



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 118 enables 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 118 supports 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

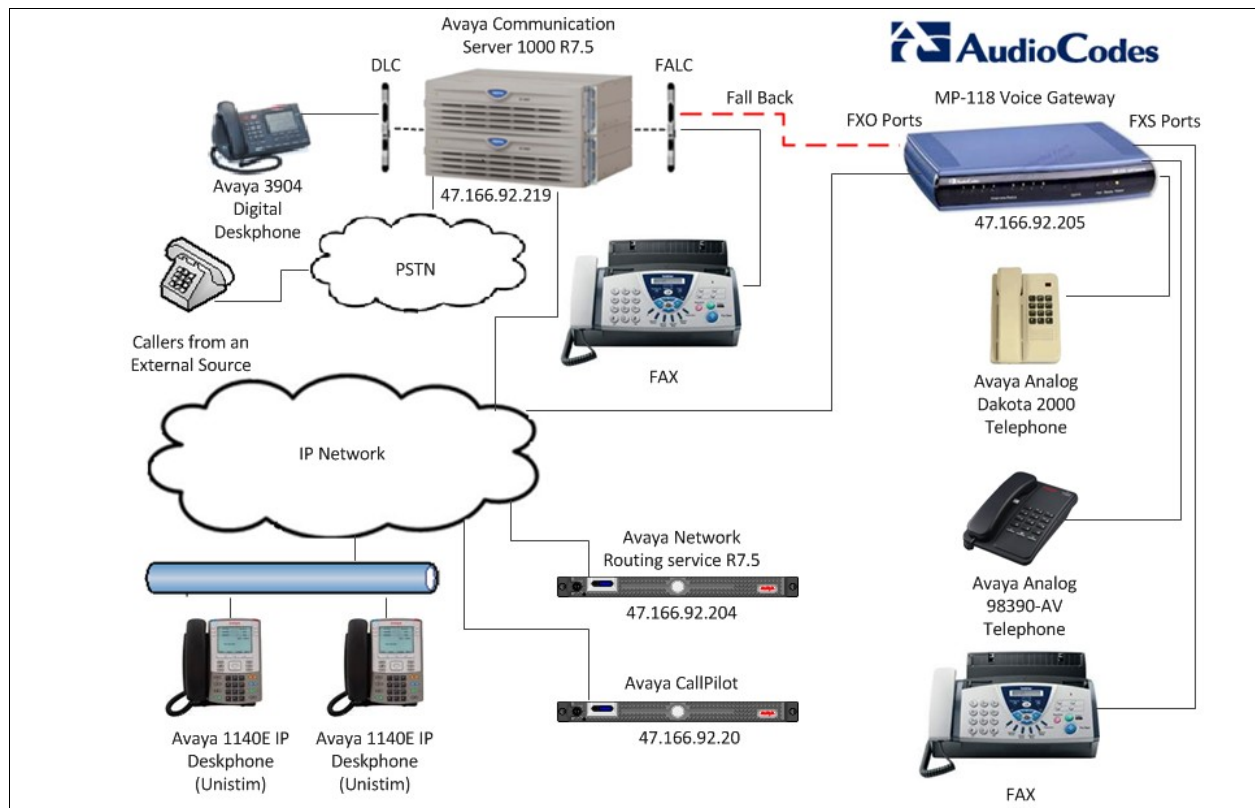


Figure 1: Avaya Communication Server R7.5 with AudioCodes MP-118 Voice Gateway Reference Configuration

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 Communication Server 1000E R7.5 SP1 |
| Avaya Media Gateway NTDW60 | FPGA AA18 |
| Avaya Flexible Analog Line Card NT5K02QC | - |
| Avaya Digital Line Card NT8D02GA | - |
| IBM System x3350 | Network Routing Service R7.50.17 |
| Avaya CallPilot 600r | Call Pilot R5 SU08S |
| Avaya 1140E IP set | 0625C8A (UNISTim 5.0) |
| Avaya 3904 Digital set | F/W 2.4 |
| Avaya Analog set | - |
| 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, deplists 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.

The screenshot displays the Avaya Network Routing Service Manager web interface. The left sidebar contains a navigation menu with categories like «UCM Network Services», System, Numbering Plans (highlighted), and Tools. Under Numbering Plans, Endpoints is selected. The main content area shows the 'Managing' section with 'Active database' and 'Standby database' radio buttons, where 'Standby database' is selected. Below this is a 'Search for Endpoints' section with a text input for 'Endpoint ID' and three dropdown menus for 'Limit results to Domain' (set to 'devcon.avayag'), protocol (set to 'udp'), and transport (set to 'cdp'). A 'Search' button is present. At the bottom, there are two tabs: 'Gateway Endpoints (4)' (active) and 'User Endpoints (0)'. Below the tabs are 'Add', 'Delete', and 'SIP phone context...' buttons, along with a 'Refresh' link.

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.

The screenshot displays the AVAYA Network Routing Service Manager web interface. The left sidebar contains a navigation menu with categories like System, Numbering Plans, Tools, and Routing Tests. The main content area is titled 'Edit Gateway Entry (devcon.avayag / udp / cdp / ACodes)'. It features a form with several fields: 'End point name' (set to 'ACodes'), 'Description' (set to 'AudioCodes'), 'Trust Node' (checked), 'Tandem gateway endpoint name' (set to 'Not Applicable'), and 'Endpoint authentication enabled' (set to 'Authentication off'). Below these are fields for 'Authentication password' and various E.164 codes. A red arrow points to the vertical scrollbar on the right side of the form, indicating the instruction to scroll down. The bottom of the form has 'Save' and 'Cancel' buttons.

In the page below fill in the following:

- **SIP support: endpoint** From the dropdown box select **Dynamic SIP**
- **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

AVAYA Network Routing Service Manager

Help | Logout

Managing: ☐ Active database 172.18.20.19
☒ Standby database [Numbering Plans > Endpoints > Gateway Endpoint](#)

Edit Gateway Entry (devcon.avayag / udp / cdp / ACodes)

Private Special number 2:

Private Special number 2 dialing code length: (0-31)

Static endpoint address type: IP version 4

Static endpoint address:

H.323 support: H.323 not supported

SIP support: Dynamic SIP endpoint

SIP mode: ☒ Proxy Mode ☐ Redirect Mode

SIP TCP transport enabled: ☒

SIP TCP port: 5060

SIP UDP transport enabled: ☒

SIP UDP port: 5060

SIP TLS transport enabled: ☐

SIP TLS port: 5061

Persistent TCP support enabled: ☒

* Required value

Save Cancel

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.

The screenshot displays the Avaya Network Routing Service Manager web interface. The left sidebar contains a navigation tree with categories like «UCM Network Services», System, Numbering Plans, and Tools. The 'Routes' option under Numbering Plans is selected and highlighted. The main content area is titled 'Network Routing Service Manager' and shows search filters for Routing Entries. The 'Limit results to Domain:' section has three dropdown menus set to 'devcon.avayag', 'udp', and 'cdp'. The 'Endpoint Name:' dropdown is set to 'ACodes'. Below the search filters, there are tabs for 'Routing Entries (1)', 'Default Routes (0)', and 'Emergency Fallback Routes (0)'. The 'Routing Entries (1)' tab is active, and the 'Add...' button is highlighted. The table below the tabs shows columns for DN Prefix, DN Type, Route Cost, SIP URI Phone Context, and Context.

Once the new page opens enter the following:

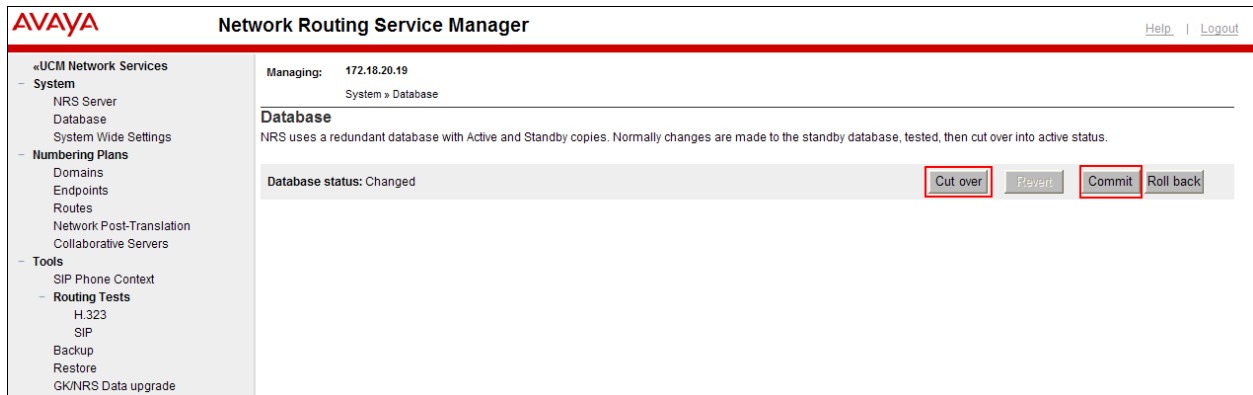
- **DN type:** From the dropdown box select **Private level 0 regional (CDP steering code)**
- **DN Prefix:** Enter the digit 6
- **Route cost:** Enter 1

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

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

The screenshot shows the Avaya Network Routing Service Manager interface. The left sidebar contains a navigation menu with categories: «UCM Network Services, System, NRS Server, Database (highlighted with a red box), System Wide Settings, Numbering Plans, Domains, Endpoints, Routes, Network Post-Translation, Collaborative Servers, Tools, SIP Phone Context, Routing Tests, H.323, SIP, Backup, Restore, and GK/NRS Data upgrade. The main content area is titled 'Network Routing Service Manager' and includes a 'Help' and 'Logout' link. Below the title, there are radio buttons for 'Active database' (selected) and 'Standby database', along with the IP address '172.18.20.19'. A breadcrumb trail shows 'Numbering Plans » Routes » Routing Entry'. The main section is titled 'Edit Routing Entry (devcon.avayag / udp / cdp / ACodes)'. It contains a form with three fields: 'DN type' (a dropdown menu showing 'Private level 0 regional (CDP steering code)' with a red box around it), 'DN prefix' (a text box containing '6' with a red box around it), and 'Route cost' (a text box containing '1' with a red box around it). A red box also highlights the 'Save' and 'Cancel' buttons at the bottom right of the form. A note at the bottom left of the form states '* Required value:'.

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.

The screenshot shows the Avaya CS1000 Element Manager web interface. The left sidebar contains a navigation tree with categories like UCM Network Services, System, and IP Network. The 'IP Network' category is expanded, and 'Nodes: Servers, Media Cards' is selected. The main content area displays the 'Node ID: 3 - Virtual Trunk Gateway Configuration Details' page. The 'SIP Gateway Services' tab is active, showing the 'SIP URI Map' section. This section has two columns: 'Public E.164 domain names' and 'Private domain names'. Under 'Public E.164 domain names', there are fields for 'National', 'Subscriber', 'Special number' (with 'PublicSpecial' selected), and 'Unknown' (with 'PublicUnknown' selected). Under 'Private domain names', there are fields for 'UDP' (with 'udp' selected), 'CDP' (with 'cdp.udp' selected), 'Special number' (with 'PrivateSpecial' selected), and 'Vacant number' (with 'PrivateUnknown' selected). The 'Unknown' field in the private domain names column is highlighted with a red box. Below the 'SIP URI Map' section is the 'SIP Gateway Services' section, which includes a checkbox for 'SIP Converged Desktop' and a 'Service DN' field. At the bottom right, there are 'Save' and 'Cancel' buttons, with the 'Save' button highlighted by a red box. A note at the bottom states: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.'

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.

The screenshot shows the AVAYA CS1000 Element Manager interface. The left sidebar contains a navigation tree with 'IP Network' and 'Nodes: Servers, Media Cards' highlighted. The main content area is titled 'Node ID: 3 - Voice Gateway (VGW) and Codecs' and has tabs for 'General', 'Voice Codecs', and 'Fax'. The 'General' tab is active, showing various configuration options. The 'Modem/Fax pass-through' checkbox is checked and highlighted with a red box. Other settings include Echo cancellation, Voice activity detection threshold, and Signaling options. The 'Save' button is highlighted with a red box.

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.

The screenshot shows the AVAYA CS1000 Element Manager interface. The left sidebar contains a navigation tree with 'IP Network' and 'Nodes: Servers, Media Cards' highlighted. The main content area is titled 'Node ID: 3 - Voice Gateway (VGW) and Codecs' and has tabs for 'General', 'Voice Codecs', and 'Fax'. The 'Voice Codecs' tab is active, showing various configuration options. The 'Codec G.729' checkbox is checked and highlighted with a red box. Other settings include Codec G.722 and Codec G.723.1. The 'Save' button is highlighted with a red box.

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

AVAYA CS1000 Element Manager

Managing: 172.18.20.14 Username: admin2

System > IP Network > IP Telephony > Nodes > Node Details

Node Details (ID: 3 - SIP Line, LTPS, PD, IP Media Services, Gateway (SIPGw, H323Gw))

Node ID: 3 * (0-9999)

Call server IP address: 172.18.20.14 * TLAN address type: ☒ IPv4 only ☐ IPv4 and IPv6

Embedded LAN (ELAN) Gateway IP address: 172.18.20.1 * Subnet mask: 255.255.255.128 *

Telephony LAN (TLAN) Node IPv4 address: 47.166.92.219 * Subnet mask: 255.255.255.224 *

Node IPv6 address: *

* Required Value.

Save Cancel

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

AVAYA CS1000 Element Manager

Managing: 172.18.20.14 Username: admin2

System > IP Network > IP Telephony > Nodes > Node Saved

Node Saved

Node ID: 3 has been saved on the call server.

The new configuration must also be transferred to associated servers and media cards.

Transfer Now... You will be given an option to select individual servers, or transfer to all.

Show Nodes You may initiate a transfer manually at a later time.

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 Analog 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 AudioCodes MP-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.



A Windows-style dialog box titled "Connect to 47.166.92.205". The title bar is blue with a question mark icon and a close button (X). The main area has a light beige background. At the top left is a yellow key icon. Below it, text reads: "The server 47.166.92.205 at Realm1 requires a username and password." Below this text is a red rectangular box containing two input fields: "User name:" with a small user icon and a dropdown arrow, and "Password:" with a text box. Below the password field is a checkbox labeled "Remember my password". At the bottom right, outside the red box, are two buttons: "OK" and "Cancel". The "OK" button is highlighted with a red rectangular box.

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**.

The screenshot displays the AudioCodes MP-118 FXS_FXO web interface. The 'Configuration' tab is active, and the 'Full' radio button is selected under the 'Basic' tab. The 'Network' section is expanded, and 'IP Settings' is selected. The 'Multiple Interface Table' shows a single entry with Index 0, Application Type 'OAMP + Media + Control', IP Address '47.166.92.205', Prefix Length '27', Gateway '47.166.92.222', VLAN ID '1', and Interface Name 'O+M+C'. The 'VLAN Mode' is set to 'Disable' and the 'Native VLAN ID' is '1'.

| Index | Application Type | IP Address | Prefix Length | Gateway | VLAN ID | Interface Name |
|-------|------------------------|---------------|---------------|---------------|---------|----------------|
| 0 | OAMP + Media + Control | 47.166.92.205 | 27 | 47.166.92.222 | 1 | O+M+C |

VLAN Mode: Disable
Native VLAN ID: 1
IP Interface Status Table

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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar shows the navigation tree with 'VoIP' and 'Network' expanded, and 'Internal DNS Table' selected. The main area displays the 'Internal DNS Table' configuration. A table with 5 columns (Domain Name, First IP Address, Second IP Address, Third IP Address, Fourth IP Address) and 10 rows is shown. The first row is populated with 'devcon.avayag', '47.166.92.205', and three '0.0.0.0' addresses. The 'Submit' button is highlighted in the top navigation bar.

| | Domain Name | First IP Address | Second IP Address | Third IP Address | Fourth IP Address |
|----|---------------|------------------|-------------------|------------------|-------------------|
| 1 | devcon.avayag | 47.166.92.205 | 0.0.0.0 | 0.0.0.0 | 0.0.0.0 |
| 2 | | | | | |
| 3 | | | | | |
| 4 | | | | | |
| 5 | | | | | |
| 6 | | | | | |
| 7 | | | | | |
| 8 | | | | | |
| 9 | | | | | |
| 10 | | | | | |

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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar shows the navigation tree with 'VoIP' and 'Media' expanded, and 'Voice Settings' selected. The main area displays the 'Voice Settings' configuration. A list of parameters is shown, including 'Voice Volume', 'Input Gain', 'Silence Suppression', 'DTMF Transport Type' (set to 'RFC2833 Relay DTMF'), 'DTMF Volume', 'NTE Max Duration', 'Enable Answer Detector', 'Answer Detector Activity Delay', 'Answer Detector Silence Time', 'Answer Detector Redirection', 'Answer Detector Sensitivity', 'DTMF Generation Twist', and 'Echo Cancellor'. The 'Submit' button is highlighted in the top navigation bar.

| Parameter | Value |
|--------------------------------|--------------------|
| Voice Volume (-32 to 31 dB) | 0 |
| Input Gain (-32 to 31 dB) | 0 |
| Silence Suppression | Disable |
| DTMF Transport Type | RFC2833 Relay DTMF |
| DTMF Volume (-31 to 0 dB) | -11 |
| NTE Max Duration | -1 |
| Enable Answer Detector | Disable |
| Answer Detector Activity Delay | 0 |
| Answer Detector Silence Time | 10 |
| Answer Detector Redirection | 0 |
| Answer Detector Sensitivity | 0 |
| DTMF Generation Twist | 0 |
| Echo Cancellor | Enable |

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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar contains a tree view with the following structure:

- Configuration
- Maintenance
- Status & Diagnostics
- Scenarios
- Search
- Basic
- Full
- System
- VoIP
- Network
- Security
- Media
- Voice Settings
- Fax/Modem/CID Settings
- RTP/RTCP Settings
- IPMedia Settings
- General Media Settings
- Analog Settings
- Media Security
- Applications Enabling
- Control Network
- SIP Definitions
- Coders And Profiles
- GW and IP to IP
- SAS

The main content area is titled "Fax/Modem/CID Settings" and contains the following settings:

| General Settings | |
|---------------------------|---------------|
| Fax Transport Mode | RelayEnable |
| Caller ID Transport Type | Mute |
| Caller ID Type | Standard ETSI |
| V.21 Modem Transport Type | Disable |
| V.22 Modem Transport Type | Enable Bypass |
| V.23 Modem Transport Type | Enable Bypass |
| V.32 Modem Transport Type | Enable Bypass |
| V.34 Modem Transport Type | Enable Bypass |
| Fax CNG Mode | Disable |
| CNG Detector Mode | Disable |

| Fax Relay Settings | |
|-------------------------------------|----------|
| Fax Relay Redundancy Depth | 0 |
| Fax Relay Enhanced Redundancy Depth | 4 |
| Fax Relay ECM Enable | Enable |
| Fax Relay Max Rate (bps) | 14400bps |

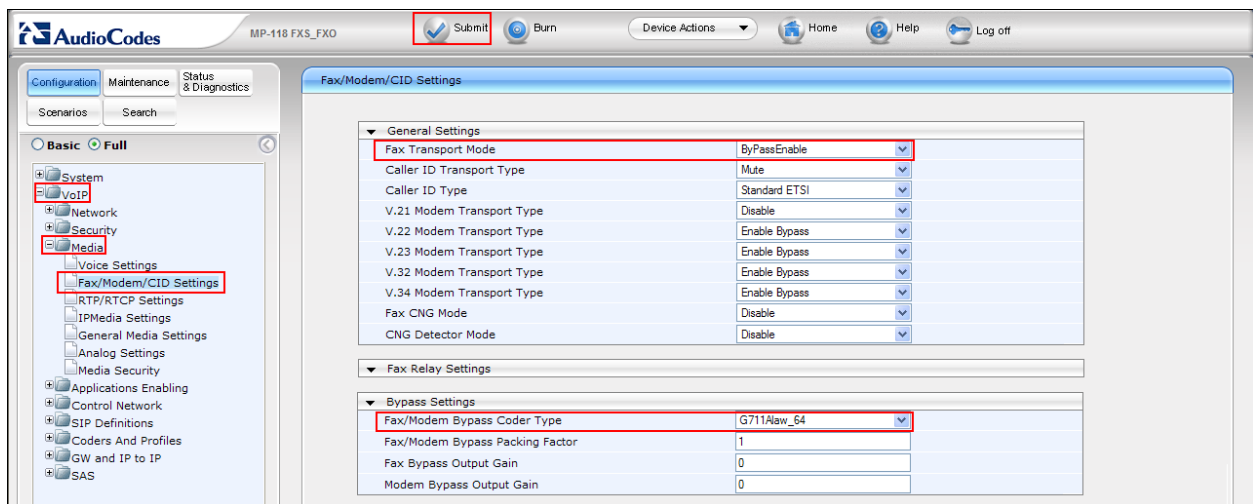
Below the Fax Relay Settings is a section for Bypass Settings, which is currently collapsed.

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 Codec Type** drop down box in the **Bypass Settings** window select **G.711Alaw_64**

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

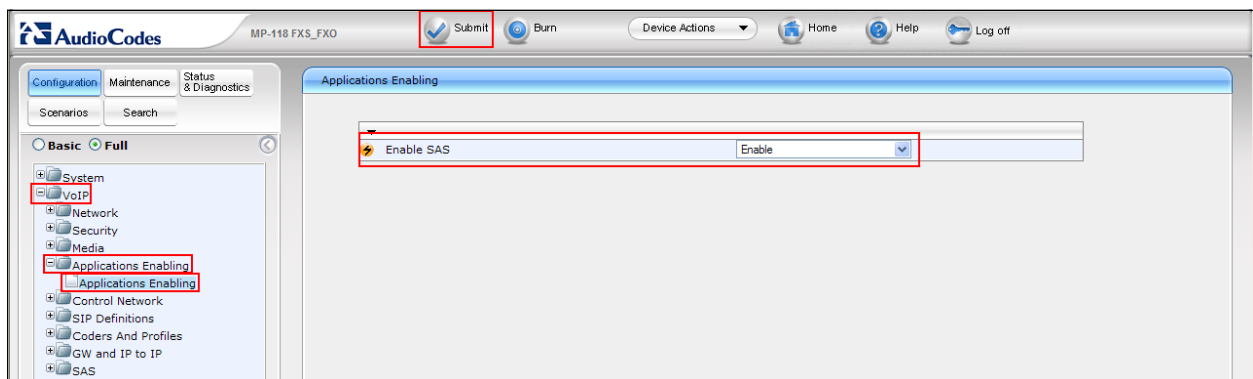


6.6. Configure Application Enabling

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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The top bar includes a 'Submit' button (highlighted with a red box), a 'Burn' button, and a 'Device Actions' dropdown. The left sidebar shows the navigation tree with 'IP Group Table' selected under 'Control Network'. The main area shows the configuration form for the 'IP Group Table'.

IP Group Table Configuration:

- Index:** 1
- Description:** NRS
- Proxy Set ID:** 1
- SIP Group Name:** devcon.avayag
- Contact User:**
- IP Profile ID:** 0
- Gateway Parameters:**
 - Always Use Route Table: No
 - Routing Mode: Not Configured
 - SIP Re-Routing Mode: Standard
 - Enable Survivability: Disable
 - Serving IP Group ID:

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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar shows the navigation tree with 'VoIP' selected, and 'Control Network' > 'Proxy Sets Table' highlighted. The main area is titled 'Proxy Sets Table'. At the top, 'Proxy Set ID' is set to 0. Below this is a table with two columns: 'Proxy Address' and 'Transport Type'. The table contains two entries: 1. 47.166.92.204, TCP and 2. 47.166.92.205:5070, TCP. Below the table are several configuration options: 'Enable Proxy Keep Alive' (Using Options), 'Proxy Keep Alive Time' (60), 'Proxy Load Balancing Method' (Disable), 'Is Proxy Hot Swap' (Yes), 'Proxy Redundancy Mode' (Homing), 'SRD Index' (0), and 'Classification Input' (IP only). The 'Submit' button is highlighted in the top navigation bar.

| Proxy Set ID | Proxy Address | Transport Type |
|--------------|----------------------|----------------|
| 0 | 1 47.166.92.204 | TCP |
| | 2 47.166.92.205:5070 | TCP |
| | 3 | |
| | 4 | |
| | 5 | |

| | |
|-----------------------------|---------------|
| Enable Proxy Keep Alive | Using Options |
| Proxy Keep Alive Time | 60 |
| Proxy Load Balancing Method | Disable |
| Is Proxy Hot Swap | Yes |
| Proxy Redundancy Mode | Homing |
| SRD Index | 0 |
| Classification Input | IP only |

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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar contains a tree view with categories like System, Network, Security, Media, Applications Enabling, Control Network, IP Group Table, SIP Definitions, Coders And Profiles, GW and IP to IP, and SAS. The 'Proxy Sets Table' is selected under 'Control Network'. The main area displays the configuration for Proxy Set ID 1. A table lists proxy addresses and transport types. Below the table, several settings are configured, including 'Enable Proxy Keep Alive' set to 'Using Options', 'Proxy Keep Alive Time' set to 60, 'Proxy Load Balancing Method' set to 'Disable', 'Is Proxy Hot Swap' set to 'No', 'Proxy Redundancy Mode' set to 'Not Configured', 'SRD Index' set to 0, and 'Classification Input' set to 'IP only'. A 'Submit' button is highlighted in the top navigation bar.

| Proxy Set ID | Proxy Address | Transport Type |
|--------------|--------------------|----------------|
| 1 | 47.166.92.204:5060 | TCP |
| 2 | | |
| 3 | | |
| 4 | | |
| 5 | | |

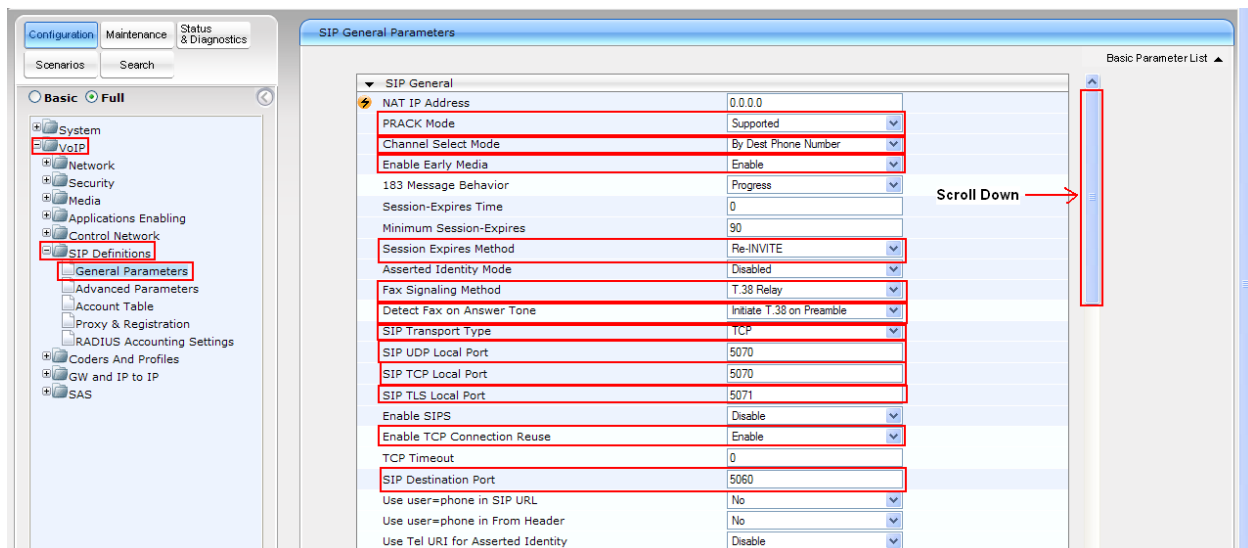
| | |
|-----------------------------|----------------|
| Enable Proxy Keep Alive | Using Options |
| Proxy Keep Alive Time | 60 |
| Proxy Load Balancing Method | Disable |
| Is Proxy Hot Swap | No |
| Proxy Redundancy Mode | Not Configured |
| SRD Index | 0 |
| Classification Input | IP only |

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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar contains a tree view with categories like System, VoIP, Network, Security, Media, Applications Enabling, Control Network, SIP Definitions, General Parameters, Advanced Parameters, Account Table, Proxy & Registration, RADIUS Accounting Settings, Coders And Profiles, GW and IP to IP, and SAS. The 'SIP Definitions' category is expanded, and 'General Parameters' is selected. The main area displays the 'SIP General Parameters' configuration table. The table has two columns: the parameter name and its value. Several rows are highlighted with red boxes, indicating the settings to be configured: 'Use user=phone in SIP URL' (No), 'Use user=phone in From Header' (No), 'Play Ringback Tone to IP' (Don't Play), 'Play Ringback Tone to Tel' (Prefer IP), and 'SDP Session Owner' (ACodes). The 'Submit' button is visible at the top of the interface.

| Parameter | Value |
|--|------------|
| Use user=phone in SIP URL | No |
| Use user=phone in From Header | No |
| Use Tel URI for Asserted Identity | Disable |
| Tel to IP No Answer Timeout | 180 |
| Enable Remote Party ID | Disable |
| Add Number Plan and Type to RPI Header | Yes |
| Enable History-Info Header | Disable |
| Use Source Number as Display Name | No |
| Use Display Name as Source Number | No |
| Enable Contact Restriction | Disable |
| Play Ringback Tone to IP | Don't Play |
| Play Ringback Tone to Tel | Prefer IP |
| Use Trgp information | Disable |
| Enable GRUU | Disable |
| User-Agent Information | |
| SDP Session Owner | ACodes |
| Subject | |

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 **Registrar 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.

The screenshot shows the AudioCodes MP-118 FXS_FXD web interface. The left sidebar contains a navigation tree with the following items: System, VoIP, Network, Security, Media, Applications Enabling, Control Network, SIP Definitions, General Parameters, Advanced Parameters, Account Table, Proxy & Registration, RADIUS Accounting Settings, Coders And Profiles, GW and IP to IP, and SAS. The 'Proxy & Registration' item is selected. The main area displays the 'Proxy & Registration' configuration page. The parameters are listed in a table with the following values:

| Parameter | Value |
|---|-------------------|
| Use Default Proxy | Yes |
| Proxy Set Table | |
| Proxy Name | devcon.avayag |
| Redundancy Mode | Homing |
| Proxy IP List Refresh Time | 60 |
| Enable Fallback to Routing Table | Enable |
| Prefer Routing Table | No |
| Use Routing Table for Host Names and Profiles | Disable |
| Always Use Proxy | Enable |
| Redundant Routing Mode | Routing Table |
| SIP ReRouting Mode | Use Routing Table |
| Enable Registration | Enable |
| Registrar Name | |
| Registrar IP Address | |
| Registrar Transport Type | TCP |
| Registration Time | 3600 |
| Re-registration Timing [%] | 50 |
| Registration Retry Time | 30 |
| Registration Time Threshold | 0 |
| Re-register On INVITE Failure | Disable |
| ReRegister On Connection Failure | Disable |
| Gateway Name | devcon.avayag |
| Gateway Registration Name | ACodes |

A red arrow points to the scroll bar on the right side of the table, indicating the need to scroll down to view the bottom parameters.

- 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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The top navigation bar includes the AudioCodes logo, the device model 'MP-118 FXS_FXO', and buttons for 'Submit' (highlighted with a red box), 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar contains a tree view with categories like 'Configuration', 'Maintenance', and 'Status & Diagnostics'. Under 'Configuration', the 'Full' tab is selected, and the 'Proxy & Registration' section is highlighted. The main panel displays the 'Proxy & Registration' settings. The 'Subscription Mode' and 'Registration Mode' dropdowns are both set to 'Per Gateway' and are highlighted with red boxes. Other settings include 'Enable Registration' (checked), 'Registrar Name', 'Registrar IP Address', 'Registrar Transport Type' (TCP), 'Registration Time' (3600), 'Re-registration Timing [%]' (50), 'Registration Retry Time' (30), 'Registration Time Threshold' (0), 'Re-register On INVITE Failure' (Disable), 'ReRegister On Connection Failure' (Disable), 'Gateway Name' (dpp.notel), 'Gateway Registration Name' (ACodes), 'DNS Query Type' (A-Record), 'Proxy DNS Query Type' (A-Record), 'Number of RTX Before Hot-Swap' (3), 'Use Gateway Name for OPTIONS' (No), 'User Name', 'Password' (Default_Passwd), 'Cnonce' (Default_Cnonce), 'Set Out-Of-Service On Registration Failure' (Disable), and 'Challenge Caching Mode' (None).

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.

The screenshot shows the AudioCodes configuration interface. The left sidebar contains a tree view with the following items: System, VoIP, Network, Security, Media, Applications Enabling, Control Network, SIP Definitions, Coders And Profiles (highlighted), Coders Group Settings, Tel Profile Settings, IP Profile Settings, GW and IP to IP, and SAS. The 'Coders And Profiles' item is expanded, showing 'Coders' and 'Profiles'. The 'Coders' item is selected. The main area displays the 'Coders Table' with the following columns: Coder Name, Packetization Time, Rate, Payload Type, and Silence Suppression. The first row is highlighted with a red box and contains the following values: G.711A-law, 20, 64, 8, and Disabled. The top of the interface shows a 'Submit' button (highlighted with a red box) and other navigation buttons like 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'.

| Coder Name | Packetization Time | Rate | Payload Type | Silence Suppression |
|------------|--------------------|------|--------------|---------------------|
| G.711A-law | 20 | 64 | 8 | Disabled |
| | | | | |
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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.

The screenshot shows the AudioCodes configuration interface. The left sidebar has a tree view with 'VoIP' and 'Coders And Profiles' highlighted. The 'Coders' sub-item is also highlighted. The main area displays the 'Coders Table' with the following data:

| Coder Name | Packetization Time | Rate | Payload Type | Silence Suppression |
|------------|--------------------|------|--------------|---------------------|
| G.729 | 20 | 8 | 18 | Disabled |
| | | | | |
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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.

The screenshot shows the AudioCodes MP-118 FXS_FXO web interface. The left sidebar contains a tree view with 'VoIP' expanded, and 'Coders And Profiles' selected. The main area is titled 'Tel Profile Settings'. It features a 'Profile ID' dropdown set to '1' and a 'Profile Name' text field. Below, the 'Profile Parameters' section includes a 'Fax Signaling Method' dropdown set to 'T.38 Relay'. Other parameters like 'Dynamic Jitter Buffer Minimum Delay', 'RTP IP DiffServ', and 'Signaling DiffServ' are also visible. A 'Submit' button is highlighted with a red box at the top of the interface.

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.

This screenshot is similar to the one above, showing the 'Tel Profile Settings' page. In this configuration, the 'Fax Signaling Method' dropdown is set to 'G.711 Transport'. The 'Profile ID' remains '1'. The 'Submit' button is again highlighted with a red box.

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.

The screenshot shows the AudioCodes configuration web interface for an MP-118 FXS_FXO device. The left sidebar shows a tree view with 'VoIP' expanded, and 'IP Profile Settings' selected under 'Coders And Profiles'. The main panel is titled 'IP Profile Settings'. At the top, the 'Profile ID' dropdown is set to '1'. Below, the 'Common Parameters' section includes fields for RTP IP DiffServ (46), Signaling DiffServ (40), Disconnect on Broken Connection (Yes), Dynamic Jitter Buffer Minimum Delay (10), Dynamic Jitter Buffer Optimization Factor (10), RTP Redundancy Depth (0), Echo Canceled (Enable), Input Gain (0), and Voice Volume (0). The 'Gateway Parameters' section shows 'Fax Signaling Method' set to 'T.38 Relay', 'Play Ringback Tone to IP' set to 'Don't Play', and 'Enable Early Media' set to 'Enable'. A 'Submit' button is highlighted with a red box at the top of the interface.

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.

This screenshot is similar to the previous one, showing the 'IP Profile Settings' for Profile ID 1. However, in the 'Gateway Parameters' section, the 'Fax Signaling Method' is now set to 'G.711 Transport'. Other parameters remain the same. The 'Submit' button is again highlighted with a red box.

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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar contains a tree view with the following items: Configuration, Maintenance, Status & Diagnostics, Scenarios, Search, Basic, Full, System, VoIP, Network, Security, Media, Applications Enabling, Control Network, SIP Definitions, Coders And Profiles, GW and IP to IP, Hunt Group, Endpoint Phone Number, Hunt Group Settings, Manipulations, Routing, DTMF and Supplementary, Analog Gateway, Advanced Applications, and SAS. The 'Endpoint Phone Number' item is selected. The main area displays a table titled 'Endpoint Phone Number Table' with the following data:

| | Channel(s) | Phone Number | Hunt Group ID | Tel Profile ID |
|---|------------|--------------|---------------|----------------|
| 1 | 1 | 6000 | 1 | 0 |
| 2 | 2 | 6001 | 1 | 0 |
| 3 | 3 | 6111 | 1 | 0 |
| 4 | | | | |
| 5 | 5 | 5111 | 2 | 0 |
| 6 | | | | |
| 7 | | | | |
| 8 | | | | |

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.

MP-118 FXS_FXO

Submit Burn Device Actions Home Help Log off

Configuration Maintenance Status & Diagnostics

Scenarios Search

Basic Full

System

VoIP

Network

Security

Media

Applications Enabling

Control Network

SIP Definitions

Coders And Profiles

GW and IP to IP

Hunt Group

EndPoint Phone Number

Hunt Group Settings

Manipulations

Routing

DTMF and Supplementary

Analog Gateway

Advanced Applications

SAS

Hunt Group Settings

Basic Parameter List

Index 1-12

| Hunt Group ID | Channel Select Mode | Registration Mode | Serving IP Group ID | Gateway Name | Contact User |
|---------------|----------------------|-------------------|---------------------|--------------|--------------|
| 1 | By Dest Phone Number | Per Gateway | | | |
| 2 | Cyclic Ascending | Don't Register | | | |
| 3 | | | | | |
| 4 | | | | | |
| 5 | | | | | |
| 6 | | | | | |
| 7 | | | | | |
| 8 | | | | | |
| 9 | | | | | |
| 10 | | | | | |
| 11 | | | | | |
| 12 | | | | | |

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.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The 'Tel to IP Routing' configuration page is displayed. The 'Submit' button is highlighted in red. The configuration table shows a single entry for routing index 1 with the following values:

| Routing Index | Src. Trunk Group ID | Dest. Phone Prefix | Source Phone Prefix | Dest. IP Address | Port | Transport Type | Dest. IPGroup ID | Dest. SRD | IP Profile ID |
|---------------|---------------------|--------------------|---------------------|------------------|------|----------------|------------------|-----------|---------------|
| 1 | * | * | * | 47.166.92.205 | 5070 | TCP | -1 | -1 | 0 |
| 2 | | | | | | Not Configured | | | |
| 3 | | | | | | Not Configured | -1 | | |
| 4 | | | | | | Not Configured | -1 | | |
| 5 | | | | | | Not Configured | -1 | | |
| 6 | | | | | | Not Configured | -1 | | |
| 7 | | | | | | Not Configured | -1 | | |
| 8 | | | | | | Not Configured | -1 | | |
| 9 | | | | | | Not Configured | -1 | | |
| 10 | | | | | | Not Configured | -1 | | |

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 | Source 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.

The screenshot shows the AudioCodes MP-118 FXS_FXO web interface. The left sidebar contains a navigation tree with the following items: System, VoIP, Network, Security, Media, Applications Enabling, Control Network, SIP Definitions, Coders And Profiles, GW and IP to IP, Hunt Group, Manipulations, Routing, General Parameters, Tel to IP Routing, IP to Trunk Group Routing (selected), Alternative Routing Reasons, Forward On Busy Trunk, DTMF and Supplementary, Analog Gateway, Advanced Applications, and SAS. The main area displays the 'IP To Hunt Group Routing Table' configuration page. The table has 12 rows and 8 columns: Dest. Host Prefix, Source Host Prefix, Dest. Phone Prefix, Source Phone Prefix, Source IP Address, Hunt Group ID, IP Profile ID, and IP. The first four rows are populated with values from the table above, and the rest are empty. A red box highlights the 'Submit' button in the top navigation bar and the first four rows of the table.

| | Dest. Host Prefix | Source Host Prefix | Dest. Phone Prefix | Source Phone Prefix | Source IP Address | Hunt Group ID | IP Profile ID | IP |
|----|-------------------|--------------------|--------------------|---------------------|-------------------|---------------|---------------|----|
| 1 | | | 6000 | * | * | 1 | 0 | -1 |
| 2 | | | 6001 | * | * | 1 | 0 | -1 |
| 3 | | | 6111 | * | * | 1 | 0 | -1 |
| 4 | * | * | * | * | * | 2 | 0 | -1 |
| 5 | | | | | | | | |
| 6 | | | | | | | | |
| 7 | | | | | | | | |
| 8 | | | | | | | | |
| 9 | | | | | | | | |
| 10 | | | | | | | | |
| 11 | | | | | | | | |
| 12 | | | | | | | | |

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.

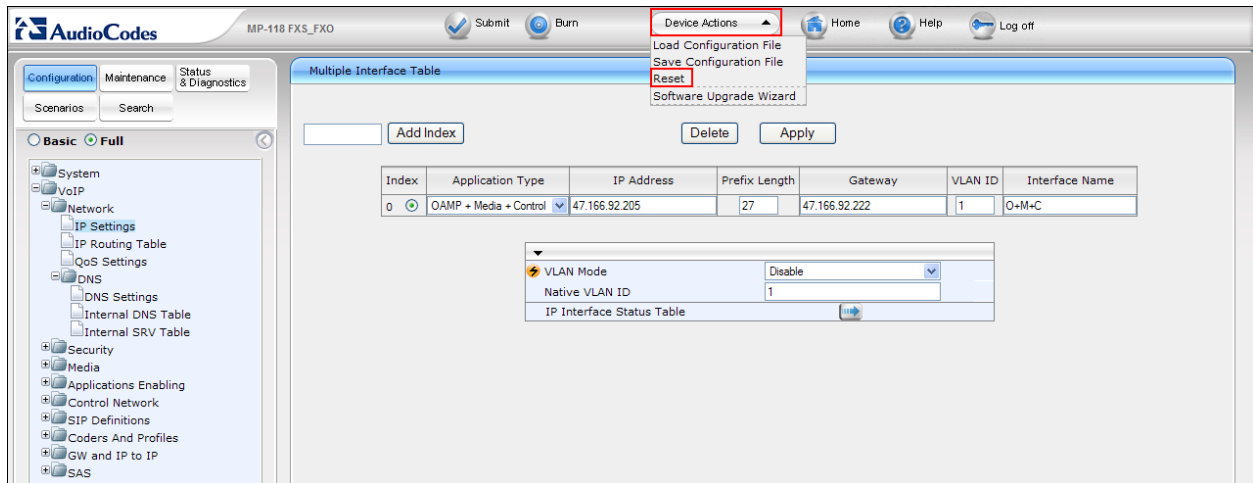
The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar contains a tree view with the following items: System, VoIP, Network, Security, Media, Applications Enabling, Control Network, SIP Definitions, Coders And Profiles, GW and IP to IP, Hunt Group, Manipulations, Routing, DTMF and Supplementary, Analog Gateway, Keypad Features, Metering Tones, Charge Codes, FXO Settings, Authentication, Automatic Dialing, Caller Display Information, Call Forward, Caller ID Permissions, Call Waiting, Advanced Applications, and SAS. The 'FXO Settings' item is selected. The main panel displays the 'FXO Settings' configuration page. At the top of the main panel, there are buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The 'Submit' button is highlighted with a red box. The 'FXO Settings' table has the following rows:

| Setting | Value |
|--|-----------|
| Dialing Mode | One Stage |
| Waiting for Dial Tone | No |
| Time to Wait before Dialing [msec] | 1000 |
| Ring Detection Timeout [sec] | 8 |
| Reorder Tone Duration [sec] | 255 |
| Answer Supervision | No |
| Rings before Detecting Caller ID | 1 |
| Send Metering Message to IP | No |
| Disconnect Call on Busy Tone Detection (CAS) | Disable |
| Disconnect On Dial Tone | Disable |
| Guard Time Between Calls | 1 |
| FXO AutoDial Play BusyTone | Disable |

6.19. Resetting the Gateway

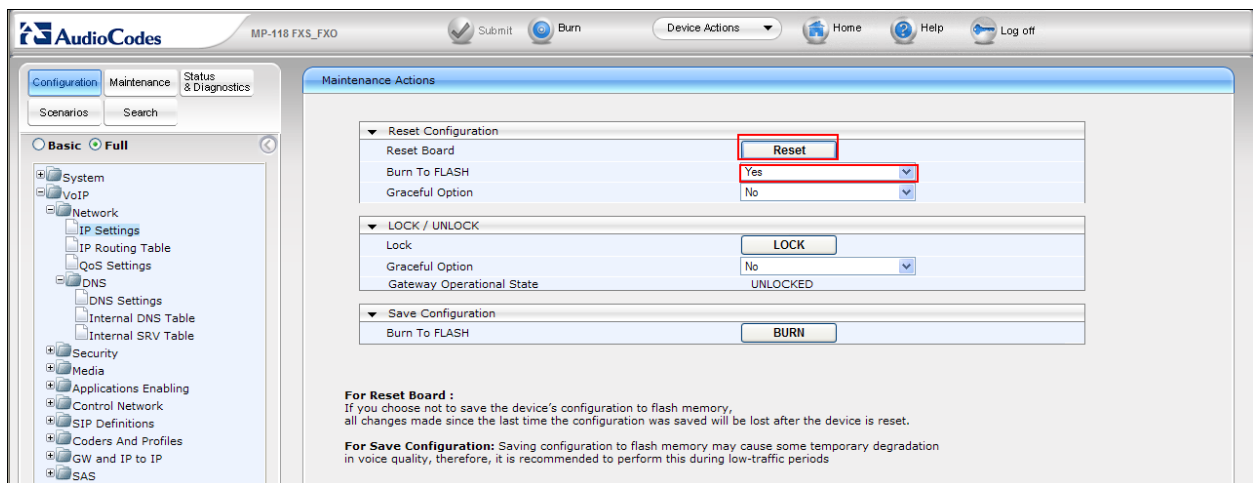
After configuring **IP Settings (Section 6.2)** and **Application Enabling (section 6.6)** a Gateway Reset needs to be preformed. The following steps are required:

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



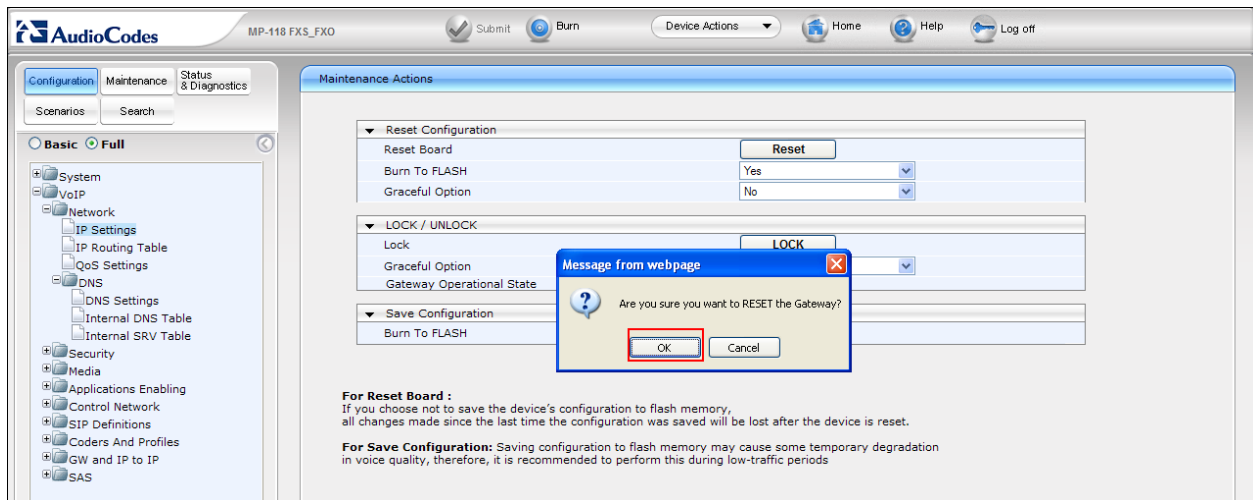
Once the next page opens the following steps are required:

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



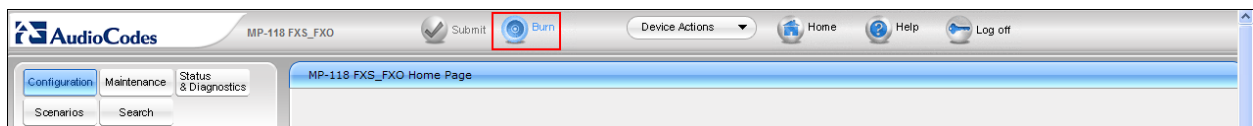
Once the **Message from webpage** window opens click the **OK** button.

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

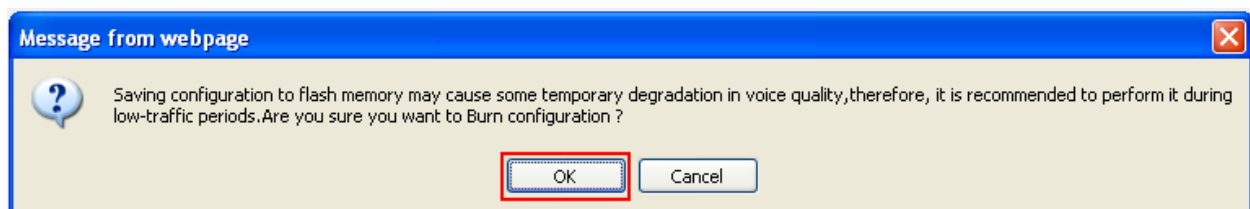


6.20. Saving all Configurations to Flash Memory

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

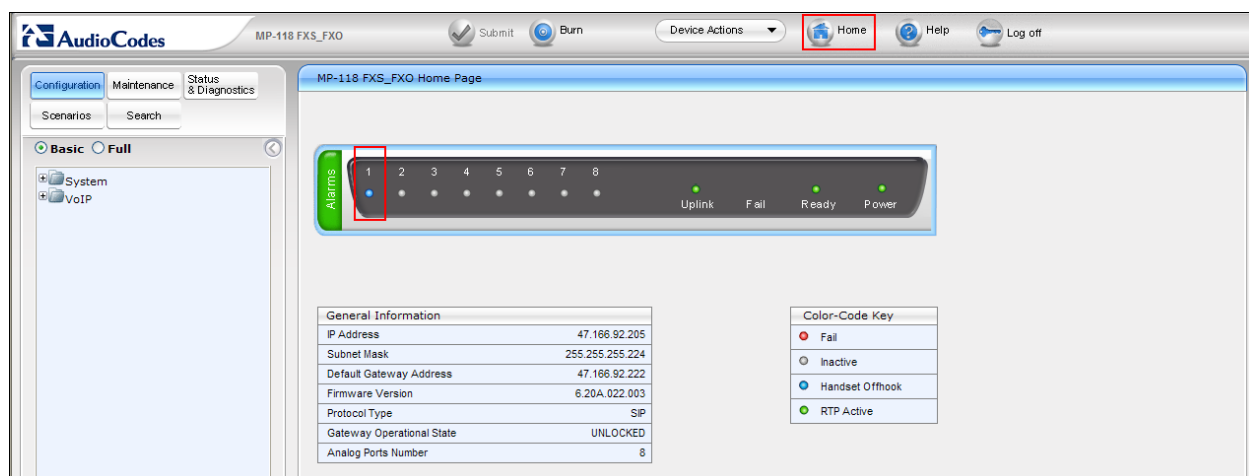
In Normal Mode with the Ethernet link connected check the status of the D-channel created in **Section 5.1.1** by running the command **STAT DCH** in Overlay 96 (**LD 96**) as shown below. The example below shows that D-Channel 66 is operational and established.

LD 96

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

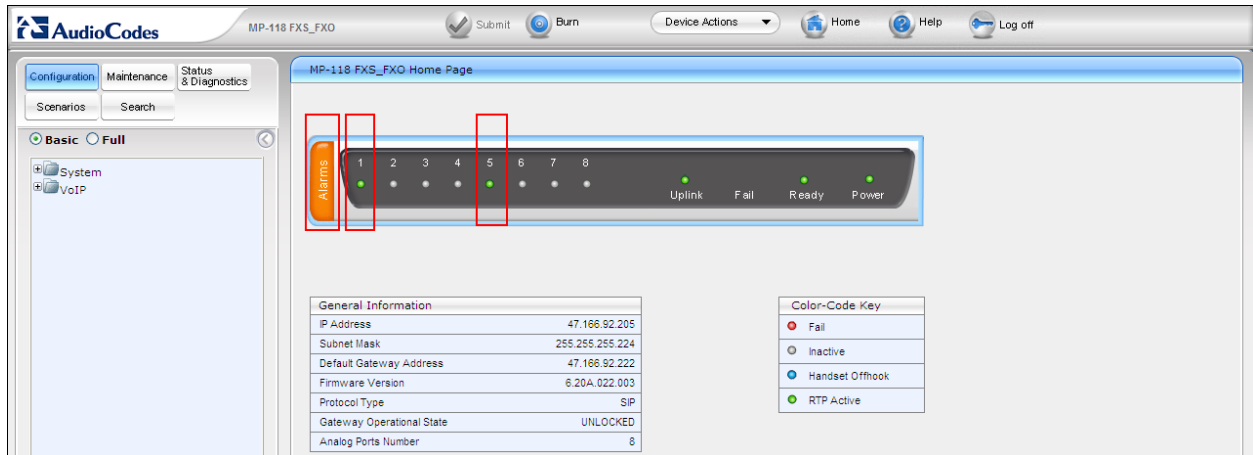
7.2. AudioCodes MP-118 Normal Mode Status

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 | ISS1:10F1 | p30595_1 | 14/06/2011 | p30595_1.cpl | NO |
| 001 | wi00835294 | ISS1:10F1 | p30565_1 | 14/06/2011 | p30565_1.cpl | NO |
| 002 | wi00832106 | ISS1:10F1 | p30550_1 | 14/06/2011 | p30550_1.cpl | NO |
| 003 | wi00837618 | ISS1:10F1 | p30594_1 | 14/06/2011 | p30594_1.cpl | NO |
| 004 | wi00852365 | ISS1:10F1 | p30707_1 | 14/06/2011 | p30707_1.cpl | NO |
| 005 | wi00843623 | ISS1:10F1 | p30731_1 | 14/06/2011 | p30731_1.cpl | YES |
| 006 | wi00839255 | ISS1:10F1 | p30591_1 | 14/06/2011 | p30591_1.cpl | NO |
| 007 | wi00832626 | ISS2:10F1 | p30560_2 | 14/06/2011 | p30560_2.cpl | NO |
| 008 | wi00857566 | ISS1:10F1 | p30766_1 | 14/06/2011 | p30766_1.cpl | NO |
| 009 | wi00841980 | ISS1:10F1 | p30618_1 | 14/06/2011 | p30618_1.cpl | NO |
| 010 | wi00837461 | ISS1:10F1 | p30597_1 | 14/06/2011 | p30597_1.cpl | NO |
| 011 | wi00839821 | ISS1:10F1 | p30619_1 | 14/06/2011 | p30619_1.cpl | NO |
| 012 | wi00842409 | ISS1:10F1 | p30621_1 | 14/06/2011 | p30621_1.cpl | NO |
| 013 | wi00838073 | ISS1:10F1 | p30588_1 | 14/06/2011 | p30588_1.cpl | NO |
| 014 | wi00850521 | ISS1:10F1 | p30709_1 | 14/06/2011 | p30709_1.cpl | YES |
| 015 | wi00860722 | ISS1:10F1 | p30784_1 | 14/06/2011 | p30784_1.cpl | YES |
| 016 | wi00839134 | ISS1:10F1 | p30698_1 | 14/06/2011 | p30698_1.cpl | YES |
| 017 | wi00836981 | ISS1:10F1 | 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

[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
SIPPREROUTINGMODE = 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_MWIAAnalog, 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_CNMode, 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_SBCDiversionsMode,
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 ]
```

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