Field Installation Guide for AudioCodes Media GW

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Table of Contents

1. Preliminary

The primary purpose of this document is a field guide to be used by Genesys personnel as an aid in integrating the AudioCodes Media Gateway with the Genesys Sip Server. It is intended to provide a high level view of sample configurations as well as show as many critical configuration details as practical without duplicating the existing installation, operation, and configuration documentation. This document may also be useful to customers or potential customers who are interested in standard installations involving these components.

This field guide provides you with supplementary information on AudioCodes SIP-based, Voice-over-IP (VoIP) devices specific to Genesys Sip Server deployments. Refer to the User Manual and other references at the end of this section for a full description of the capabilities of these products.

1.1 Applicable AudioCodes Systems

Media Gateway 3000 Series:

The AudioCodes Mediant 3000 system is a SIP-based Voice-over-IP (VoIP) media gateway. The Mediant 3000 offers integrated voice gateway functionality for voice, data, and fax streaming over IP networks. The dual board Mediant 3000 provides full system redundancy, offering an ideal solution for deploying high-density, high-availability Voice-over-Packet systems.

- Mediant 3000 gateway hosting a single or dual (High Availability) TP-8410 blade delivers up to 2,016 simultaneous voice channels supporting either 84 T1 or 63 E1 PSTN interfaces.
- Mediant 3000 gateway hosting a single or dual (High Availability) TP-6310 blade delivers up to 2,016 simultaneous voice channels supporting either STM-1/OC-3 or T3 PSTN interfaces.

Figure 1 – AudioCodes Mediant 3000 (6310 HA shown)

Mediant 2000 gateway (with TP-1610 blade):
The AudioCodes Mediant 2000 is a SIP-based Voice-over-IP (VoIP) media gateway which delivers similar functionality in a more compact 19" 1U device that supports up to 480 simultaneous voice channels supporting 16 DS1 spans (either T1 or E1).

Figure 2 - AudioCodes Mediant 2000

Mediant 1000 media gateway:
The AudioCodes Mediant 1000 is best suited for small-to-medium sized enterprises, branch offices, and residential media gateway solutions. The device is a highly scalable and modular system that matches the density requirements for smaller environments, while meeting service providers' demands for growth. The modules support a mix of digital and analog interface configurations.

The digital module supports multiples of 1, 2, or 4 E1/T1/J1 spans for connecting the PSTN/PBX to the IP network. The digital module also supports ISDN Basic Rate Interface (BRI). Each Analog module supports 4 analog RJ-11 ports with a maximum of six modules in various Foreign Exchange Office (FXO) or Foreign Exchange Station (FXS) configurations, supporting up to 24 simultaneous VoIP calls. The AudioCodes Mediant 1000 is ideal as an IP PBX for small offices not equipped with a PBX.

Figure 3 - AudioCodes Mediant 1000

AudioCodes EMS
The AudioCodes Element Management System (EMS) is an advanced solution for standards-based management of Media Gateways within VoP networks, covering all areas vital for the efficient operation, administration, management and provisioning (OAM&P) of AudioCodes' families of Media Gateways. The EMS features

Client/Server architecture, enabling customers to access the EMS from multiple, remotely located work centers and workstations.

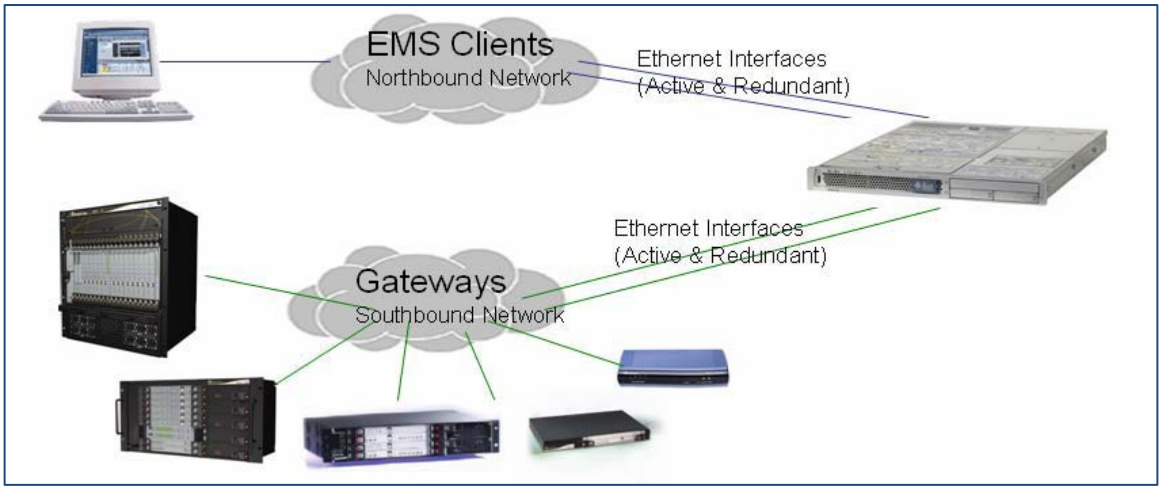

Figure 4 - AudioCodes EMS

1.2 Management by Web

The AudioCodes EMS is commonly used in solutions which involve multiple gateways or in high availability or high security applications. Each gateway is also equipped with a configuration and management interface accessible via a standard web browser via HTTP or HTTPS. The web design of the various digital gateway products in general share a common web design and the references within this field guide are applicable to any of the deployed products. The web interface provides a simple search capability which makes it easy to find the various parameters when some or all of the name is known (for example entering "Alt Routing").

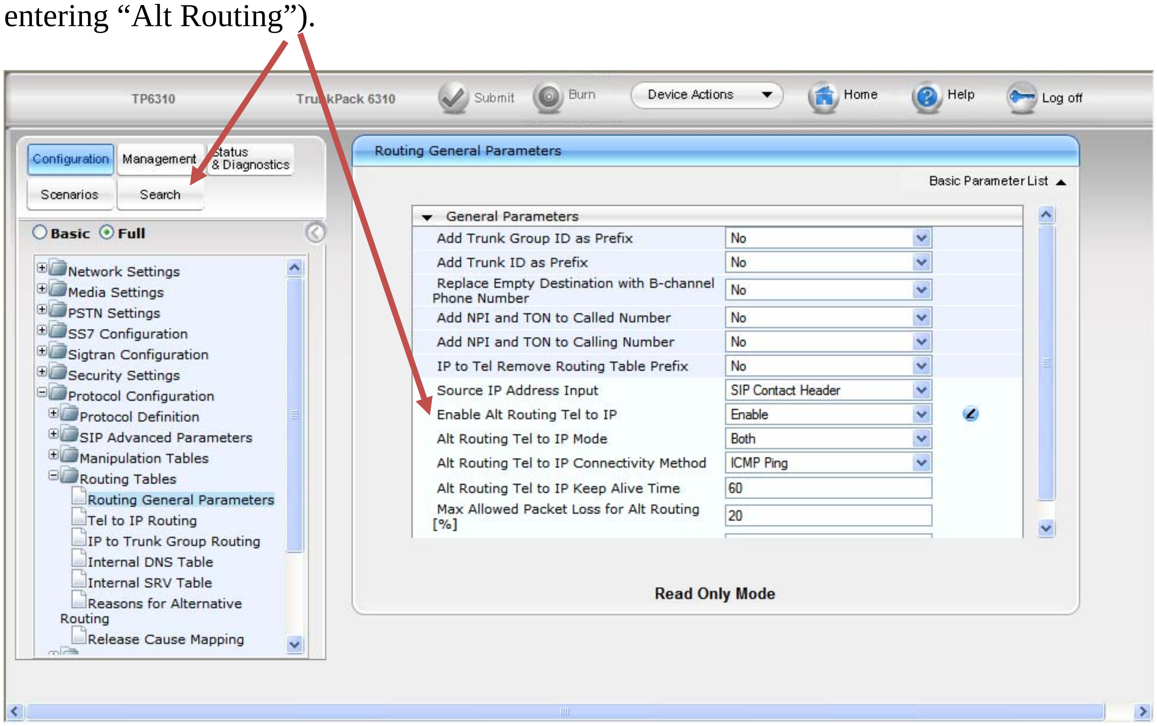

Figure 5 - Sample Web Configuration screen

1.3 Management by EMS

The sample configuration screens within this manual for management by EMS (unless otherwise noted) are for the Mediant 3000 product. The EMS supports a Help menu option which provides a search similar to the Web GUI. There is a convention used in this document to indicate navigation for menus, sub-menus, and tab headings (both top of frame and side of frame). The example shown below would be denoted in bold as follows:

SIP->Routing->Route Modes

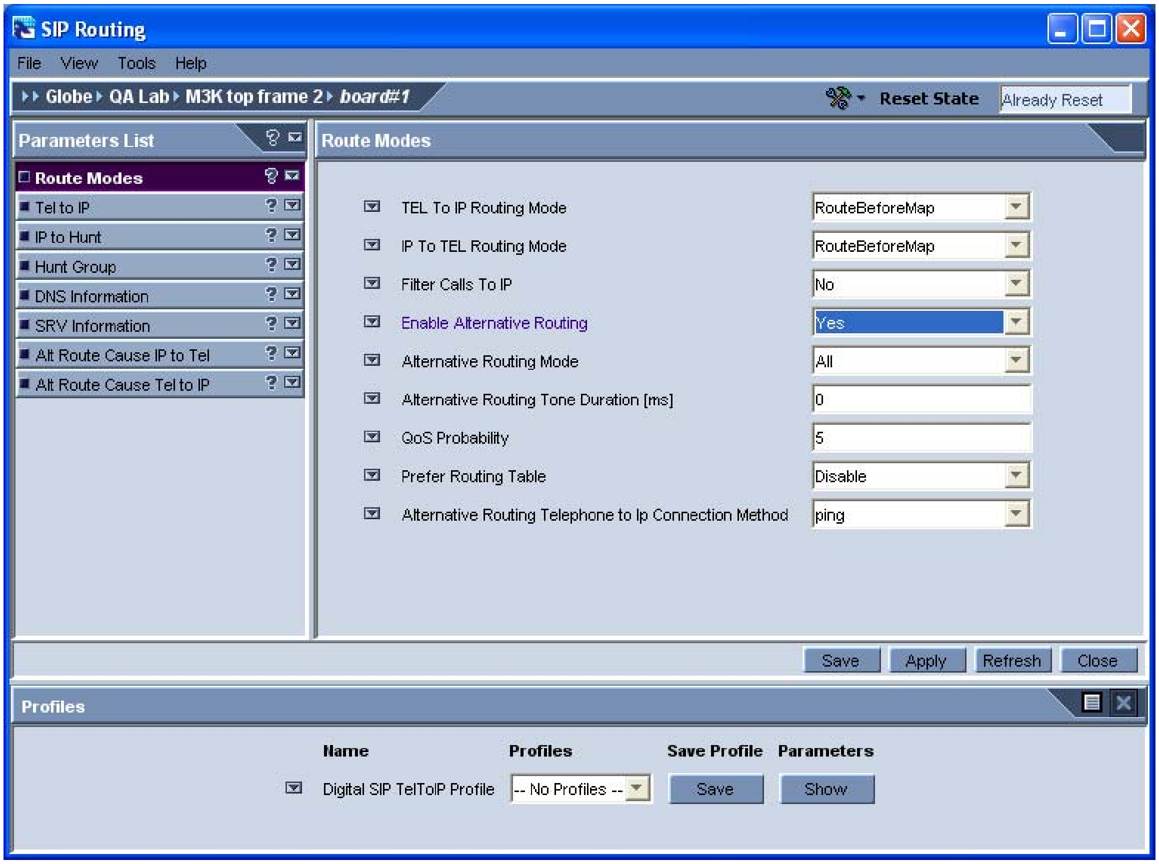

Figure 6 - Sample EMS Screen

1.4 AudioCodes Configuration File Basics

In addition to the Web and EMS view, each configuration parameter is part of an ASCII file known as the initialization file or INI file (a.k.a. the conf file). Most parameters follow in a simple NAME = VALUE, line by line sequence as shown below:

```
ALTROUTINGTEL2IPENABLE = 1
```

In general, the value 0 indicates FALSE or NO and 1 indicates TRUE or YES.

For more complex configuration parameters, the INI file supports tables. Table entries in the document are indicated by using an underscore notation as shown below:

```
SNMPManagerTableIP_2 = 0.0.0.0
SNMPManagerIsUsed_2 = 0
```

Because the INI file numbering is zero-based, this example shows that the third SNMP manager in the table has not been enabled.

The INI file can be easily retrieved from the media gateway by either the WEB or EMS.

1.5 Document Conventions

In most cases, a screen capture of the referenced parameter will be provided for both the web GUI and the EMS view. The user should reference the screen which matches the management mode for the deployment (either Web or EMS). For clarity, each referenced parameter will first be discussed by referencing the INI file name (for example, ALTROUTINGTEL2IPENABLE). The reference will be followed by a short definition which will include a description of valid ranges and default values.

1.6 AudioCodes Software Releases

Each AudioCodes software release will have four numerical indicators, for example 5.2.34.4. The first two numbers indicate the major release. This indicator progresses as follows: 5.0, 5.2, 5.4, 5.6, 5.8, 6.0, etc. The second two indicators represent the minor release. In general, the minor release indicators are not usually significant to end users.

The software version can be easily determined from either the web or EMS views. The software version can be referred to with leading or trailing zeroes which can be ignored. For example, the web GUI may display a version 05.20.034.004 for the 5.2 version referenced above.

This manual in general is applicable to the 5.6 and 5.8 versions. The screens are captured from the 5.8 version and may vary slightly from version to version.

1.7 References

The last 3 digits of the document reference number (beginning with LTRT) vary by release. These digits are replaced by xxx in the table below.

LTRT-94xxx Mediant 3000 HA EMS-SIP Configuration Guide Ver 5.x.pdf
LTRT-89xxx Mediant 3000, TP-8410 and TP-6310 SIP User's Manual Ver 5.x.pdf
LTRT-32xxx Mediant 3000 SIP-MGCP-MEGACO EMS Parameter Perform Alarms Guide Ver 5.x.pdf
LTRT-69xxx Mediant 3000, Mediant 2000, TP Series SIP Release Notes Ver 5.x.pdf
LTRT-68xxx Mediant 2000 SIP User's Manual Ver 5.x.pdf

LTRT-70xxx Mediant 2000 SIP-MGCP-MEGACO and IPmedia 2000 SIP Installation Manual Ver 5.x.pdf
LTRT-831xx Mediant 1000 SIP-MEGACO Installation Manual Ver 5.x.pdf
LTRT-833xx Mediant 1000 and Mediant 600 SIP User's Manual Ver 5.x.pdf
LTRT-83xxx Mediant 1000 and Mediant 600 SIP Release Notes Ver 5.x.pdf
LTRT-40xxx Mediant Series IP-to-IP SIP Call Routing Application Note Ver 5.x.pdf
LTRT-52xxx Product Reference Manual for SIP Gateways and Media Servers Ver 5.x.pdf
LTRT-94xxx EMS Server Installation, Operation, and Maintenance (IOM) Manual v5.x.pdf
LTRT-91xxx EMS User's Manual v5.x.pdf
LTRT-90xxx EMS Release Notes v5.x.pdf

A complete set of applicable AudioCodes documentation is shipped by CD/DVD with the product. Updated versions can be obtained by registering at www.audiocodes.com in the Support section.

2. Overview

In Genesys Sip Server deployments, the AudioCodes (AC) Media Gateway (MGW) is well suited to one of two IP-to-PSTN applications. The first application is as an interface between the SIP server and the conventional phones. Typically, an existing PBX system provides T1, E1 or DS3 spans to the PSTN side of the MGW. In some cases, a multiplexer (MUX) can be used to make the PSTN connection. The Genesys Sip Server typically manages the call control and directs the AudioCodes MGW to originate and terminate IP calls over SIP trunks using the existing (non-IP) PBX equipment. Typically, this arrangement yields considerable savings over the conventional system.

The second basic application involves conventional trunks which come to the location from a large service provider, for example Verizon or Quest. In this case, the AudioCodes MGW is used to route external calls into a state of the art SIP application environment supported by the Genesys SIP server. This is ideal for either an inbound call center environment or an outbound telemarketing center.

The AudioCodes MGW also supports IP-to-IP applications in conjunction with the AudioCodes Multi-Service Business Gateway (MSBG) which acts as a Session Border Controller (SBC).

2.1 IP-to-PBX trunk GW applications

The following section shows further interfacing SIP trunks to existing PBX equipment using the AudioCodes MGW and the Genesys Sip Server by referencing some of the typical end customer equipment seen in standard installations.

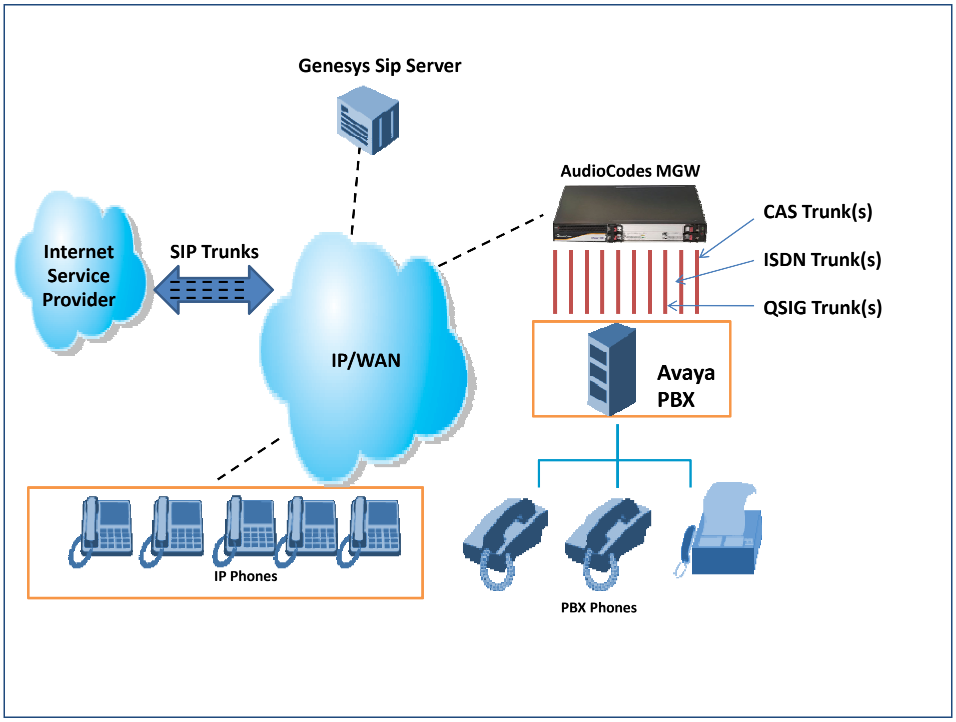

Figure 7 - Typical Single Office PBX Interface Application (Avaya PBX shown)

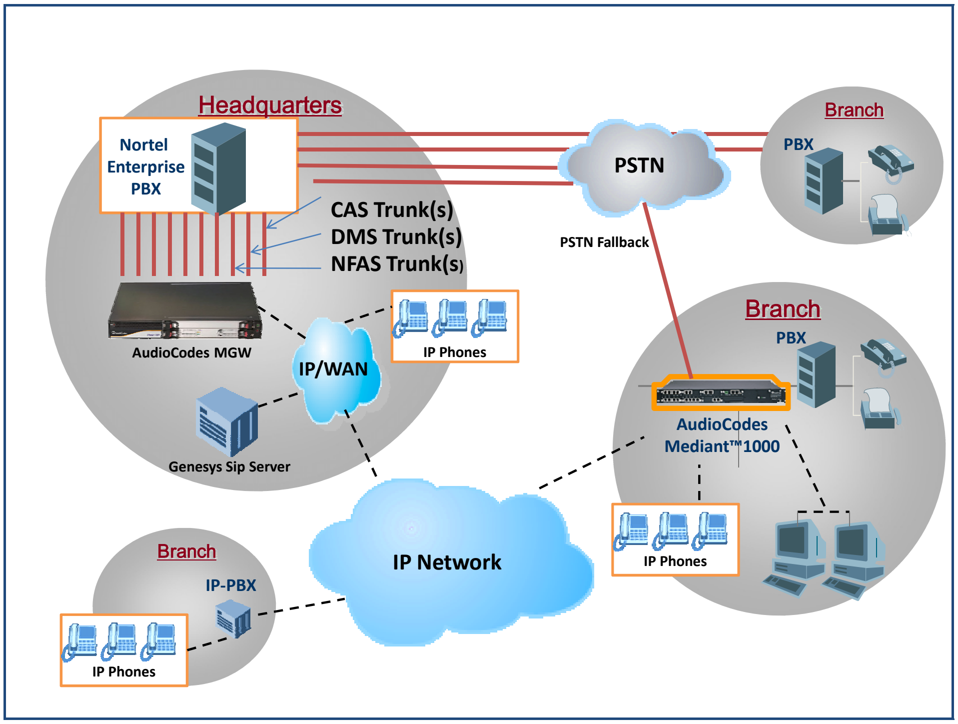

Figure 8 - Typical Enterprise PBX Application (Nortel PBX shown)

2.2 PSTN-to-IP Call Center Applications

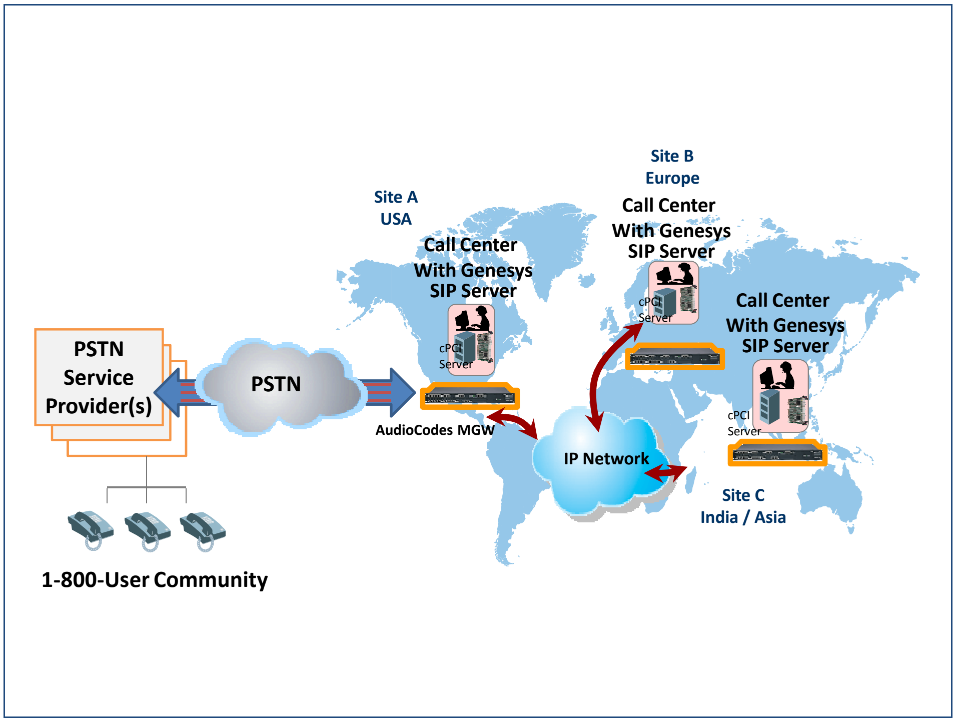

Figure 9 - PSTN to IP Call Center Application

2.3 IP-to-PSTN Outbound telemarketing applications

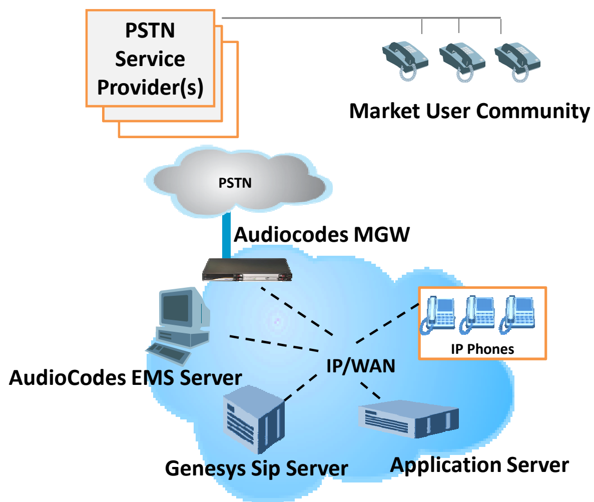

Figure 10 - IP to PSTN Telemarketing Application

3. AudioCodes Media Gateway Installation

Aside from the physical connections of the power, trunks, and Ethernet, the basic steps for AudioCodes MGW installation involve assigning an IP address to enable access by the web browser or with the EMS. There are several ways to assign the IP address. The IP can be assigned by serial access (CLI) or by creating an Ethernet connection using the gateway's default IP address. The most common method of assigning the IP address is by using the AudioCodes BOOTP application.

There are slight variations in the installation process. Reference the applicable installation guide or User Manual for the complete procedure.

LTRT-89xxx Mediant 3000, TP-8410 and TP-6310 SIP User's Manual Ver 5.x.pdf
LTRT-68xxx Mediant 2000 SIP User's Manual Ver 5.x.pdf
LTRT-70xxx Mediant 2000 SIP-MGCP-MEGACO and IPmedia 2000 SIP Installation Manual Ver 5.x.pdf
LTRT-831xx Mediant 1000 SIP-MEGACO Installation Manual Ver 5.x.pdf
LTRT-833xx Mediant 1000 and Mediant 600 SIP User's Manual Ver 5.x.pdf

3.1 AudioCodes BootP Application

BootP is normally used to initially configure the device. Thereafter, BootP is no longer required as all parameters can be stored in the gateway's non-volatile memory. BootP can then be enabled again later if the IP address of the device needs to be changed.

For convenience, AudioCodes provides a Windows PC based BootP server with each gateway in the Utilities section of the software CD/DVD. The BootP procedure can also be performed using any other standard compatible BootP server.

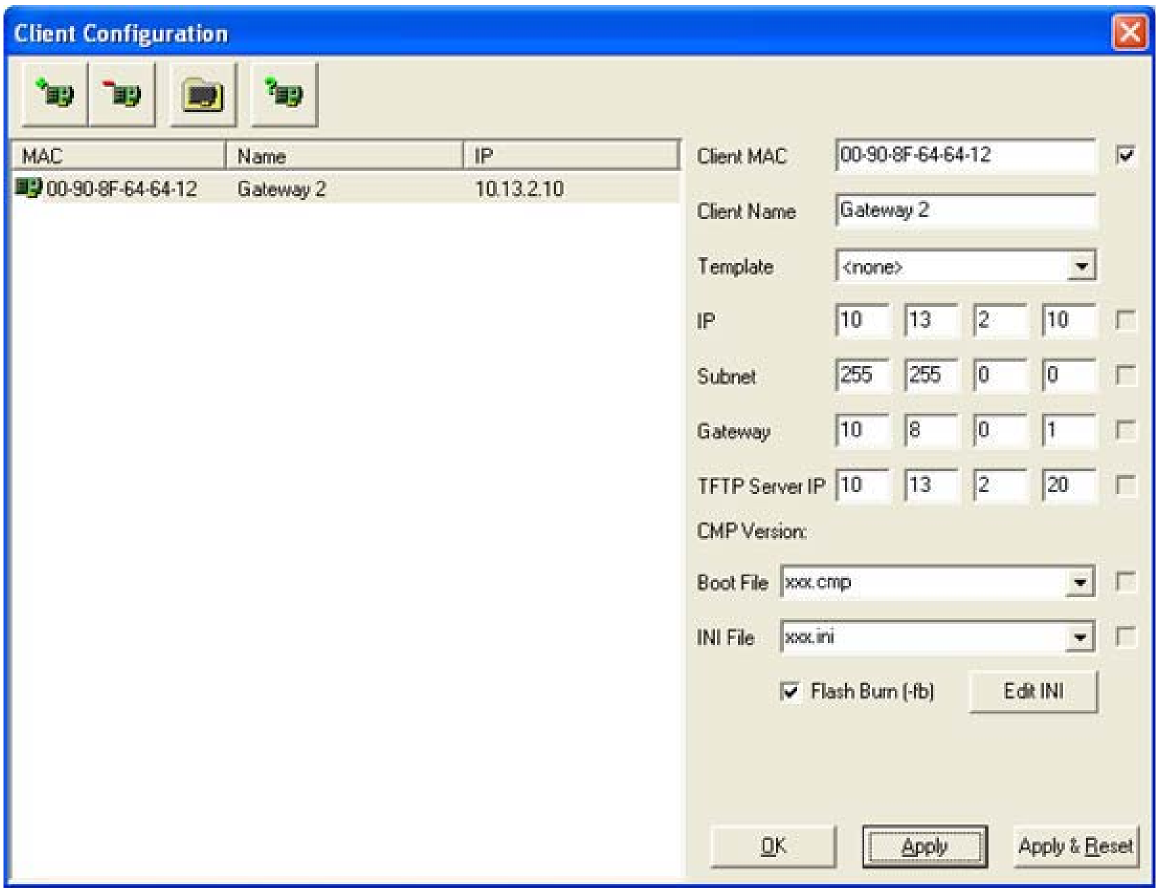

Figure 11 - Screen capture from the AudioCodes BootP server

The AudioCodes BootP application can also be used to change the software version and loaded configuration of the gateway by downloading the compressed binary software load file (CMP file) and initial configuration (INI) file. These fields are normally left blank for the initial install.

In an HA system, when the VoP blades are initially configured using BootP, the BootP assigns the two blades using two different local IP addresses, one for the active and one for the redundant blade. In a later step, a third global IP address within the same subnet is defined by the M3KGlobalIPAddr configuration parameter. Refer to the following:

LTRT-94xxx Mediant 3000 HA EMS-SIP Configuration Guide Ver 5.x.pdf

3.2 AudioCodes Syslog Application

AudioCodes also provides a Windows based syslog application in the Utilities section of the installation disk. This is described in further detail in the Troubleshooting section at the end of this document. It is mentioned in this section since it can be useful during the installation to confirm the basic IP configuration of the gateway.

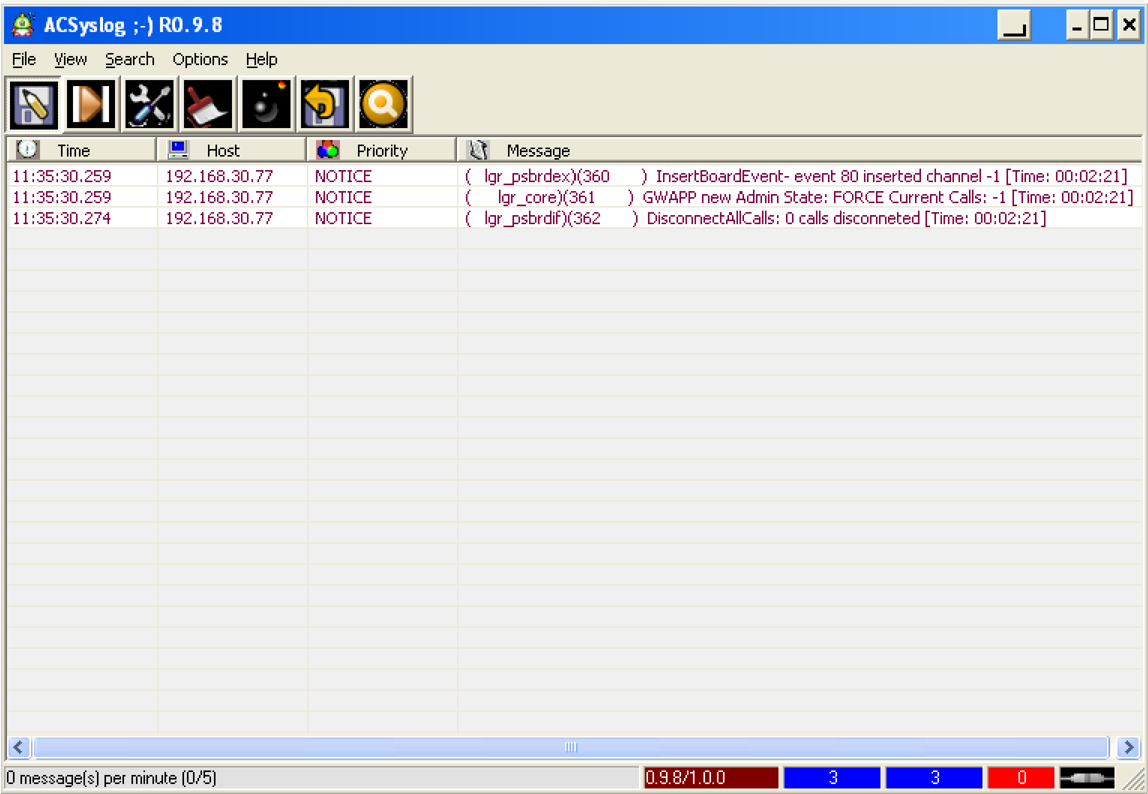

Figure 12 - AudioCodes Syslog Application

The following INI file configuration parameters are used:

```
SyslogServerIP = 192.168.10.57
```

~ SyslogServerIP: This parameter defines the IP address in dotted format notation. e.g. 192.10.1.255, Range = Legal IP address.

```
EnableSyslog = 1
```

~ EnableSyslog: This parameter is used to enable the Syslog protocol log.
1 = Activate, 0 = Deactivate.

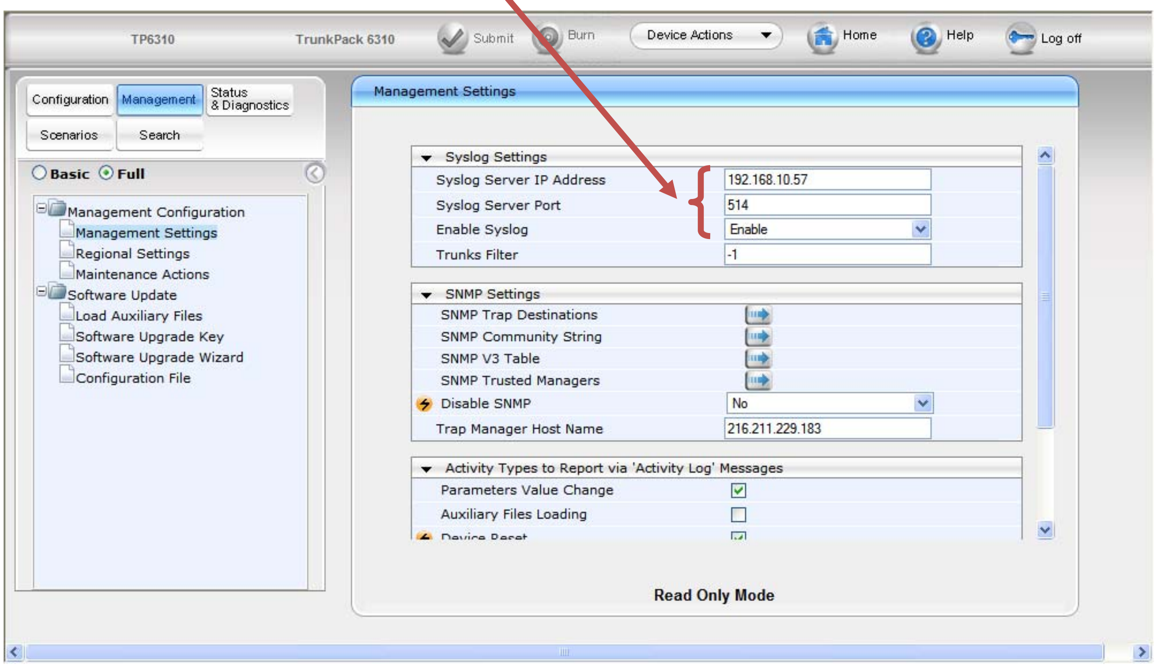

Figure 13 - Syslog Server IP Address Configuration by web

4. SIP Settings

4.1 SIP Trunking

The combination of an AudioCodes Media Gateways in conjunction with the Genesys Sip Server support Tel-IP/IP-Tel and IP-to-IP Voice over IP (VoIP) call routing (or SIP Trunking). This enables Enterprises to seamlessly connect their legacy PBX, key system, or IP-based PBX (IP-PBX) to SIP trunks, typically provided by an Internet Telephony Service Provider (ITSP). Enterprises can then communicate (using AudioCodes gateway) with PSTN networks (local and overseas) through ITSP's, which interface directly with the PSTN.

The gateways also support multiple SIP Trunking. This can be useful if a connection to one ITSP goes down, the call can immediately be transferred to another ITSP. In addition, by allowing multiple SIP trunks, where each trunk is designated a specific ITSP, the gateway can route calls to an ITSP based on call destination (e.g., country code).

At the same time, AudioCodes gateways can also provide an interface with the traditional PSTN network, enabling PSTN fallback in case of IP connection failure with the ITSP's.

An overview of the configuration settings which support these features are described in the following sections. The following settings are also applicable to Call Center PSTN interface (not just SIP Trunking).

4.2 General Settings

The values shown are examples since a variety of settings are appropriate for different applications. Other than some that are highlighted below, most of the default values are usually used in the SIP General Parameters.

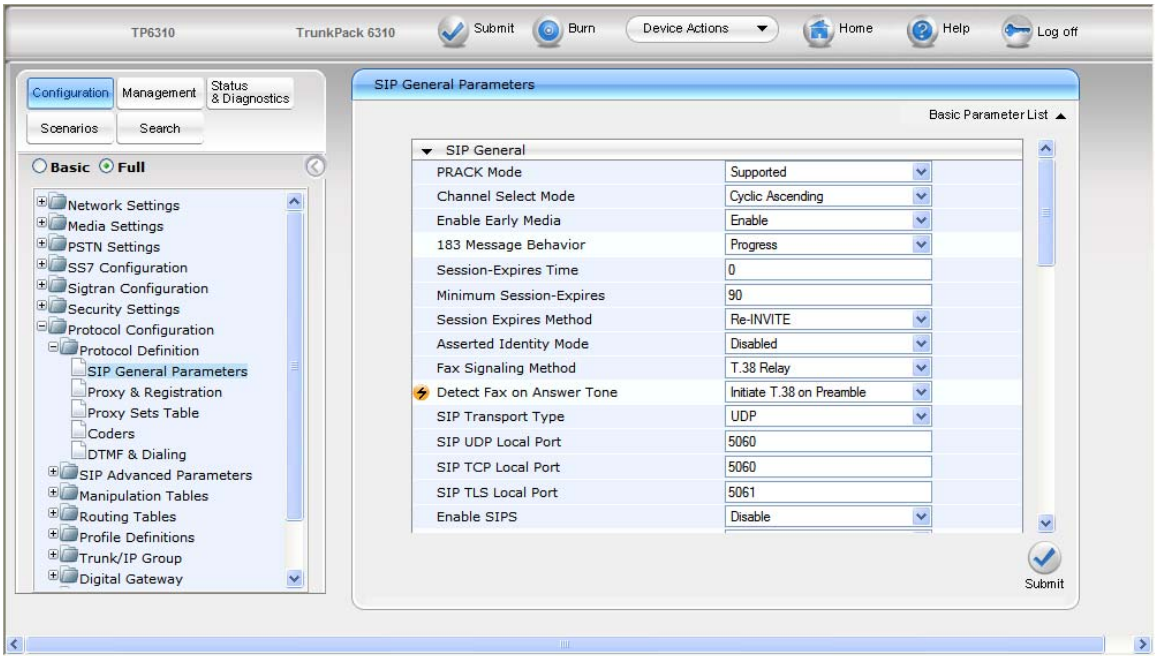

Figure 14 – Web view of SIP General Parameters

```
ChannelSelectMode = 1
```

~Channel Select Mode - Channel allocation algorithm for IP-to-Tel calls.

[0] By Dest Phone Number = Selects the device's channel according to the called number. (default.)

[1] Cyclic Ascending = Selects the next available channel in an ascending cyclic order. When the device reaches the highest channel number in the trunk group, it selects the lowest channel number in the trunk group and then starts ascending again.

[2] Ascending = Selects the lowest available channel.

[3] Cyclic Descending = Selects the next available channel in descending cyclic order. When the device reaches the lowest channel number in the trunk group, it selects the highest channel number in the trunk group and then starts descending again.

[4] Descending = Selects the highest available channel.

[5] Dest Number + Cyclic Ascending

[6] By Source Phone Number

[7] Trunk Cyclic Ascending = Selects the device's port from the first channel of the next trunk (next to the trunk from which the previous channel was allocated.

```
EnableEarlyMedia = 1
```

~ Enable Early Media - Enables the device to send a 183 Session Progress response with SDP (instead of 180 Ringing), allowing the media stream to be established prior to the answering of the call.

[0] Disable = Early Media is disabled (default).

[1] Enable = Enables Early Media.

```
IsFaxUsed = 1
```

~ Fax Signaling Method - Determines the SIP signaling method for establishing and transmitting a fax session after a fax is detected.

[0] No Fax = No fax negotiation using SIP signaling. Fax transport method is according to the parameter FaxTransportMode (default).
[1] T.38 Relay = Initiates T.38 fax relay.
[2] G.711 Transport = Initiates fax / modem using the coder G.711 Alaw/ μ-law with adaptations (refer to Note below).
[3] Fax Fallback = Initiates T.38 fax relay. If the T.38 negotiation fails, the device re-initiates a fax session using the coder G.711 A-law/μ-law with adaptations (refer to the Note in the User Manual).

```
SIPTransportType = 0
```

~ SIP Transport Type - Determines the default transport layer for outgoing SIP calls initiated by the device.

[0] UDP (default)
[1] TCP
[2] TLS (SIPS)

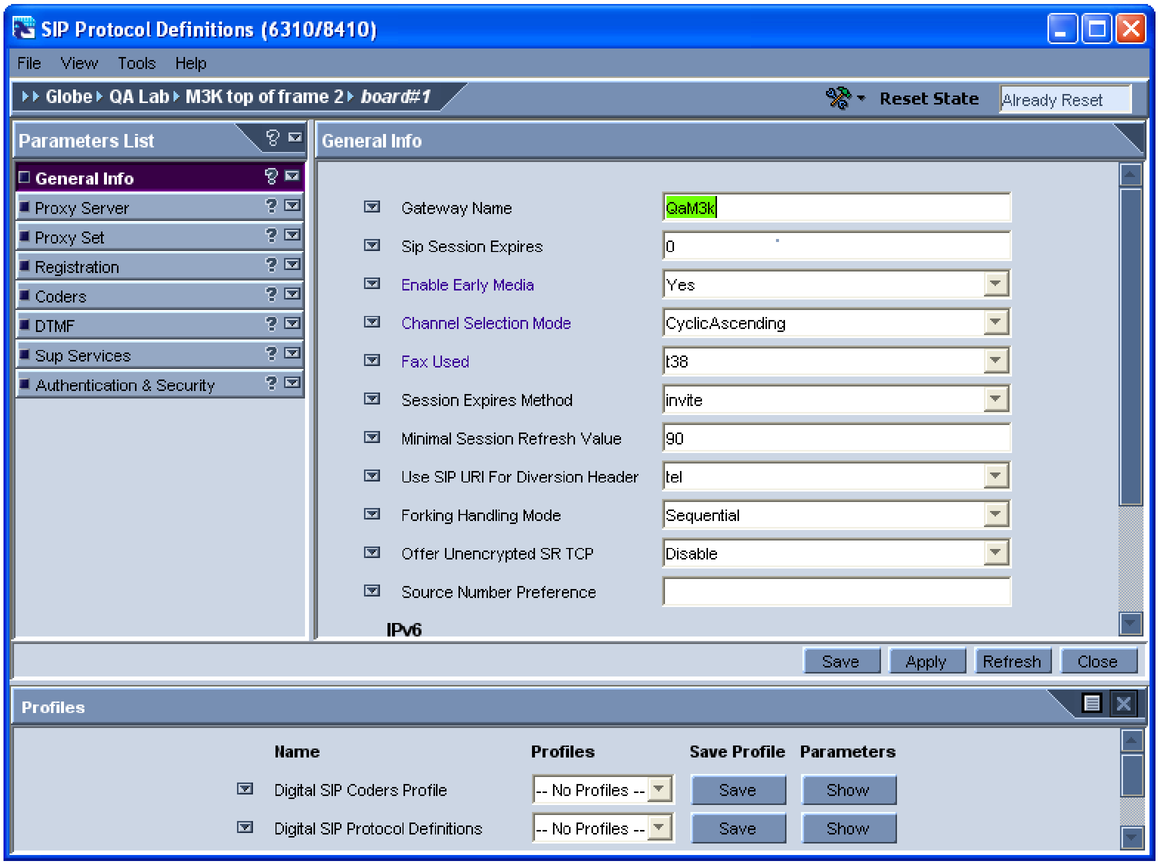

Figure 15 - EMS view of SIP General Settings

The SIP General Settings is found in the **SIP->Protocol Definition->General Info** tab in the EMS.

4.3 Coders

The 'Coders' page allows you to configure up to five coders (and their attributes) for the device. The first coder in the list is the highest priority coder and is used by the device whenever possible. If the far-end device cannot use the first coder, the device attempts to use the next coder in the list, and so forth. The coder table is simple, but an empty table in a new installation will prevent any successful calls.

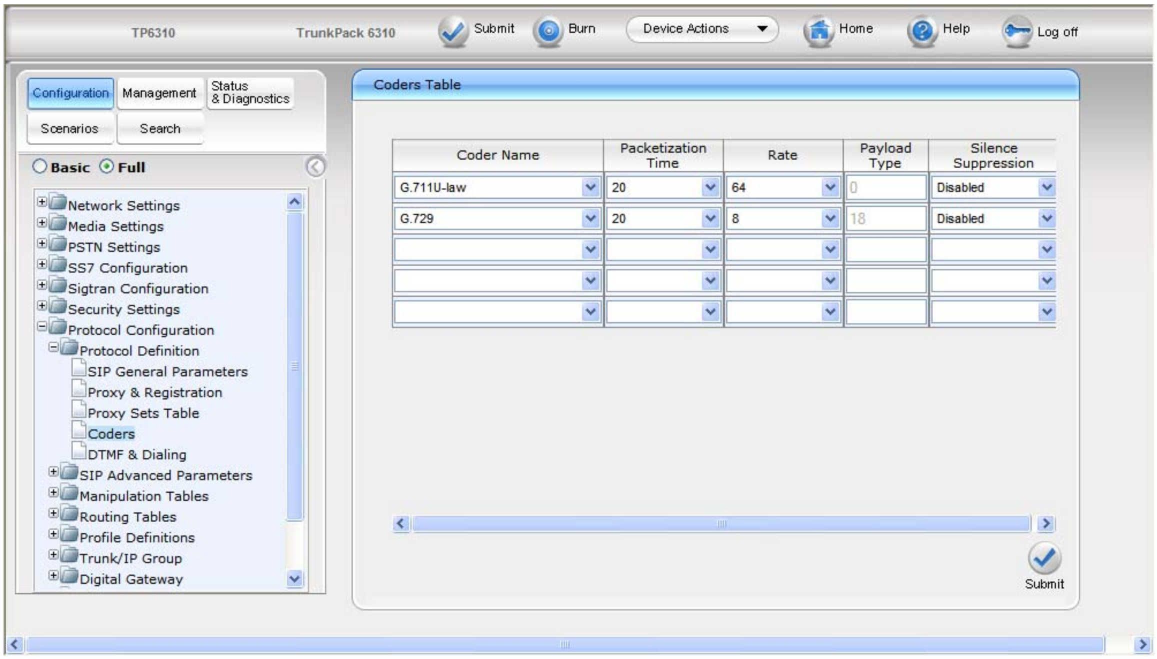

Figure 16 - Web view of Coder Table

Refer to the User’s Manual for a list of all the coders supported.

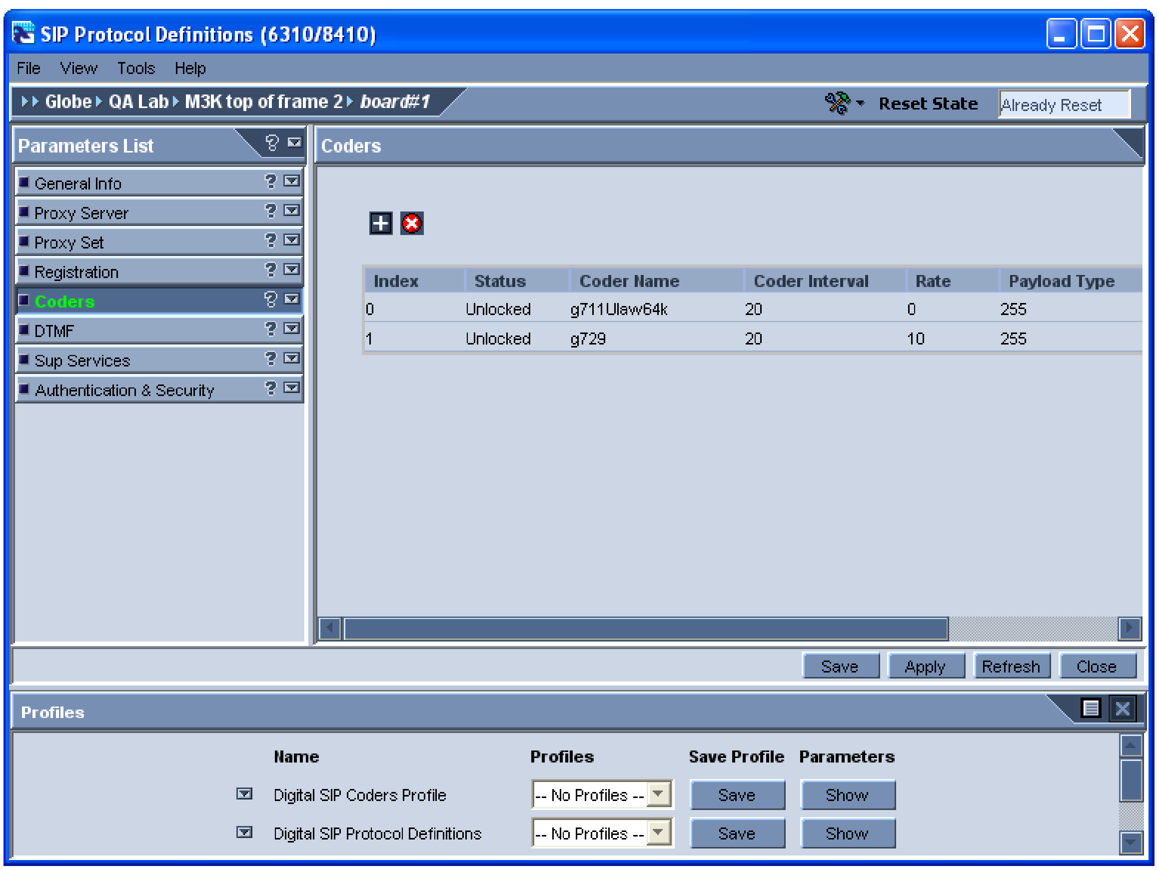

Figure 17 - EMS view of the coder table

The Coders Table is found in the **SIP->Protocol Definition->Coders** tab in the EMS. Use the [+] to add rows and the [x] to delete rows. Keep in mind that the order of the table is significant.

4.4 Proxy Settings

SIP REGISTER messages are sent to the to the Proxy's IP address (or to a separate Registrar IP address or FQDN if one is defined). A single message is sent once per device, or messages are sent per B-channel according to the parameter AuthenticationMode. There is also an option to configure registration mode per Trunk Group using the TrunkGroupSettings table. The registration request is resent according to the parameter RegistrationTimeDivider. For example, if RegistrationTimeDivider = 70 (%) and Registration Expires time = 3600, the device resends its registration request after 3600 x 70% = 2520 sec. The default value of RegistrationTimeDivider is 50%.

Below is an example of Proxy and Registrar Registration:

```
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
```

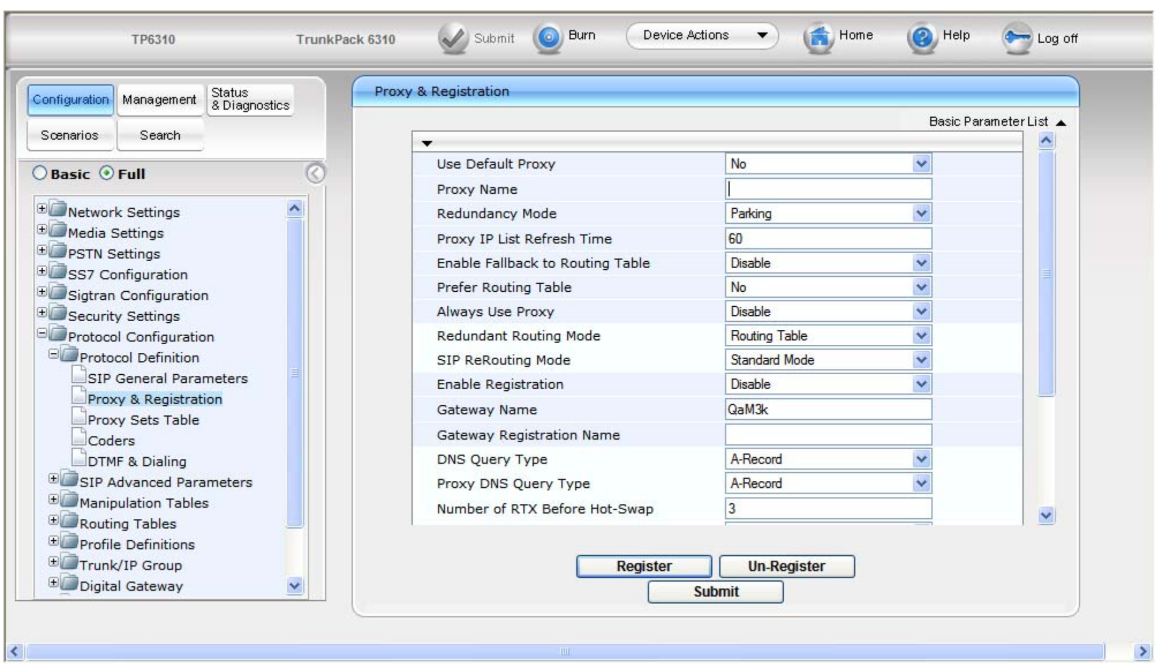

Figure 18 - Web view Proxy & Registration (defaults shown)

There is a wide variety of proxy and registration options supported. The 'servername' string is defined according to the following rules; The "servername" is equal to "RegistrarName" if configured. Otherwise, the "servername" is equal to "RegistrarIP" (either FQDN or numerical IP address), if configured. Otherwise, the "servername" is equal to "ProxyName" if configured. Otherwise, the "servername" is equal to "ProxyIP" (either FQDN or numerical IP address).

The parameter GWRegistrationName can be any string. This parameter is used only if registration is per device. If the parameter is not defined, the parameter UserName is used instead. If the registration is per endpoint, the endpoint phone number is used.

```
IsProxyUsed = 0
```

~Use Default Proxy - Enables the use of a SIP Proxy server.

[0] No = Proxy isn't used - the internal routing table is used instead (default).

[1] Yes = Proxy is used. Parameters relevant to Proxy configuration are displayed.

ProxyName = 'englab1.audiocodesusa.net'

~Proxy Name - Defines the Home Proxy Domain Name. If specified, the Proxy Name is used as the Request-URI in REGISTER, INVITE, and other SIP messages, and as the host part of the To header in INVITE messages. If not specified, the Proxy IP address is used instead.

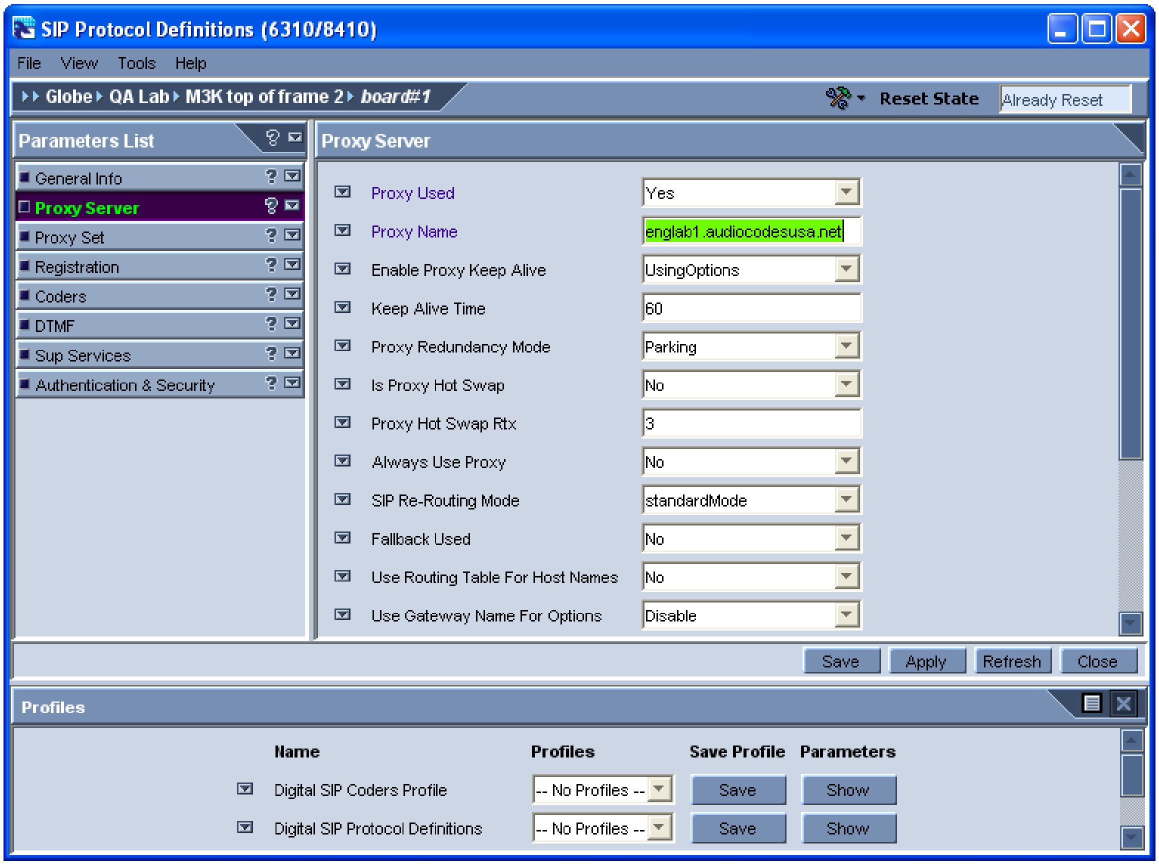

Figure 19 - EMS view Proxy Server (enabled)

The Proxey Server settings are found in the **SIP->Protocol Definition->Proxy Server** tab in the EMS (see also **Registration** tab).

Proxy Examples

Proxy IP Address = 192.168.10.51
Proxy Name is blank
Outgoing INVITE Request URI
INVITE sip:301@192.168.10.51;user=phone SIP/2.0

Proxy IP Address = 192.168.10.51
Proxy Name = corp.audiocodes.com
Outgoing INVITE Request URI:
INVITE sip:301@corp.audiocodes.com;user=phone SIP/2.0

Redundancy is enabled by setting the 'Enable Proxy Keep Alive' value to 'OPTIONS' or 'REGISTER'. A 'keep alive' SIP message is sent out periodically (dependent on whether OPTIONS or REGISTER is used). Any response back (200 OK, 4xx message) is considered validation that the proxy is still alive. When proxy does not respond to INVITE messages, the gateway treats it as a 'keep alive' failure.

Gateway operates in one of two modes:

- Parking
  - o Stay with redundant proxy until next failure (default)
- Homing
  - o Prefer to work with primary proxy (switch back when available)

4.5 Proxy Sets

The Proxy Sets Table page allows you to define Proxy Sets. A Proxy Set is a group of Proxy servers defined by IP address or fully qualified domain name (FQDN). You can define up to six Proxy Sets, each having a unique ID number and each containing up to five Proxy server addresses.

Generally speaking, the Proxy Set Table is used to implement multiple SIP trunks. The Proxy IP table is one way to implement a Sip Server redundancy, for example a geographically distributed redundancy. There is also a configuration for DNS Server IP and there is a DNS lookup table within the gateway which can be used to implement proxy redundancy using Fully Qualified Domain Names (FQDN).

The Proxy Sets are referred to later in the Call Routing section of this document. For example, once a Proxy Set is defined, all 800 or 1-800 calls can be routed to this proxy.

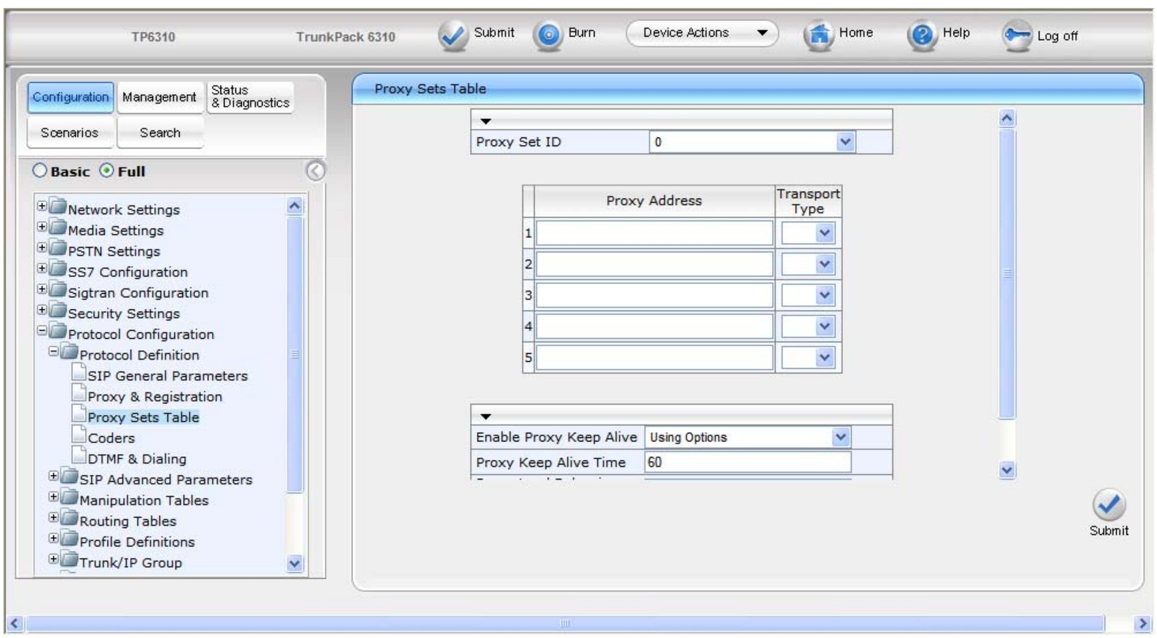

Figure 20 - Web view Proxy Set Table

```
[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress,
ProxyIp_TransportType, ProxyIp_ProxySetId;
ProxyIp 0 = 192.168.64.215, 0, 1;
ProxyIp 2 = 192.168.50.44, 0, 2;
ProxyIp 3 = 192.168.50.48, 0, 3;
```

```
[ \ProxyIp ]

[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap;
ProxySet 0 = 0, 60, 0, 0;
ProxySet 1 = 0, 60, 0, 0;
ProxySet 2 = 0, 60, 0, 0;
ProxySet 3 = 0, 60, 0, 0;
```

The Proxy Set Table is found in the **SIP->Protocol Definition->Proxy Sets** tab in the EMS.

4.6 Advanced SIP Settings

Answering Machine Detection

Answering Machine Detection can be useful in automatic dialing applications. In some of these applications, it is important to detect if a human voice or answering machine is answering the call. Answering Machine Detection can be activated and de-activated only after a channel is already open.

The direction of the detection (PSTN or IP) can be configured (using the parameter AMDDetectionDirection - refer to "Media Server Parameters" in the User Guide., as well as the detector detection sensitivity using the parameter AMDDetectionSensitivity.

Upon every Answering Machine Detection activation, the device can send a SIP INFO message to an Application server, notifying it of one of the following(Human voice has been detected, Answering machine has been detected, Silence has been detected) as shown in this example:

```
INFO sip:sipp@172.22.2.9:5060 SIP/2.0
Via: SIP/2.0/UDP 172.22.168.249;branch=z9hG4bKac1566945480
Max-Forwards: 70
From: sut <sip:3000@172.22.168.249:5060>;tag=1c1505895240
To: sipp <sip:sipp@172.22.2.9:5060>;tag=1
Call-ID: 1-29758@172.22.2.9
CSeq: 1 INFO
Contact: <sip:56700@172.22.168.249>
Supported: em,timer,replaces,path,resource-priority
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,I
NFO,SUB
SCRIBE,UPDATE
User-Agent: Audiocodes-Sip-Gateway-IPmedia
260_UN/v.5.20A.040.004
```

```
Content-Type: application/x-detect
Content-Length: 30
Type= AMD
SubType= AUTOMATA
```

If the device detects voice and not an answering machine, the SIP INFO message includes: Type= AMD SubType= VOICE. If the device detects silence, the SIP INFO message includes the SubType SILENT.

A pre-requisite for enabling the AMD feature is to set the INI file parameter EnableDSPIPMDetectors to 1. In addition, to enable voice detection, the INI file parameter EnableVoiceDetection must be set to 1.

```
EnableDSPIPMDetectors = 1
```

~ Enable DSP IPMedia Detectors - Enables or disables the device's DSP detectors.
[0] = Disable (default).
[1] = Enable.

```
EnableAnswerDetector = 1
```

~ EnableAnswerDetector - Enables or disables activation of the AD (Answer Detector).
[0] = Disable,
[1] = Enable.

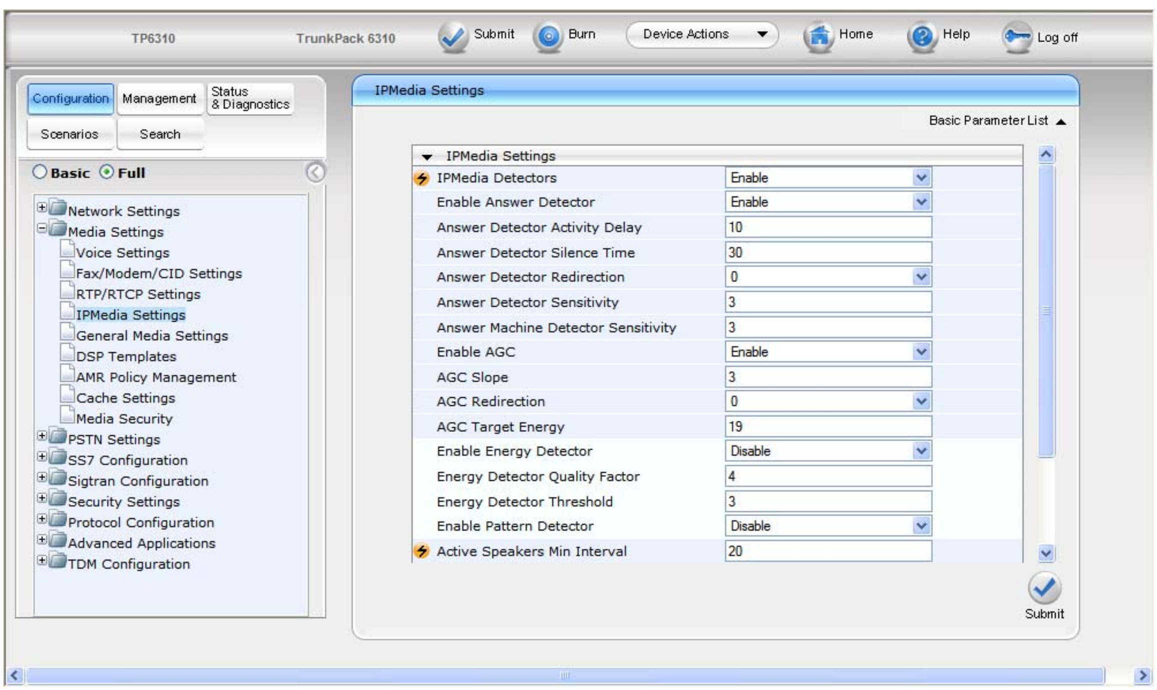

Figure 21 - Web View IP Media Settings

5. Trunks Settings

5.1 Clocking Considerations

As described earlier, the AudioCodes Media Gateway can either provide IP-to-Tel service to an existing PBX Tel-to-IP or provide connections from a PSTN central office provider for a call center or telemarketing center. Generally speaking, the PSTN clock signal is sent from the central office out to the branch circuits to maintain synchronization.

Following this convention, the MGW usually transmits clock (master clock) in PBX connected applications. This is because the PBX normally is not set up to transmit clock sync. Alternatively, the MGW should almost always recover clock (or slave) in PSTN service provider connected applications such as call centers and telemarketing centers.

5.2 External Clocking from PSTN Service Provider Interfaces (Web Example)

Prior to setting up the T1 trunks, the gateway is normally set up for DS3 connections (TP-6310 systems only). In some cases, particularly outside of North America, SONET/SDH fiber connections may be used (STM-1/OC3). The following web example shows a slave clock driven by the external network.

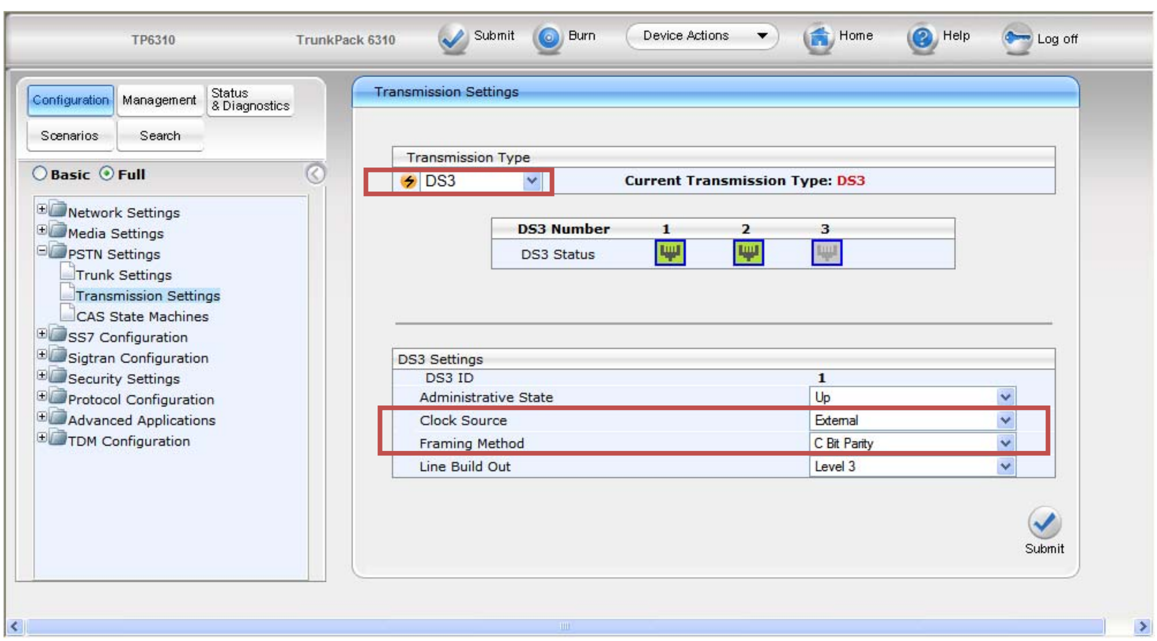

Figure 22 - Web view of the Transmission Settings (DS3)

```
PSTNTransmissionType = 2
```

~ Transmission Type - Defines the PSTN transmission type for the device.

[0] None = PSTN Transmission type is not defined (default).

[1] SONET/SDH = Optical SONET or SDH transmission type.

[2] DS3 = Copper T3 transmission type.

`DS3Config_ClockSource = 0`

~ Clock Source - Selects the T3 clock mode blade for the interface.
[0] External = DS3 clock is recovered from the line (default).
[1] Local Board = DS3 trunk clock source is provided by the blade's internal clock.

The MGW supports two common DS3 framing types. This critical setting should match your far end.

`DS3Config_FramingMethod = 1`

~ Framing Method - Determines the physical T3 framing method for the interface.
[0] M23 = M23 framing (default).
[1] C Bit Parity = C Bit Parity.

The Line Build Out determines the level of signal transmission. This level influences the cable length that can be used with the device. Consult your User's Guide for a complete description of the procedure for properly setting up the DS3 transmission settings.

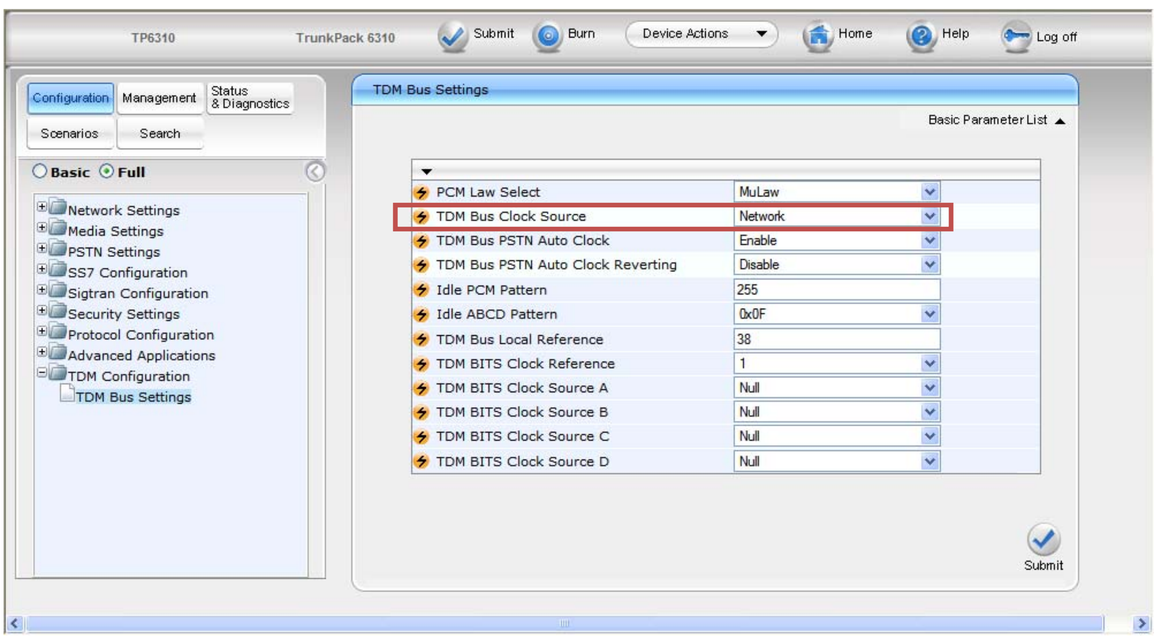

Figure 23 - Web view TDM Bus Settings

The following settings control the T1 PSTN clock synchronization (slave shown):

`TDMBusClockSource = 4`

TDM Bus Clock Source - Selects the clock source to which the device synchronizes.
[1] Internal = Generate clock from local source (default).
[4] Network = Recover clock from PSTN line.

```
ClockMaster = 1
```

Clock Master - Determines the Tx clock source of the E1/T1 line.
[0] Recovered = Generate the clock according to the Rx of the E1/T1 line (default).
[1] Generated

The TDM Bus Clock Source has a single board level setting. The Clock Master is set at the Trunk level but is normally set consistently throughout all trunks.

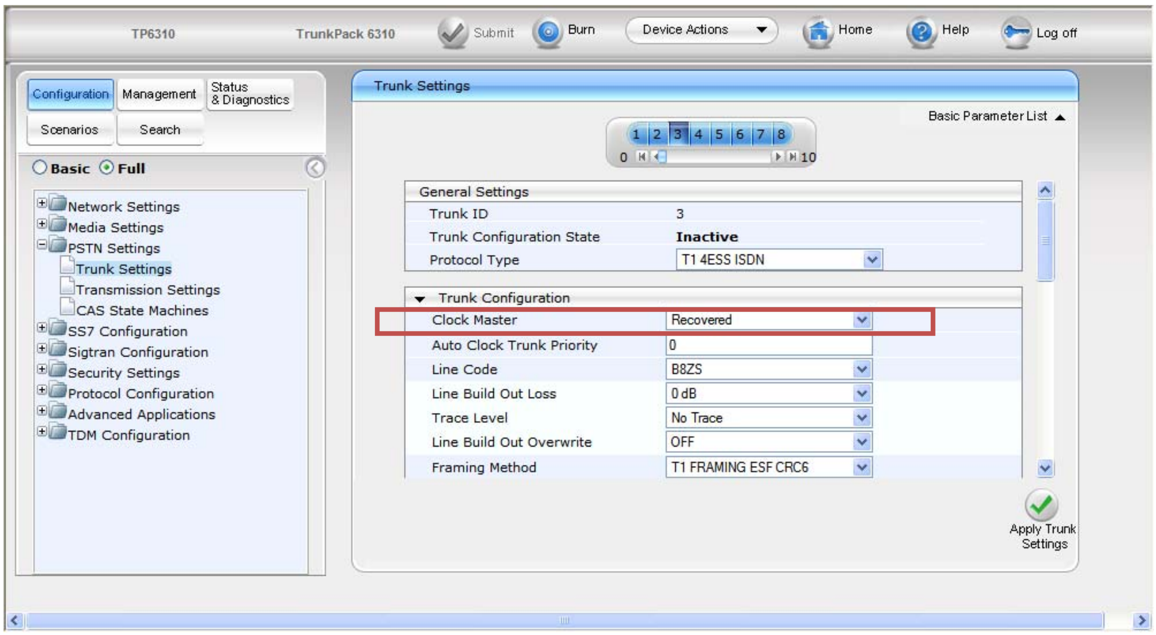

Figure 24 - Web View Trunk Settings

For detailed information on configuring the device's TDM clocking, refer to your User's Guide.

Some examples of the more commonly seen of T1 Protocol types are discussed in the next section.

```
ProtocolType_0 = 11
```

~ ProtocolType - Used to set the PSTN protocol to be used for this trunk.
Either: NONE = 0, E1_EURO_ISDN = 1, T1_CAS = 2, T1_RAW_CAS = 3, T1_TRANSPARENT = 4, E1_TRANSPARENT_31 = 5, E1_TRANSPARENT_30 = 6, E1_MFCR2 = 7, E1_CAS = 8, E1_RAW_CAS = 9, T1_NI2_ISDN = 10, T1_4ESS_ISDN = 11, T1_5ESS_9_ISDN = 12, T1_5ESS_10_ISDN = 13, T1_DMS100_ISDN = 14, J1_TRANSPARENT = 15, T1_NTT_ISDN = 16, E1_AUSTEL_ISDN = 17, E1_HKT_ISDN = 18, E1_KOR_ISDN = 19, T1_HKT_ISDN = 20, E1_QSIG = 21, E1_TNZ_ISDN = 22, T1_QSIG = 23, V5_2_AN = 26, T1_IUA = 28, E1_IUA = 29, E1_FRENCH_VN6_ISDN = 30, E1_FRENCH_VN3_ISDN = 31, T1_EURO_ISDN = 34, T1_DMS100_MERIDIAN_ISDN = 35, T1_NI1_ISDN = 36,

E1_DUA = 37, E1_Q931_PACKETS = 38, T1_Q931_PACKETS = 39, E1_NI2_ISDN = 40.

For T1 trunks, the most common settings are shown below:

```
LineCode = 0
```

~ Line Code - Use to select B8ZS or AMI for T1 spans, and HDB3 or AMI for E1 spans.
    [0] B8ZS = use B8ZS line code (for T1 trunks only) default.
    [1] AMI = use AMI line code.
    [2] HDB3 = use HDB3 line code (for E1 trunks only).

The framing method should also exactly match your far end (most common setting shown):

```
Framing Method = D
```

~ FramingMethod - Determines the physical framing method for the trunk.
    [C] = T1 Extended SuperFrame without CRC6 (not used)
    [D] = T1 Extended SuperFrame with CRC6 (T1 Framing ESF CRC6)

There are many other choices for Framing Method (especially for E1 trunks). Consult your User's Guide for a complete description of the procedure for properly setting up the trunk settings.

5.3 Generating Clock for PBX Interfaces (EMS Example)

The following EMS example shows master clock settings (generated). These settings are typical in PBX interfacing applications. These are the same settings discussed above in the external clock slave example (Web).

Prior to setting up the T1 trunks, the gateway is normally set up for DS3 connections (TP-6310 systems only).

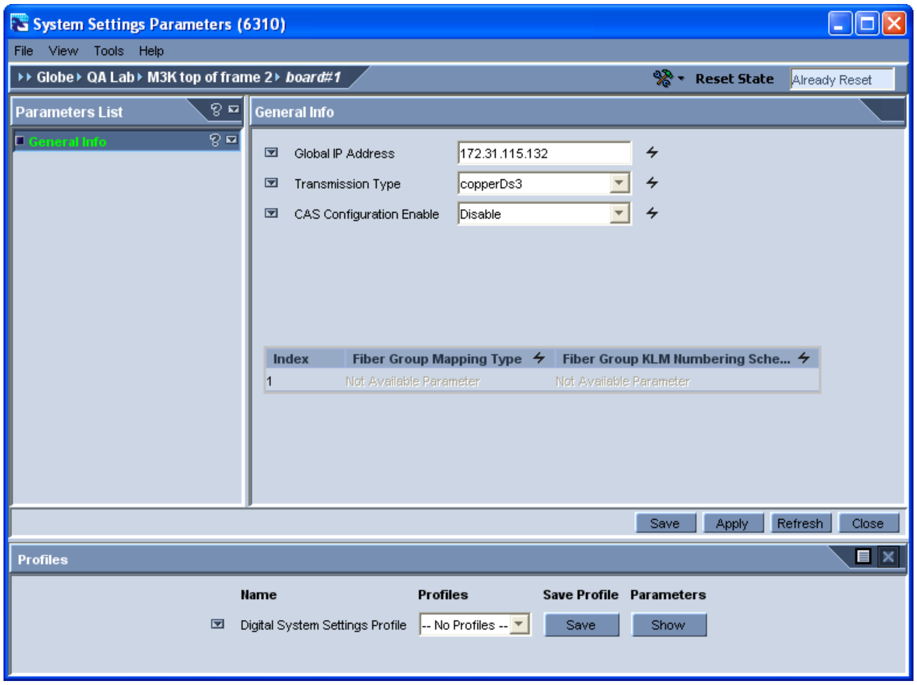

Figure 25 - EMS View Transmission Type

The **Transmission Type** setting is found by double clicking on the active TP-6310 board and choosing **General Config->System Settings->General Info** tab.

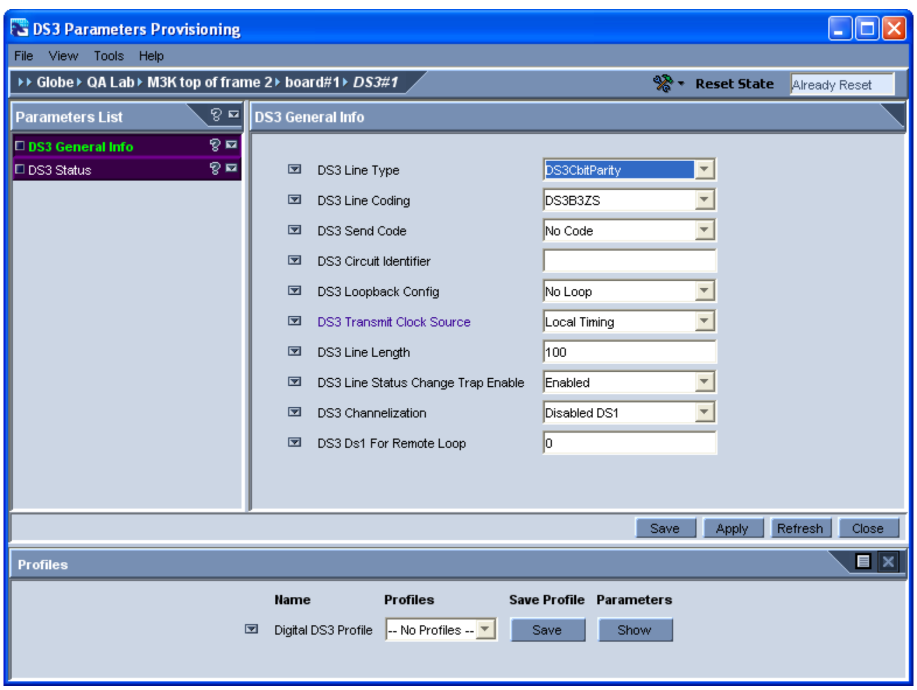

Figure 26 - EMS View DS3 General Info (Local Timing shown)

The **DS3 Transmit Clock Source** and **DS3 Line Type** can be found in the **General Info** tab by double clicking on the TP-6310 board and choosing the **Fiber Group** tab. Then highlight one of the three DS3s in the table and right-click and choose **Properties**.

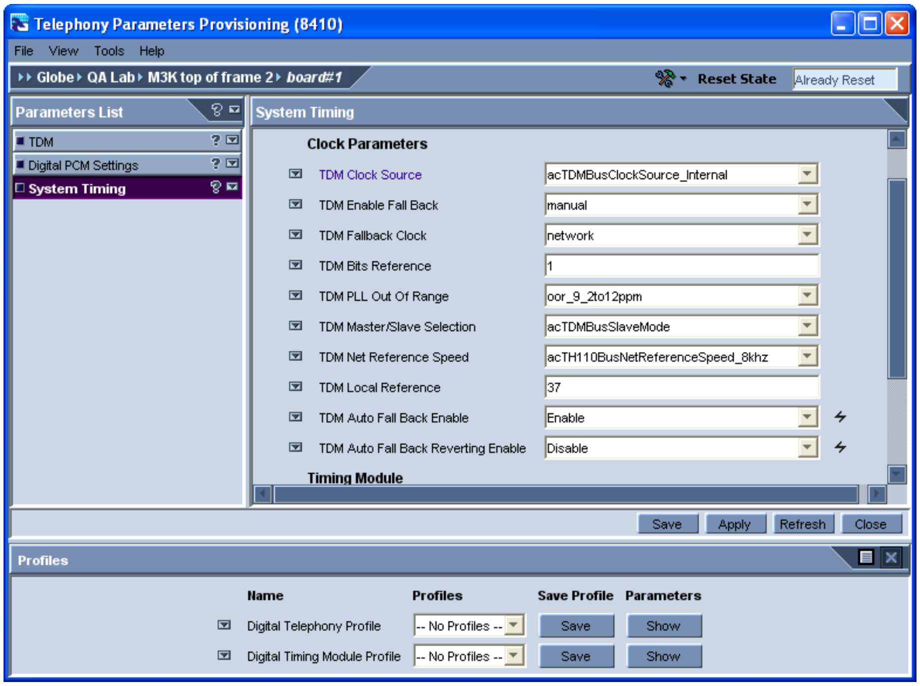

Figure 27 - EMS view TDM Clock Source

The **System Timing** settings are found by double clicking on the active TP-6310 board and choosing **General Config->Telephony->System Timing** tab.

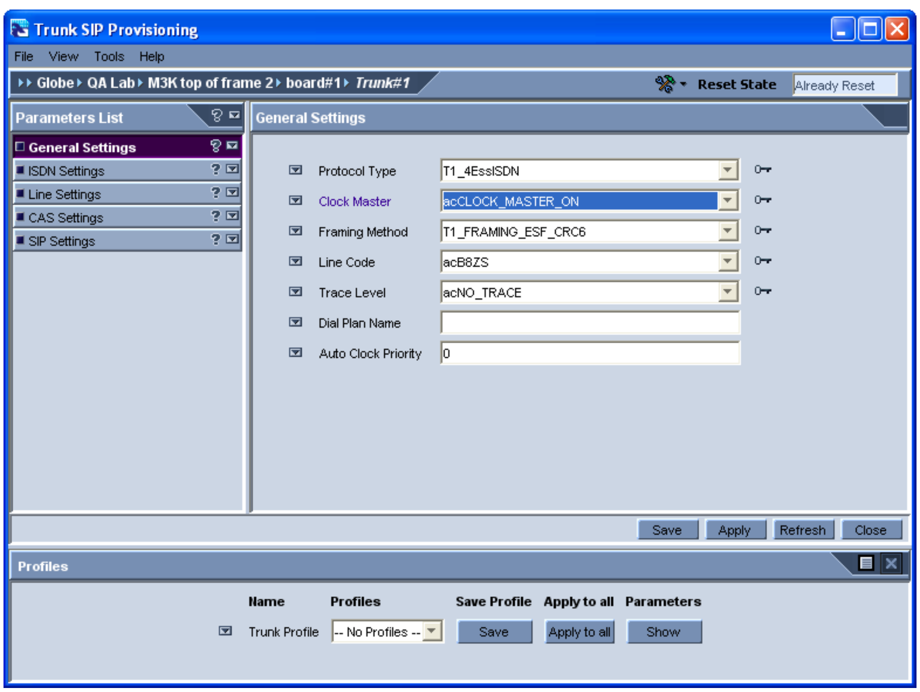

Figure 28 - EMS View T1 Clock Master

The **Clock Master** setting can be found in the **T1 General Settings** by double clicking on the TP-6310 board and choosing the **E1/T1 Trunks** tab. Then highlight one of the DS1s in the table and right-click and choose **Configuration ->Properties**.

5.4 NI2 Trunk Example (Aspect PBX shown)

One of the more common T1 trunk protocol types seen in the field is the NI2 ISDN trunk.

```
ProtocolType_0 = 10
```

T1_NI2_ISDN = 10

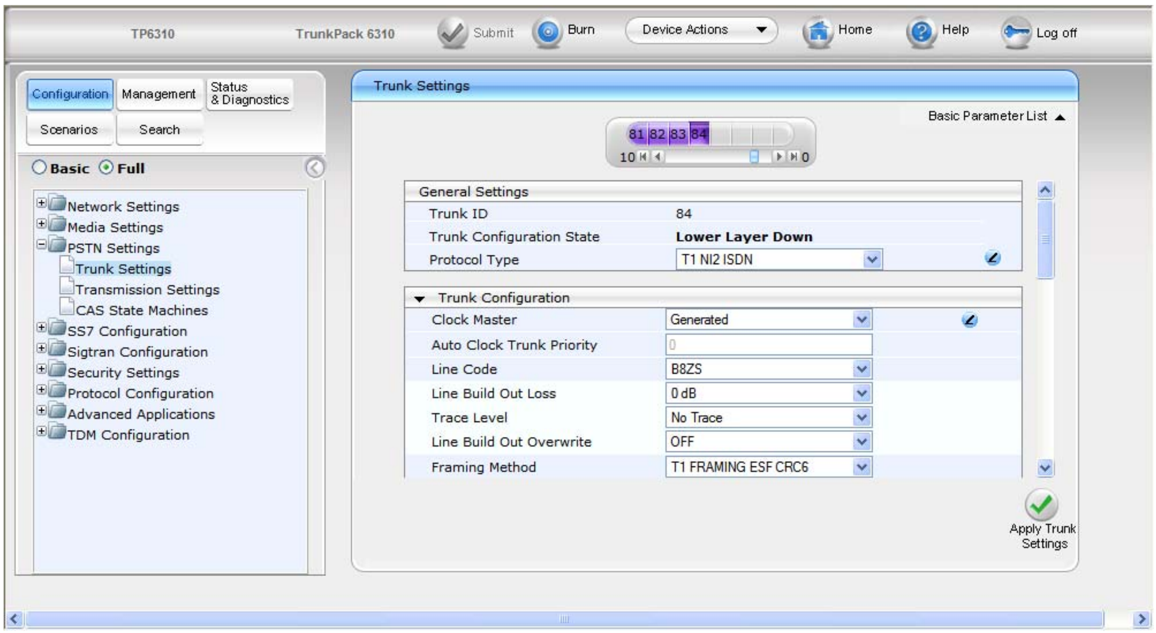

Figure 29 - Web view Ni2 Trunk Settings

In the web view, scroll down to view the ISDN Configuration Settings.

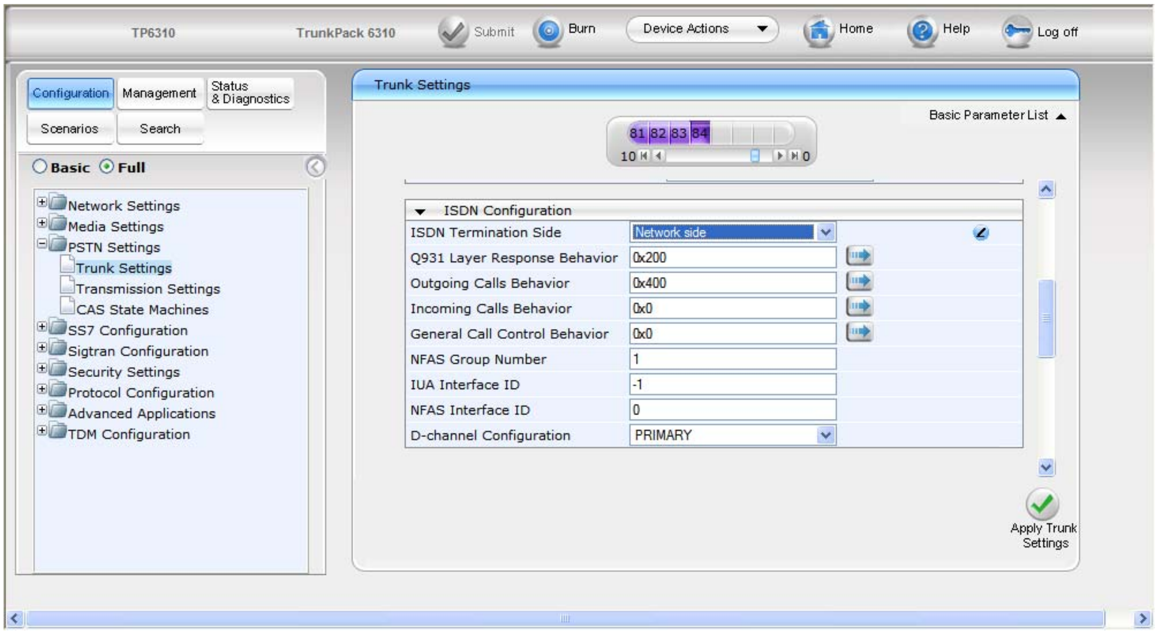

Figure 30 - Web View ISDN Configuration (Network side shown)

The ISDN Termination side should be coordinated with the far end as described in the note below.

```
TerminationSide = 1
```

~ ISDN Termination Side - Selects the ISDN termination side. Applicable only to ISDN protocols.

[0] User side = ISDN User Termination Equipment (TE) side (default)
[1] Network side = ISDN Network Termination (NT) side

NOTE: Select 'User side' when the PSTN or PBX side is configured as 'Network side' and vice versa. If you don't know the device's ISDN termination side, choose 'User side'. If the D-channel alarm is indicated, choose 'Network Side'.

There are some ISDN settings which can be used to alter the outgoing and incoming calls behavior. The default settings are correct unless customized behavior is desired.

```
ISDNOutCallsBehavior = 1024
```

~ Outgoing Calls Behavior - This parameter determines several behaviour options that influence the behaviour of the ISDN Stack outgoing calls. To select options, click the arrow button, and then for each required option, select 1 to enable. The default is 0 (i.e., disable).

[1024] = Numbering plan / type for T1 IP-to-Tel calling numbers are defined according to the manipulation tables or according to the RPID header (default). Otherwise, the plan / type for T1 calls are set according to the length of the calling number.

NOTE: When using the INI file to configure the device to support several ISDNOutCallsBehavior features, add the individual feature values. For example, to support both [2] and [16] features, set ISDNOutCallsBehavior = 18 (i.e., 2 + 16).

Check boxes are used in the EMS to set the ISDN behavior. Refer to these check boxes for other ISDN behavior settings. For a complete description, refer to the User Guide.

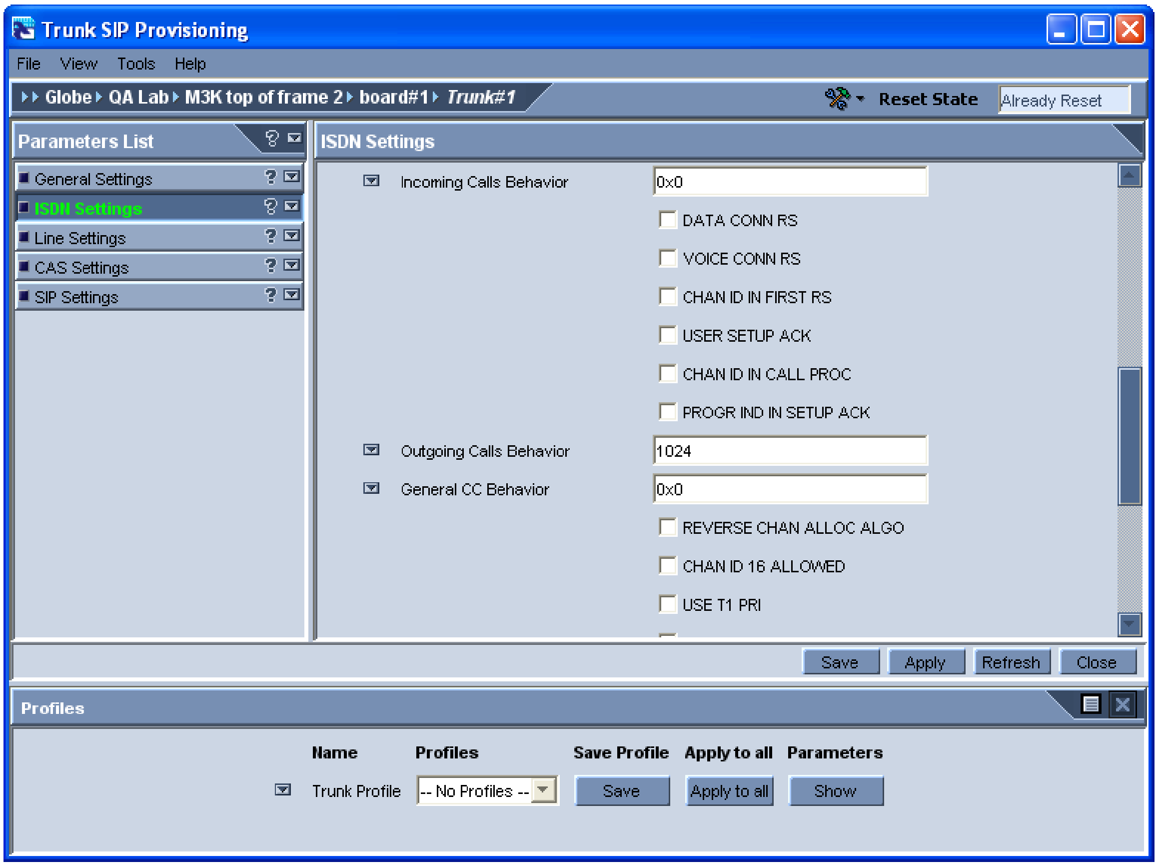

Figure 31 - EMS View ISDN Settings

The **ISDN Settings** can be found in the **T1 General Settings** by double clicking on the TP-6310 board and choosing the **E1/T1 Trunks** tab. Then highlight one of the DS1s in the table and right-click and choose **Configuration ->Properties**. Then choose the **ISDN Settings** tab.

5.5 QSIG Trunk Example (Siemens PBX shown)

When the Trunk Protocol type is T1 QSIG, the only modification is in the Trunk Settings. QSIG is a peer-to-peer signaling system used in corporate voice networking and often seen with Siemens and Avaya PBX equipment. Internationally, QSIG is known as Private Signaling System No. 1 (PSS1).

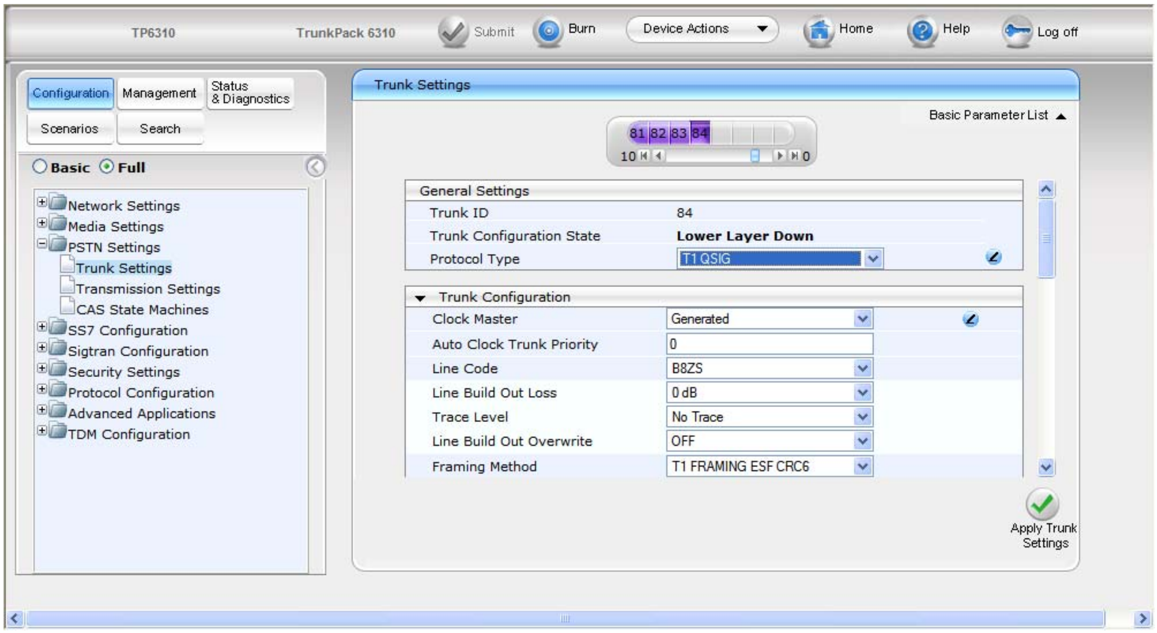

Figure 32 - Web view T1 QSIG trunks

5.6 CAS Trunk Example (Avaya PBX Shown)

Often, some or all of the trunks in a PBX connection will use Channel Associated Signaling (also called Robbed Bit Signaling). There are many variants of CAS trunks for example Wink Start, delay dial, immediate start, FGB, FGD, etc.

The AudioCodes Media Gateway uses CAS Protocol auxiliary files which contain the CAS Protocol state machine definitions that are used for CAS terminated trunks. AudioCodes provides many state machine definitions which have been seen in standard installations on the software distribution media (CD/DVD) in the \Auxiliary_Files folder. In addition to the supplied files, custom definitions can be compiled as needed. Up to eight files can be loaded and different files can be assigned to different trunks. The CAS files can be loaded to the device using the Web interface, the EMS or by INI files. Refer to "Loading Auxiliary Files" in the User’s Guide for more information about loading CAS Protocol Auxiliary files.

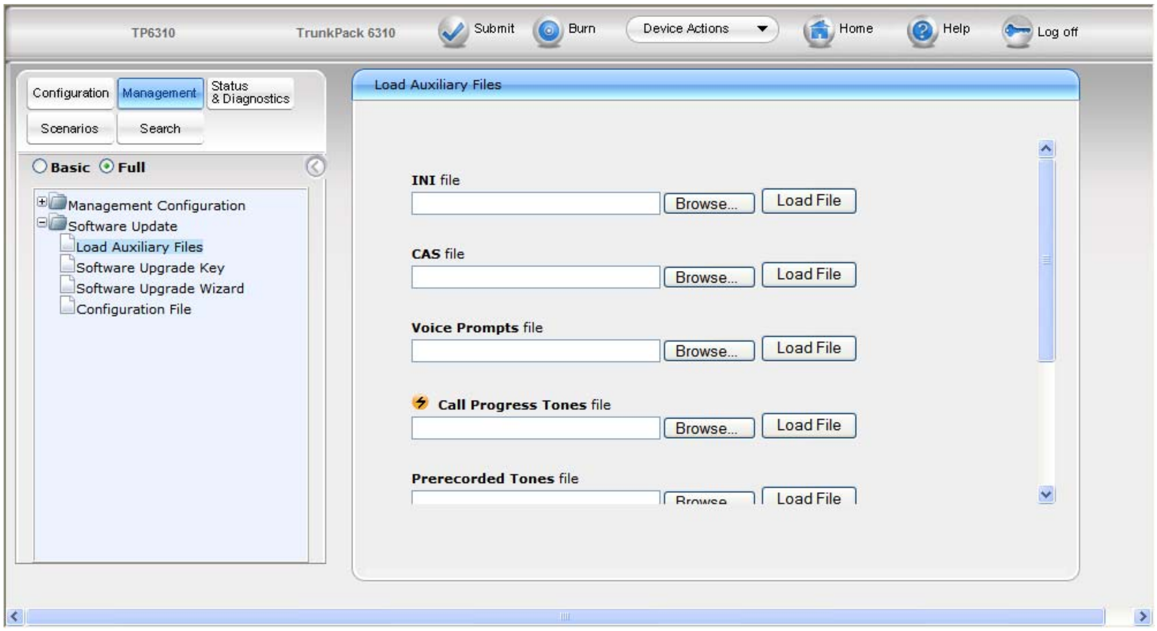

Figure 33 - Web View Load Auxiliary Files

When working directly with the MGW in the web, loading a CAS Protocol Auxiliary file is a two step process. First the file(s) must be downloaded to the MGW. Second, each T1_CAS trunk must be assigned a state machine.

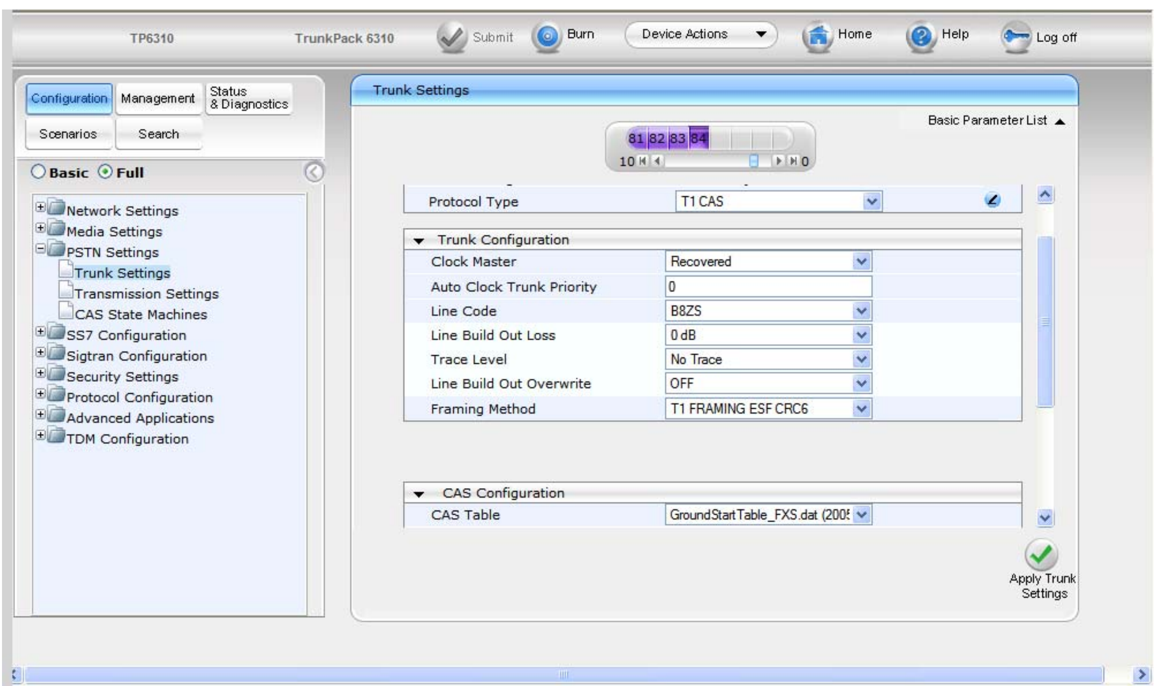

Figure 34 - Web view of CAS Configuration

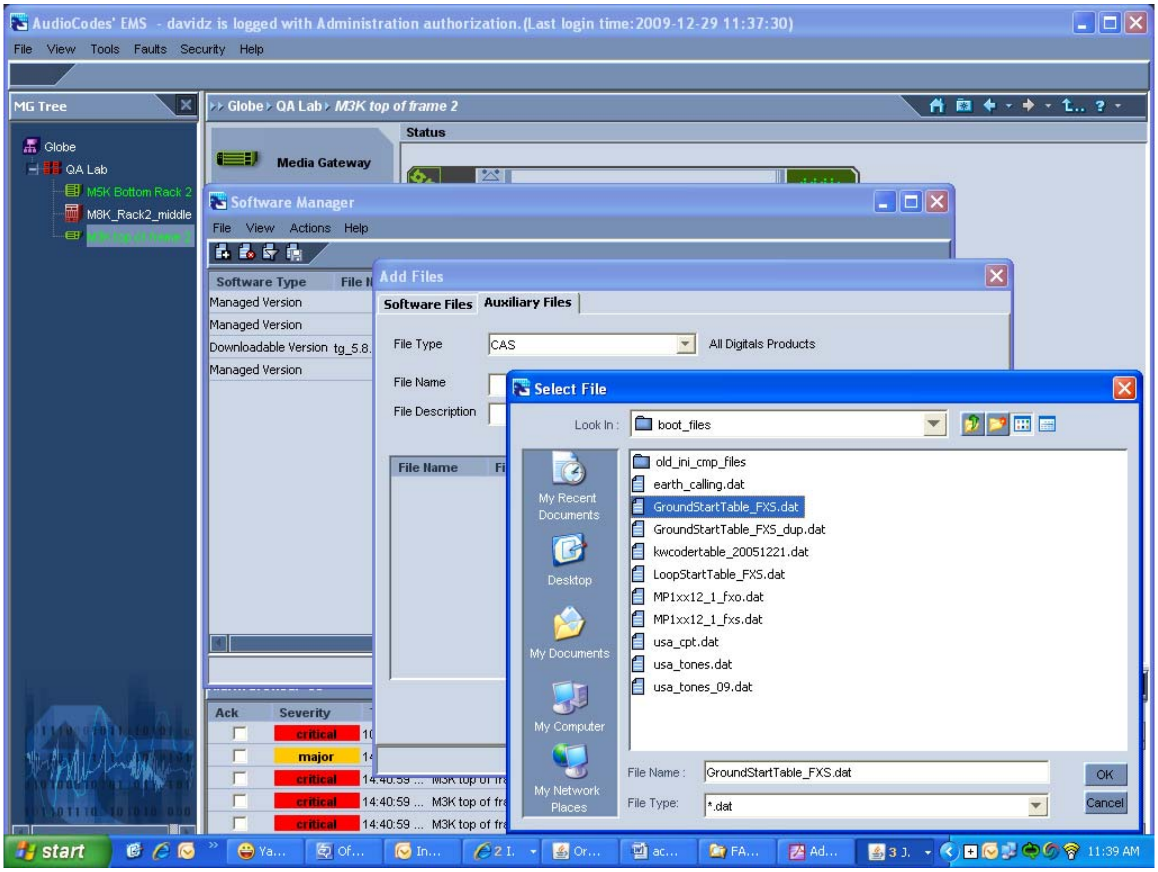

Figure 35 - EMS View Add CAS File to Software Manager

When using the EMS, there is a three step procedure (step 1 shown above).

When using the EMS, the procedure is as follows:

1. Upload the file from your desktop PC to the EMS server using the **Add Auxiliary Files** dialog of the EMS Client by choosing **Software Manager** from the Tools menu at the top of the application.
2. Upgrade the MGW software by downloading the file(s) to the MGW by pressing the **Software Upgrade** button to bring up the **Files Manager** dialog.
3. Each T1_CAS trunk must be assigned a state machine.

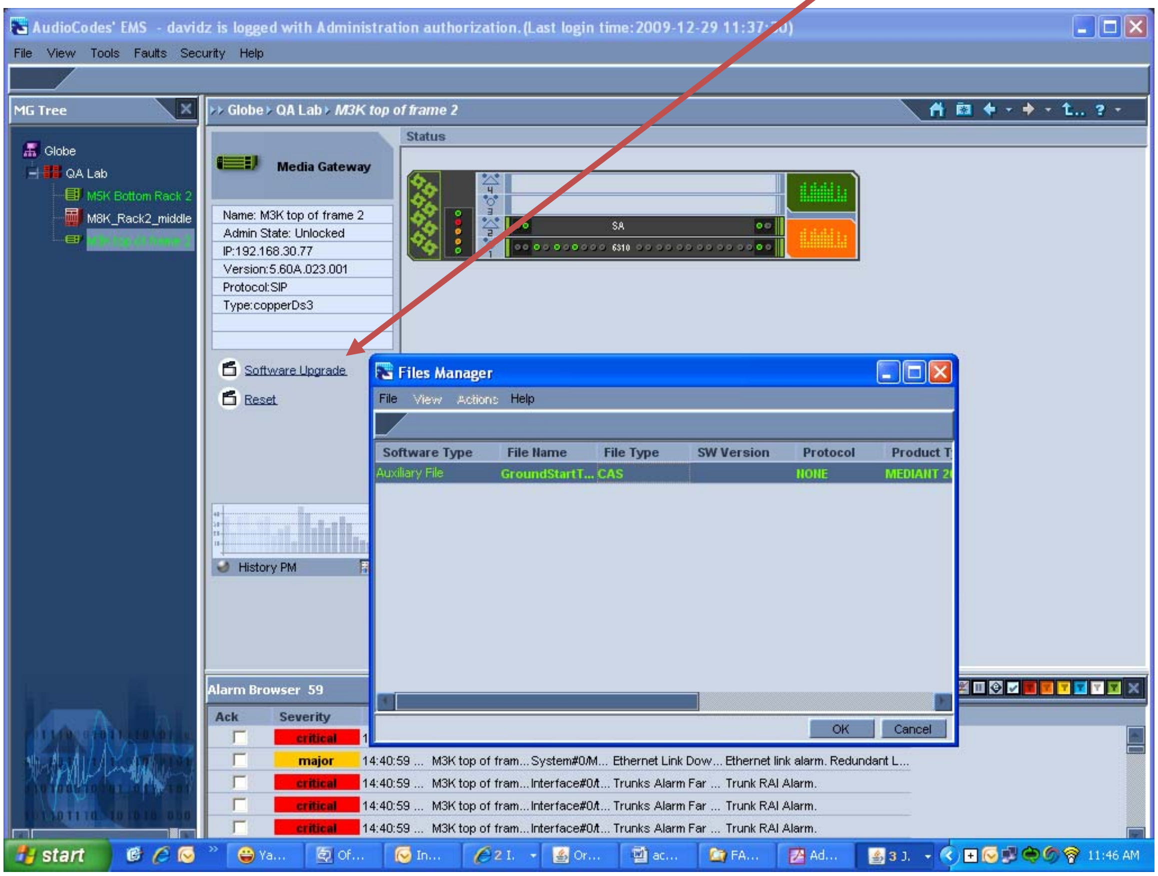

Figure 36 - EMS download of CAS file using the Files Manager dialog

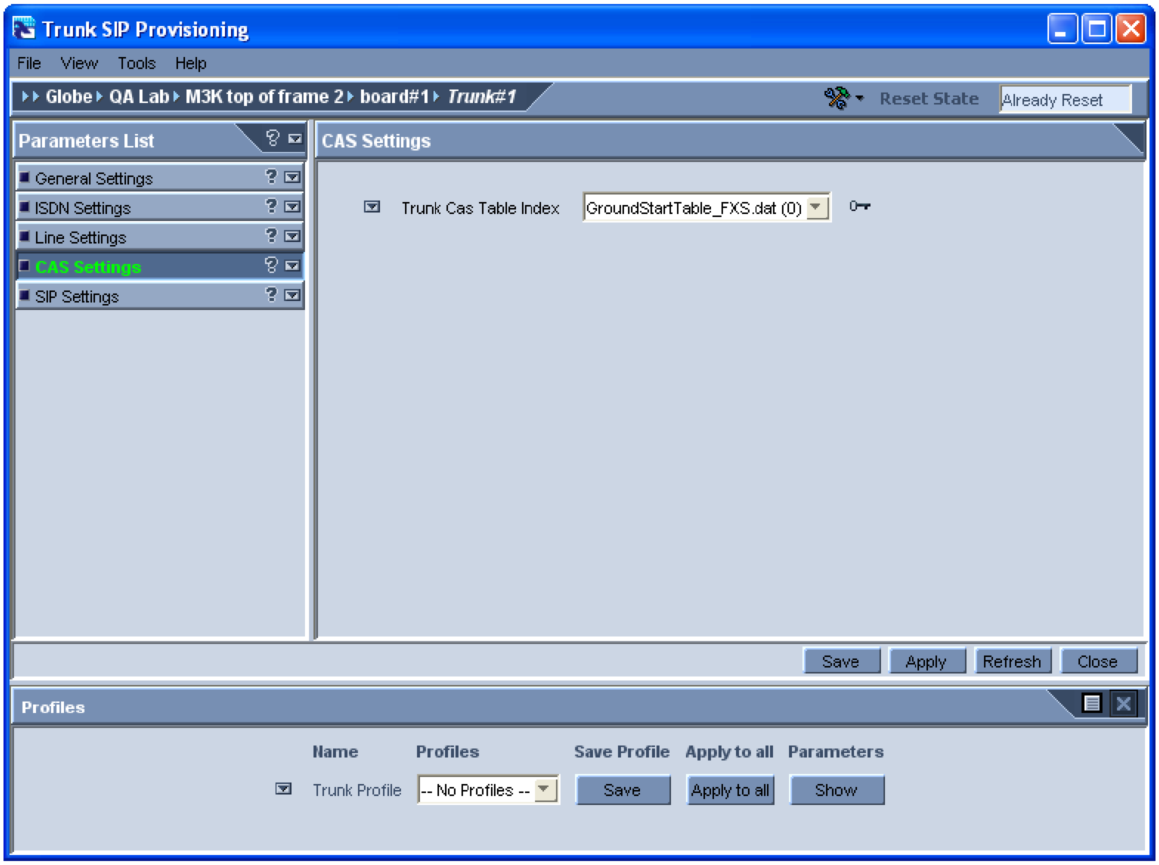

Figure 37 - EMS view CAS Settings

CAS Table Index Example, Trunk 84 shown (INI file trunk 83):

```
ProtocolType_83 = 2
```

T1_CAS = 2

NOTE: The value T1_CAS (or E1_CAS) is normally used for PBX interface (e.g. do not use RAW_CAS).

```
CASTableIndex_83 = 0
```

~ CAS Table Index - This parameter determines which CAS protocol file to use on a specific trunk. The index value corresponds to the number configured for the parameter CASFileName_X. Range = not greater than the parameter defining the PSTN CAS Table Num.

```
CASTablesNum = 2
```

~ CAS Tables Number - This parameter defines the quantity of CAS tables that are loaded to the device during a reset. The quantity of CAS tables defined should match the value configured for parameter CASFILENAME_X (zero when there is no CAS table to be loaded, Range = 0 to 8.

```
CASFileName_0 = 'GroundStartTable_FXS.dat'
CASFileName_1 = 'LoopStartTable_FXS.dat'
```

NOTE: The 'CAS State Machine' page allows you to modify various timers and other basic parameters to define the initialization of the CAS state machine without changing the state machine itself (no compilation is required). It's strongly recommended that you don't modify the default values unless you fully understand the implications of the changes and know the default values. Every change affects the configuration of the state machine parameters and the call process related to the trunk you are using with this state machine.

5.7 NFAS Trunk Example (Nortel PBX Shown)

In regular T1 ISDN trunks, a single 64 kbps channel carries signaling for the other 23 Bchannels of that particular T1 trunk. This channel is called the D-channel and usually resides on timeslot # 24. The ISDN Non-Facility Associated Signaling (NFAS) feature enables the use of a single D-channel to control multiple PRI interfaces.

With NFAS it is possible to define a group of T1 trunks, called an NFAS group, in which a single D-channel carries ISDN signaling messages for the entire group. The NFAS group’s B-channels are used to carry traffic such as voice or data. The NFAS mechanism also enables definition of a backup D-channel on a different T1 trunk, to be used if the primary D-channel fails.

The NFAS group can comprise up to ten T1 trunks. Each T1 trunk is called an ‘NFAS member’. The T1 trunk whose D-channel is used for signaling is called the ‘Primary NFAS Trunk’. The T1 trunk whose D-channel is used for backup signaling is called the ‘Backup NFAS Trunk’. The primary and backup trunks each carry 23 B-channels while all other NFAS trunks each carry 24 B-channels.

The device supports up to 9 NFAS groups.

For example, to assign the first four T1 trunks to NFAS group #1, in which trunk #0 is the primary trunk and trunk #1 is the backup trunk, use the following configuration:

```
NFASGroupNumber 0 = 1
NFASGroupNumber_1 = 1
NFASGroupNumber_2 = 1
NFASGroupNumber_3 = 1
DchConfig_0 = 0 ;Primary T1 trunk
DchConfig_1 = 1 ;Backup T1 trunk
DchConfig_2 = 2 ;24 B-channel NFAS trunk
DchConfig_3 = 2 ;24 B-channel NFAS trunk
```

NFASGroupNumber_x

NFAS Group Number - Indicates the NFAS group number (NFAS member) for the selected trunk. 'x' identifies the Trunk ID. Trunks that belong to the same NFAS group have the same number.

0 = Non NFAS trunk (default)
1 to 9 = NFAS group number

DChConfig_x

~ D-channel Configuration - Defines primary, backup (optional), and B-channels only. The INI file parameter x represents the Trunk ID.

[0] PRIMARY= Primary Trunk (default) - contains a D-channel that is used for signaling.
[1] BACKUP = Backup Trunk - contains a backup D-channel that is used if the primary D-channel fails.
[2] NFAS = NFAS Trunk - contains only 24 B-channels, without a signaling D-channel.

Several ISDN switches require an additional configuration parameter per T1 trunk that is called ‘Interface Identifier’. In NFAS T1 trunks, the Interface Identifier is sent explicitly in Q.931 Setup / Channel Identification IE for all NFAS trunks, except for the B-channels of the Primary trunk (refer to note below).

The Interface ID can be defined per member (T1 trunk) of the NFAS group, and must be coordinated with the configuration of the Switch. The default value of the Interface ID is identical to the number of the physical T1 trunk (0 for the first trunk, 1 for the second T1 trunk, and so on, up to the maximum number of trunks).

The Nortel switch requires the following NFAS Interface ID definitions:

InterfaceID #0 for the Primary trunk
InterfaceID #1 for the Backup trunk
InterfaceID #2 for a 24 B-channel T1 trunk
InterfaceID #3 for a 24 B-channel T1 trunk, and so on for subsequent T1 trunks

For example, if four T1 trunks on a device are configured as a single NFAS group with Primary and Backup T1 trunks that is used with a Nortel switch, the following Additional parameters should be used:

```
ISDNNFASInterfaceID_0 = 0
ISDNNFASInterfaceID_1 = 1
ISDNNFASInterfaceID_2 = 2
ISDNNFASInterfaceID_3 = 3
ISDNIBehavior = 512
```

ISDNIBehavior

Q.931 Layer Response Behavior - Bit-field used to determine several behavior options that influence the behaviour of the Q.931 protocol. To select the options, click the arrow button, and then for each required option, select 1 to enable. The default is 0 (i.e., disable).

[512] EXPLICIT INTERFACE ID = Enables to configure T1 NFAS Interface ID (refer to the parameter ISDNNFASInterfaceID_x).

`ISDNNFASInterfaceID_x`
~ NFAS Interface ID - Defines a different Interface ID for each T1 trunk. The valid range is 0 to 100. The default interface ID equals to the trunk's ID.

The NFAS parameters described in this section can be set using the web GUI or EMS provided the following procedure is used:

- To create an NFAS Group, take these 3 steps:

1. If there's a backup ('secondary') trunk for this group, it must be configured first.
2. Configure the primary trunk before configuring any NFAS ('slave') trunk.
3. Configure NFAS ('slave') trunks.

- To stop / delete an NFAS Group, take these 3 steps:

1. Stop or delete (by setting ProtocolType to 0, i.e., 'None') all NFAS ('slave') trunks.
2. Stop or delete (by setting ProtocolType to 0, i.e., 'None') the backup trunk if a backup trunk exists.
3. Stop or delete (by setting ProtocolType to 0, i.e., 'None') the primary trunk.

In practice, due to the large number of configuration steps necessary, it is often more convenient to edit an INI file with the proper settings and use the 'Load Configuration from File' feature and reset the gateway to apply the NFAS configuration.

6. Call Routing

6.1 Trunk Group

Calls to the PSTN side of the AudioCodes MGW are routed by a pre-defined channel select algorithm to a pre-defined trunk or b-channel. PSTN Trunks / B-channels must be assigned to a 'Trunk Group' for the gateway to route IP-to-Tel calls. Stated another way, resources (T1 spans, B-channels) left out of any trunk group configuration are not utilized by the gateway.

Trunk Groups achieve granular control of routing on per span/channel basis. Trunk groups can be as little as one B-channel or as large as all trunks and B-channels on the gateway. Trunk groups are independent of the T1/E1 protocol on any given PSTN interface. For example, CAS and PRI trunks can be combined in trunk groups.

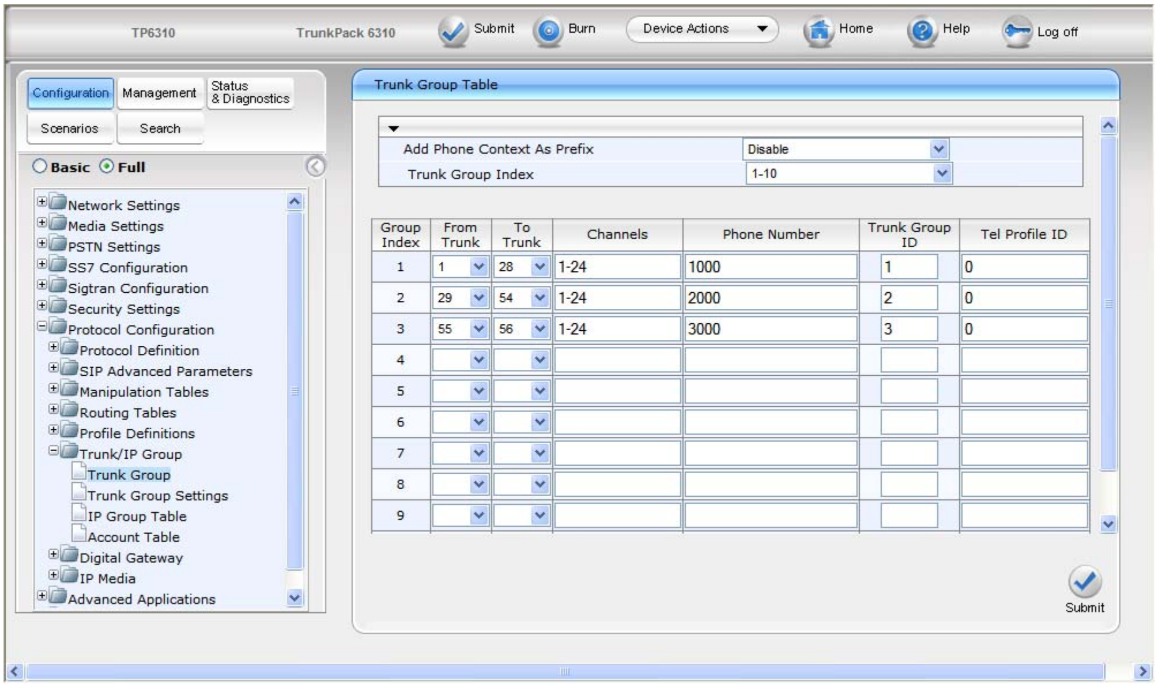

Figure 38 - Web view of Trunk Group table

Trunk Groups and Hunt Groups are synonymous in the EMS.

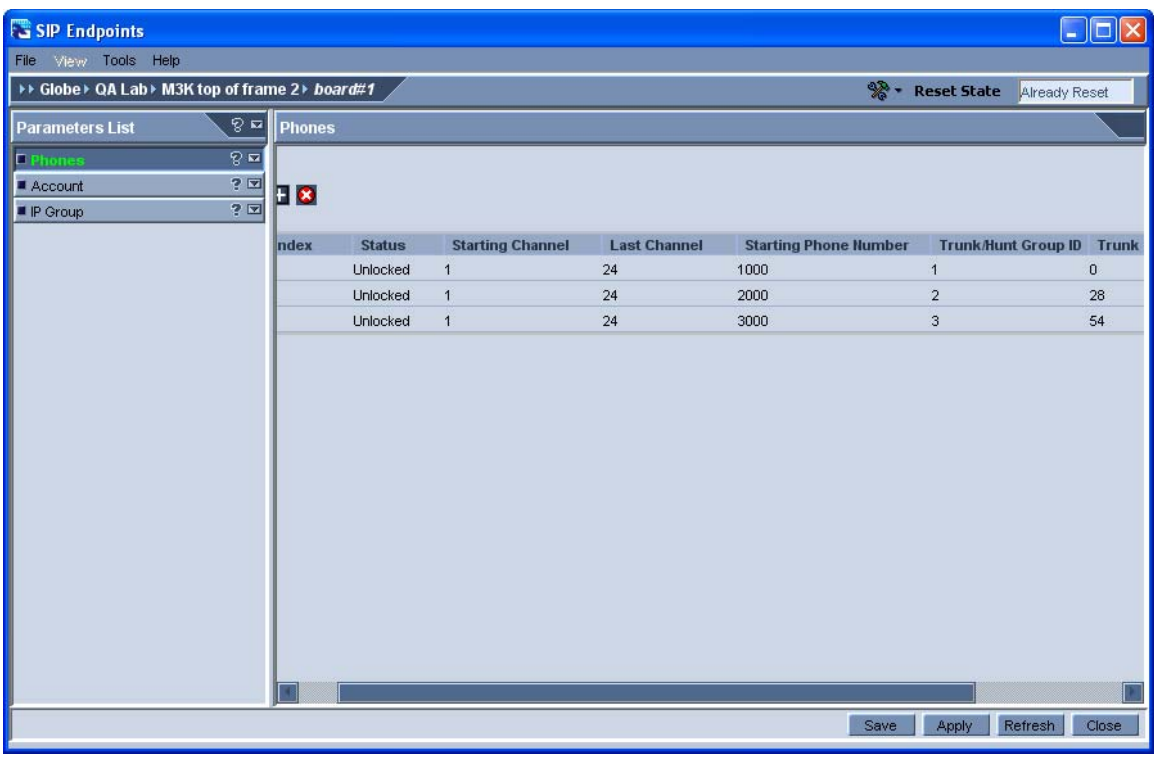

Figure 39 - EMS view of the Trunk/Hunt Groups

The **Trunk Group** table can be found in the EMS by double clicking on the TP-6310 and choosing **SIP->EndPoints->Phones** tab.

6.2 Trunk Group Settings

First, set up Trunk Groups and then choose a channel select mode in the Trunk Group Settings table.

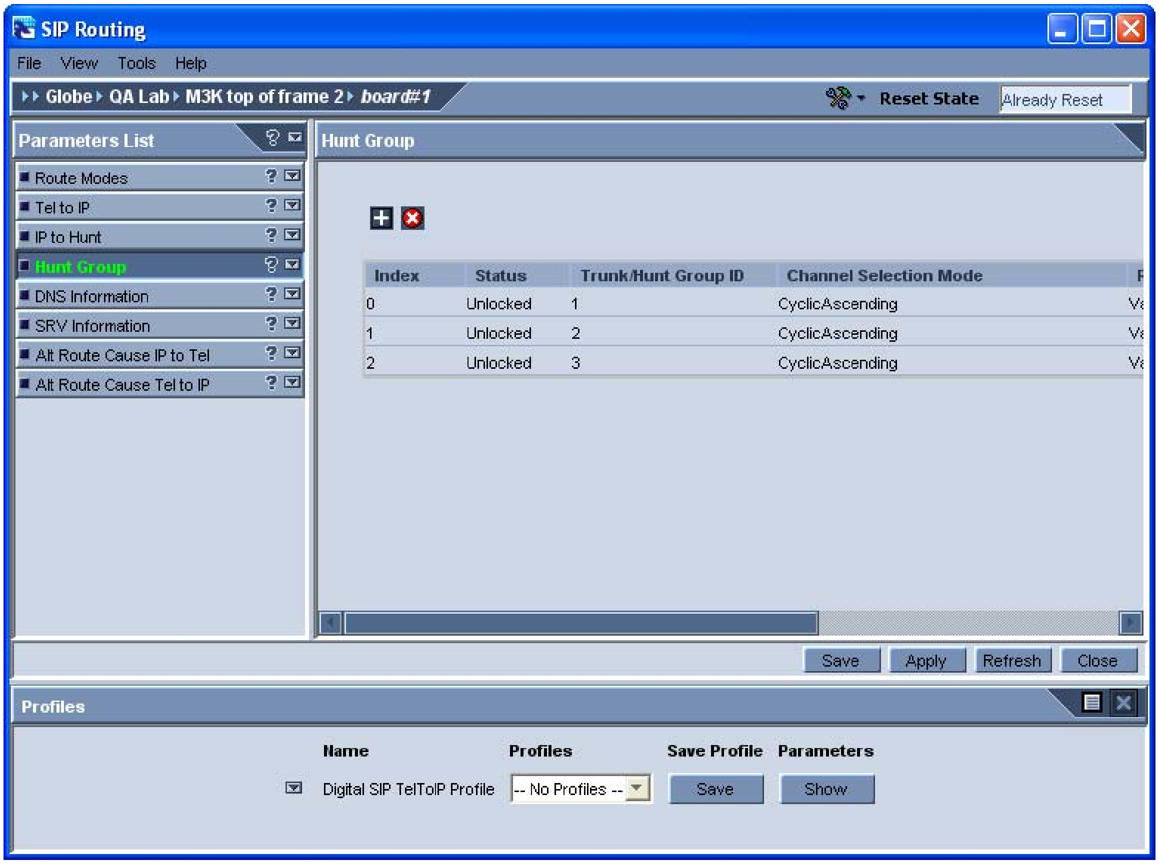

Figure 40 - EMS view of the Trunk Group (Hunt Group) settings

Then, Configure IP-to-Tel Routing tables.

6.3 Routing Table Overview

The Routing and Manipulation tables map incoming called/calling telephone numbers to destination IP address or FQDN. When a proxy is defined, that proxy is used for primary call routing. There are several reasons that the routing tables would still be used with a proxy including:

- Fallback to routing tables – in case Proxy is down

```
PreferRouteTable = 0
```

; ~ Prefer Routing Table - Determines if the device's internal routing table takes precedence over a Proxy for routing calls.

[0] No = Only a Proxy server is used to route calls (default).

[1] Yes = The device checks the routing rules in the 'Tel to IP Routing' table for a match with the Tel-to-IP call. Only if a match is not found is a Proxy used.

- Filter Calls to IP - 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.

```
FilterCalls2IP = 1
```

; ~ Filter Calls to IP - Enables filtering of Tel-to-IP calls when a Proxy is used
    [0] Don't Filter = device doesn't filter calls when using a Proxy.(default)
    [1] Filter = Filtering is enabled.

- IP Security Features - Gateway accepts only those IP/Tel calls with a source IP address identical to one of the IP addresses entered in the Telephone to IP Routing Table.

```
SecureCallsFromIP = 1
```

; ~ IP Security - Determines whether the device accepts SIP calls received from only IP addresses defined in the 'Tel to IP Routing' table. This is useful in preventing unwanted SIP calls or messages and/or VoIP spam.
    [0] Disable = device accepts all SIP calls (default).
    [1] Enable = device accepts SIP calls only from IP addresses

- Obtain different SIP URI host names per called number – multiple SIP trunks (see example in Tel/IP Routing Section below)
- Disable Proxy Routing - Routing Tables take precedence over Proxy by design (Set PreferRouteTable=1)
- Alternative routing - More than one address for a phone number provides a fallback in case the primary route is down.

```
AltRoutingTel2IPEnable = 1
```

; ~ Enable Alt Routing Tel to IP - Enables the Alternative Routing feature for Tel-to-IP calls.
    [0] Disable = Disables the Alternative Routing feature (default).
    [1] Enable = Enables the Alternative Routing feature.
    [2] Status Only = The Alternative Routing feature is disabled, but read-only information on the Quality of Service of the destination IP addresses is provided.

- Profiles

Choose one of two modes to determine the order of manipulation and routing (Applicable to both IP to Tel and Tel to IP tables:

```
RouteModeTel2IP = 0
```

~ Tel to IP Routing Mode - Determines whether to route Tel calls to IP before or after

Manipulation of destination number.

[0] Route calls before manipulation = Tel-to-IP calls are routed before the number manipulation rules are applied (default).

[1] Route calls after manipulation = Tel-to-IP

```
RouteModeIP2Tel = 0
```

~ IP to Tel Routing Mode - Determines whether to route IP calls to the Trunk Group before or after manipulation of destination number.

[0] Route calls before manipulation = IP-to-Tel calls are routed before the number manipulation rules are applied (default).

[1] Route calls after manipulation = IP-to-Tel calls are routed after the number manipulation rules are applied.

6.4 Tel to IP Routing

Route calls based on Destination Phone Number Prefix or Source Phone Number Prefix. Route calls to a specific IP address or FQDN. Routing table pattern matching is the same as Manipulation tables (see next section). Hierarchical order for matching the prefix from specific to general (Top Down).

Calls that match prefixes in table are routed, otherwise call is not completed. Up to 50 entries are allowed.

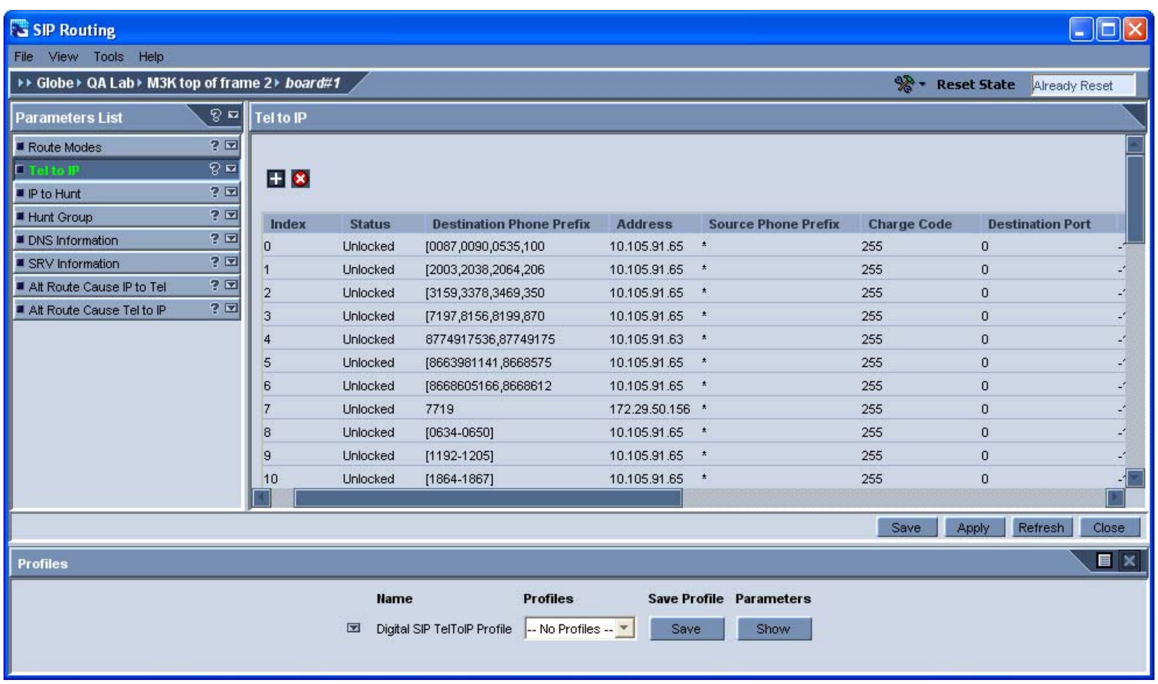

Figure 41 - EMS view Tel to IP Routing Table

The **Tel-to-IP** routing table can be found in the EMS by double clicking on the TP-6310 board and choosing the **SIP->Routing->Tel to IP** tab.

6.5 IP to Tel Routing

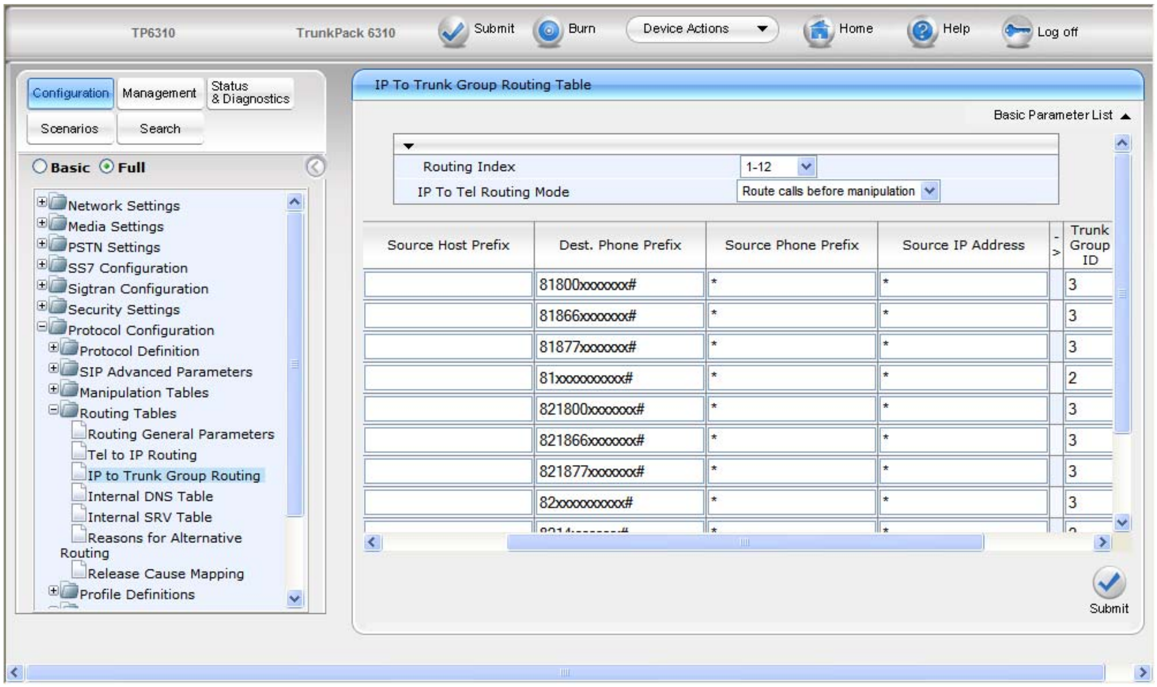

Figure 42 - Web View IP-to-Tel Routing

Prior to defining your IP-toTel routing, define trunk groups as shown above.

6.6 Manipulation

The Manipulation tables allow manipulation of the destination and source phone numbers for IP-to-Tel and Tel-to-IP Calls. They can take effect before or after routing and can be used to Strip or Add digit(s) from/to dialed numbers to conform to dial plan. They can also be used to Allow/Disallow caller ID presentation.

There are four tables provided: IP-to-Tel Destination Numbers, Tel-to-IP Destination Numbers, IP-to-Tel Source Numbers, and Tel-to-IP Source Numbers.

Incoming calls (IP-to-Tel and Tel-to-IP) are matched against the associated Source and Destination manipulation tables. Tables are parsed from top to bottom for a match. Since the table is hierarchical, the more specific patterns should reside at the top and the more general ones at the bottom. In the case where a pattern match is made, the manipulation occurs and the specific manipulation table is exited. Patterns are matched from LEFT to RIGHT. No pattern match will result in nothing changing in the number.

Digits can be removed from the left (prefix) or right (suffix) of the telephone number. A value of ‘0’ indicates that there should not be any action.

Digits can be added to the left (prefix) or right (suffix) of the telephone number. Leave the field blank to invoke no changes.

[n-m]
Represents a range of numbers
Ex: '[1-4]' matches any single digit 1-4 inclusive

[n,m]
Represents multiple numbers.
Note that this notation only supports single digit numbers
Ex: '[9,4]' will match either a 9 OR a 4
x
Represents any single digit
'0-9'

#
(Terminates the number) represents the end of a number

*
A single asterisk (*) represents any number

Patterns can also be used in the IP Address field of both Manipulation Tables and Routing Tables to define ranges or classes of IP Addresses to match.

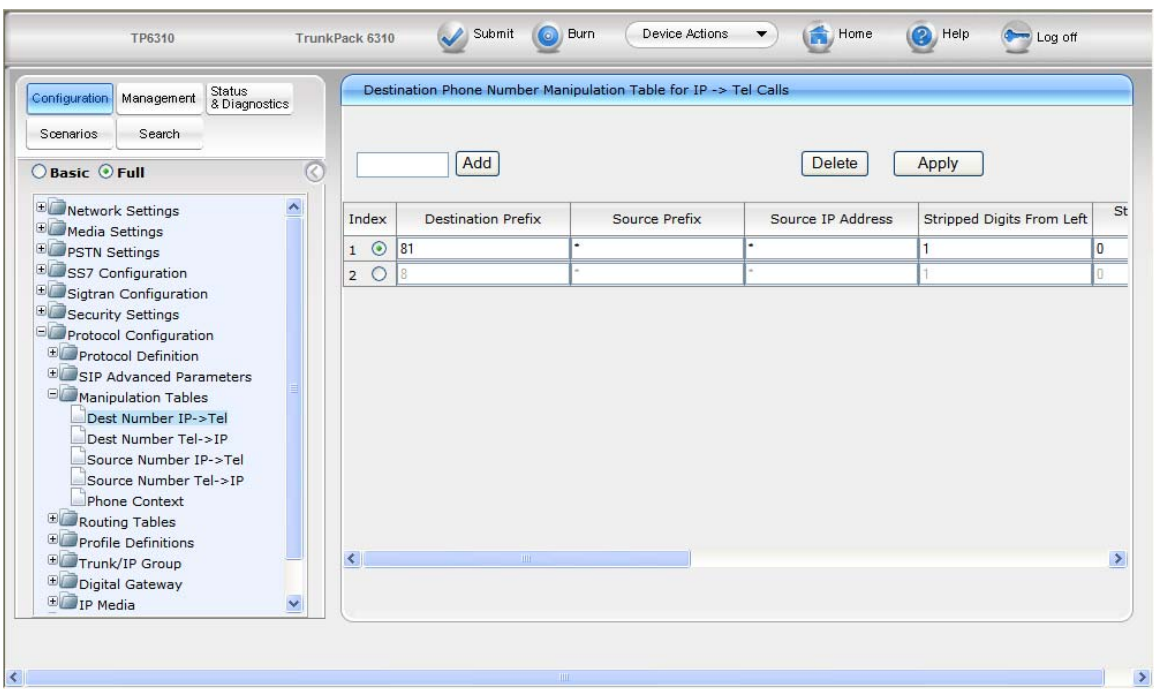

Figure 43 - Web view IP-to-Tel Manipulation table

7. Advanced Settings

7.1 DTMF Settings

The AudioCodes MGW supports several ways of transporting DTMF tones:

- In-Band

  - Transparent
  - DTMF relay (RFC 2833)
- Out-Of-Band
  - INFO (Cisco)
  - INFO (Nortel)
  - NOTIFY

With the transparent setting, the DTMF is carried within the voice stream which works well with a high bit rate coder (for example G.711). It is common to make the setting for DTMF digits to be carried to the remote side as part of the RTP stream in accordance with RFC 2833 standard to accommodate lower bit rate coders.

```
DTMFTransportType = 3
```

DTMF Transport Type - Determines the DTMF transport type.

[0] DTMF Mute = Erases digits from voice stream and doesn't relay to remote.
[2] Transparent DTMF = Digits remain in voice stream.
[3] RFC 2833 Relay DTMF = Erases digits from voice stream and relays to remote according to RFC 2833 (default).
[7] RFC 2833 Relay Rcv Mute = DTMFs are sent according to RFC 2833 and muted when received.

For more information about the Out-of-Band modes, consult the User's Guide. There is also some custom configuration of the RFC 2833 settings controlling the SDP negotiation.

7.2 FAX Settings

The AudioCodes MGW supports several ways of transporting FAX and Modem data:

- T.38 (FAX Only)
- Transparent (FAX and Modem)
- Transparent with events
- Bypass (FAX and Modem)
- NSE (FAX and Modem)
- G711 Transport
- Fax Fallback (from 4.6)
- V.152 (Voice Band Data from 5.0)

The most common setting is T.38 FAX.

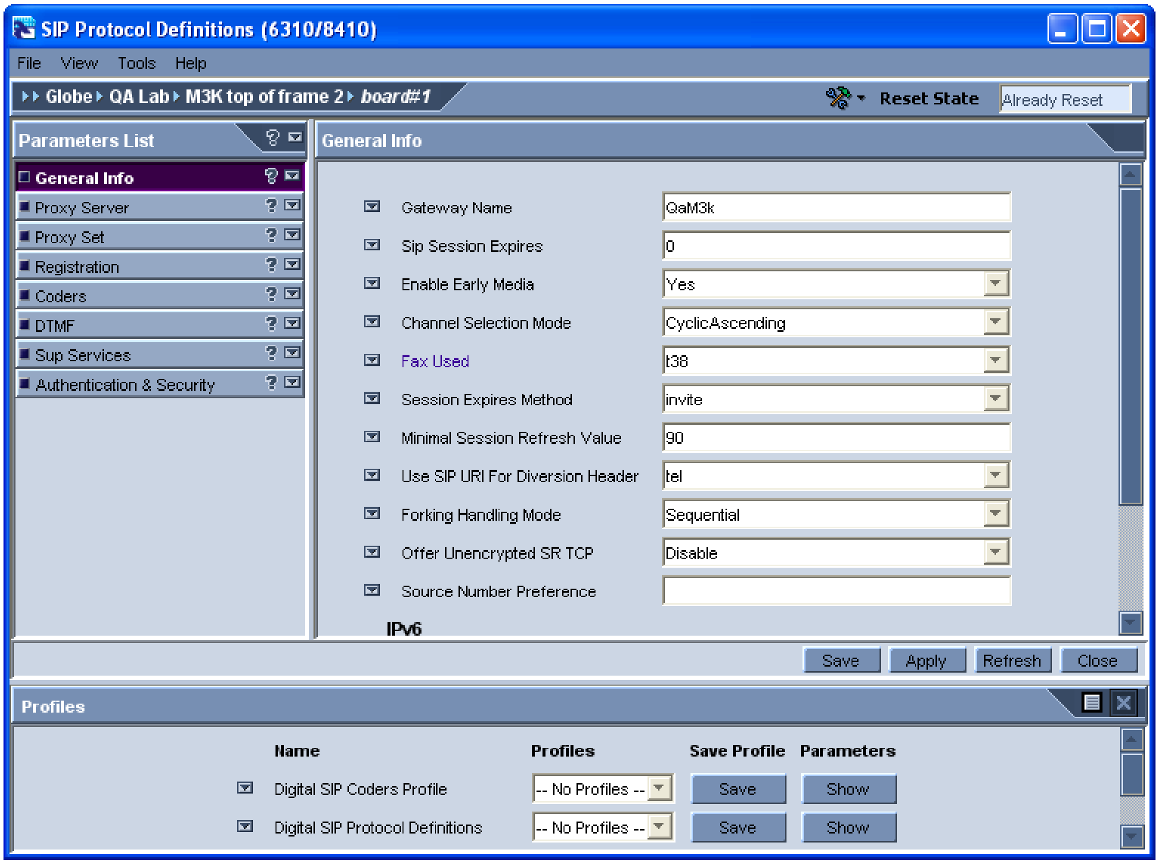

Figure 44 - EMS View General Info (FAX Used)

IsFaxUsed = 1

~ Fax Signaling Method - Determines the SIP signaling method for establishing and transmitting a fax session after a fax is detected.

[0] No Fax = No fax negotiation using SIP signaling. Fax transport method is according to the parameter FaxTransportMode (default).
[1] T.38 Relay = Initiates T.38 fax relay.
[2] G.711 Transport = Initiates fax / modem using the coder G.711 Alaw/µ-law with adaptations (refer to Note below).
[3] Fax Fallback = Initiates T.38 fax relay. If the T.38 negotiation fails, the device re-initiates a fax session using the coder G.711 A-law/µ-law with adaptations (refer to the User Guide for more info).

There are several inter-related parameters involved in changing the T.38 behavior from the most common settings shown in which the terminating endpoint detects the FAX tone and is responsible for sending the originator a RE-INVITE with a T.38 SDP.

Refer to the user guide for a complete description of the different alternatives.

8. Security

Refer to the User Guide for details on the various forms of enhanced security available on the AudioCodes MGW.

- Web Interface – Enhanced security on the web interface using HTTPS.
- Media Security - Enabling Media security using SRTP
- Call Control Security - Enabling Call Control Security using TLS
- OAM Security – Enabling Call Control Security using SNMP V3

The gateway can also be operated in a network with separate Media, Control and Management LANs.

9. OAM

The AudioCodes MGW MIBs can be directly accessed using a MIB browser or other third party custom application. There is also a configuration parameter to send traps to a up to five other destinations (in addition to the AudioCodes EMS).

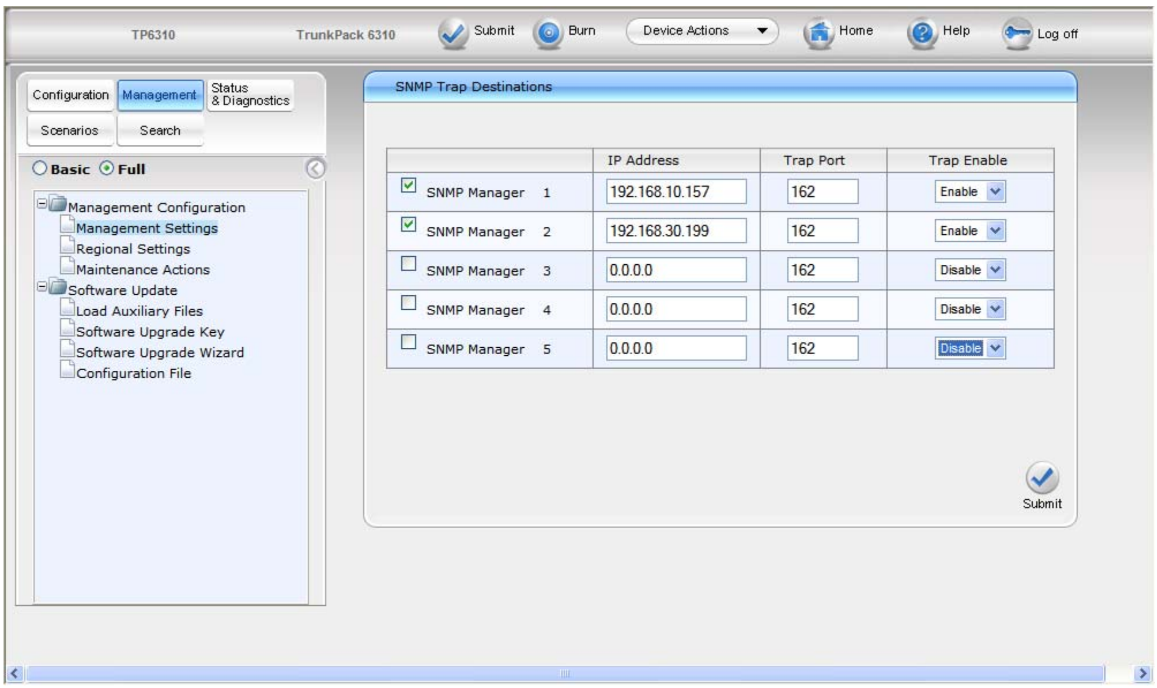

Figure 45 - Web view of SNMP Trap Destinations table

10. Troubleshooting

10.1 Basic Syslogs

Refer to the Installation section of this document for information on configuring basic syslogs and using the AudioCodes syslog server.

10.2 Debug Mode Syslogs

During initial configuration and during troubleshooting sessions, it can be helpful to enable Debug Mode level syslogs. Logging can be modified between five normal debug levels. In practice, most users either disable the feature or use level 5 (not using intermediate levels).

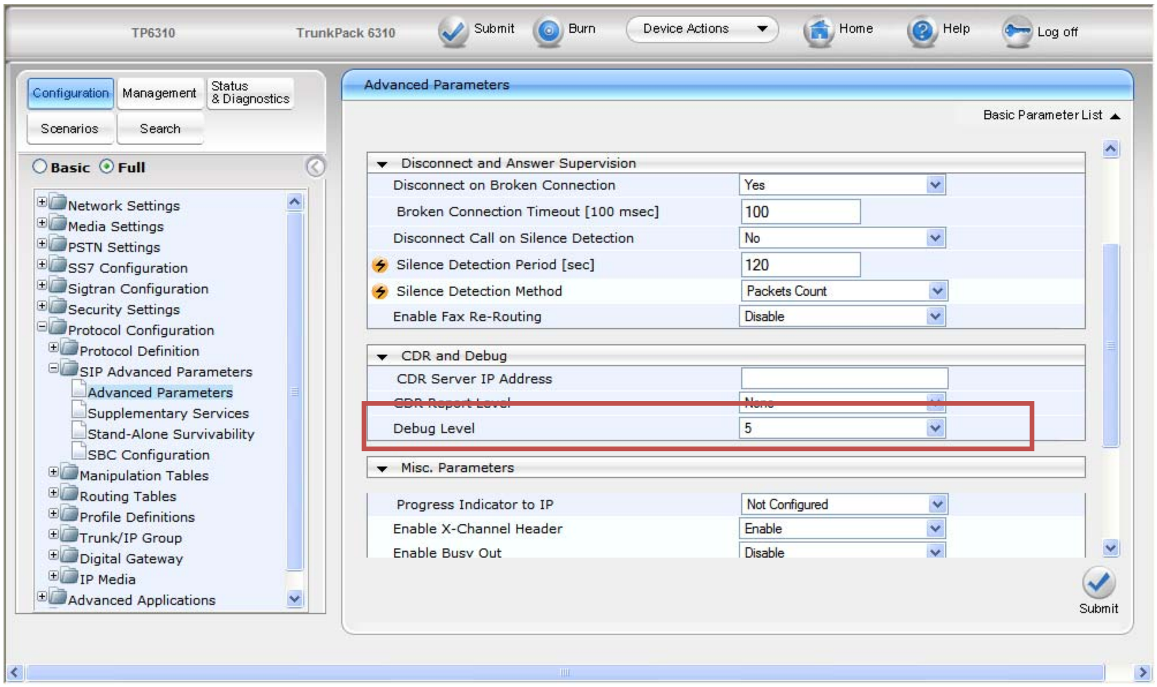

Figure 46 - Web View of Debug Level

```
GwDebugLevel = 5
```

~ Debug Level - Syslog debug logging level.
- [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.

When working with the Gateway Debug Level, be sure to return the setting to DISABLED. The volume of syslog messaging in Debug Mode can affect call processing at high levels of traffic.

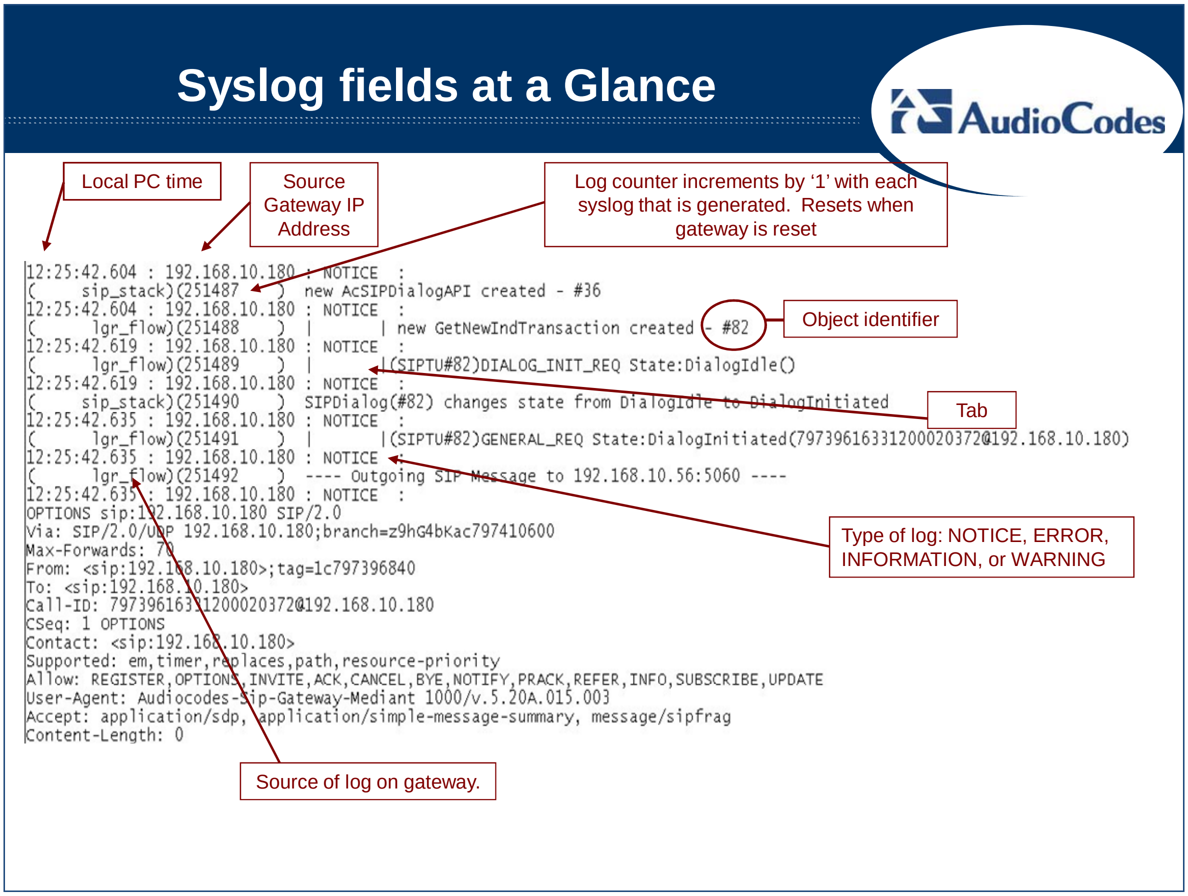

Figure 47 - Syslog fields at a glance (Debug Mode 5)

10.3 Debug Recording Overview

AudioCodes TP Board Debug Recording traces are a powerful debugging tool that provides ability to capture traffic being handled by a specific Media Gateway board. They may be used for analyzing different inter-op scenarios or specific Media Gateway board malfunctions:

- Voice quality issues
- TDM connectivity issues
- Signaling issues

The tool enables forwarding of the specific packets being handled by the Media Gateway board to the user-specified remote IP address. The remote IP address may belong to one of the following:

- External PC (with Wireshark)
- EMS (using snoop)

Flexible user-defined filtering rules apply to selected traffic to be forwarded. For example, it is possible to select traffic between the Media Gateway board and specific remote Media Gateway.   Alternatively, select traffic that belongs to the specific call. Multiple filtering rules may be applied simultaneously.

When Debug Recording Traces are forwarded to the External PC, simply use a packet capturing application (for example Ethereal or Wireshark) for trace collection. An AudioCodes plug-in for Wireshark is supplied with the gateway on the distribution media (CD/DVD) which is used for analyzing the trace (or contact an AudioCodes support representative).