Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E R7.5 with AudioCodes MediaPack 118 over SIP Trunks using a Network Routing Service – Issue 1.0

Abstract

These Application Notes describe how to configure an Avaya Communication Server 1000E to interface with an Audiocodes MediaPack 118. The MediaPack is configured for Gateway Registration and interoperates with the Avaya Communication Server 1000E over SIP trunks using Network Routing Service.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the test configuration for Avaya Communication Server 1000E R7.5 with AudioCodes MediaPack 118. The AudioCodes MediaPack 118 is a standalone analog Voice-over-IP Session Initiation Protocol Media Gateway for connecting legacy analog telephones, fax machines and Private Branch Exchange systems with IP-based telephony networks. The AudioCodes MediaPack 118 is used in small and medium-sized enterprises, branch offices, or residential media gateway solutions. The AudioCodes MediaPack 118 consists of 4 Foreign eXchange Subscriber (FXS) ports to allow the connection of analog phones, modems and faxes and 4 Foreign eXchange Office (FXO) ports to interface to the PSTN. The AudioCodes MediaPack 118enables users to make local or international telephone and/ or fax calls over the Internet between distributed company offices, using existing telephones and fax. These calls are routed over the existing network ensuring that voice traffic uses minimum bandwidth. The AudioCodes MediaPack 118 also provides SIP Trunking capabilities for Enterprises operating with multiple Internet Telephony Service Providers for VoIP services. The AudioCodes MediaPack 118supports the SIP protocol, enabling the deployment of VoIP solutions in environments where each enterprise or residential location is provided with a simple media gateway. This provides the enterprise with a telephone connection (i.e., RJ-11 connector) and also the capability to transmit voice and telephony signals over a packet network. The AudioCodes MediaPack 118 is also equipped with a 10/100Base-TX Ethernet port for connection to the IP network. The AudioCodes MediaPack 118 provides LEDs for indicating operating status of the various interfaces. The AudioCodes MediaPack 118 is a compact unit that can be mounted on a desktop, wall, or in a 19-inch rack. This provides a variety of management and provisioning tools, including an HTTP based embedded Web server, Telnet, Element Management System, and Simple Network Management Protocol the Web interface provides remote configuration using any standard Web browser.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using an Avaya Communication Server 1000E (CS1000E). The AudioCodes MP-118 in normal mode connects to the CS1000E via SIP trunks configured on a Network Routing Service (NRS) and is configured as Gateway Registration, See **Figure 1** for a network diagram. The interoperability compliance test included both feature functionality and serviceability tests. The feature functionality testing focused on a variety of inbound and outbound call handling scenarios to verify successful handling of calls between the CS1000E and the Audiocodes MP-118. The serviceability testing focused on verifying the ability of calls to be routed over the FXO port in the event of the Ethernet connection being lost. A basic Distance Steering Code configuration (DSC) was configured on the CS1000E to route calls to the Audiocodes MP-118. Various Avaya telephones (see **Section 4** for a full list) were used to make and receive calls between the CS1000E and the Audiocodes MP-118. The AudioCodes MP-118 had analog telephones connected to 2 FXS ports. A Fax machine was also connected to both the CS1000E and Audiocodes MP-118. Avaya Call Pilot was configured to test basic voice mail functionality. A simulated PSTN was configured to enable external calls to be made and received.

Note: Although not tested, these configuration steps can also be applied to the AudioCodes MediaPack 114 and 112 using the AudioCodes firmware version specified in **Section 4**. The Audiocodes MP-114 has 2 FXS and FXO ports. The Audiocodes MP-112 has only 2 FXS ports therefore the sections in this document pertaining to Fallback are not relevant.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

Feature functionality testing included

- Basic Calls (using both Codec's G.711 and G.729A)
- Transfers
- Forwarding
- Conferencing
- Call Waiting
- Accessing CallPilot Voicemail
- Sending and receiving Fax's (using both T.38 and G.711 Pass through)
- Call Waiting
- Hold/Unhold
- Calls to and from the PSTN

Serviceability testing verified that basic call functionality continued after an Ethernet outage and full functionality resumed once the Ethernet connection recovered.

2.2. Test Results

All testcases were executed and passed successfully.

2.3. Support

Technical support for AudioCodes can be found as follows:

- Web Portal: http://crm.audiocodes.com/ iSupport Log In required

Enter assigned user name and password. If no access, contact AudioCodes support manager or account manager.

- Email: support@audiocodes.com or applications@audiocodes.com

Describe problem and provide requested information.

- Phone contact: Once a Service Request (SR) has been opened and it is assigned to an AudioCodes Support Engineer, contact is via phone and/or email.

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of a Communication Server 1000E, a Network Routing Service which uses a SIP Trunk to connect to the Audiocodes MP-118. An Avaya CallPilot is included to allow for testing of basic voicemail. The Audiocodes MP-118 in Normal Mode is registered to the Network Routing Service as a Gateway Endpoint. The Communication Server 1000E has one Avaya digital 3904 telephone, two Avaya 1140E IP telephones and a Fax machine configured. Two Analog telephones and a Fax machine are connected to the FXS ports on the Audiocodes MP-118. In Fallback mode one FXO port was connected to a unit on the Avaya Flexible Analog Line Card on the Communication Server 1000E

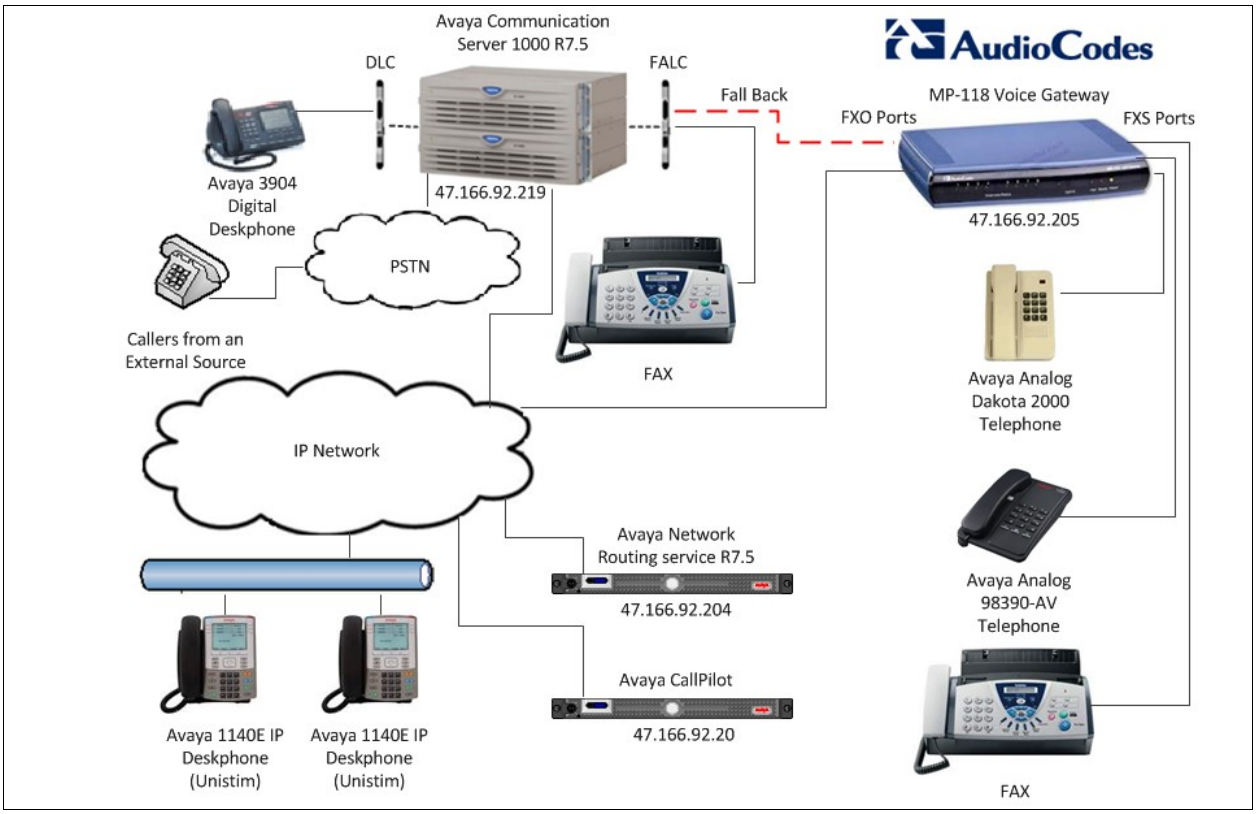


Figure 1: Avaya Communication Server R7.5 with AudioCodes MP-118 Voice Gateway Reference Configuration

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Avaya Equipment	**Software / Firmware Version**
Call Processor Pentium Mobile (CPPM) Avaya Media Gateway NTDW60 Avaya Flexible Analog Line Card NT5K02QC Avaya Digital Line Card NT8D02GA	Avaya Communication Server 1000E R7.5 SP1 FPGA AA18 - -
IBM System x3350	Network Routing Service R7.50.17
Avaya CallPilot 600r	Call Pilot R5 SU08S
Avaya 1140E IP set Avaya 3904 Digital set Avaya Analog set	0625C8A (UNIStim 5.0) F/W 2.4 -
Brother Fax Machine	T106
AudioCodes Equipment	**Software / Firmware Version**
AudioCodes MP-118 FXs-FXO	R6.20A.022.003

5. Configure Avaya Communication Server 1000E

The configuration operations illustrated in this section were performed using terminal access to the CS1000E over a telnet session. It is implied a working system is already in place. For all other provisioning information such as Installation and Configuration, please refer to the product documentation in **Section 9**. **Appendix A** has a list of all CS1000E patches, deplist and service packs loaded on the system. The configuration operations described in this section can be summarized as follows:

- Creating a SIP Trunk to Audiocodes MP-118
- Creating a Coordinated Dialing Plan
- Fax configuration for T.38
- Fax configuration for G.711 Pass Through

Note: In the telnet screenshots below only the unique prompt inputs are shown. To accept default values carriage return at all other prompts.

5.1. Creating a SIP Trunk to AudioCodes MP118

To configure the SIP connection to the Audiocodes MP-118 a number of steps are required:

- Create a D-channel for the SIP trunk
- Create Route Data Block
- Adding TIE Trunks

5.1.1. Create a D-channel for the SIP trunk

Use the **CHG** command in **LD 17** to create a D-channel for the SIP connection. In the example below D-Channel 66 (**DCH 66**) was created. At the **CTYP** prompt enter **DCIP** this signifies SIP D-Channel.

LD 17

Prompt	Response	Description
>	**LD 17**	Enter Overlay 17
REQ	**CHG**	Change
TYPE	ADAN	Change the Action Device and Number
ADAN	NEW	Create New Action Device and Number
TYPE	**DCH 66**	Create new D-Channel 66
CTYP	**DCIP**	Card type is IP D-Channel
USR	ISDL	Integrated Services Digital Line
IFC	SL1	D-Channel interface type

5.1.2. Create Route Data Block

Use the **NEW** command in **LD 16** to create a Route Data Block. In the example below **Route 20** was created. The **TKTP** is a **TIE** route in order to connect to the AudioCodes MP-118. **DCH** is **66** as was created in **Section 5.1.1**. **PCID** is **SIP** and the **Node** is **3** as previously configured.

LD 16

Prompt	**Response**	**Description**
>	**LD 16**	Enter Overlay 16
REQ	**NEW**	Create new
TYPE	**RDB**	Route Data block
CUST	0	Customer Number as defined in LD15
ROUT	**20**	Route Number
TKTP	**TIE**	Route Type
VTRK	YES	Virtual Route
ZONE	1	Zone used
PCID	**SIP**	Protocol ID for route
Node	**3**	Node ID
DTRK	NO	Digital Trunk Route
ISDN	YES	Integrated Services Digital Network
MODE	ISLD	mode of operation
DCH	**66**	D-Channel number as created in Section 5.1
IFC	SL1	Interface type
ICOG	IAO	Incoming and/or Outgoing trunk
ACOD	8020	Access Code for trunk route

5.1.3. Adding TIE Trunks

Use the **NEW** command in **LD 14** to add (**IPTI**) **TIE** trunks. The route number of **RTMB** is **20** as was created in **Section 5.1.2**. If adding multiple trunks use **NEW XX**, where XX is the number of trunks. In the example below **10** trunks were added.

LD 14

Prompt	Response	Description
>	**LD 14**	Enter Overlay 14
REQ	**NEW 10**	Create New
TYPE	IPTI	IP TIE trunk
TN	096 0 3 0	Loop Shelf Card Unit
CUST	0	Customer Number as defined in LD15
TRK	ANLG	Trunk type
RTMB	**20 1**	Route number and Member number
CHID	31	Channel ID for this trunk
TGAR	1	Trunk Group Access Restriction
STRI	IMM	Start arrangement Incoming
STRO	IMM	Start arrangement Outgoing

5.2. Creating a Coordinated Dialing Plan

There are a number of ways to setup a dialing plan to call the AudioCodes MP-118. For compliance testing a Coordinated Dialing Plan (CDP) was used. To configure the CDP a number of steps are required:

- Create a Route List Index
- Create a Distant Steering Code

5.2.1. Create a Route List Index

Use the **NEW** command in **LD 86** to create a new **RLI**. Enter **20** for **ROUT** as was created in **Section 5.1.2.**

LD 86

Prompt	Response	Description
>**LD 86**	Enter Overlay 86	
REQ	**NEW**	Create New
CUST	0	Customer Number as defined in LD15
FEAT	RLB	Route list Block
TYPE	**RLI**	Route list Index
RLI	20	Route list Index number
ENTR	0	First entry for the RLI
ROUT	**20**	Enter the route number

5.2.2. Create a Distant Steering Code

Use the **NEW** command in **LD 87** to create a Distant Steering Code (DSC) entry for the AudioCodes MP-118. For each extension a DSC entry needs to be created. During compliance testing all the telephone extensions on the AudioCodes MP-118 began with the digits 60. In the example below the **DSC 60** was used. All extensions on the AudioCodes MP-118 were four digits in length, therefore the **FLEN** is set to **4.** The **RLI** is **20** as created in **Section 5.2.1**.

Note: During compliance testing the Fax machine had an extension 6111, therefore another DSC of 6111 was created (not shown).

LD 87

Prompt	**Response**	**Description**
>	**LD 87**	Enter Overlay 87
REQ	**NEW**	Create new
CUST	0	Customer Number as defined in LD15
FEAT	**CDP**	Coordinated dialing plan
TYPE	DSC	Distance Steering code
DSC	60	Distant Steering code
FLEN	4	Flexible Length number of digits
RLI	20	Route list index Number

5.3. Fax Configuration

During compliance testing T.38 and G.711 Pass Through were both tested. The configuration operations described in this section can be summarized as follows:

- T.38 Fax Configuration
- G.711 Pass Through Configuration

5.3.1. T.38 Fax Configuration

To ensure that T.38 is used by the Fax machine, the unit that the Fax is configured on must have the Class of Service (**CLS**) Fax Allowed (**FAXA**) and Modem Pass Through Denied (**MPTD**). Use the **CHG** command in **LD 10** to change the CLS. During compliance testing **TN 4 0 7 0** was the unit that the Fax machine was configured.

LD 10

Prompt	**Response**	**Description**
>	**LD 10**	Enter Overlay 10
REQ	**CHG**	Change
TYPE	**500**	Set Type
TN	**4 0 7 0**	Loop Shelf Card Unit
ECHG	**YES**	Easy Change
ITEM	**CLS**	Item to change
CLS FAXA MPTD		

5.3.2. G.711 Pass Through Fax Configuration

To ensure that G.711 Pass Through is used by the Fax machine, the unit that the Fax is configured on must have the Class of Service (**CLS**) Fax Denied (**FAXD**) and Modem Pass Through Allowed (**MPTA**). Use the **CHG** command in **LD 10** to change the CLS. During compliance testing **TN 4 0 7 0** was the unit that the Fax machine was configured.

Note: Pass Through must also be enabled on the Node, see **Section 5.6.2** for more details.

LD 10

Prompt	Response	Description
>	**LD 10**	Enter Overlay 10
REQ	**CHG**	Change
TYPE	**500**	Set Type
TN	**4 0 7 0**	Loop Shelf Card Unit
ECHG	**YES**	Easy Change
ITEM	**CLS**	Item to change
CLS FAXD MPTA		

5.4. AudioCodes MP-118 as a Endpoint on the NRS

To route calls to and from the AudioCodes MP-118 it is configured as a Dynamic Endpoint and Routing Entries need to be created on the NRS. Access the web GUI of the NRS Manager, using the URL https://<fqdn> or http://<ip-address>. Where the <fqdn> is the fully qualified domain name of the NRS and the <ipaddress> is the IP address of the NRS. At the Log in page log in with the appropriate credentials. The steps required on the NRS can be summarized as follows:

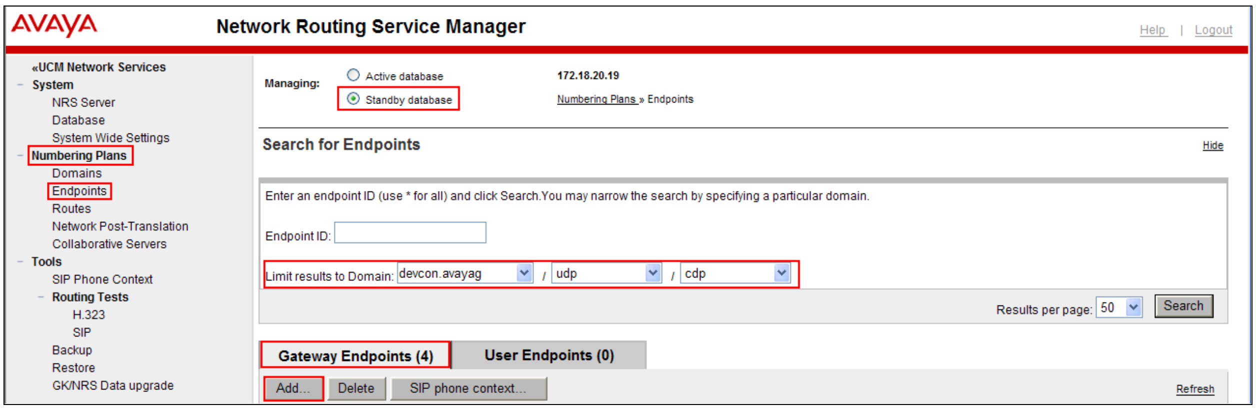
- Create a SIP Dynamic Endpoint
- Create Routing Entries for the AudioCodes MP-118

5.4.1. Create a SIP Dynamic Endpoint

From the **Network Routing Service Manager** page, navigate to **Numbering Plans → Endpoints**. Click on the **Standby database** radio button. From the **Limit results to Domain:** drop down boxes select the following:

- Drop down box 1 Select the domain the NRS resides. i.e., **devcon.avayag**
- Drop down box 2 Select **udp**
- Drop down box 3 Select **cdp**

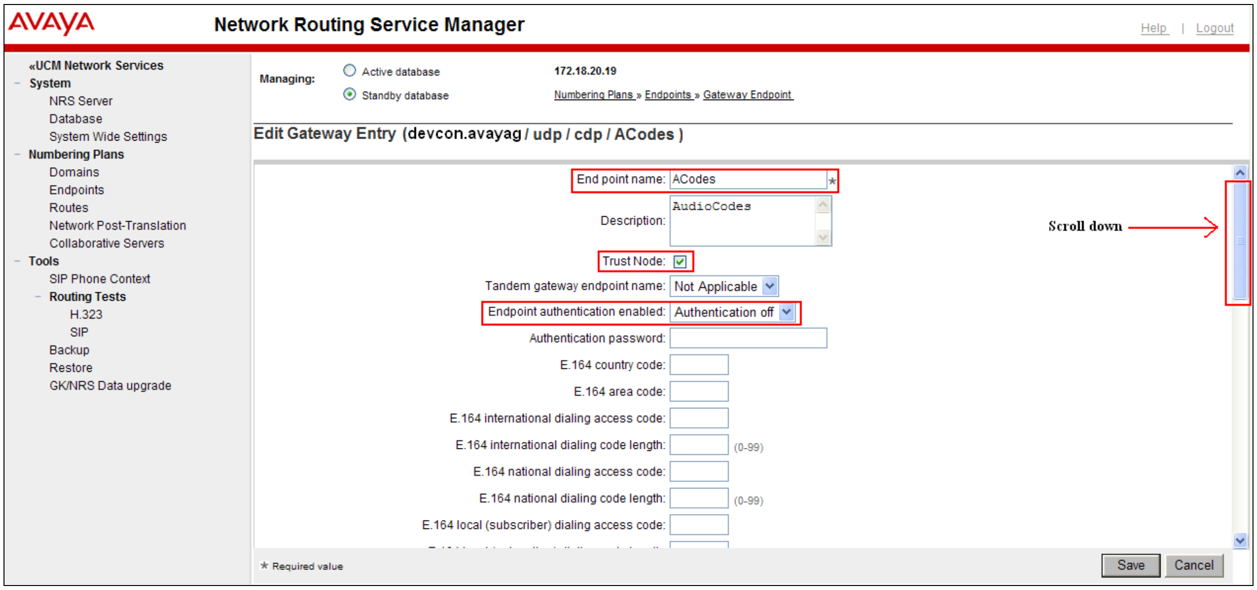
Select the **Gateway Endpoints** tab and click on the **Add** button.



Once the new page opens enter the following:

- **End point name:** Enter an informative name. i.e., **ACodes**
- **Trust Node:** Check the checkbox
- **Endpoint authentication enabled:** From the dropdown box select **Authentication off**

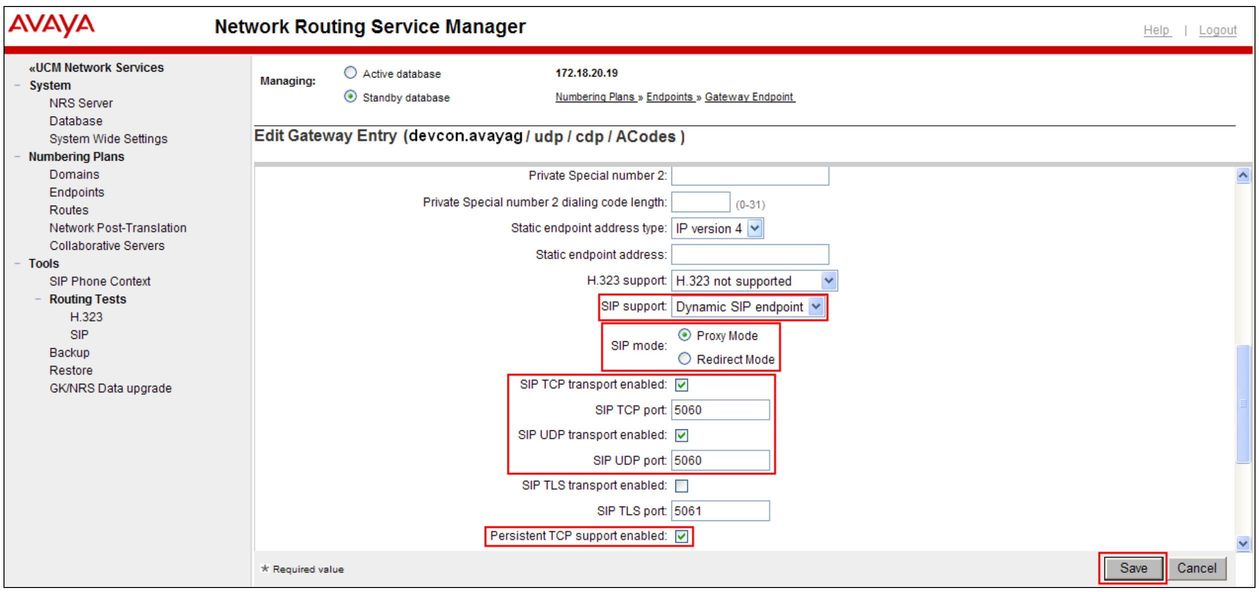
Scroll down using the scroll bar as shown in the screen shot.



In the page below fill in the following:

- **SIP support:** From the dropdown box select **Dynamic SIP endpoint**
- **SIP Mode:** Click on the **Proxy Mode** radio button
- **SIP TCP transport enabled:** Check the checkbox
- **SIP TCP port:** Enter **5060**
- **SIP UDP transport enabled:** Check the checkbox
- **SIP UDP port:** Enter **5060**
- **Persistent TCP support enabled:** Check the checkbox

Click on the **Save** button



5.4.2. Create Routing Entries for the AudioCodes MP-118

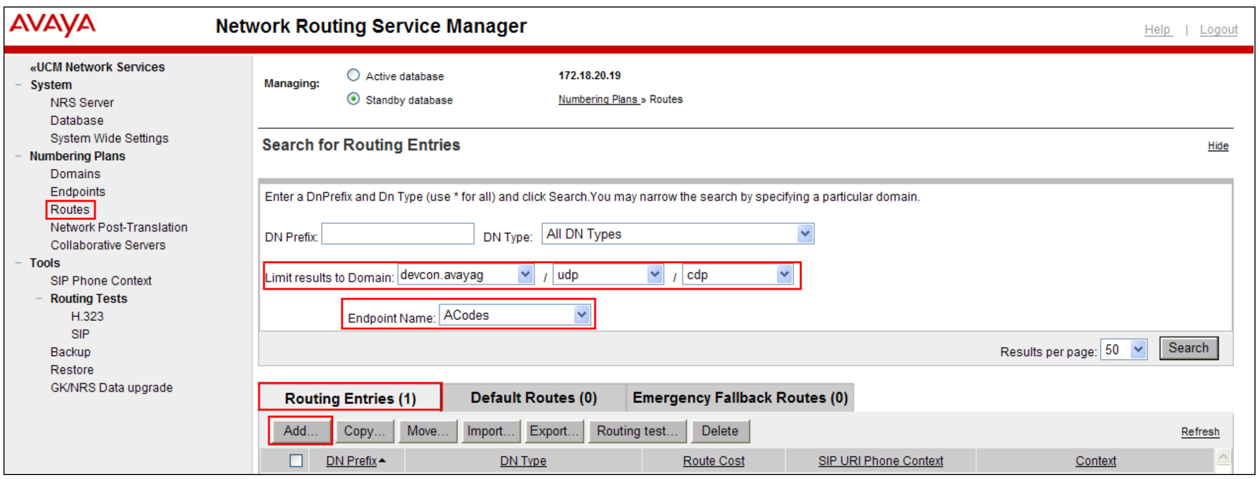
To route calls to extensions on the AudioCodes MP-118 a Routing Entry needs to be created. During compliance testing only calls beginning with the digit 6 were routed the Audiocodes MP-118 therefore only one routing entry beginning with the digit 6 was required.

From the **Network Routing Server Manager** page, navigate to **Numbering Plans → Routes**.

From the **Limit results to Domain:** drop down boxes select the following:

- Drop down box 1 Select the domain the NRS resides. i.e., **devcon.avayag**
- Drop down box 2 Select **udp**
- Drop down box 3 Select **cdp**

Select **ACodes** from **the Endpoint Name:** dropdown box and click on the **Add** button.

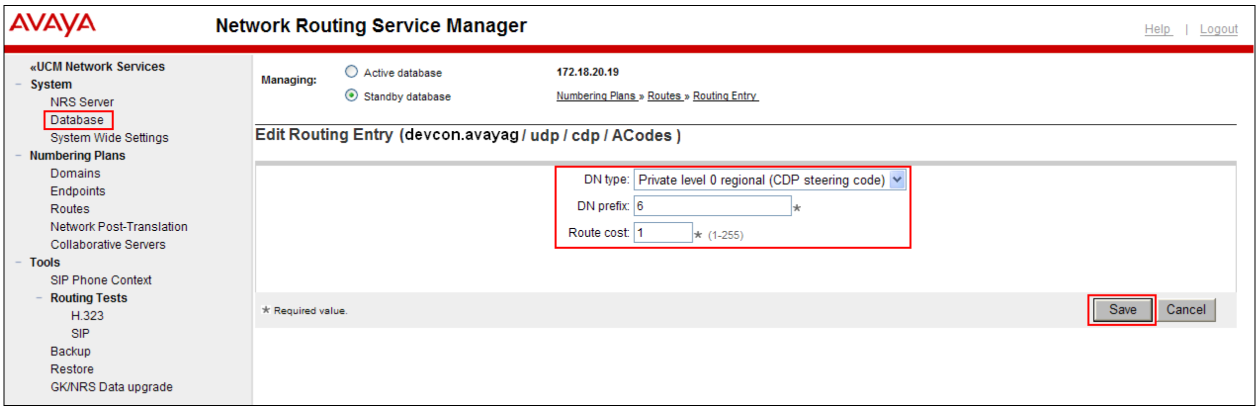


Once the new page opens enter the following:

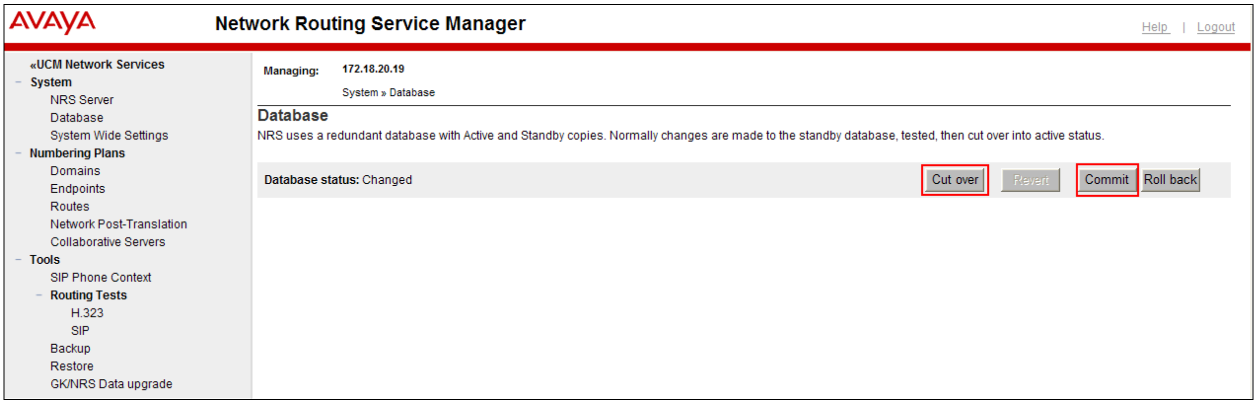
- **DN type:** From the dropdown box select **Private level 0 regional (CDP steering code)**
- **DN Prefix:** Enter the digit 6
- **Route cost:** Enter **1**

Click on the **Save** button and select **Database**.

Note: To save the changes to the Active Database a Cut over is required.



Once the new page opens click on the **Cutover** button followed by the **Commit** button. It may take a few minutes for the databases to synchronise.



5.5. Avaya Element Manager Configurations

During compliance testing some configuration changes were require using the Element Manager. The configuration changes are made using the Element Manager of the Nodes where the CS1000E resides. At the Log in page of Element Manager log in with the appropriate credentials. The changes can be summarised as follows:

- Modify SIP URI Map
- Configure Modem/Fax pass-through
- Configure Voice Codec G.729
- Transmitting Node Changes

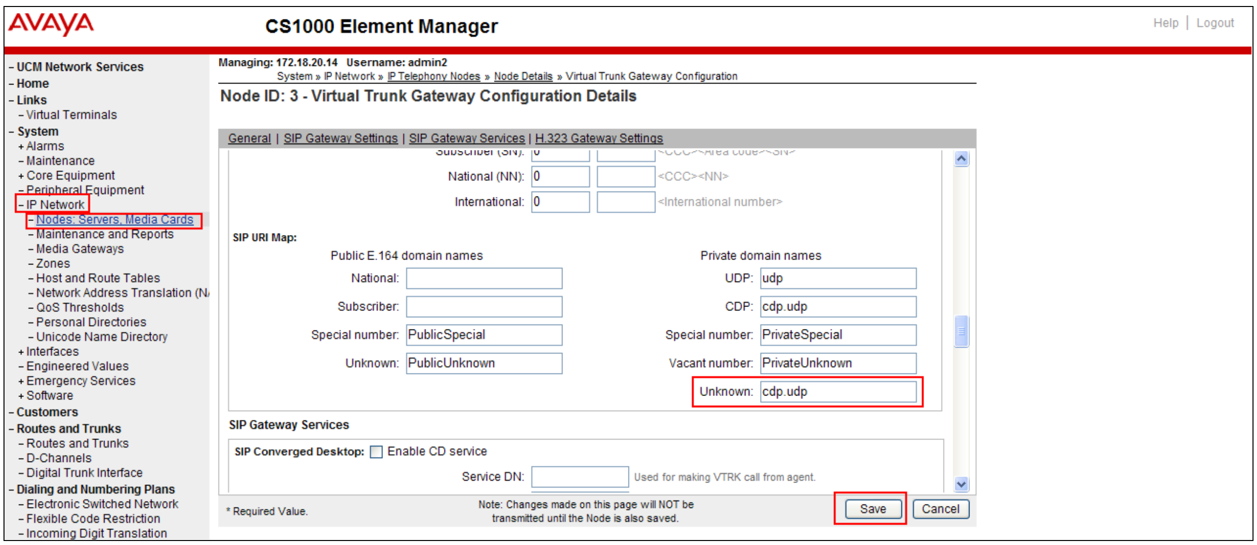
Note: As G.711 is the default Voice Codec it is not shown in the configuration.

5.5.1. Modify SIP URI Map

The SIP URI Map needs to be modified so as to allow for Calls originating from the CS1000E to be transferred back to another extension on CS1000E where the transfer was initiated by an extension on the AudioCodes MP-118. Once the Element Manager page is opened navigate to **IP Network → Nodes: Servers, Media Cards** select the appropriate **Node** (not shown)**,** on the **Node Details** page select **Gateway (SIP Gw & H323 Gw)** (not shown) and scroll to **SIP URI Map:** and enter the following:

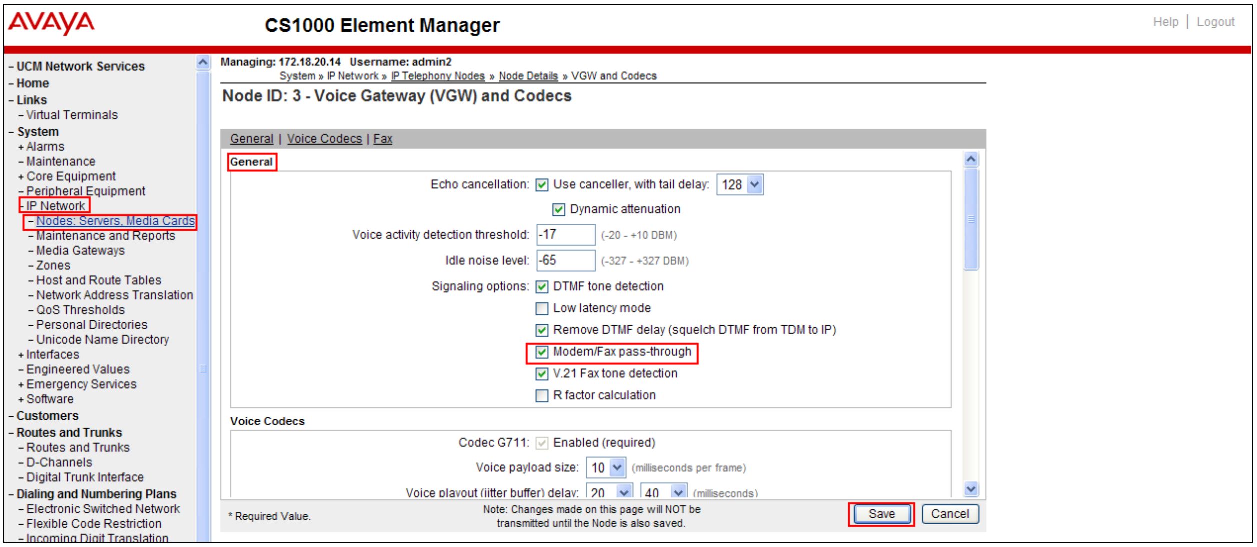
- **Unknown** Enter **cdp.udp**

Click on the **Save** button.



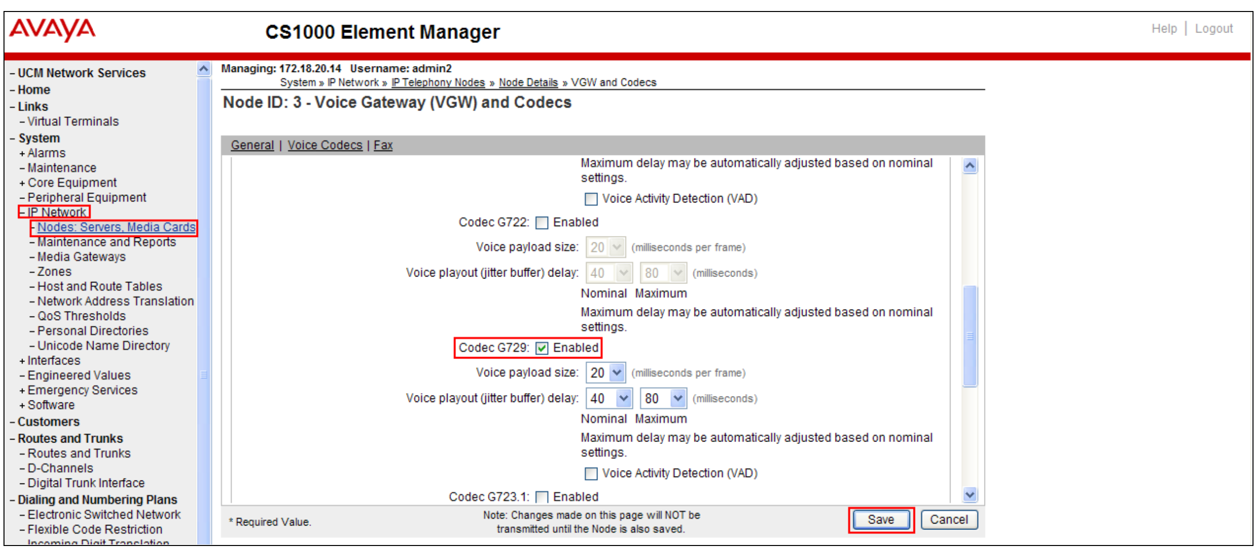
5.5.2. Configure Modem/Fax pass-through

During compliance testing Modem/Fax pass-through was tested. Once the Element Manager page is opened navigate to **IP Network → Nodes: Servers, Media Cards** select the appropriate **Node** (not shown), on the **Node Details** page select **Voice Gateway (VGW) and Codecs** (not shown)**,** in **General** settings check the **Modem/Fax pass-through** check box and click on the **Save** button.



5.5.3. Configure Voice Codec G.729

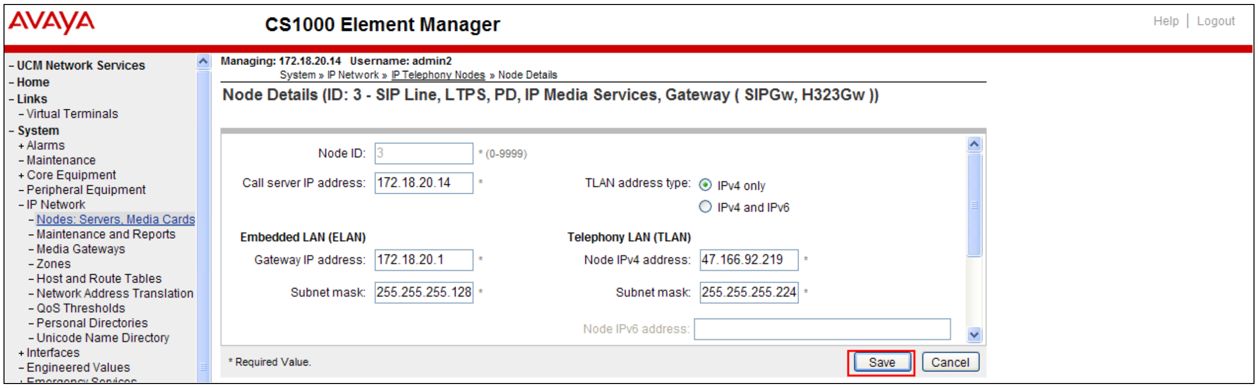
During compliance testing the voice codec G.729 was tested. Once the Element Manager page is opened navigate to **IP Network → Nodes: Servers, Media Cards** select the appropriate **Node** (not shown), on the **Node Details** page select **Voice Gateway (VGW) and Codecs** (not shown). Scroll to **Voice Codecs** and check the **Codec G.729** check box and click on the **Save** button.



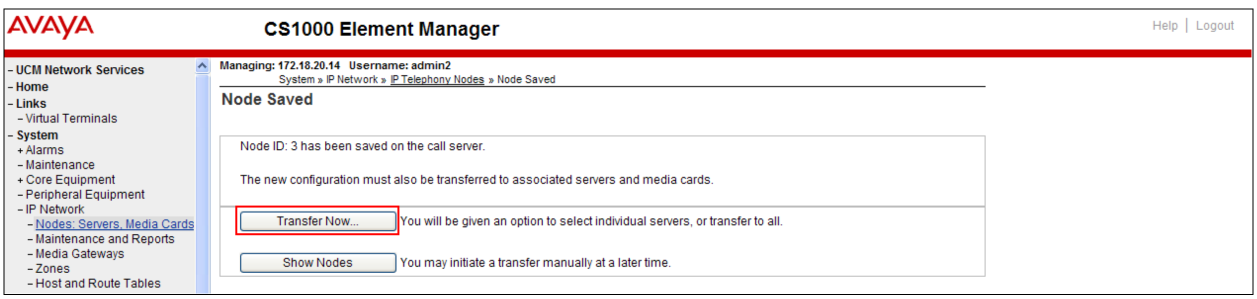
5.5.4. Transmitting Node Changes

Configurations made on the Element Manager must be transmitted to the Node before the changes take effect. To transmit the Node changes the following step is required:

- On the **Node Details** page click on the **Save** button



Once the following page opens click on the **Transfer Now** button.



6. Configure AudioCodes MP-118

During compliance testing the AudioCodes MP-118 was configured with Gateway Registration Both G.711 and G.729 codecs were tested as well as Fax using T.38 and G.711 Pass through. In Normal Mode the MP-118 was connected to the NRS using a SIP Trunk. In Fallback mode when the Ethernet connection was broken communication between the AudioCodes MP-118 and the CS1000E was with an Anolog connection from a FXO port on the Audiocodes MP-118 and unit on a FALC on the CS1000E.The following sections describe the configuration for all these scenarios. It is implied a working system is already in place. These configurations can be summarised as follows:

- Logging into the AudioCodes MP-118
- Configure IP Settings
- Configure Internal DNS Table
- Configure Voice Settings
- Configure Fax/Modem/CID Settings (T.38)
- Configure Fax/Modem/CID Settings (G.711 Pass Through)
- Configure Application Enabling
- Configure IP Group Table
- Configure Proxy Sets Table (Proxy Set ID 0)
- Configure Proxy Sets Table (Proxy Set ID 1)
- Configure Proxy & Registration
- Configure Coders
- Configure Profile for Fax
- Configure Endpoint Phone numbers
- Configure Hunt group Settings
- Configure Tel to IP Routing
- Configure IP to Trunk Group Routing
- Configure FXO Settings
- Resetting the Gateway
- Saving all Configurations to Flash Memory

Note: During compliance testing a standard WEB browser was used for the complete configuration of the AudioCodesMP-118. Some pre-configuration can be done by modifying a Configuration File which can be loaded on to the MP-118. An example of the Configuration File used is in **Appendix C**.

6.1. Logging into the AudioCodes MP-118

Enter the IP address of the AudioCodes MP-118 into a web browser. At first time log in enter the appropriate credentials and click on the **OK** button.

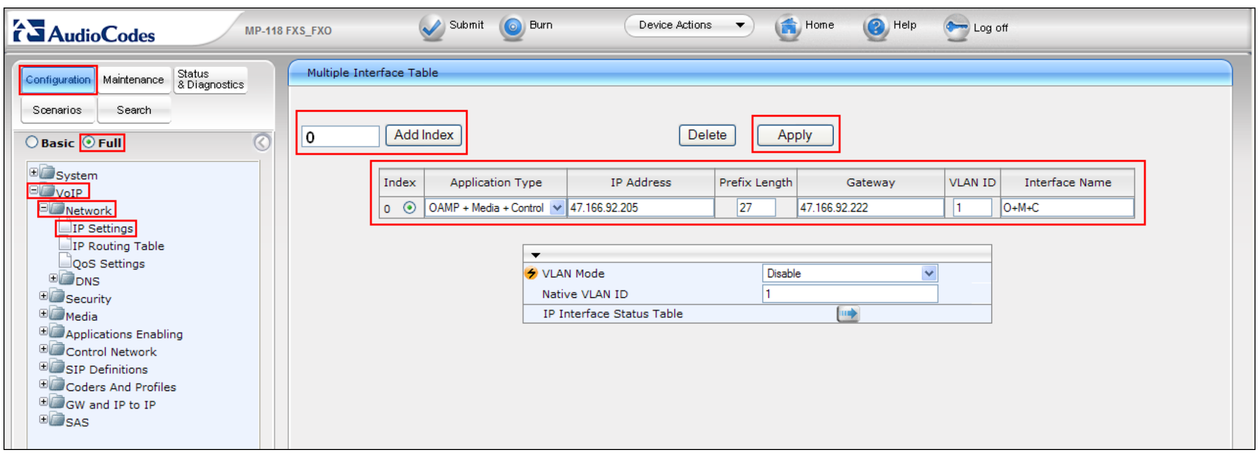
6.2. Configure IP Settings

Once the web page opens click on the **Configuration** button and check the **Full** radio button and navigate to **VoIP → Network → IP Settings**. The following steps are required:

- Enter **0** in the Add Index box and click the **Add Index button**
- In the Application Type drop down box select **OAMP+Media+Control**
- Enter the **IP address** assigned to the AudioCodes MP-118. i.e., **47.166.92.205**
- Enter **27** for **Prefix Length** Number of IP addresses available in subnet, supplied by site IT administrator
- Enter the IP address of the AudioCodes MP-118 **Gateway**. i.e., **47.166.92.222**
- Enter **1** for **VLAN ID**
- Enter **O+M+C** for the **Interface Name**

Click on the **Apply** button to save.

Note: Once the configuration is saved a **Gateway Reset** is required. See **Section 6.19**.

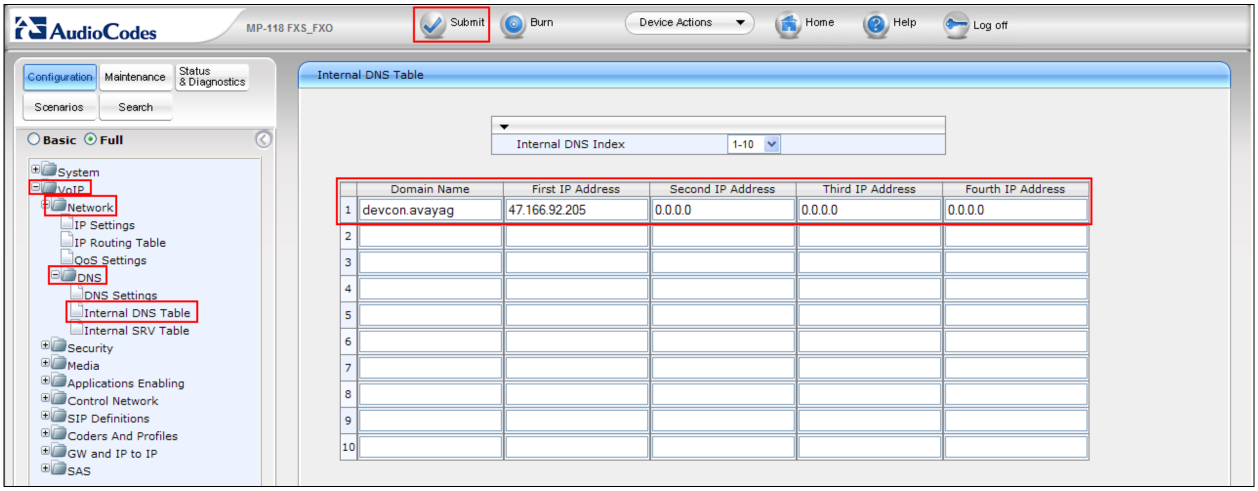


Configure Internal DNS Table

Navigate to **VoIP → Network →DNS→ Internal DNS Table**. The following steps are required:

- Enter the Domain of the NRS in **Domain Name** field. i.e., **devcon.avayag**
- Enter the IP address assigned to the AudioCodes MP-118 in the **First IP Address** field. i.e., **47.166.92.205**. The other IP Addresses can remain as **0.0.0.0**

Click on the **Submit** button to save.

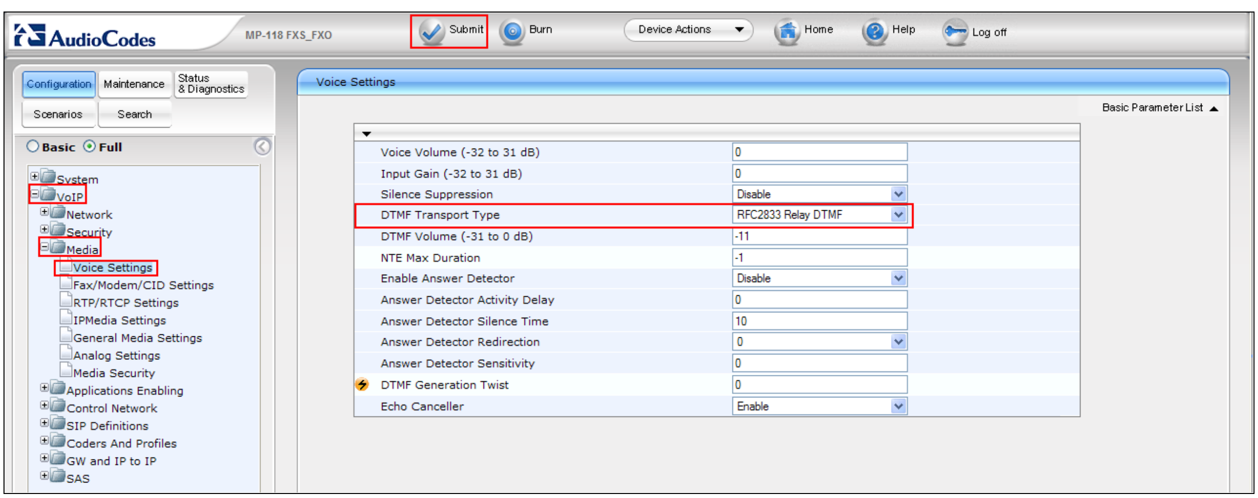


6.3. Configure Voice Settings

Navigate to **VoIP → Media → Voice Setting**. The following steps are required:

- From the **DTMF Transport Type** drop down box select **RFC2833 Relay DTMF**

Click on the **Submit** button to save.

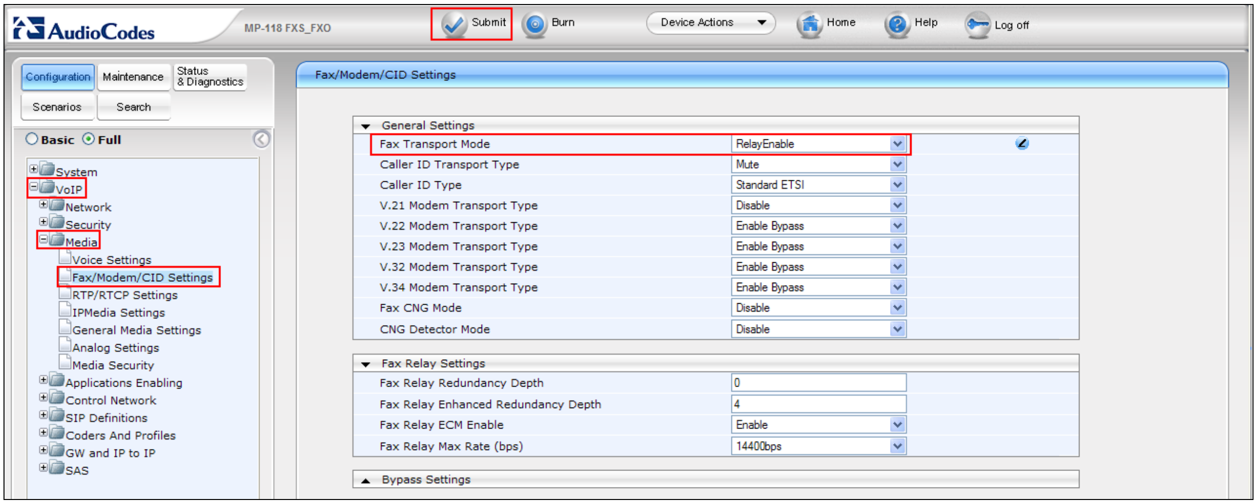


6.4. Configure Fax/Modem/CID Settings (T.38)

During compliance testing the T.38 Fax configuration was as follows: Navigate to **VoIP → Media → Fax/Modem/CID Settings**. The following steps are required:

- From the **Fax Transport Mode** drop down box in the **General Settings** window select **RelayEnable**

Click on the **Submit** button to save.

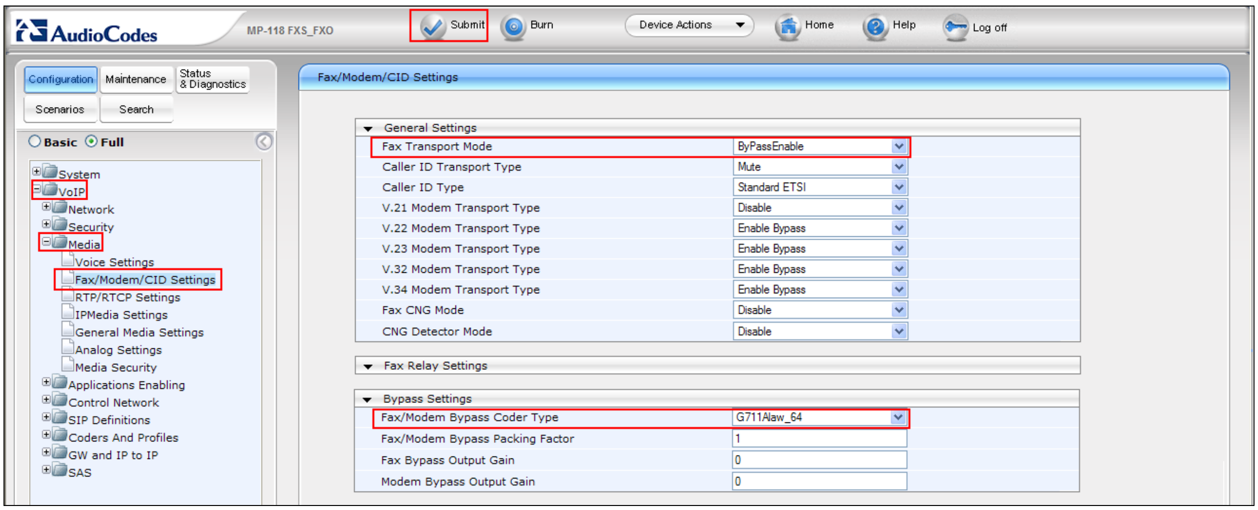


6.5. Configure Fax/Modem/CID Settings (G.711 Pass Through)

During compliance testing the G.711 Pass Through Fax configuration was as follows: Navigate to **VoIP → Media → Fax/Modem/CID Settings**. The following steps are required:

- From the **Fax Transport Mode** drop down box in the **General Settings** window select **ByPassEnable**
- From the **Fax/Modem Bypass Coder Type** drop down box in the **Bypass Settings** window select **G.711Alaw_64**

Click on the **Submit** button to save.

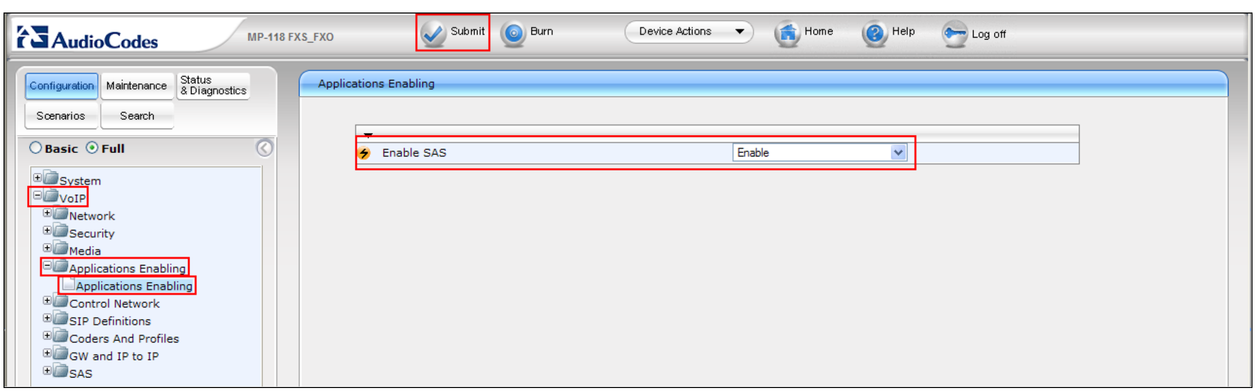


6.6. Configure Application Enabling

During compliance test the Stand Alone Survivability (SAS) feature was enabled. Navigate to **VoIP →Applications Enabling→ Applications Enabling**. From the **Enable SAS** dropdown box select **Enable**.

Click on the **Submit** button to save.

Note: Once the configuration is saved a **Gateway Reset** is required. See **Section 6.19**.

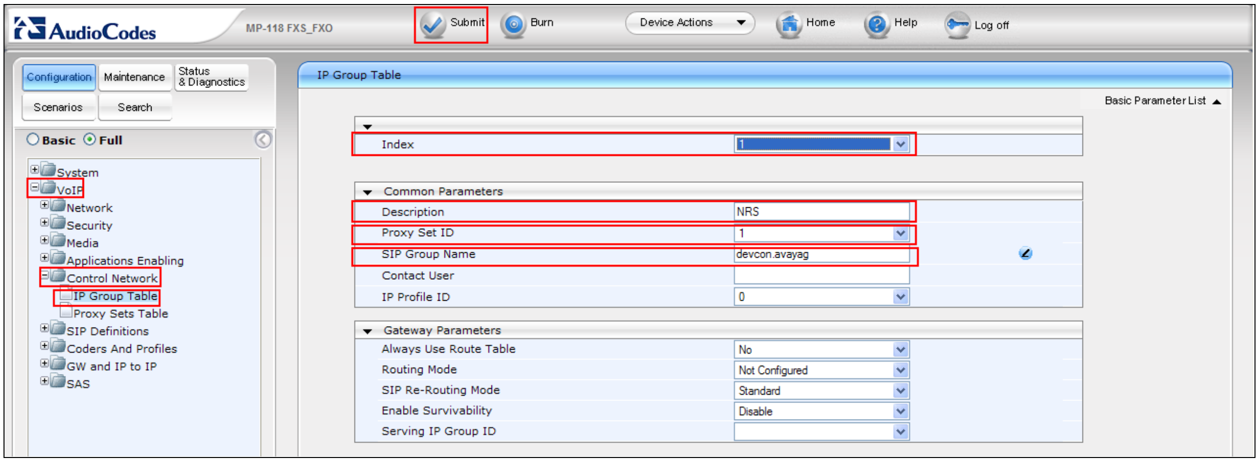


6.7. Configure IP Group Table

Navigate to **VoIP → Control network → IP Group Table**. The following steps are required:

- Select a free index from the **Index** drop down box. i.e., **1**
- Enter a description in the **Description** field. i.e., **NRS**
- Select **1** from the **Proxy Set ID** drop down box
- Enter the Domain of the NRS in the **SIP Group Name** field. i.e., **devcon.avayag**

Click on the **Submit** button to save.



6.8. Configure Proxy Sets Table (Proxy Set ID 0)

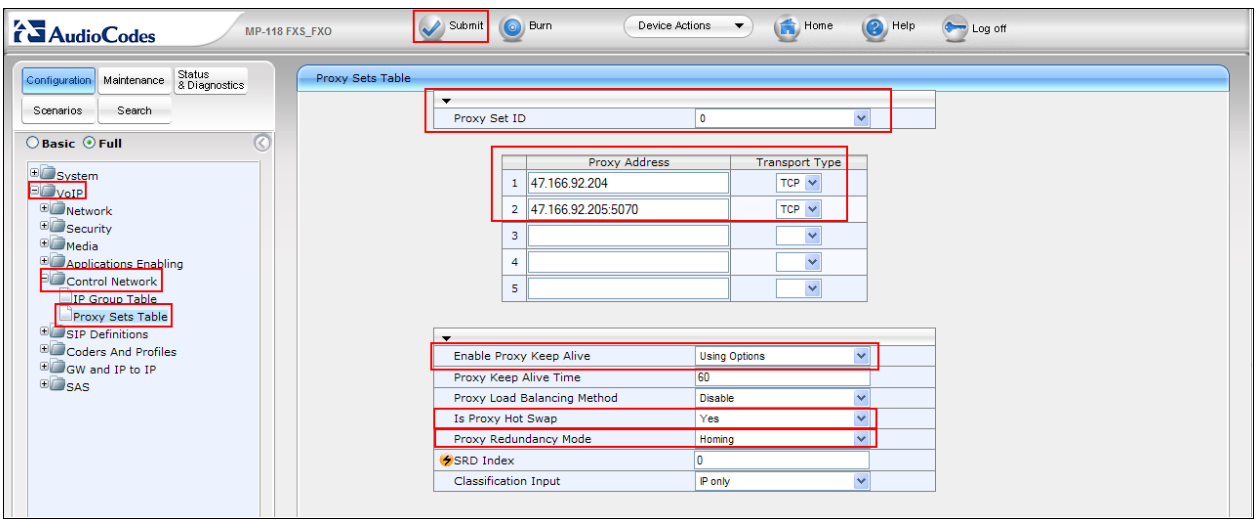
Proxy Set ID 0 was configured to allow for failure of the Ethernet connection. When the connection to the NRS (47.166.92.204) was lost then the Proxy used was of the AudioCodes MP-118 (47.166.92.205). Navigate to **VoIP → Control network → Proxy Sets Table**. From the **Proxy Set ID** drop down box select **0**.The table below shows the entries added during compliance testing:

	Proxy Address	Transport Type
1	47.166.92.204	TCP
2	47.166.92.205:5070	TCP

The following steps are also required:

- Select **Using Options** from the **Enable Proxy Keep Alive** drop down box
- Select **Yes** from the **Is Proxy Hot Swap** drop down box
- Select **Homing** from the **Proxy Redundancy Mode** drop down box

Click on the **Submit** button to save.

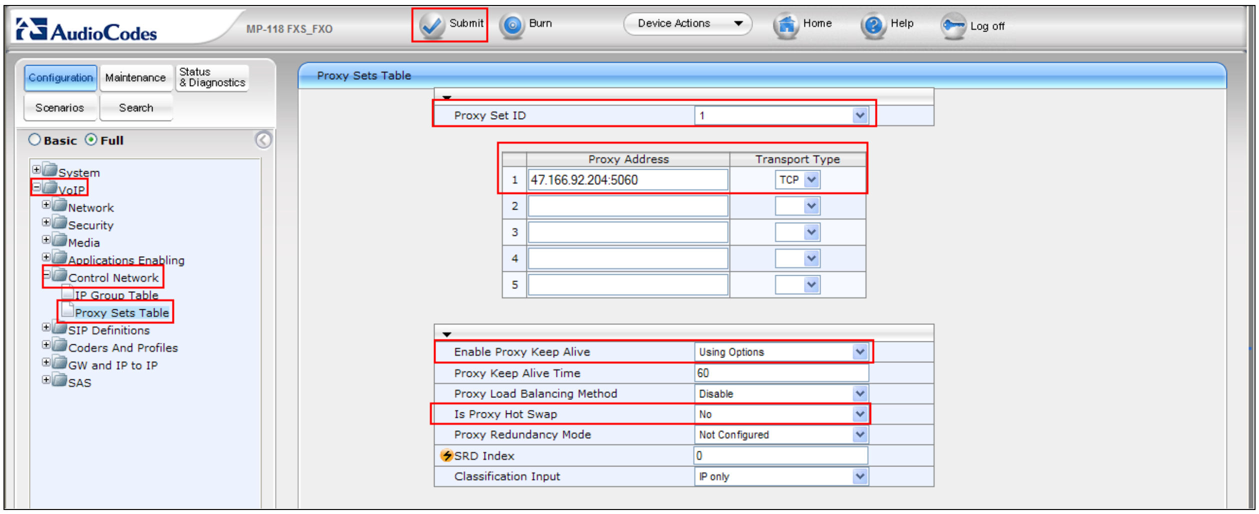


6.9. Configure Proxy Sets Table (Proxy Set ID 1)

Proxy Set ID 1 was configured for when the Ethernet connection to the NRS was available. From the **Proxy Set ID** drop down box select **1**.The following steps are required:

- Enter **47.166.92.204:5060** in the first **Proxy Address** field. This is the IP Address and Port of the NRS.
- Select **TCP** from the **Transport Type** drop down box
- Select **Using Options** from the **Enable Proxy Keep Alive** drop down box
- Select **No** from the **Is Proxy Hot Swap** drop down box

Click on the **Submit** button to save.

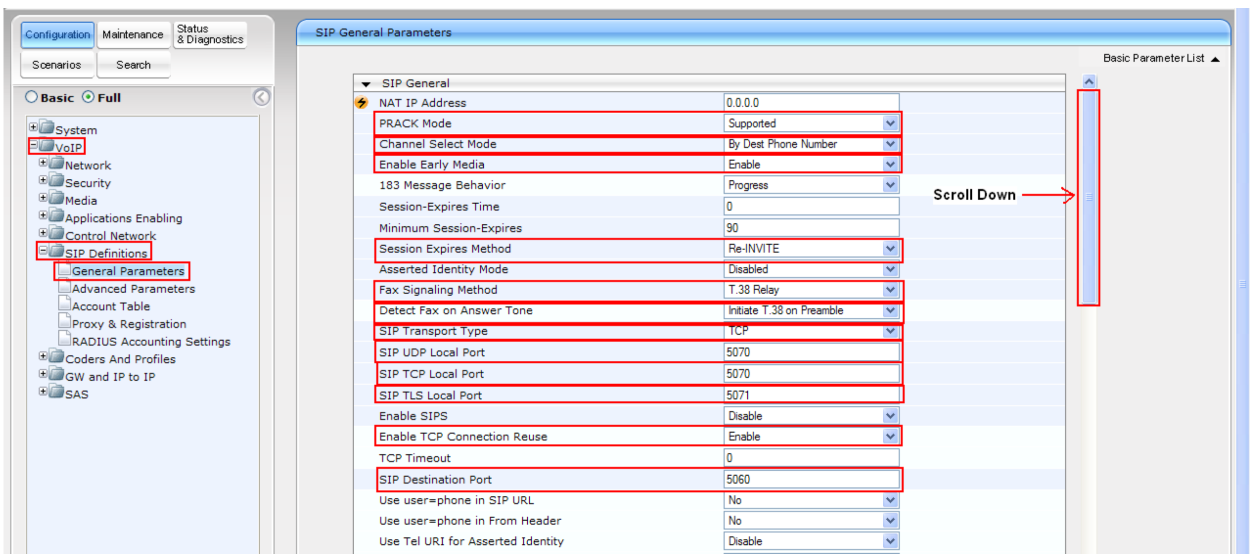


6.10. Configure General Parameters

Navigate to **VoIP → SIP Definitions → General Parameters**. The following steps are required:

- Select **Supported** from the **PRACK Mode** drop down box
- Select **By Dest Phone Number** from the **Channel Select Mode** drop down box
- Select **Enable** from the **Enable Early Media** drop down box
- Select **Re-INVITE** from the **Session-Expires Method** drop down box
- Select **G.711 Transport** or **T.38 Relay** from the **Fax Signaling Method** dropdown box depending on which signaling method is required.
- Select **Initiate T.38 on Preamble** from the **Detect Fax on Answer Tone** drop down box
- Select **TCP** from the **SIP Transport Type** drop down box
- Enter **5070** in the **SIP UDP Local Port** field
- Enter **5070** in the **SIP TCP Local Port** field
- Enter **5071** in the **SIP TLS Local Port** field
- Select **Enable** from the **Enable TCP Connection rescue** drop down box
- Enter **5060** in the **SIP Destination Port** field

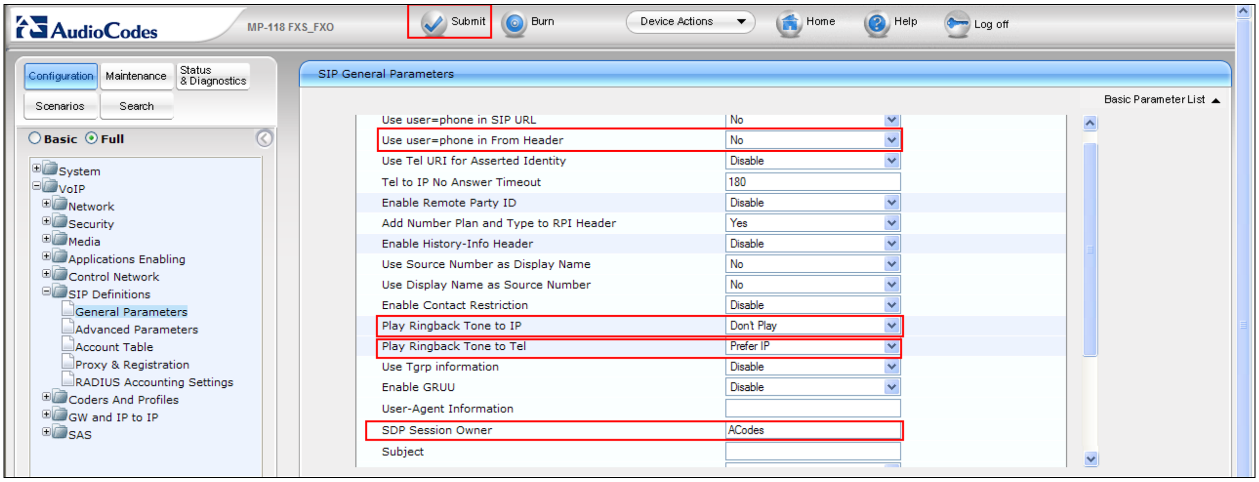
Scroll down using the scroll bar as shown in the screen shot.



Continue to fill in the following:

- Select **No** from the **Use user = phone in From Header** drop down box
- Select **Don't Play** from the **Play Ringback Tone to IP** drop down box
- Select **Prefer IP** from the **Play Ringback Tone to Tel** drop down box
- Enter **ACodes** in **SDP Session Owner** field

Click on the **Submit** button to save.

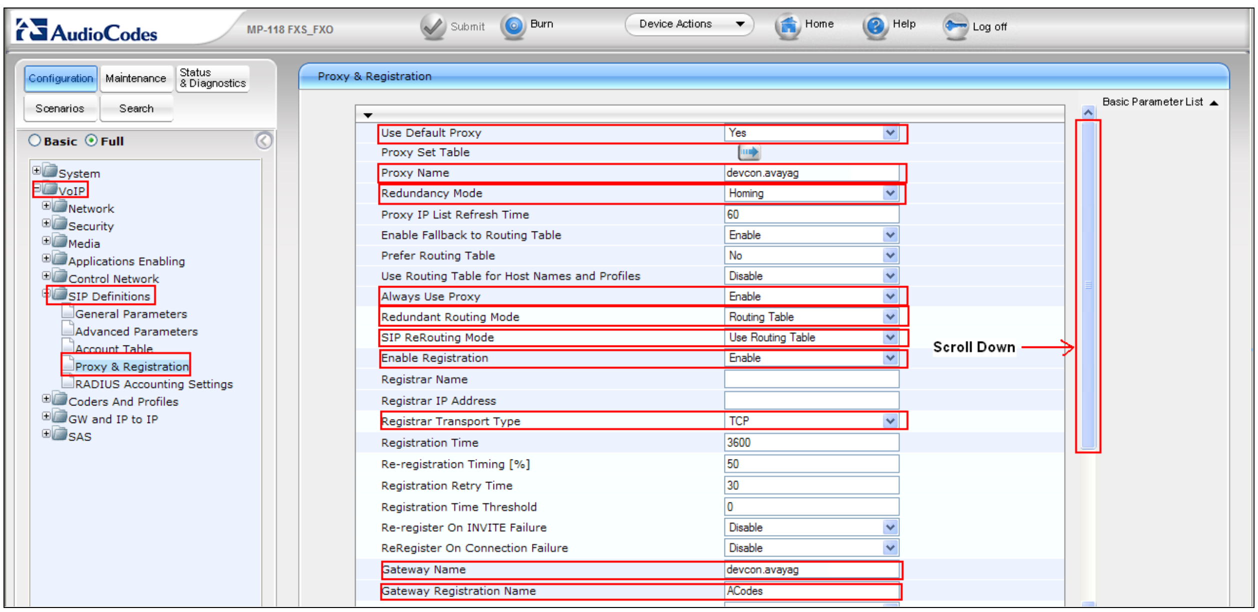


6.11. Configure Proxy & Registration

Navigate to **VoIP → SIP Definitions → Proxy & Registration**. The following steps are required:

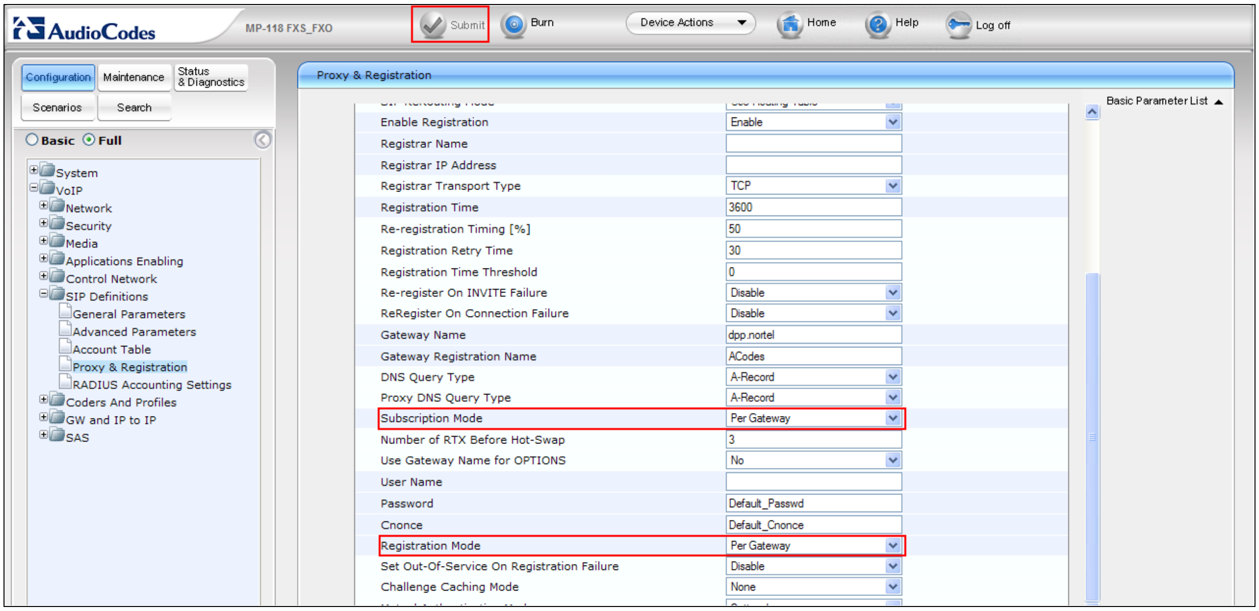
- Select a **Yes** from the **Use Default Proxy** drop down box
- Enter the Domain of the NRS in the **SIP Group Name** field. i.e., **devcon.avayag**
- Select **Homing** from the **Redundancy Mode** dropdown box
- Select **Enable** from the **Always Use Proxy** drop down box
- Select **Routing Table** from the **Redundant Routing Mode** drop down box
- Select **Use Routing Table** from the **SIP ReRouting Mode** drop down box
- Select **Enable** from the **Enable Registration** drop down box
- Select **TCP** from the **Rigistar Transport Type** drop down box
- Enter the Domain of the NRS in the **Gateway Name** field. i.e., **devcon.avayag**
- Enter **ACodes** in the **Gateway Registration Name** field

Scroll down using the scroll bar as shown in the screen shot.



- Select **Per Gateway** from the **Subscription Mode** type dropdown box
- Select **Per Gateway** from the **Registration Mode** type dropdown box
  Note: Per Endpoint is not supported on the NRS

Click on the **Submit** button to save.



6.12. Configure Coders

During compliance testing the both Codec G.711 and G.729 was used. The following section shows both configurations.

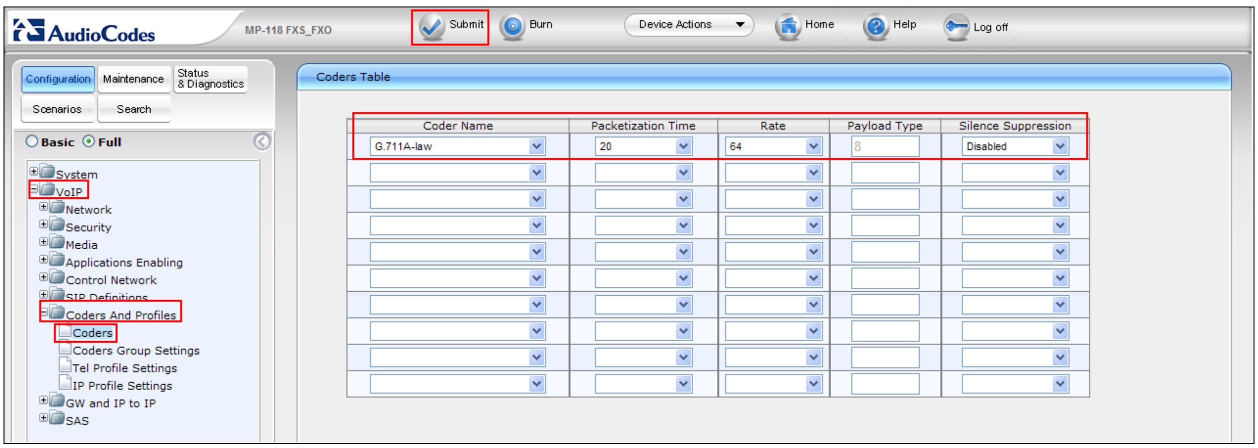
Note: Both Codecs were tested exclusively.

6.12.1. Configure G.711 Codec

Navigate to **VoIP → Coders and Profiles → Coders**. The following steps are required:

- Select **G.711A-law** from the **Coder Name** drop down box
- Select **20** from the **Packetization Time** drop down box
- Select **64** from the **Rate** drop down box
- Select **Disabled** from the **Silence Suppression** drop down box

Click on the **Submit** button to save.

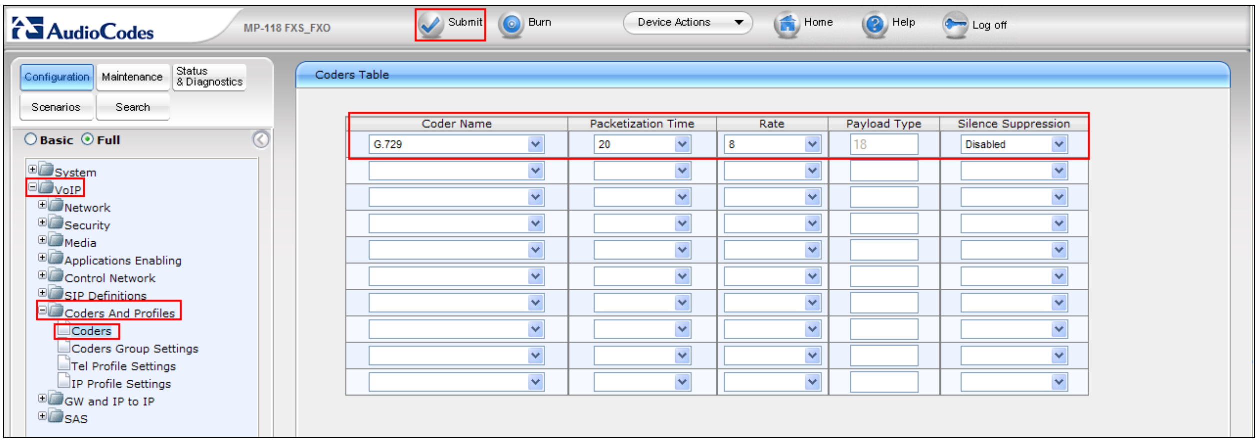


6.12.2. Configure G.729 Codec

Navigate to **VoIP → Coders and Profiles → Coders**. The following steps are required:

- Select **G.729** from the **Coder Name** drop down box
- Select **20** from the **Packetization Time** drop down box
- Select **8** from the **Rate** drop down box
- Select **Disabled** from the **Silence Suppression** drop down box

Click on the **Submit** button to save.



6.13. Configure Profile for Fax

During compliance testing both T.38 and G.711 Transport fax signaling methods were used. The following section shows both configurations.

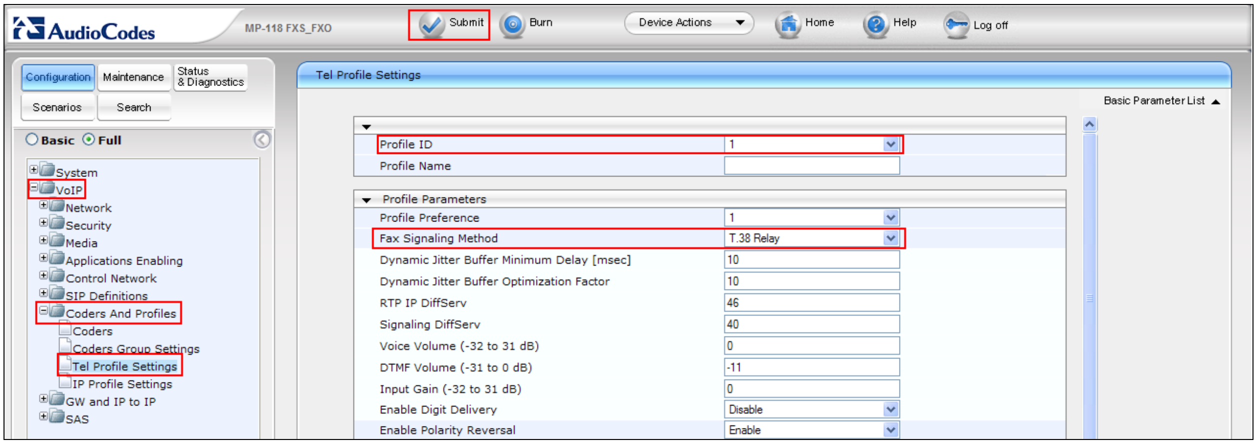
Note: Both fax signaling methods were tested exclusively.

6.13.1. Configure Tel Profile T.38 for Fax

Navigate to **VoIP → Coders and Profiles → Tel Profile Settings**. The following step is required:

- Select **1** from the **Profile ID** drop down box
- Select **T.38 Relay** from the **Fax signaling Method** drop down box

Click on the **Submit** button to save.

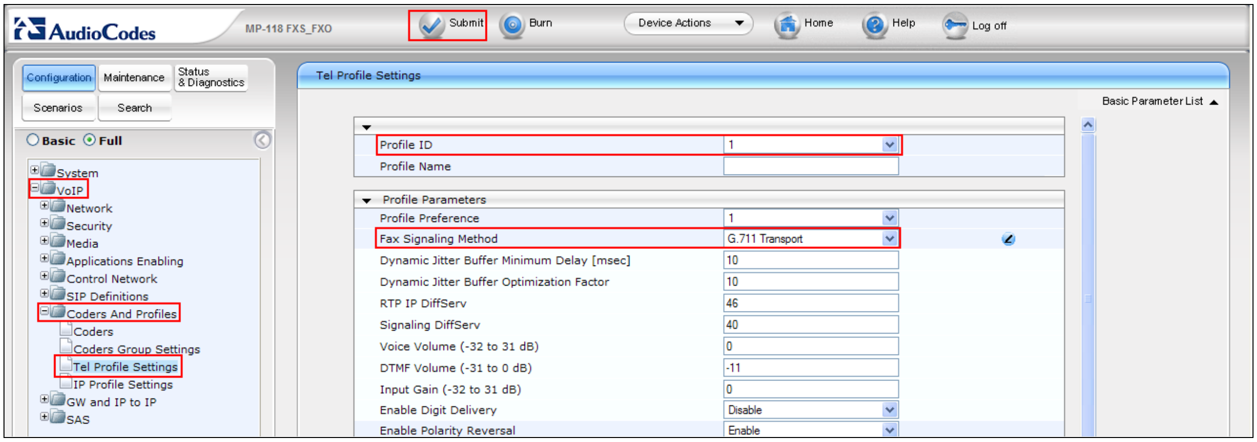


6.13.2. Configure Tel Profile G.711 Transport for Fax

Navigate to **VoIP → Coders and Profiles → Tel Profile Settings**. The following step is required:

- Select **1** from the **Profile ID** drop down box
- Select Tel Profile **G.711 Transport** from the **Fax signalling Method** drop down box

Click on the **Submit** button to save.

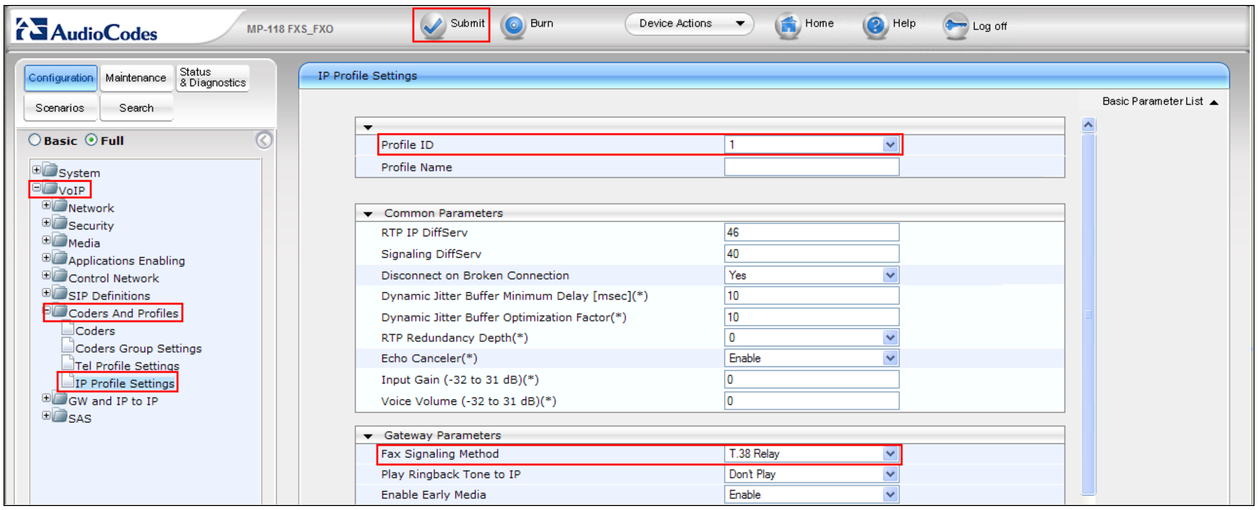


6.13.3. Configure IP Profile T.38 for Fax

Navigate to **VoIP → Coders and Profiles → IP Profile Settings**. The following step is required:

- Select **1** from the **Profile ID** drop down box
- Select **T.38 Relay** from the **Fax signalling Method** drop down box

Click on the **Submit** button to save.

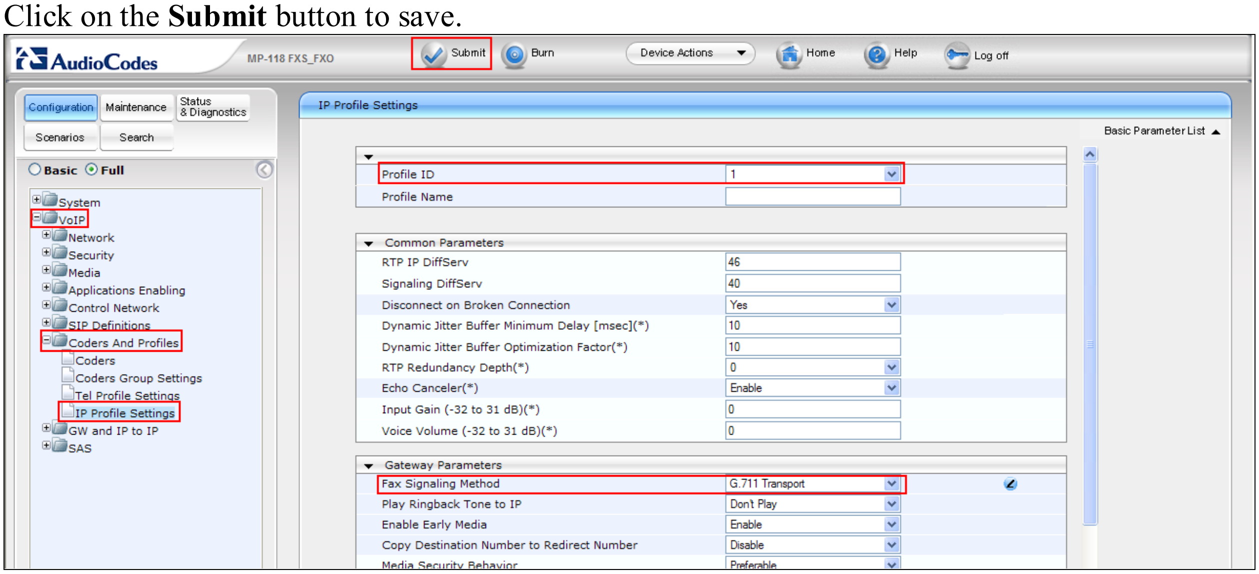


6.13.4. Configure IP Profile G.711 Transport for Fax

Navigate to **VoIP → Coders and Profiles → IP Profile Settings**. The following step is required:

- Select **1** from the **Profile ID** drop down box
- Select **G.711 Transport** from the **Fax signalling Method** drop down box

Click on the **Submit** button to save.



6.14. Configure Endpoint Phone numbers

During compliance testing 4 Phone numbers were used. Channels 1 and 2 were Telephones and Channel 3 was a Fax machine on the AudioCodes. Channel 5 is the extension on the FXO port used for Fallback. Navigate to **VoIP → GW and IP to IP → Hunt Group → Endpoint Phone Number**. The following steps are required:

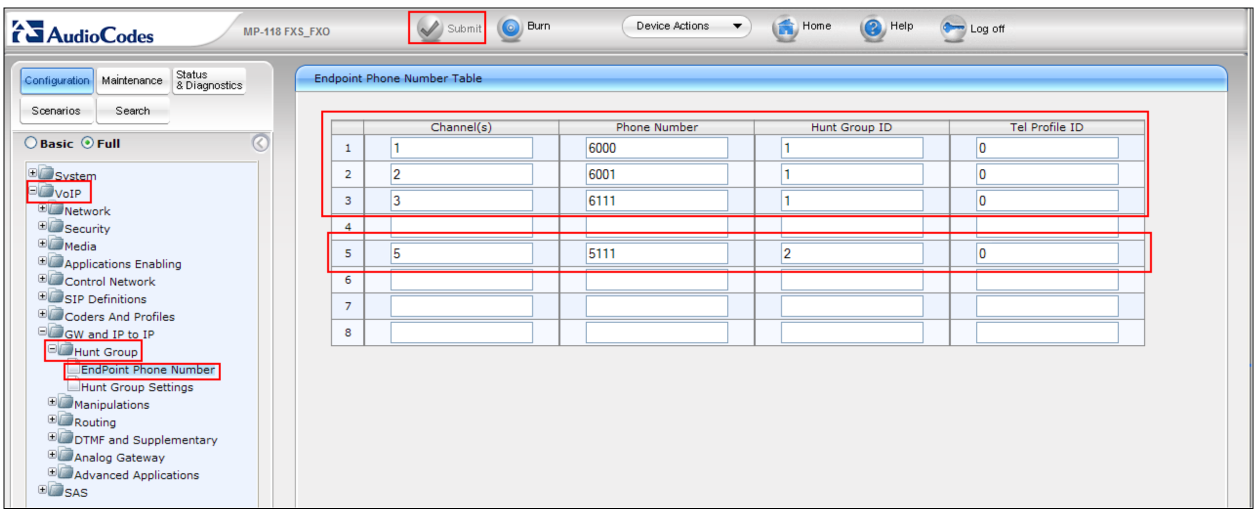
- Enter the values as per table below

The table below shows the entries added during compliance testing:

Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	6000	1	0
2	6001	1	0
3	6111	1	0
5	5111	2	0

Note: Channels 1-4 are FXS ports and channels 5-8 are FXO ports

Click on the **Submit** button to save.

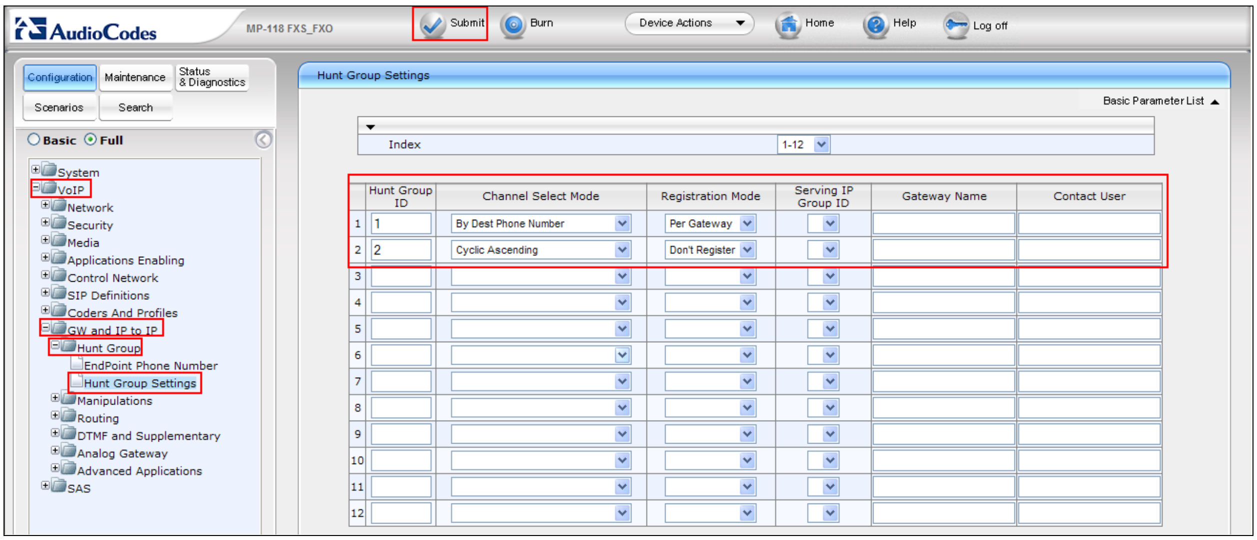


6.15. Configure Hunt group Settings

During compliance testing 2 Hunt Groups were configured. Hunt Group 1 was used for normal mode and Hunt Group 2 was used for fallback mode. Navigate to **VoIP → GW and IP to IP → Hunt Group → Hunt Group Settings**. The following steps are required:

- For **Hunt Group ID 1** select **By Dest Phone Number** from the **Channel Select mode** drop down box and select **Per Gateway** from the **Registration Mode** drop down box.
- For **Hunt Group ID 2** select **Cyclic Ascending** from the **Channel Select mode** drop down box and select **Don't Register** from the **Registration Mode** drop down box.

Click on the **Submit** button to save.

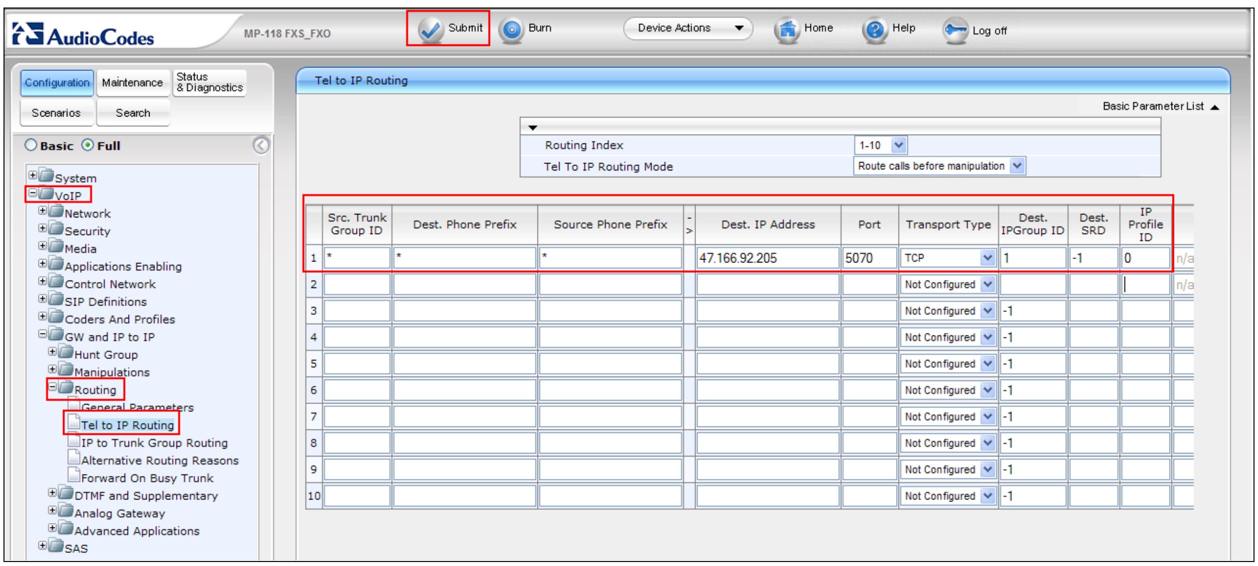


6.16. Configure Tel to IP Routing

Navigate to **VoIP → GW and IP to IP → Routing → Tel to IP Routing.** The following steps are required:

- Enter * in the **Src. Trunk Group ID** field
- Enter * in the **Dest. Phone Prefix** field
- Enter * in the **Source Phone Prefix** field
- Enter the IP address of the Audiocodes MP-118 in the **Dest. IP Address** field. i.e., **47.166.92.205**
- Enter **5070** in the **Port** field
- Select **TCP** from the **Transport** drop down box
- Enter **-1** in the **Dest. IPGroup ID** field
- Enter **-1** in the **Dest. SRD** field
- Enter **0** in the **IP Profile** field

Click on the **Submit** button to save.



6.17. Configure IP to Trunk Group Routing

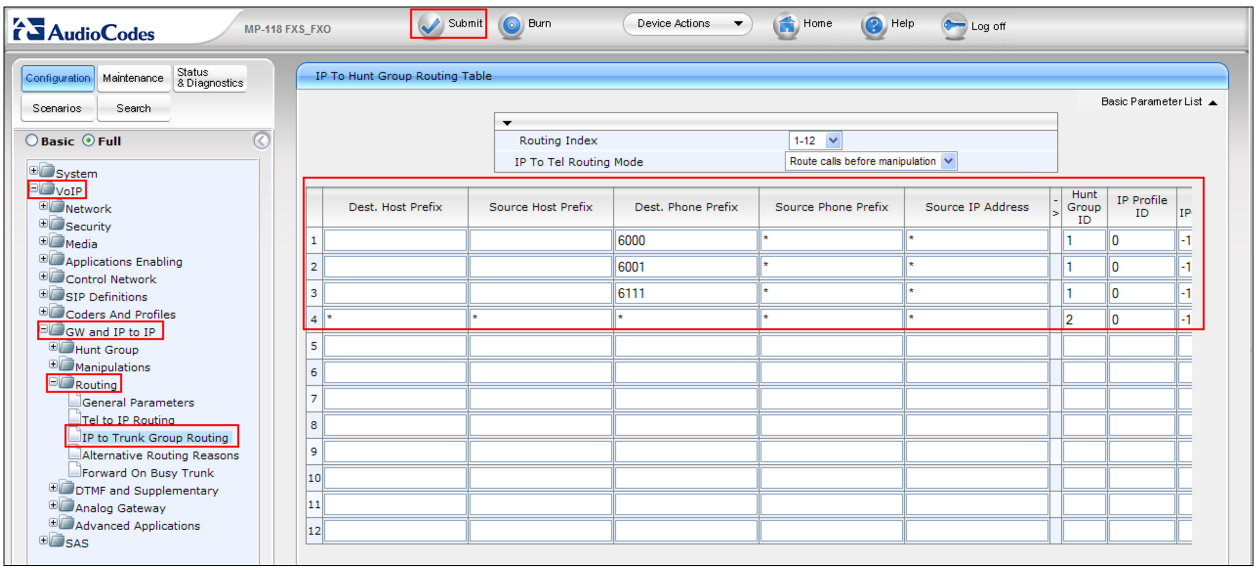
Navigate to **VoIP → GW and IP to IP → IP to Trunk Group Routing.** The following steps are required:

- Enter the values as per the table below

Note: where there are no values in a field leave blank.

Dest. Host Prefix	Dest. Host Prefix	Dest. Host Prefix	Dest. Host Prefix	Surce IP Address	Hunt Group ID	IP Profile ID	Source IPGroup ID
		6000	*	*	1	0	-1
		6001	*	*	1	0	-1
		6111	*	*	1	0	-1
*	*	*	*	*	2	0	-1

Click on the **Submit** button to save.

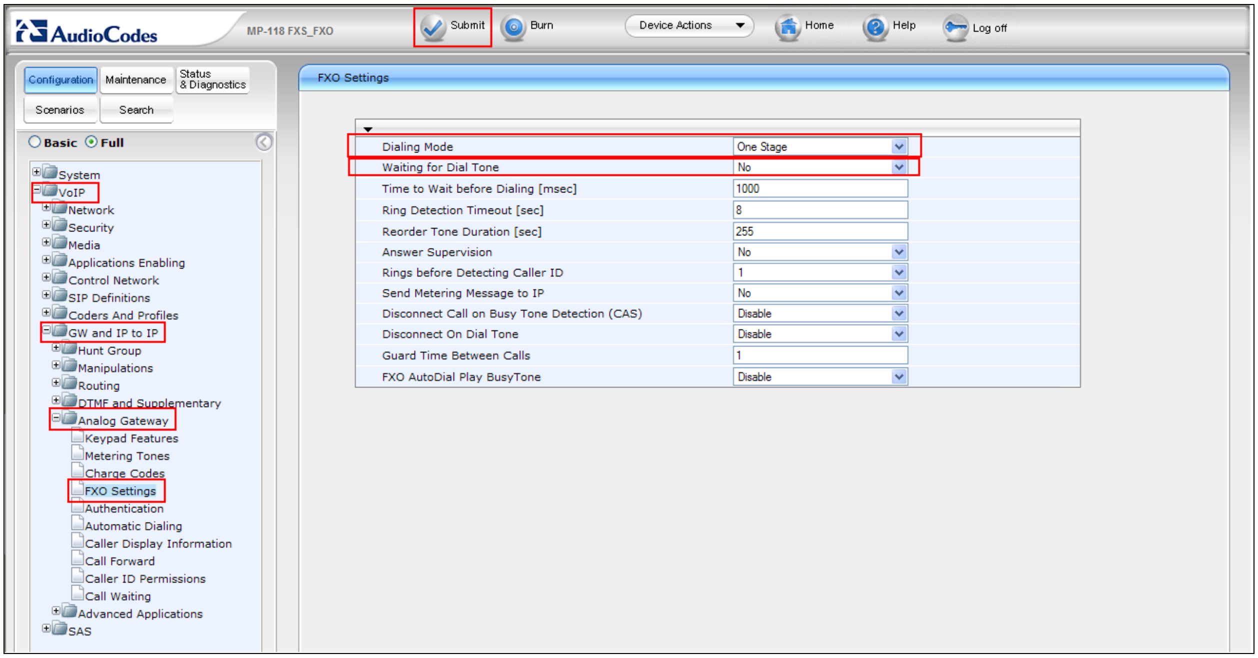


6.18. Configure FXO Settings

Navigate to **VoIP → GW and IP to IP → Analog Gateway → FXO Settings.** The following step is required:

- Select **One Stage** from the **Dialing Mode** drop down box
- Select **No** from the **Waiting for Dial Tone** drop down box

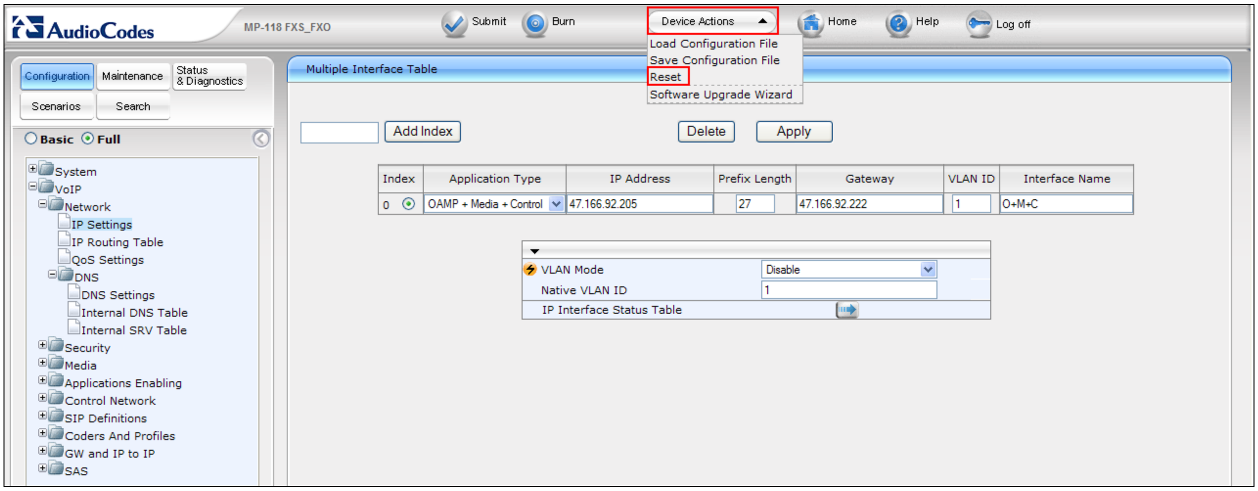
Click on the **Submit** button to save.



6.19. Resetting the Gateway

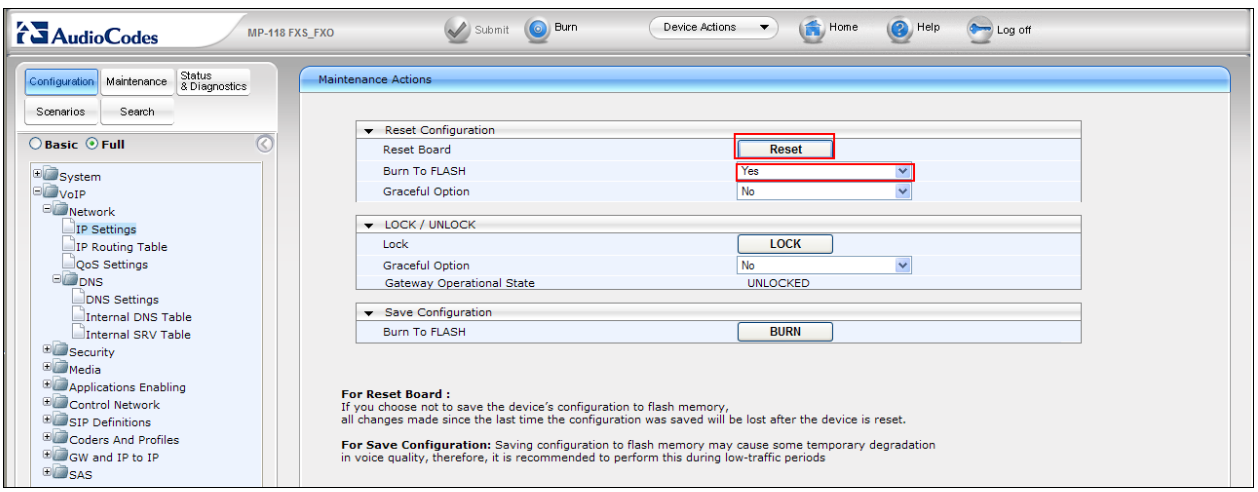
After configuring **IP Settings** (**Section 6.2**) and **Application Enabling** (**section 6.6**) a Gateway Reset needs to be preformed. The following steps are required:

- Click on the **Device Actions** drop down box
- Select **Reset**



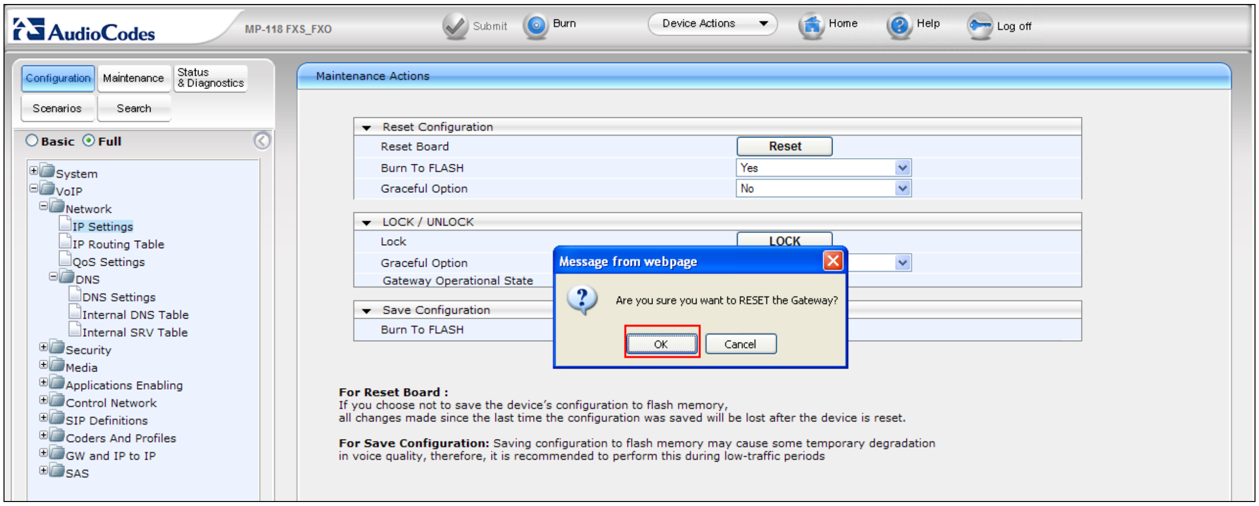
Once the next page opens the following steps are required:

- Select **Yes** from the **Burn To FLASH** drop down button
- Click on the **Reset** button



Once the **Message from webpage** window opens click the **OK** button.

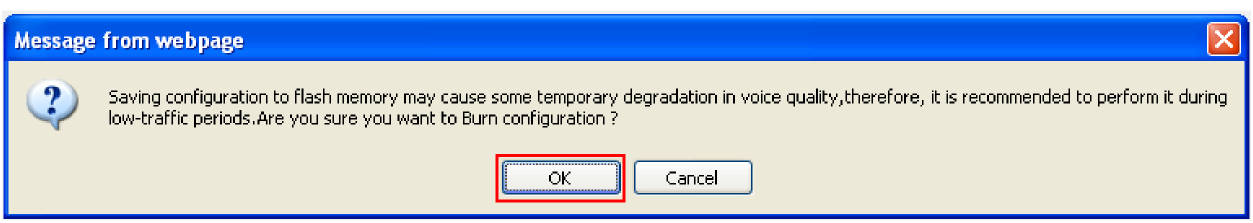
Note: It will take up to 60 seconds for the Gateway to reset.



6.20. Saving all Configurations to Flash Memory

To save the configuration changes to Flash Memory click on the **Burn** button.

When the **Message from webpage** appears click on the **OK** button



7. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the CS1000E and the AudioCodes MP-118

7.1. D-channel Status

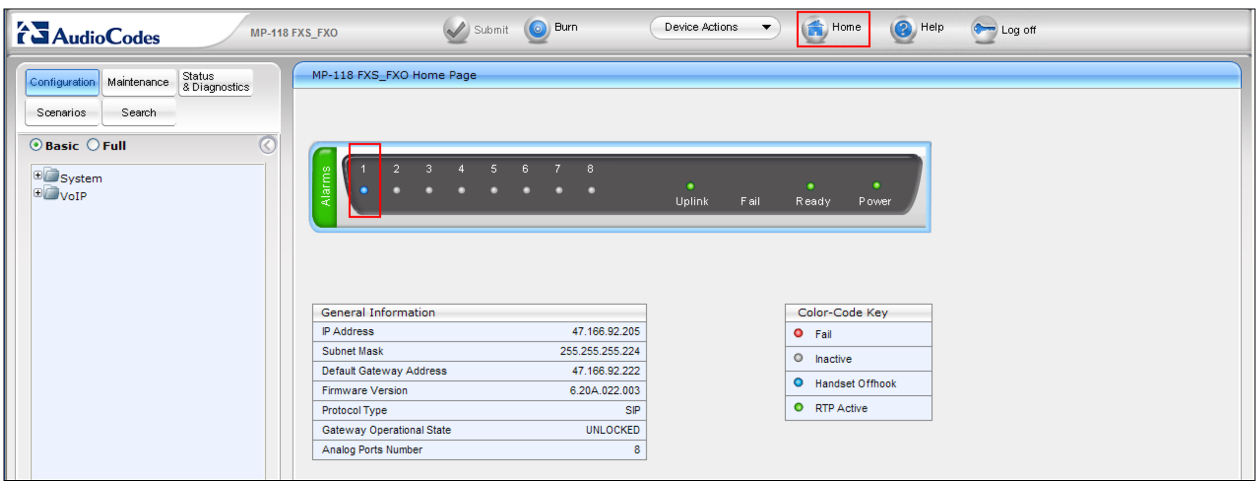
In Normal Mode with the Ethernet link connected check the status of the D-channel created in **Section 5.1.1** by running the command **STAT DCH** in Overlay 96 (**LD 96**) as shown below. The example below shows that D-Channel 66 is operational and established.

LD 96

Prompt	Response	Description
>LD 96		Enter Overlay 96
STAT DCH 66		Check status of D-Channels
DCH 066	OPER EST	

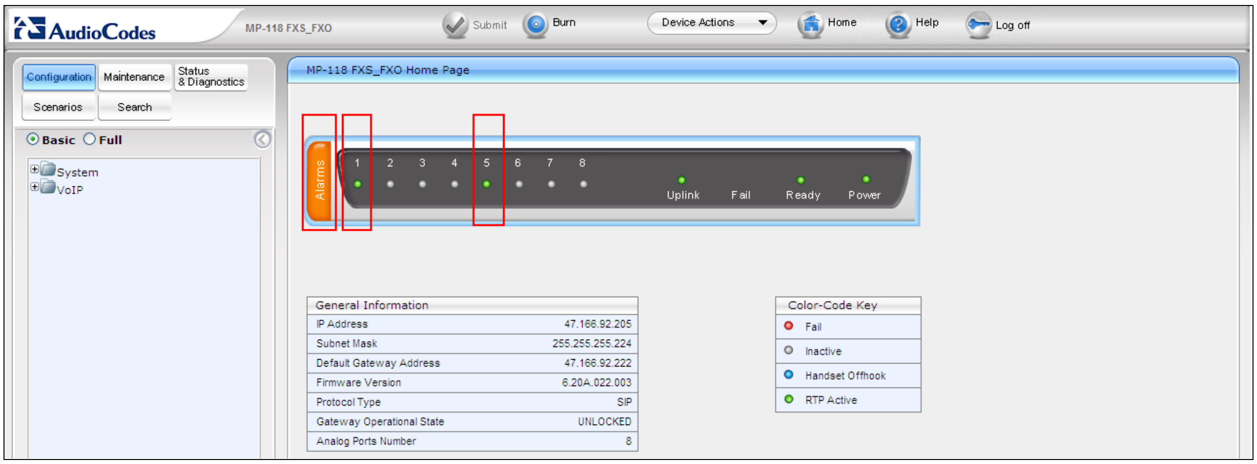
7.2. AudioCodes MP-118 Normal Mode Status

In Normal Mode with the Ethernet link connected make a call from a telephone on the AudioCodes MP-118 to an extension on the CS1000E, the LED of the FXS port which the telephone is configured should be illuminated. In the example below the telephone 6000 which is configured on FXS port 1, calls the extension on the CS1000E, FXS port 1 is illuminated.



7.3. AudioCodes MP-118 Fallback Mode Status

In Fallback Mode with the Ethernet link connected make a call from a telephone on the AudioCodes MP-118 to an extension on the CS1000E, the LED's of the FXS port which the telephone is configured and the FXO port the Fallback connection is configured should be illuminated In the example below the telephone 6000 which is configured on FXS port 1 calls an extension on the CS1000E, FXO port 5 is configured for Fallback. Both FXS port 1 and FXO port 5 is illuminated. Note also that the Alarms indicator is orange to highlight that the Ethernet connection is down.



8. Conclusion

These Application Notes describe the configuration steps required for Avaya Communication Server 1000E R7.5 to successfully interoperate with AudioCodes MediaPack 118 R6.20A. AudioCodes MediaPack 118 R6.20A is considered compliant with Avaya Communication Server 1000E R7.5 All test cases have passed and met the objectives outlined in **Section 2.1.**

9. Additional References

This section references the Avaya and AudioCodes documentation that is relevant to these Application Notes. Product documentation for Avaya products may be found at http://support.avaya.com.

[1] Software Input Output Reference — Administration Avaya Communication Server 1000 7.5, NN43001-611, 05.09 September 2011

[2] Network Routing Service Fundamentals Avaya Communication Server 1000 7.5, NN43001-130, 03.10 September 2011

Software and technical documentation for at the AudioCodes can be found at www.audiocodes.com

Login is required to obtain software and technical product documentation. If no login privileges are assigned they may be requested from the website.

Appendix A: Avaya Communication Server 1000E Software

```
              Avaya Communication Server 1000E call server deplists
VERSION 4121
RELEASE 7
ISSUE 50 Q +
DepList 1: core Issue: 01 (created: 2011-03-15 10:26:33 (est))
IN-SERVICE PEPS
PAT# CR #            PATCH REF #     NAME      DATE        FILENAME
SPECINS
000  wi00688505     ISS1:1OF1     p30595_1  14/06/2011  p30595_1.cpl   NO
001  wi00835294     ISS1:1OF1     p30565_1  14/06/2011  p30565_1.cpl   NO
002  wi00832106     ISS1:1OF1     p30550_1  14/06/2011  p30550_1.cpl   NO
003  wi00837618     ISS1:1OF1     p30594_1  14/06/2011  p30594_1.cpl   NO
004  wi00852365     ISS1:1OF1     p30707_1  14/06/2011  p30707_1.cpl   NO
005  wi00843623     ISS1:1OF1     p30731_1  14/06/2011  p30731_1.cpl   YES
006  wi00839255     ISS1:1OF1     p30591_1  14/06/2011  p30591_1.cpl   NO
007  wi00832626     ISS2:1OF1     p30560_2  14/06/2011  p30560_2.cpl   NO
008  wi00857566     ISS1:1OF1     p30766_1  14/06/2011  p30766_1.cpl   NO
009  wi00841980     ISS1:1OF1     p30618_1  14/06/2011  p30618_1.cpl   NO
010  wi00837461     ISS1:1OF1     p30597_1  14/06/2011  p30597_1.cpl   NO
011  wi00839821     ISS1:1OF1     p30619_1  14/06/2011  p30619_1.cpl   NO
012  wi00842409     ISS1:1OF1     p30621_1  14/06/2011  p30621_1.cpl   NO
013  wi00838073     ISS1:1OF1     p30588_1  14/06/2011  p30588_1.cpl   NO
014  wi00850521     ISS1:1OF1     p30709_1  14/06/2011  p30709_1.cpl   YES
015  wi00860722     ISS1:1OF1     p30784_1  14/06/2011  p30784_1.cpl   YES
016  wi00839134     ISS1:1OF1     p30698_1  14/06/2011  p30698_1.cpl   YES
017  wi00836981     ISS1:1OF1     p30613_1  14/06/2011  p30613_1.cpl   NO
```

```
   Avaya Communication Server 1000E Peripheral Software Version (PSWV) data
PSWV  VERSION: PSWV 100
LCRI:       VERSION NUMBER: AA02
XNET:       VERSION NUMBER: AC23
XPEC:       VERSION NUMBER: AC43
FNET:       VERSION NUMBER: AA07
FPEC:       VERSION NUMBER: AA08
MSDL:       VERSION NUMBER: AJ73
SDI:        VERSION NUMBER: AH51
DCH:        VERSION NUMBER: AA72
AML:        VERSION NUMBER: AK81
BRIL:       VERSION NUMBER: AK83
BRIT:       VERSION NUMBER: AK82
MISP:       VERSION NUMBER: AJ71
MPH:        VERSION NUMBER: AH51
BRSC:       VERSION NUMBER: AJ71
BBRI:       VERSION NUMBER: AH54
PRIE:       VERSION NUMBER: AA87
BRIE:       VERSION NUMBER: AK89
ISIG:       VERSION NUMBER: AA33
SWE1:       VERSION NUMBER: BA53
UKG1:       VERSION NUMBER: BA51
AUS1:       VERSION NUMBER: BA49
DEN1:       VERSION NUMBER: BA48
FIN1:       VERSION NUMBER: BA49
```

```
GER1:        VERSION NUMBER: BA54
ITA1:        VERSION NUMBER: AA54
NOR1:        VERSION NUMBER: BA49
POR1:        VERSION NUMBER: BA49
DUT1:        VERSION NUMBER: BA50
EIR1:        VERSION NUMBER: BA49
SWI1:        VERSION NUMBER: BA53
BEL1:        VERSION NUMBER: BA49
SPA1:        VERSION NUMBER: BA51
NET1:        VERSION NUMBER: BA48
FRA1:        VERSION NUMBER: BA52
CIS1:        VERSION NUMBER: BA48
ETSI:        VERSION NUMBER: BA48
E403:        VERSION NUMBER: BA07
N403:        VERSION NUMBER: BA05
JTTC:        VERSION NUMBER: AC08
TCNZ:        VERSION NUMBER: AA13
AUBR:        VERSION NUMBER: AA14
AUPR:        VERSION NUMBER: AA04
HKBR:        VERSION NUMBER: AA06
HKPR:        VERSION NUMBER: AA08
SING:        VERSION NUMBER: AA15
THAI:        VERSION NUMBER: AA07
NI02:        VERSION NUMBER: AA26
T1IS:        VERSION NUMBER: AA10
T1ES:        VERSION NUMBER: AA09
ESGF:        VERSION NUMBER: AC30
ISGF:        VERSION NUMBER: AC31
ESGFTI:      VERSION NUMBER: AC29
ISGFTI:      VERSION NUMBER: AC31
INDO:        VERSION NUMBER: AA06
JAPN:        VERSION NUMBER: AA16
MSIA:        VERSION NUMBER: AA04
CHNA:        VERSION NUMBER: AA04
INDI:        VERSION NUMBER: AA03
PHLP:        VERSION NUMBER: AA02
TAIW:        VERSION NUMBER: AA03
EAUS:        VERSION NUMBER: AA02
EGF4:        VERSION NUMBER: AC14
DCH3:        VERSION NUMBER: AA10
PUP3:        VERSION NUMBER: AA14
T1E1:        VERSION NUMBER: AA19
DITI:        VERSION NUMBER: AA40
CLKC:        VERSION NUMBER: AA20
3902:        VERSION NUMBER: AA84
3903:        VERSION NUMBER: AA91
3904:        VERSION NUMBER: AA94
3905:        VERSION NUMBER: AA94
MGC, MGX and MGS:  CSP  VERSION: MGCC CD01
  MSP  VERSION: MGCM AB01
  APP  VERSION: MGCA BA07
  FPGA VERSION: MGCF AA18
  BOOT VERSION: MGCB BA07
  DSP1 VERSION: DSP1 AB03
  DSP2 VERSION: DSP2 AB03
  DSP3 VERSION: DSP3 AB03
```

```
  DSP4 VERSION: DSP4 AB01
  DSP5 VERSION: DSP5 AA01
UDT VERSION NUMBER: AA42
```

Appendix B: Avaya Network Routing Service Software

```
                    Avaya Network Routing Service Software
Product Release: 7.50.17.00
Base Applications
   base                         7.50.17     [patched]
   NTAFS                        7.50.17
   sm                           7.50.17
   cs1000-Auth                  7.50.17
   Jboss-Quantum                7.50.17     [patched]
   lhmonitor                    7.50.17
   baseAppUtils                 7.50.17
   dfoTools                     7.50.17
   nnnm                         7.50.17
   cppmUtil                     7.50.17
   oam-logging                  7.50.17
   dmWeb                        n/a         [patched]
   baseWeb                      n/a         [patched]
   ipsec                        7.50.17
   Snmp-Daemon-TrapLib          7.50.17
   ISECSH                       7.50.17
   patchWeb                     7.50.17
   EmCentralLogic               7.50.17
Application configuration: NRS
Packages:
NRS
Configuration version:     7.50.17-00
   dbcom                        7.50.17
   nrsm                         7.50.17
   cslogin                      7.50.17
   nrsmWebService               7.50.17
   managedElementWebService     7.50.17
   sigServerShare               7.50.17
   sps                          7.50.17
   ncs                          7.50.17
   gk                           7.50.17
```

```
                    Avaya Network Routing Service Patches
In System service updates: 4
PATCH#  IN_SERVICE   DATE       SPECINS   REMOVABLE   NAME
0       Yes          12/04/11   NO        YES         cs1000-baseWeb-
                                                      7.50.17.01-1.i386.000
1       Yes          12/04/11   NO        YES         cs1000-linuxbase-
                                                      7.50.17.04-00.i386.000
2       Yes          12/04/11   NO        YES         cs1000-Jboss-Quantum-
                                                      7.50.17.01-1.i386.000
3       Yes          12/04/11   NO        YES         cs1000-dmWeb-
                                                      7.50.17.04-00.i386.001
```

Appendix C: AudioCodes MP-118 Configuration File

```
;**************
;** Ini File **
;**************

;Board: MP-118 FXS_FXO
;Serial Number: 2410047
;Slot Number: 1
;Software Version: 6.20A.022.003
;DSP Software Version: 204IM => 620.08
;Board IP Address: 47.166.92.205
;Board Subnet Mask: 255.255.255.224
;Board Default Gateway: 47.166.92.222
;Ram size: 32M   Flash size: 8M
;Num of DSP Cores: 2  Num DSP Channels: 8
;Profile: NONE
;License Key limits aren't active full features capabilities are available !;
;----------------------------------------------


[SYSTEM Params]

SyslogServerIP = 10.1.1.89

[BSP Params]

PCMLawSelect = 3
RoutingTableHopsCountColumn = 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0,
0, 0, 0, 0, 0, 0, 0, 0, 0

[Analog Params]

PolarityReversalType = 1
MinFlashHookTime = 100
FXSLoopCharacteristicsFilename = 'MP11x-02-1-FXS_16KHZ.dat'
CurrentDisconnectDuration = 1500
FlashHookPeriod = 1500
TimeToSampleAnalogLineVoltage = 100

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]

EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 0
EP_Num_3 = 0
EP_Num_4 = 0
DIGITMAPPING = '2xxx|3xxx|4xxx|6xxx|7xxx|5xxx'
```

```
[Voice Engine Params]

CallProgressTonesFilename = 'usa_tones_13.dat'
BrokenConnectionEventTimeout = 18000
CallerIDType = 1
FaxTransportMode = 2
RFC2833TxPayloadType = 101
RFC2833RxPayloadType = 101
AnalogSignalTransportType = 1
RTPAuthenticationDisableTx = 1
RTCPEncryptionDisableTx = 1
RTPEncryptionDisableTx = 1

[WEB Params]

LogoWidth = '145'
HTTPSCipherString = 'RC4:EXP'

[SIP Params]

ENABLECALLERID = 1
MAXDIGITS = 10
LOCALSIPPORT = 5070
REGISTRATIONTIME = 3600
DISCONNECTONBUSYTONE = 0
ISPROXYUSED = 1
ISREGISTERNEEDED = 1
AUTHENTICATIONMODE = 1
ISTWOSTAGEDIAL = 0
ENABLECURRENTDISCONNECT = 1
ENABLEREVERSALPOLARITY = 1
GWDEBUGLEVEL = 5
ENABLEPROXYKEEPALIVE = 2
ENABLEEARLYMEDIA = 1
ISUSERPHONE = 0
PROXYNAME = 'devcon.avayag'
SIPGATEWAYNAME = 'devcon.avayag'
ISFALLBACKUSED = 1
HOOKFLASHOPTION = 4
ALWAYSSENDTOPROXY = 1
PROXYREDUNDANCYMODE = 1
ISFAXUSED = 3
SUBSCRIPTIONMODE = 1
HOLDFORMAT = 1
SIPTRANSPORTTYPE = 1
TCPLOCALSIPPORT = 5070
GWREGISTRATIONNAME = 'ACodes'
TLSLOCALSIPPORT = 5071
HookFlashCode = '##'
ENABLE3WAYCONFERENCE = 1
CONFERENCECODE = ''
ENABLESEMIATTENDEDTRANSFER = 1
SASDEFAULTGATEWAYIP = '47.166.92.205:5070'
ENABLESAS = 1
SASLOCALSIPUDPPORT = 5060
```

```
SIPSDPSESSIONOWNER = 'ACodes'
SASREGISTRATIONTIME = 300
ENABLEHOLD2ISDN = 1
SASLOCALSIPTCPPORT = 5060
SASLOCALSIPTLSPORT = 5061
REGISTRARTRANSPORTTYPE = 1
SASPROXYSET = 1
SIPREROUTINGMODE = 2
SASBINDINGMODE = 1
SOURCEIPADDRESSINPUT = 0
SASSURVIVABILITYMODE = 1
RELIABLECONNECTIONPERSISTENTMODE = 1
FlashKeysSequenceStyle = 1
3WAYCONFERENCEMODE = 2
KEYCALLPICKUP = '#77'

[IPsec Params]


[SNMP Params]

;
;  *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;

;
;  *** TABLE PREFIX ***
;
;

[ PREFIX ]
; ** NOTE: Changes were made to active configuration.
; **       The data below is different from current values.
FORMAT PREFIX_Index = PREFIX_DestinationPrefix, PREFIX_DestAddress,
PREFIX_SourcePrefix, PREFIX_ProfileId, PREFIX_MeteringCode, PREFIX_DestPort,
PREFIX_SrcIPGroupID, PREFIX_DestHostPrefix, PREFIX_DestIPGroupID,
PREFIX_SrcHostPrefix, PREFIX_TransportType, PREFIX_SrcTrunkGroupID,
PREFIX_DestSRD;
PREFIX 0 = *, 47.166.92.205, *, 0, 255, 5070, -1, , -1, , 1, -1, -1;

[ \PREFIX ]

;
;  *** TABLE TrunkGroup ***
;
;

[ TrunkGroup ]
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum, TrunkGroup_FirstTrunkId,
TrunkGroup_FirstBChannel, TrunkGroup_LastBChannel,
TrunkGroup_FirstPhoneNumber, TrunkGroup_ProfileId, TrunkGroup_LastTrunkId,
TrunkGroup_Module;
TrunkGroup 0 = 1, 255, 1, 1, 6000, 0, 255, 255;
```

```
TrunkGroup 1 = 1, 255, 2, 2, 6001, 0, 255, 255;
TrunkGroup 2 = 1, 255, 3, 3, 6111, 0, 255, 255;
TrunkGroup 4 = 2, 255, 5, 5, 5111, 0, 255, 255;

[ \TrunkGroup ]

;
;  *** TABLE SourceNumberMapIp2Tel ***
;
;

[ SourceNumberMapIp2Tel ]
FORMAT SourceNumberMapIp2Tel_Index = SourceNumberMapIp2Tel_DestinationPrefix,
SourceNumberMapIp2Tel_SourcePrefix, SourceNumberMapIp2Tel_SourceAddress,
SourceNumberMapIp2Tel_NumberType, SourceNumberMapIp2Tel_NumberPlan,
SourceNumberMapIp2Tel_RemoveFromLeft, SourceNumberMapIp2Tel_RemoveFromRight,
SourceNumberMapIp2Tel_LeaveFromRight, SourceNumberMapIp2Tel_Prefix2Add,
SourceNumberMapIp2Tel_Suffix2Add,
SourceNumberMapIp2Tel_IsPresentationRestricted,
SourceNumberMapIp2Tel_SrcTrunkGroupID, SourceNumberMapIp2Tel_SrcIPGroupID;
SourceNumberMapIp2Tel 2 = *, *, *, 255, 255, 0, 0, 255, , , 255, -1, -1;
SourceNumberMapIp2Tel 3 = 951xx, *, *, 255, 255, 1, 0, 255, , , 255, -1, -1;

[ \SourceNumberMapIp2Tel ]

;
;  *** TABLE PstnPrefix ***
;
;

[ PstnPrefix ]
; ** NOTE: Changes were made to active configuration.
; **       The data below is different from current values.
FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId,
PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = 6000, 1, *, *, 0, -1, , ;
PstnPrefix 1 = 6001, 1, *, *, 0, -1, , ;
PstnPrefix 2 = 6111, 1, *, *, 0, -1, , ;
PstnPrefix 3 = *, 2, *, *, 0, -1, *, *;

[ \PstnPrefix ]

;
;  *** TABLE Dns2Ip ***
;
;

[ Dns2Ip ]
FORMAT Dns2Ip_Index = Dns2Ip_DomainName, Dns2Ip_FirstIpAddress,
Dns2Ip_SecondIpAddress, Dns2Ip_ThirdIpAddress, Dns2Ip_FourthIpAddress;
Dns2Ip 0 = devcon.avayag, 47.166.92.205, 0.0.0.0, 0.0.0.0, 0.0.0.0;

[ \Dns2Ip ]
```

```
;
;  *** TABLE ProxyIp ***
;
;

[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = 47.166.92.204, 1, 0;
ProxyIp 1 = 47.166.92.204:5060, 1, 1;
ProxyIp 2 = 47.166.92.205:5070, 1, 0;

[ \ProxyIp ]
;
;  *** TABLE TxDtmfOption ***
;
;
[ TxDtmfOption ]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]
;
;  *** TABLE TrunkGroupSettings ***
;
;
[ TrunkGroupSettings ]
FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId,
TrunkGroupSettings_ChannelSelectMode, TrunkGroupSettings_RegistrationMode,
TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup, TrunkGroupSettings_MWIInterrogationType;
TrunkGroupSettings 0 = 1, 0, 1, , , -1, 255;
TrunkGroupSettings 1 = 2, 1, 4, , , -1, 255;

[ \TrunkGroupSettings ]
;
;  *** TABLE TelProfile ***
;
;
[ TelProfile ]
FORMAT TelProfile_Index = TelProfile_ProfileName, TelProfile_TelPreference,
TelProfile_CodersGroupID, TelProfile_IsFaxUsed, TelProfile_JitterBufMinDelay,
TelProfile_JitterBufOptFactor, TelProfile_IPDiffServ,
TelProfile_SigIPDiffServ, TelProfile_DtmfVolume, TelProfile_InputGain,
TelProfile_VoiceVolume, TelProfile_EnableReversePolarity,
TelProfile_EnableCurrentDisconnect, TelProfile_EnableDigitDelivery,
TelProfile_EnableEC, TelProfile_MWIAnalog, TelProfile_MWIDisplay,
TelProfile_FlashHookPeriod, TelProfile_EnableEarlyMedia,
TelProfile_ProgressIndicator2IP, TelProfile_TimeForReorderTone,
TelProfile_EnableDIDWink, TelProfile_IsTwoStageDial,
TelProfile_DisconnectOnBusyTone, TelProfile_EnableVoiceMailDelay,
TelProfile_DialPlanIndex, TelProfile_Enable911PSAP,
TelProfile_SwapTelToIpPhoneNumbers, TelProfile_EnableAGC,
TelProfile_ECNlpMode, TelProfile_DigitalCutThrough;
```

```
TelProfile 1 = , 1, 0, 1, 10, 10, 46, 40, -11, 0, 0, 1, 0, 0, 1, 0, 0, 150,
1, -1, 255, 0, 0, 0, 1, -1, 0, 0, 0, 0, 0;

[ \TelProfile ]

;
;  *** TABLE IpProfile ***
;
;

[ IpProfile ]
FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupID, IpProfile_IsFaxUsed, IpProfile_JitterBufMinDelay,
IpProfile_JitterBufOptFactor, IpProfile_IPDiffServ, IpProfile_SigIPDiffServ,
IpProfile_SCE, IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode,
IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP, IpProfile_EnableEarlyMedia,
IpProfile_ProgressIndicator2IP, IpProfile_EnableEchoCanceller,
IpProfile_CopyDest2RedirectNumber, IpProfile_MediaSecurityBehaviour,
IpProfile_CallLimit, IpProfile_DisconnectOnBrokenConnection,
IpProfile_FirstTxDtmfOption, IpProfile_SecondTxDtmfOption,
IpProfile_RxDTMFOption, IpProfile_EnableHold, IpProfile_InputGain,
IpProfile_VoiceVolume, IpProfile_AddIEInSetup,
IpProfile_SBCExtensionCodersGroupID, IpProfile_MediaIPVersionPreference,
IpProfile_TranscodingMode, IpProfile_SBCAllowedCodersGroupID,
IpProfile_SBCAllowedCodersMode, IpProfile_SBCMediaSecurityBehaviour,
IpProfile_SBCRFC2833Behavior, IpProfile_SBCAlternativeDTMFMethod,
IpProfile_SBCAssertIdentity, IpProfile_AMDSensitivityParameterSuit,
IpProfile_AMDSensitivityLevel, IpProfile_AMDMaxGreetingTime,
IpProfile_AMDMaxPostSilenceGreetingTime, IpProfile_SBCDiversionMode,
IpProfile_SBCHistoryInfoMode;
IpProfile 1 = , 1, 0, 1, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 1, -1, 1, 0,
0, -1, 1, 0, -1, 1, 1, 0, 0, , -1, 0, 0, -1, 0, 0, 0, 0, -1, 0, 8, 300, 400,
-1, -1;

[ \IpProfile ]

;
;  *** TABLE CallerDisplayInfo ***
;
;

[ CallerDisplayInfo ]
FORMAT CallerDisplayInfo_Index = CallerDisplayInfo_DisplayString,
CallerDisplayInfo_IsCidRestricted;
CallerDisplayInfo 0 = ACodes6000, 0;
CallerDisplayInfo 1 = ACodes6001, 0;

[ \CallerDisplayInfo ]

;
;  *** TABLE Authentication ***
;
;
```

```
[ Authentication ]
FORMAT Authentication_Index = Authentication_UserId,
Authentication_UserPassword;
Authentication 0 = 6000, *;
Authentication 1 = 6001, *;

[ \Authentication ]

;
;  *** TABLE ProxySet ***
;
;

[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap, ProxySet_SRD, ProxySet_ClassificationInput,
ProxySet_ProxyRedundancyMode;
ProxySet 0 = 1, 60, 0, 1, 0, 0, 1;
ProxySet 1 = 1, 60, 0, 0, 0, 0, -1;

[ \ProxySet ]

;
;  *** TABLE IPGroup ***
;
;

[ IPGroup ]
FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description, IPGroup_ProxySetId,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_EnableSurvivability,
IPGroup_ServingIPGroup, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_RoutingMode, IPGroup_SRD,
IPGroup_MediaRealm, IPGroup_ClassifyByProxySet, IPGroup_ProfileId,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_ContactName;
IPGroup 1 = 0, NRS, 1, devcon.avayag, , 0, -1, 0, 0, -1, 0, , 1, 0, -1, -1, -
1, ;

[ \IPGroup ]

;
;  *** TABLE SASRegistrationManipulation ***
;
;

[ SASRegistrationManipulation ]
FORMAT SASRegistrationManipulation_Index =
SASRegistrationManipulation_RemoveFromRight,
SASRegistrationManipulation_LeaveFromRight;
SASRegistrationManipulation 0 = 0, 0;

[ \SASRegistrationManipulation ]

;
```

```
;  *** TABLE CodersGroup0 ***
;
;

[ CodersGroup0 ]
FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = g711Alaw64k, 20, 0, -1, 0;
CodersGroup0 1 = g711Alaw64k, 20, 0, -1, 0;

[ \CodersGroup0 ]
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.