name=Busy tone
```

- Information related to matches of *all* tones recorded with the CPTWizard's internal database. The database is scanned to find one or more PBX definitions that match all recorded tones (i.e., dial tone, busy tone, ringing tone, reorder tone and any other manually-recorded tone – all match the definitions of the PBX). If a match is found, the entire PBX definition is reported (as comments) in the *ini* file using the same format.

Figure E-12: Full PBX/Country Database Match

```
## Some tones matched PBX/country Audc US
## Additional database tones guessed below (remove #'s to use).
#
# # Audc US, US Ringback tone
# [CALL PROGRESS TONE #5]
# Tone Type=2
# Low Freq [Hz]=450
# High Freq [Hz]=500
# Low Freq Level [-dBm]=0
# High Freq Level [-dBm]=0
# First Signal On Time [10msec]=180
# First Signal Off Time [10msec]=450
# Second Signal On Time [10msec]=0
# Second Signal Off Time [10msec]=0
```

Notes:

- If a match is found in the database, consider using the database's definitions instead of the recorded definitions, as they might be more accurate.
- For full operability of the MediaPack/FXO gateway, it may be necessary to edit this file and add more Call Progress Tone definitions. Sample Call Progress Tones *ini* files are available in the release package.
- When the CPT *ini* file is complete, use the TrunkPack Downloadable Conversion utility to create a loadable CPT *dat* file. After loading this file to the gateway, repeat the automatic detection procedure discussed above, and verify that the gateway detects all four Call Progress Tones correctly.

E.2.7 Adding a Reorder Tone to the CPT File

The following procedure describes how to add a Reorder tone that a PBX generates to indicate a disconnected call, to the CPT file.

➢ To add a Reorder tone to the CPT file, take these 11 steps:

1. Make a call (using G.711) between the MP FXO, which is connected to the PBX, and a remote entity in the IP network.
2. Capture the call using a "network sniffer" such as Wireshark.
3. Disconnect the call from the PBX side, and then wait approximately 30 seconds before stopping the Wireshark recording.
4. In the network trace, locate the RTP stream sent from the FXO.
5. Save the RTP payload on your PC as a *.pcm file by clicking **Save Payload** (**Statistics** menu > **RTP** > **Stream Analysis**. (Note: ensure that you select the ‘forward’ option.)
6. Open the *.pcm file in a voice recording utility such as CoolEdit.
7. Locate the tone that the PBX played to indicate the disconnected call (if such a tone exists).
8. Locate the attributes of the tone -- its frequency and interval (on / off time).
9. In the Call Progress Tones file, add a new Reorder Tone with the attributes you found in the previous step. Ensure that you update the numbers of the successive tones and the total number of tones in the beginning of the file.
10. Create a Call Progress Tones.dat file using the DConvert Utility (refer to Section E.1 on page 363).
11. Load the new file to the gateway, and then reset the gateway.

F SNMP Traps

This section provides information on proprietary SNMP traps currently supported by the gateway. There is a separation between traps that are alarms and traps that are not (logs). Currently all have the same structure made up of the same 11 varbinds (Variable Binding) (1.3.6.1.4.1.5003.9.10.1.21.1).

The source varbind is composed of a string that details the component from which the trap is being sent (forwarded by the hierarchy in which it resides). For example, an alarm from an SS7 link has the following string in its source varbind:

```
acBoard#1/SS7#0/SS7Link#6
```

In this example, the SS7 link number is specified as 6 and is part of the only SS7 module in the device that is placed in slot number 1 (in a chassis) and is the module to which this trap relates. For devices where there are no chassis options the slot number of the gateway is always 1.

F.1 Alarm Traps

The following tables provide information on alarms that are raised as a result of a generated SNMP trap. The component name (described in each of the following headings) refers to the string that is provided in the 'acBoardTrapGlobalsSource' trap varbind. To clear a generated alarm the same notification type is sent but with the severity set to 'cleared'.

F.1.1 Component: Board#<n>

The source varbind text for all the alarms under this component is Board#<n> where n is the slot number. For MediaPack, <n> = 1.

Table F-1: acBoardFatalError Alarm Trap

Alarm:	acBoardFatalError
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.1
Default Severity:	Critical
Event Type:	equipmentAlarm
Probable Cause:	underlyingResourceUnavailable (56)
Alarm Text:	Board Fatal Error: <text>
Status Changes:	
Condition:	Any fatal error
Alarm status:	Critical
<text> value:	A run-time specific string describing the fatal error
Condition:	After fatal error
Alarm status:	Status stays critical until reboot. A clear trap is not sent.
Corrective Action:	Capture the alarm information and the Syslog clause, if active. Contact your first-level support group. The support group will likely want to collect additional data from the device and perform a reset.

Table F-2: acBoardTemperatureAlarm Alarm Trap

Alarm:	acBoardTemperatureAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.3
Default Severity	Critical
Event Type:	equipmentAlarm
Probable Cause:	temperatureUnacceptable (50)
Alarm Text:	Board temperature too high
Status Changes:	
Condition:	Temperature is above 60 degrees C (140 degrees F)
Alarm status:	Critical
Condition:	After raise, temperature falls below 55 degrees C (131 degrees F)
Alarm status:	Cleared
Corrective Action:	Inspect the system. Determine if all fans in the system are properly operating.

Table F-3: acgwAdminStateChange Alarm Trap

Alarm:	acgwAdminStateChange
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.7
Default Severity	Major
Event Type:	processingErrorAlarm
Probable Cause:	outOfService (71)
Alarm Text:	Network element admin state change alarm Gateway is <text>
Status Changes:	
Condition:	Admin state changed to shutting down
Alarm status:	Major
<text value>:	Shutting down. No time limit.
Condition:	Admin state changed to locked
Alarm status:	Major
<text value>:	Locked
Condition:	Admin state changed to unlocked
Alarm status:	Cleared
Corrective Action:	A network administrator has taken an action to lock the device. No corrective action is required.

Table F-4: acOperationalStateChange Alarm Trap

Alarm:	acOperationalStateChange
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.15
Default Severity	Major
Event Type:	processingErrorAlarm
Probable Cause:	outOfService (71)
Alarm Text:	Network element operational state change alarm. Operational state is disabled.
Note:	This alarm is raised if the operational state of the node goes to disabled. The alarm is cleared when the operational state of the node goes to enabled. In IP systems, the operational state of the node is disabled if the device fails to properly initialize.
Status Changes:	
Condition:	Operational state changed to disabled
Alarm status:	Major
Condition:	Operational state changed to enabled
Alarm status:	Cleared
Corrective Action:	In IP systems, check for initialization errors. Look for other alarms and Syslogs that might provide additional information about the error.

Table F-5: acBoardEvResettingBoard Alarm Trap

Alarm:	acBoardEvResettingBoard
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.5
Default Severity:	critical
Event Type:	equipmentAlarm
Probable Cause:	outOfService (71)
Alarm Text:	User resetting board
Status Changes:	
Condition:	When a soft reset is triggered via the Web interface or SNMP.
Alarm status:	Critical
Condition:	After raise
Alarm status:	Status stays critical until reboot. A clear trap is not sent.
Corrective Action:	A network administrator has taken action to reset the device. No corrective action is required.

Table F-6: acBoardCallResourcesAlarm Alarm Trap

Alarm:	acBoardCallResourcesAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.8
Default Severity:	Major
Event Type:	processingErrorAlarm
Probable Cause:	softwareError (46)
Alarm Text:	Call resources alarm
Status Changes:	
Condition:	Number of free channels exceeds the predefined RAI *high* threshold.
Alarm status:	Major
Note:	To enable this alarm the RAI mechanism must be activated (EnableRAI = 1).
Condition:	Number of free channels falls below the predefined RAI *low* threshold.
Alarm status:	Cleared

Table F-7: acBoardControllerFailureAlarm Alarm Trap

Alarm:	acBoardControllerFailureAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.9
Default Severity:	Minor
Event Type:	processingErrorAlarm
Probable Cause:	softwareError (46)
Alarm Text:	Controller failure alarm
Status Changes:	
Condition:	Proxy has not been found
Alarm status:	Major
Additional info:	Proxy not found. Use internal routing or Proxy lost. looking for another Proxy
Condition:	Proxy is found. The clear message includes the IP address of the located Proxy.
Alarm status:	Cleared

Table F-8: acBoardOverloadAlarm Alarm Trap

Alarm:	acBoardOverloadAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.11
Default Severity:	Major
Event Type:	processingErrorAlarm
Probable Cause:	softwareError (46)
Alarm Text:	Board overload alarm
Status Changes:	
Condition:	An overload condition exists in one or more of the system components.
Alarm status:	Major
Condition:	The overload condition passed
Alarm status:	Cleared

F.1.2 Component: AlarmManager#0

The source varbind text for all the alarms under this component is *Board#*<n>*/AlarmManager#0* where n is the slot number.

Table F-9: acActiveAlarmTableOverflow Alarm Trap

Alarm:	acActiveAlarmTableOverflow
OID:	1.3.6.1.4.15003.9.10.1.21.2.0.12
Default Severity:	Major
Event Type:	processingErrorAlarm
Probable Cause:	resourceAtOrNearingCapacity (43)
Alarm Text:	Active alarm table overflow
Status Changes:	
Condition:	Too many alarms to fit in the active alarm table
Alarm status:	Major
Condition:	After raise
Alarm status:	Status stays major until reboot. A clear trap is not sent.
Note:	The status stays major until reboot as it denotes a possible loss of information until the next reboot. If an alarm is raised when the table is full, it is possible that the alarm is active, but does not appear in the active alarm table.
Corrective Action:	Some alarm information may have been lost, but the ability of the device to perform its basic operations has not been impacted. A reboot is the only way to completely clear a problem with the active alarm table. Contact your first-level group.

F.1.3 Component: EthernetLink#0

The source varbind text for all the alarms under this component is *Board#*<n>*/EthernetLink#0* where n is the slot number. This trap relates to the Ethernet Link Module (the #0 numbering doesn't apply to the physical Ethernet link).

Table F-10: acBoardEthernetLinkAlarm Alarm Trap

Alarm:	acBoardEthernetLinkAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.10
Default Severity:	Critical
Event Type:	equipmentAlarm
Probable Cause:	underlyingResourceUnavailable (56)
Alarm Text:	Ethernet link alarm: <text>
Status Changes:	
Condition:	Fault on single interface
Alarm status:	major
<text> value:	Redundant link is down
Condition:	Fault on both interfaces
Alarm status:	critical
<text> value:	No Ethernet link
Condition:	Both interfaces are operational
Alarm status:	cleared
Corrective Action:	Ensure that both Ethernet cables are plugged into the back of the system. Inspect the system's Ethernet link lights to determine which interface is failing. Reconnect the cable or fix the network problem

F.1.4 Log Traps (Notifications)

This section details traps that are not alarms. These traps are sent with the severity varbind value of 'indeterminate'. These traps don't 'clear', they don't appear in the alarm history or active tables. One log trap that does send clear is acPerformanceMonitoringThresholdCrossing.

Table F-11: acKeepAlive Log Trap

Trap:	acKeepAlive
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.16
Default Severity:	Indeterminate
Event Type:	other (0)
Probable Cause:	other (0)
Trap Text:	Keep alive trap
Status Changes:	
Condition:	The STUN client in is enabled and identified a NAT device or doesn't locate the STUN server. The *ini* file contains the following line: 'SendKeepAliveTrap=1'
Trap status:	Trap is sent
Note:	Keep-alive is sent every 9/10 of the time defined in the parameter NatBindingDefaultTimeout.

Table F-12: acPerformanceMonitoringThresholdCrossing Log Trap

Trap:	acPerformanceMonitoringThresholdCrossing
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.27
Default Severity:	Indeterminate
Event Type:	other (0)
Probable Cause:	other (0)
Trap Text:	"Performance: Threshold trap was set", with source = name of performance counter which caused the trap
Status Changes:	
Condition:	A performance counter has crossed the high threshold
Trap status:	Indeterminate
Condition:	A performance counter has crossed the low threshold
Trap status:	cleared

Table F-13: acHTTPDownloadResult Log Trap

Trap:	acHTTPDownloadResult
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.28
Default Severity:	Indeterminate
Event Type:	processingErrorAlarm (3) for failures and other (0) for success.
Probable Cause:	other (0)
Status Changes:	
Condition:	Successful HTTP download.
Trap text:	HTTP Download successful
Condition:	Failed download.
Trap text:	HTTP download failed, a network error occurred.
Note:	There are other possible textual messages describing NFS failures or success, FTP failure or success.

F.1.5 Other Traps

The following are provided as SNMP traps and are not alarms.

Table F-14: coldStart Trap

Trap Name:	coldStart
OID:	1.3.6.1.6.3.1.1.5.1
MIB:	SNMPv2-MIB
Note:	This is a trap from the standard SNMP MIB.

Table F-15: authenticationFailure Trap

Trap Name:	authenticationFailure
OID:	1.3.6.1.6.3.1.1.5.5
MIB:	SNMPv2-MIB

Table F-16: acBoardEvBoardStarted Trap

Trap Name:	acBoardEvBoardStarted
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.4
MIB:	AcBoard
Severity:	cleared
Event Type:	equipmentAlarm
Probable Cause:	Other(0)
Alarm Text:	Initialization Ended
Note:	This is the AudioCodes Enterprise application cold start trap.

F.1.6 Trap Varbinds

Each trap described above provides the following fields (known as 'varbinds'). Refer to the AcBoard MIB for additional details on these varbinds.

- acBoardTrapGlobalsName
- acBoardTrapGlobalsTextualDescription
- acBoardTrapGlobalsSource
- acBoardTrapGlobalsSeverity
- acBoardTrapGlobalsUniqID
- acBoardTrapGlobalsType
- acBoardTrapGlobalsDateAndTime
- acBoardTrapGlobalsProbableCause
- acBoardTrapGlobalsAdditionalInfo1
- acBoardTrapGlobalsAdditionalInfo2
- acBoardTrapGlobalsAdditionalInfo3

Note that 'acBoardTrapGlobalsName' is actually a number. The value of this varbind is 'X' minus 1, where 'X' is the last number in the trap's OID. For example, the 'name' of 'acBoardEthernetLinkAlarm' is '9'. The OID for 'acBoardEthernetLinkAlarm' is 1.3.6.1.4.1.5003. 9.10.1.21.2.0.10.

Reader's Notes

G Installation and Configuration of Apache HTTP Server

This appendix describes the installation and configuration of Apache's HTTP server with Perl script environment (required for recording).

G.1 Windows 2000/XP Operation Systems

Note: For detailed installation information, refer to http://perl.apache.org/docs/2.0/os/win32/config.html.

Additional required software: an uploading script (put.cgi), supplied with the software package.

➢ To configure the Apache HTTP server and mod_perl version 2.0 software, take these 9 steps:

1. Download the third party Perl-5.8-win32-bin.exe, installation file, from the following link: www.apache.org/dist/perl/win32-bin/Perl-5.8-win32-bin.exe. The installation file includes: Apache 2.0.46, Perl 5.8.0 and mod_perl-1.99 (the content of the file and the software version are subject to modification and changes in the future). For full installation instructions refer to www.apache.org/dist/perl/win32-bin/Perl-5.8-win32-bin.readme.
2. To start the installation wizard run the Perl-5.8-win32-bin.exe file.
3. During the installing, you are prompt to determine the Destination Folder under which the package is installed, it is advised to provide a non-spaced path (such as: c:\\directory_name_without_spaces).
4. In the following screen (configuration): uncheck the "Build html docs" and "Configure CPAN pm" checkboxes, if they are present. If you are prompted to bring "nmake" answer no.
5. After the installation is complete, add the "/path/perl/bin" and "/path/apache2/bin" (*path* stands for the path that was previously specified in the "Destination Folder") directories to the system known path. Open the Control Panel→System→Advanced→Environment Variables, inside the System Variables dialog box choose "Path" and click the Edit button; in the opened Variable Value checkbox append both of the paths to the existing list. Restart window in order to activate the new paths.
6. Open the Apache2/conf/httpd.conf file for editing and set the parameter MaxKeepAliveRequests to 0 (enables an unlimited number of requests during a persistent connection – required for multiple consecutive HTTP PUT requests for uploading the file).

7. Open the Apache2/conf/perl.conf file for editing and add the line “Script PUT /perl/put.cgi” after the last line in the following section (note that if the following section is omitted or different in the file, insert it into the file or change it there accordingly):

```
Alias /perl/ "C:/Apache2/perl/
<Location /perl>
SetHandler perl-script
PerlResponseHandler ModPerl::Registry
Options +ExecCGI
PerlOptions +ParseHeaders
</Location>
```

8. Locate the file put.cgi on the supplied software package and copy it into the Apach2\perl\ directory. Change the first line in this file from c:/perl/bin/perl to your perl executable file (this step is required only if mod_perl is not included in your Apache installation).
9. In the apach2\bin directory, from a DOS prompt, type the following commands:

   Apache.exe -n Apache2 -k install
   Apache.exe -n Apache2 -k start

The installation and configuration are finished. You are now ready to start using the HTTP server.

G.2 Linux Operation Systems

> **Note:** It is assumed that the installing of Linux already includes: Apache server (for example, Apache 1.3.23), perl and mod_perl (for example mod_perl 1.26).

Additional required software: an uploading script (put.cgi), supplied with the software package.

➢ To configure Apache HTTP server, take these 4 steps:

1. Inside the Apache directory, create the directory /perl (for example /var/www/perl). Locate the file put.cgi on the supplied software package and copy it to that directory.
2. In the put.cgi script, change the first line from c:/perl/bin/perl to your perl executable file (this step is required only if mod_perl is not included in your Apache installation).
3. Enable access to the following directories and files by typing:
   - >chmod 777 perl
   - >chmod 755 put.cgi
   - >chmod 777 html (the name of the server’s shared files directory)
4. Configure the Apache sever:
   a. Open etc/httpd/conf/httpd.conf (or a similar file) for editing.
   b. Set the KeepAlive parameter to true.

c. Set the MaxKeepAliveRequests parameter to 0 (enables an unlimited number of requests during a persistent connection – required for multiple consecutive HTTP POST requests for uploading the file).

d. Set MaxClients to 250.

e. Change the mod_perl module lines to:

```
<IfModule mod perl.c>
    Alias /perl/ /var/www/perl/
    <Directory /var/www/perl>
        SetHandler perl-script
        PerlHandler Apache::Registry
        Options +ExecCGI
        PerlSendHeader On
    </Directory>
</IfModule>

Script PUT /perl/put.cgi
```

Reader's Notes

H Regulatory Information

Declaration of Conformity

Application of Council Directives:	73/23/EEC (including amendments) 89/336/EEC (including amendments) 1999/5/EC Annex-II of the Directive	
Standards to which Conformity is Declared:	EN55022: 1998 + A1: 2000 + A2: 2003 EN55024:1998 + A1: 2001 + A2: 2003 EN61000-3-2: 2000 + A2: 2005 EN61000-3-3: 1995 + A1: 2001 EN60950-1: 2001	
Manufacturer's Name:	AudioCodes Ltd.	
Manufacturer's Address:	1 Hayarden Street, Airport City, Lod 70151, Israel.	
Type of Equipment:	Analog VoIP System	
Model Numbers:	**MP-11x/FXS+FXO Mixed Series:**	MP-114/ 2FXS/2FXO; MP-118/ 4FXS/4FXO
	MP-11x/FXS Series:	MP-112/ 2FXS; MP-114/ 4FXS; MP-118/ 8FXS
	MP-11x/FXO Series:	MP-112/ 2FXO; MP-114/ 4FXO; MP-118/ 8FXO
	MP-124/FXS Series:	MP-124D/FXS

I, the undersigned, hereby declare that the equipment specified above conforms to the above Directives and Standards.

	27th June, 2006	Airport City, Lod, Israel
Signature	*Date (Day/Month/Year)*	*Location*

I. Zusmanovich, Compliance Engineering Manager

Czech	[AudioCodes Ltd] tímto prohlašuje, že tento [MP-11x/FXS & FXO Series & MP-124] je ve shodě se základními požadavky a dalšími příslušnými ustanoveními směrnice 89/336/EEC, 73/23/EEC; 1999/5/ES.
Danish	Undertegnede [AudioCodes Ltd] erklærer herved, at følgende udstyr [MP-11x/FXS & FXO Series & MP-124] overholder de væsentlige krav og øvrige relevante krav i direktiv 89/336/EEC, 73/23/EEC; 1999/5/ES.
Dutch	Hierbij verklaart [AudioCodes Ltd] dat het toestel [MP-11x/FXS & FXO Series & MP-124] in overeenstemming is met de essentiële eisen en de andere relevante bepalingen van richtlijn 89/336/EEC, 73/23/EEC; 1999/5/ES
English	Hereby, [AudioCodes Ltd], declares that this [MP-11x/FXS & FXO Series & MP-124] is in compliance with the essential requirements and other relevant provisions of Directive 89/336/EEC, 73/23/EEC; 1999/5/ES.
Estonian	Käesolevaga kinnitab [AudioCodes Ltd] seadme [MP-11x/FXS & FXO Series & MP-124] vastavust direktiivi 89/336/EEC, 73/23/EEC; 1999/5/ES põhinõuetele ja nimetatud direktiivist tulenevatele teistele asjakohastele sätetele.
Finnish	[AudioCodes Ltd] vakuuttaa täten että [MP-11x/FXS & FXO Series & MP-124] tyyppinen laite on direktiivin 89/336/EEC, 73/23/EEC; 1999/5/ES oleellisten vaatimusten ja sitä koskevien direktiivin muiden ehtojen mukainen.
French	Par la présente [AudioCodes Ltd] déclare que l'appareil [MP-11x/FXS & FXO Series & MP-124] est conforme aux exigences essentielles et aux autres dispositions pertinentes de la directive 89/336/EEC, 73/23/EEC; 1999/5/ES
German	Hiermit erklärt [AudioCodes Ltd], dass sich dieser/diese/dieses [MP-11x/FXS & FXO Series & MP-124] in Übereinstimmung mit den grundlegenden Anforderungen und den anderen relevanten Vorschriften der Richtlinie 89/336/EEC, 73/23/EEC; 1999/5/ES befindet".
Greek	ΜΕ ΤΗΝ ΠΑΡΟΥΣΑ [AudioCodes Ltd] ΔΗΛΩΝΕΙ ΟΤΙ [MP-11x/FXS & FXO Series & MP-124] ΣΥΜΜΟΡΦΩΝΕΤΑΙ ΠΡΟΣ ΤΙΣ ΟΥΣΙΩΔΕΙΣ ΑΠΑΙΤΗΣΕΙΣ ΚΑΙ ΤΙΣ ΛΟΙΠΕΣ ΣΧΕΤΙΚΕΣ ΔΙΑΤΑΞΕΙΣ ΤΗΣ ΟΔΗΓΙΑΣ 89/336/EEC, 73/23/EEC; 1999/5/ES

Hungarian	Alulírott, [AudioCodes Ltd] nyilatkozom, hogy a [MP-11x/FXS & FXO Series & MP-124] megfelel a vonatkozó alapvetõ követelményeknek és az 89/336/EEC, 73/23/EEC; 1999/5/ES irányelv egyéb elõírásainak
Icelandic	æki þetta er í samræmi við tilskipun Evrópusambandsins 89/336/EEC, 73/23/EEC; 1999/5/ES
Italian	Con la presente [AudioCodes Ltd] dichiara che questo [MP-11x/FXS & FXO Series & MP-124] è conforme ai requisiti essenziali ed alle altre disposizioni pertinenti stabilite dalla direttiva 89/336/EEC, 73/23/EEC; 1999/5/ES.
Latvian	Ar šo [AudioCodes Ltd] deklarē, ka [MP-11x/FXS & FXO Series & MP-124] atbilst Direktīvas 89/336/EEC, 73/23/EEC; 1999/5/ES būtiskajām prasībām un citiem ar to saistītajiem noteikumiem.
Lithuanian	[AudioCodes Ltd] deklaruoja, kad irenginys [MP-11x/FXS & FXO Series & MP-124] tenkina 89/336/EEC, 73/23/EEC; 1999/5/ES Direktyvos esminius reikalavimus ir kitas sios direktyvos nuostatas
Maltese	Hawnhekk, [AudioCodes Ltd], jiddikjara li dan [MP-11x/FXS & FXO Series & MP-124] jikkonforma mal-ħtiġijiet essenzjali u ma provvedimenti oħrajn relevanti li hemm fid-Dirrettiva 89/336/EEC, 73/23/EEC; 1999/5/ES
Norwegian	Dette produktet er i samhørighet med det Europeiske Direktiv 89/336/EEC, 73/23/EEC; 1999/5/ES
Polish	[AudioCodes Ltd], deklarujemy z pelna odpowiedzialnoscia, ze wyrób [MP-11x/FXS & FXO Series & MP-124] spelnia podstawowe wymagania i odpowiada warunkom zawartym w dyrektywie 89/336/EEC, 73/23/EEC; 1999/5/ES
Portuguese	[AudioCodes Ltd] declara que este [MP-11x/FXS & FXO Series & MP-124] está conforme com os requisitos essenciais e outras disposições da Directiva 89/336/EEC, 73/23/EEC; 1999/5/ES.
Slovak	[AudioCodes Ltd] týmto vyhlasuje, že [MP-11x/FXS & FXO Series & MP-124] spĺňa základné požiadavky a všetky príslušné ustanovenia Smernice 89/336/EEC, 73/23/EEC; 1999/5/ES.
Slovene	Šiuo [AudioCodes Ltd] deklaruoja, kad šis [MP-11x/FXS & FXO Series & MP-124] atitinka esminius reikalavimus ir kitas 89/336/EEC, 73/23/EEC; 1999/5/ES Direktyvos nuostatas.
Spanish	Por medio de la presente [AudioCodes Ltd] declara que el [MP-11x/FXS & FXO Series & MP-124] cumple con los requisitos esenciales y cualesquiera otras disposiciones aplicables o exigibles de la Directiva 89/336/EEC, 73/23/EEC; 1999/5/ES
Swedish	Härmed intygar [AudioCodes Ltd] att denna [MP-11x/FXS & FXO Series & MP-124] står I överensstämmelse med de väsentliga egenskapskrav och övriga relevanta bestämmelser som framgår av direktiv 89/336/EEC, 73/23/EEC; 1999/5/ES.

Safety Notices

1. Installation and service of this gateway must only be performed by authorized, qualified service personnel.
2. To avoid risk of fire use 26 AWG or higher wiring to connect the FXS or FXO telecom ports.
3. The equipment must be connected by service personnel to a socket-outlet with a protective earthing connection.
4. The protective earth terminal on the back of the MP-124 must be permanently connected to protective earth.

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-TX)	SELV
FXS	TNV-3
FXO	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.

Industry Canada Notice

This equipment meets the applicable Industry Canada Terminal Equipment technical specifications. This is confirmed by the registration numbers. The abbreviation, IC, before the registration number signifies that registration was performed based on a declaration of conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

FXO Ports: The Ringer Equivalence Number (REN) for this terminal is 0.05 The REN assigned to each terminal equipment provides an indication of the maximum number of terminals allowed to be connected to a telephone interface. The termination on an interface may consist of any combination of devices subject only to the requirement that the sum of Ringer Equivalence Number of all devices do not exceed five.

MP-11x FXO Notice

The MP-11x FXO Output Tones and DTMF level should not exceed -9 dBm (AudioCodes setting #23) in order to comply with FCC 68, TIA/EIA/IS-968 and TBR-21.

The maximum allowed gain between any 2 ports connected to the PSTN should be set to 0 dB in order to comply with FCC 68, TIA/EIA/IS-968 Signal power limitation.

Network Compatibility of FXO Ports

The products support the Telecom networks in EU that comply with TBR21.

FCC Statement

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.

ACTA Customer information

1. This equipment, the VoIP Analog Gateway, models MP-118, MP-114 and MP-112 complies with Part 68 of the FCC Rules and the requirements adopted by the ACTA. On the bottom of the unit of this equipment is a label, that contains among other information, a product identifier in the format US:AC1IT00BMP11X3AC. If requested, this number must be provided to the telephone company.
2. This equipment is designed to be connected to the telephone network using an RJ-11C connector, which is Part 68 compliant. The service order code (SOC) is 9.0Y and the Facility interface code (FIC) is 02LS2.
3. The REN is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. The REN for this product is part of the product identifier that has the format US:AC1IT00BMP11X3AC The digits represented by 00 are the REN without a decimal point.
4. Should the product causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. If advance notice is not practical, you will be notified as soon as possible. Also, you will be advised of your right to file a compliant with the FCC if it is necessary.
5. The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.
6. If trouble is experienced with this equipment, for repair or warranty information please contact AudioCodes Inc., 2099 Gateway Place, Suite 500, San Jose, CA, 95110, phone number 1-408-441-1175. If the equipment is causing harm to the telephone network, the telephone company may request to disconnect the equipment until the problem is resolved.
7. Installation is described in the Product User's manual. Connection to Telephone Company-provided coin service is prohibited. Connection to party lines service is subject to State tariffs.

AudioCodes™ CPE & Access Analog Gateways

SIP

MediaPack™ MP-124 & MP-11x

User's Manual Version 5.0

AudioCodes

www.audiocodes.com