Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Interalia XMU+ with Avaya Communication Server 1000E R7.5 and Avaya Aura® Session Manager R6.1using a SIP connection via AudioCodes MediaPack 118 Gateway – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Interalia's XMU+ system to successfully interoperate with Avaya Communication Server 1000E R7.5 and Avaya Aura® Session Manager R6.1 using a SIP connection. An AudioCodes MediaPack 118 is used to connect the Interalia XMU+ to Avaya Communication Server 1000E using a SIP trunk. Interalia's XMU+ is a voice application platform that supports Recorded Announcements, Music on Hold and basic IVR technology.

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 compliance tested configuration using Interalia's XMU+ solution which includes AudioCodes MediaPack 118 with Avaya Communication Server 1000E (CS1000E) R7.5, Avaya Aura® System Manager R6.1 and Avaya Aura® Session Manager R6.1. The Interalia XMU+ has an analog connection to the AudioCodes MediaPack 118 which in turn is connected to the Avaya Aura® Session Manager R6.1 using a SIP trunk

The Interalia's XMU+ is a microprocessor-based voice application platform that supports multiple applications simultaneously on a port-by-port basis. Included with every XMU+ is the XMUCOM+ administration software, a windows-based configuration and communication software that helps administrators directly manage XMU+ systems onsite. The software has a GUI interface, editor browser and pull-down menus with tools administrators need to schedule messages, simultaneously download configuration files/messages to multiple XMU+ systems, and review statistics.

Typical Interalia's XMU+ applications include.

- ACD/UCD announcements
- Auto attendant
- Information Lines
- Music on hold (MOH)

The Interalia system used for the test consists of an Interalia XMU+ server, an AudioCodes MediaPack 118 is used to provide a SIP trunk to the Avaya Communication Server 1000E via a SIP connection managed by Avaya Aura® Session Manager R6.1.

2. General Test Approach and Test Results

2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature and functionality testing. The feature and functionality testing focused on verifying that the voice application response is activated on the XMU+ in various scenarios.
The testing includes:

- Verification of connectivity between the MP118 and CS1000E using SIP trunks
- Verification that basic Recorded Announcements (RAN) in various telephony operations using RAN announcement applications on the XMU+
- Verification that interactive voice response occurs in various telephony operations using Information Lines application on the XMU+
- Failover testing of the XMU+ and MP118 systems

The compliance testing focused on testing a SIP trunk connection to the CS1000E. The testing was performed using a series of group hunt lists associated with various XMU+ unique applications. The failover testing focused on verifying the ability of the XMU+ and MP118 systems to recover from disconnection such as power supply failure.

Note: The CS1000E was configured for A Companding Law

2.2. Test Results

The test approach was to validate the correct operation of typical interactive voice response applications such as ACD Announcements, etc. The following results were obtained:

- Confirmation that interactive voice messages are played as expected in different call scenarios
- Confirmation that messages are routed successfully as expected
- Confirmation of good quality audio in all test cases
- Successful recovery of XMU+ after failover testing

The tests were all functional in nature and performance testing was not included. All the test cases passed successfully.

2.3. Support

Technical support for Avaya products can be obtained from http://support.avaya.com. Technical support for the XMU+ and AudioCodes MP118 can be obtained as follows;

- Email: support@interalia.com
- Website: www.interalia.com
- Phone: +1 800 531 0115 (Toll Free)

3. Reference Configuration

Figure 1 shows the network topology during compliance testing. System Manager R6.1, Session Manager R6.1 and Communication Server 1000E running R7.5 software on a CPPM card were used as the hosting PBX. The XMU+ is connected to the hosting PBX using a SIP connection from the MediaPack 118 Gateway. XMUCOM+ is management software installed on a client PC. For compliance testing the XMUCOM+ was installed on a Windows XP operating system.

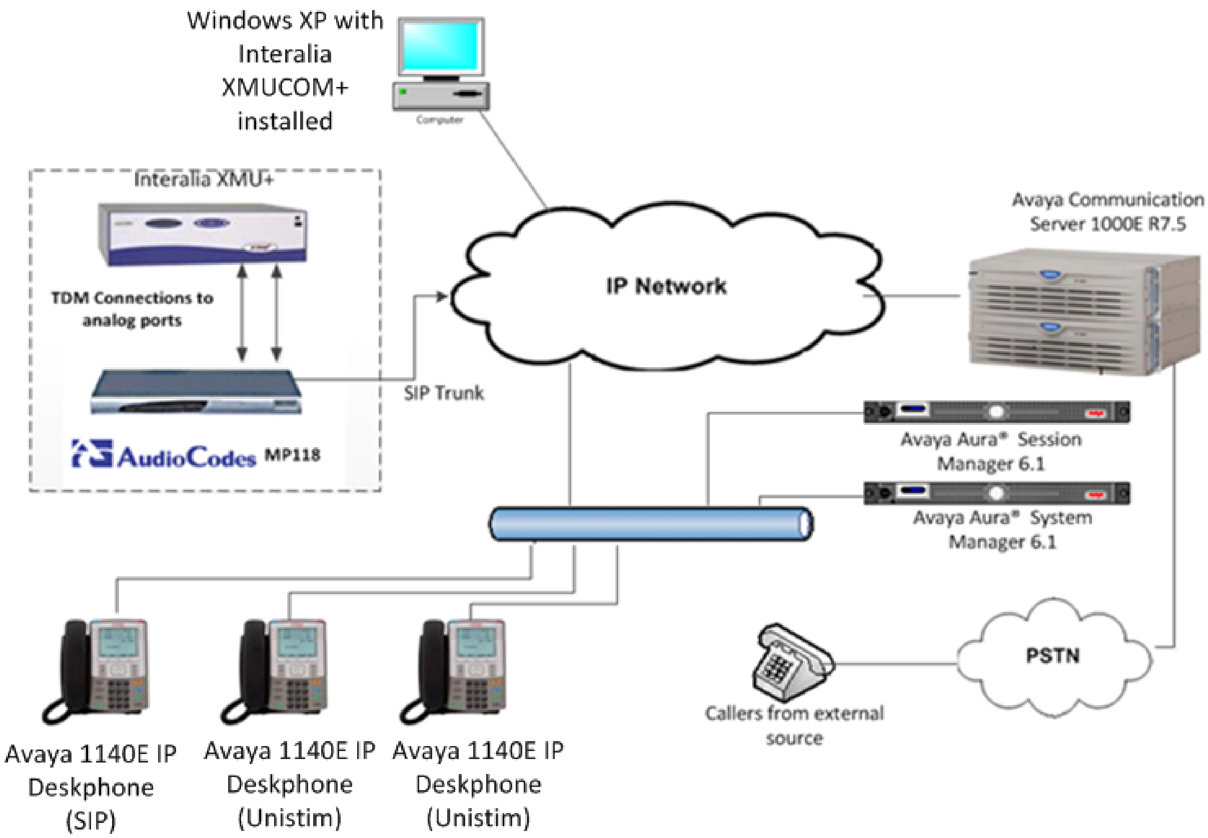


Figure 1: Network Topology and Connectivity for Interalia XMU+ and Avaya CS1000E

4. Equipment and Software Validated

All the hardware and associated software used in the compliance testing is listed below.

Equipment	Software/Firmware
Avaya PBX Products	
Avaya Communication Server 1000E CPPM	Avaya Communication Server 1000E R7.5 SP1
Avaya S8800 Media Server	Avaya Aura® System Manager R6.1 Build 6.1.0023
Avaya S8800 Media Server	Avaya Aura® Session Manager R6.1 Build 6.1.0012
Avaya 1140E IP Deskphone	UNIStim 4.3 SIP V04.00
Interalia's Equipment	
Interalia XMU+	Firmware Version: 6.85 XMUCOM+ 7.25 on Windows XP
AudioCodes MP118 Gateway	MP-118 /8FXS/3AC

5. Configure Avaya Communication Server 1000E

Configuration and verification operations on the CS1000E illustrated in this section were all performed using terminal access over a serial link to a TTY port on the CS1000E using Telnet. Configuration of the Session Manager were performed using a web GUI provided by the System Manager. The information provided in this section describes the configuration of the CS1000E for this solution. However it does not show the complete setup of ACD Queues and all external trunks and routes as it is implied a working system is already in place. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 9**. The configuration operations described in this section can be summarized as follows:

- Configuring SIP D-Channel and SIP route on CS1000E
- Adding SIP Trunks
- Creating a Coordinated Dialing Plan to access the XMU+
- Creating group hunt lists for the various XMU+ applications
- Creating a Pilot DN
- Creating the MP118 as a SIP Entity in Session Manager
- Creating a routing Pattern in Session manager for the MP118 Entity
- Creating a Routing Policy
- Creating a Dial Pattern

Note 1: Licenses are required for some configurations (example SIP Trunks).
Note 2: The configuration of the PRI interface to the PSTN is outside the scope of these Application Notes.

5.1. Creating a SIP D-Channel and SIP route

In the examples below D-channel and Route 20 were used.

5.1.1. Create D-Channel

Use the **CHG** command in **LD 17** to create a virtual D-channel for a SIP connection.

Note: In the Telnet screenshots below only the unique prompt inputs are shown. Carriage Return all other prompts to set default values.

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
CTYP	**DCIP**	Card type is IP D-Channel
USR	**ISDL**	Integrated Services Signaling Link
IFC	**SL1**	D-Channel interface type

5.1.2. Create a SIP Route

Use the **NEW** command in **LD 16** to create a new SIP route. The route created is a **TIE** route in order to connect to the MP118.

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	00001	Zone number associated with the route
PCID	SIP	Protocol for the route

5.2. Add SIP Trunks

Use the **NEW** command in **LD 14** to add virtual SIP trunks to the new route created in **Section 5.1.2.** If adding multiple trunks for each route 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	[L S C U]	Loop Shelf Card Unit
CUST	0	Customer Number as defined in LD15
RTMB	20 1	Route number and Member number

5.3. Adding a Coordinated Dialing Plan

There are a number of ways to setup a dialing plan to call the ports on the MP118. For the compliance testing a Coordinated Dialing Plan (CDP) was used. In order to create a CDP a Route List Index (RLI) in overlay 86 is required. Use the **NEW** command in **LD 86** to create a **RLI**.

Note: Enter the SIP route **(ROUT)** that 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	36	Route list Index number
ENTR	0	First entry for the RLI
ROUT	20	Enter the SIP route number

5.3.1. Create CDP

Use the **NEW** command in **LD 87** to create a CDP entry for the MP118 ports. If there are 4 ports on the MP118, 4 CDP entries are created corresponding to those port extensions. If there are 8 ports, 8 CDP entries are created.

Note: The RLI number used is the one created in **Section 5.3.**

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
RLI	36	Route list index Number

5.4. Creating New Group Hunt Lists

If there are a number of channels on the XMU+ associated with the same service then these channels will be added to a group hunt on the CS1000E in order to access using a single route point or number. Depending on the nature of the service on the XMU+, the associated channel will be a part of a specific hunt group. On the XMU+ there are up to 64 available ports for use.

For compliance testing four ports were used for IVR front line services, two ports for a first RAN message and two ports for a second RAN message. Three different hunt groups were setup with four CDP entries in the first hunt group associated with the IVR channels, two CDP entries in the second hunt group associated with the first RAN message and two CDP entries in the third hunt group associated with the second RAN message. Use the **NEW** command in **LD 18** to create a new group hunt list.

LD 18

Prompt	**Response**	**Description**
>	**LD 18**	Enter Overlay 18
REQ	**NEW**	Create new
TYPE	GHT	Group Hunt
LSNO	1	Group Hunt List Number
CUST	0	Customer Number as defined in LD15
SIZE	1-96	Amount of entries in the GHT
STOR	0 3220	x is the entry number and y is the CDP number

5.5. Creating a Pilot DN

Create a pilot DN which is the number associated with the hunt group. Use the **NEW** command in **LD 57**.

Note: The List Number **(LSNO)** used is the one created in **Section 5.4.**

LD 57

Prompt	**Response**	**Description**
>	**LD 57**	Enter Overlay 57
REQ	**NEW**	
TYPE	FFC	Flexible Feature Codes Data Block
CODE	PLDN	Pilot DN (Group hunt access DN)
USE	GPHT	Use is Group Hunt
LSNO	1	Use the list number created in section 5.1.6
HTYP	LIN/RRB	Linear or Round Robin

5.6. Creating the AudioCodes MP118 as a SIP Entity on the Avaya Aura® Session Manager

To create the AudioCodes MP118 as a SIP Entity on the Session Manager, The System Manager is used. The following must be configured.

- SIP Entity
- SIP Entity Details

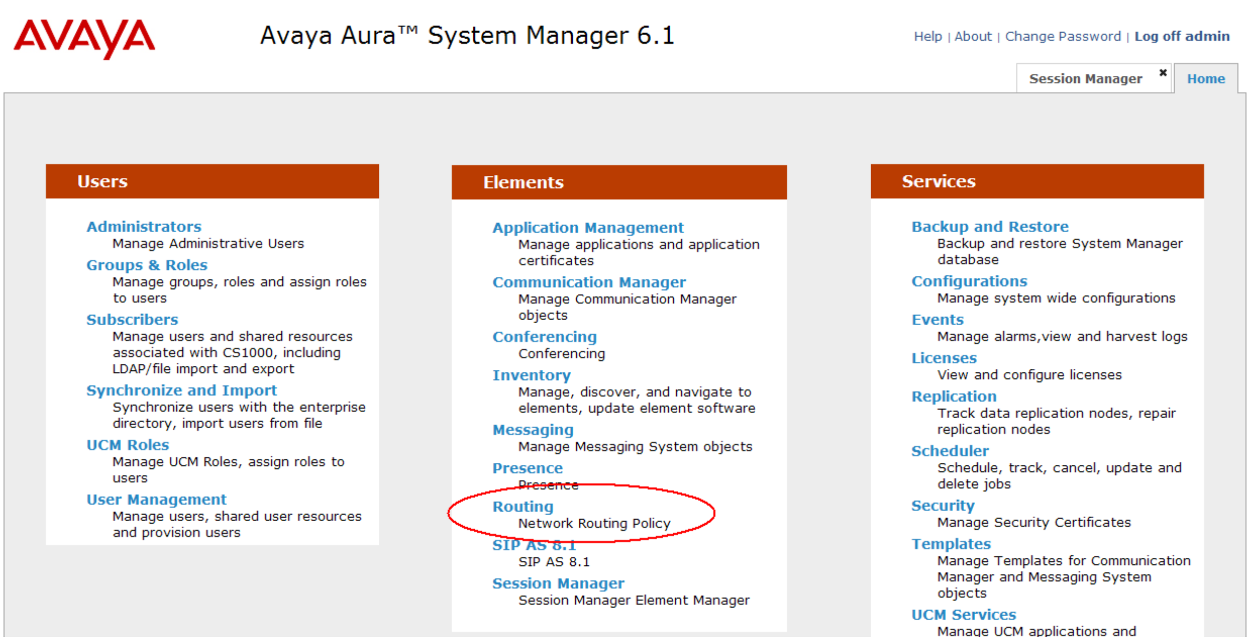
Note: To get more information for any input field you can press the **Help** link at anytime.

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager, using the URL "http://<fqdn>/SMGR" or "http://<ip-address>/SMGR", where "<fqdn> is the fully qualified domain name of Avaya Aura® System Manager or the "<ipaddress>" is the IP address of Avaya Aura® System Manager.

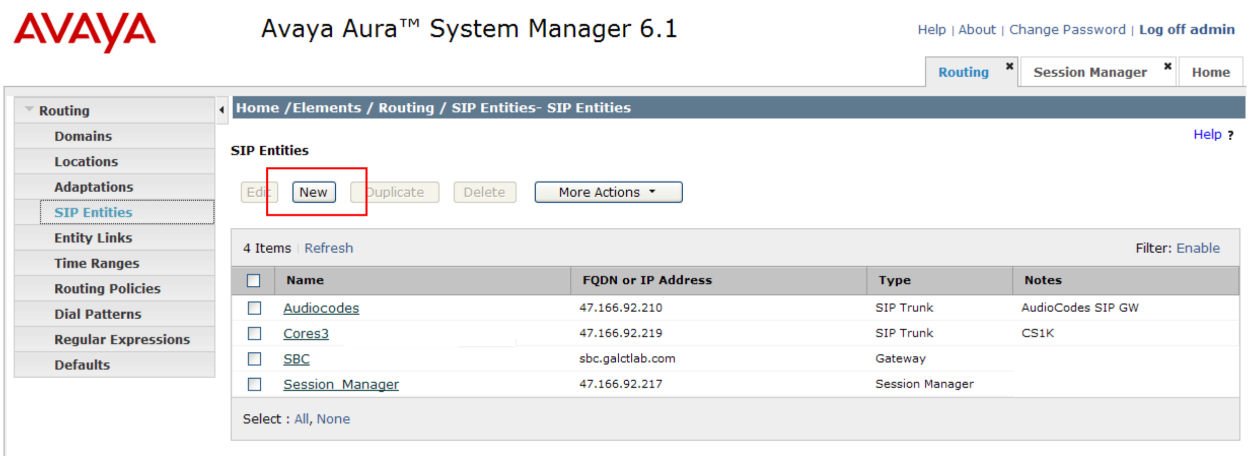
Log in with the appropriate credentials. Once logged in select the **Routing** Link under the **Elements** column.

5.6.1. Create a SIP Entity

Click **Routing** as highlighted below.



In the Routing page navigate to **SIP Entities** and click **New** as shown below.

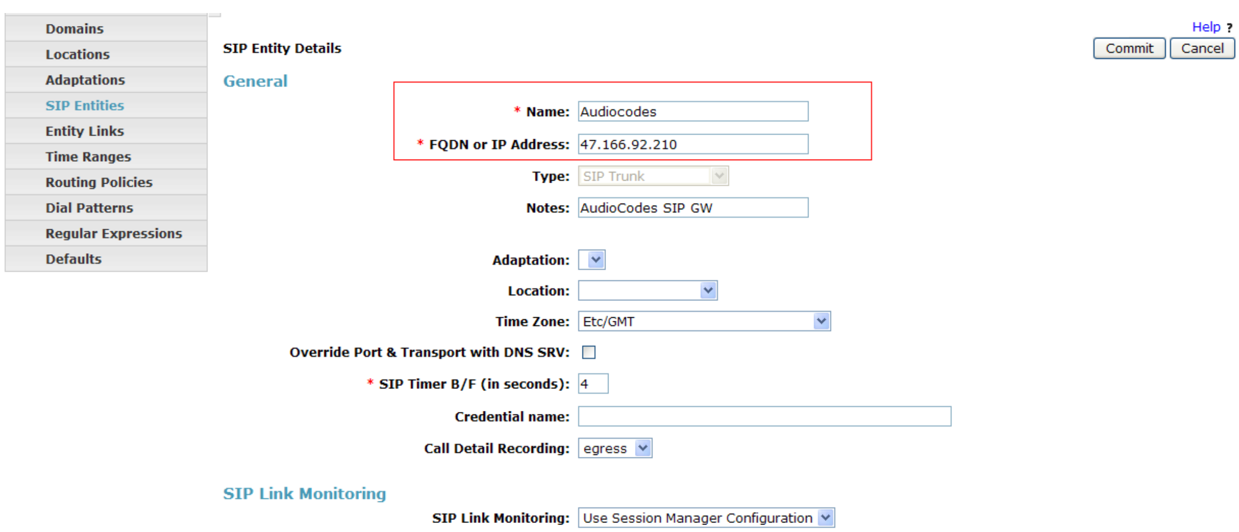


5.6.2. SIP Entity Details

To create a SIP Entity for the MP118, enter the following.

- **Name** Input descriptive name
- **FQDN or IP Address** FQDN or IP Address of the MP118

Click **Commit** when all information is filled in.



5.7. Routing Calls to the AudioCodes MP118 SIP Entity

To route calls to the MP118 create an Entity Link.

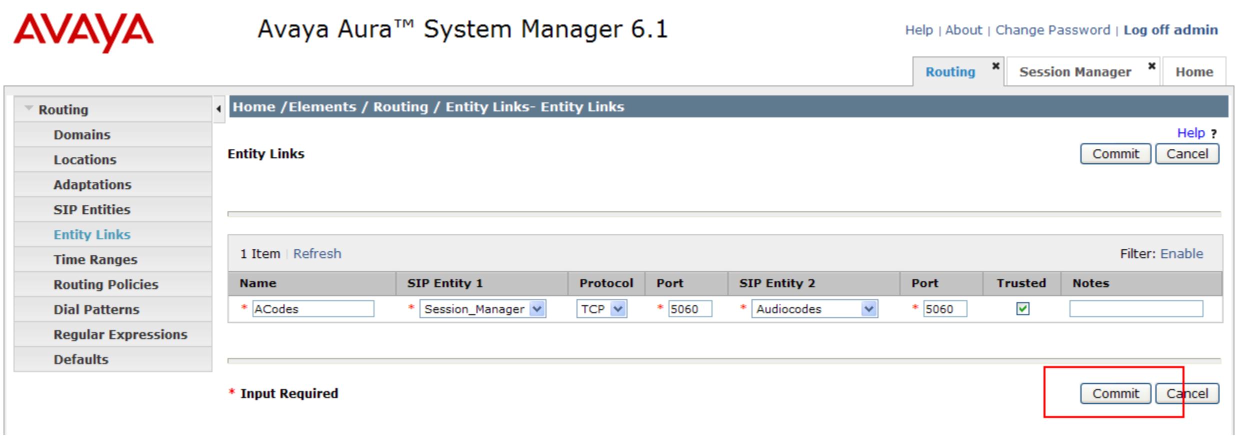
- Create an Entity Link

5.7.1. Create an Entity Link

Enter the following information below.

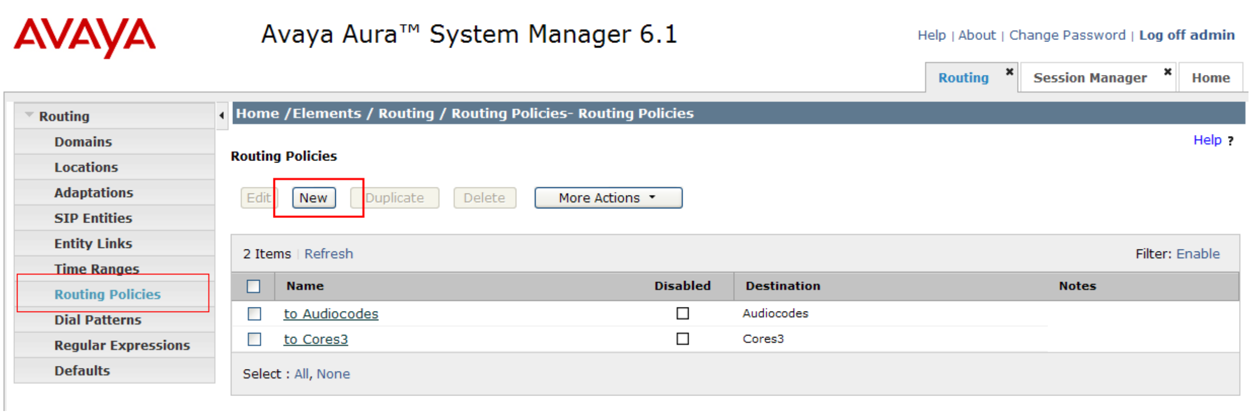
- **Name** Enter a Name for the Entity Link
- **SIP Entity 1** **Session Manager**
- **Protocol** **TCP**
- **Port** **5060**
- **SIP Entity 2** SIP Entity **Name** as created in **Section 5.6.2**

Click **Commit** when all information is filled in.



5.8. Create a Routing Policy

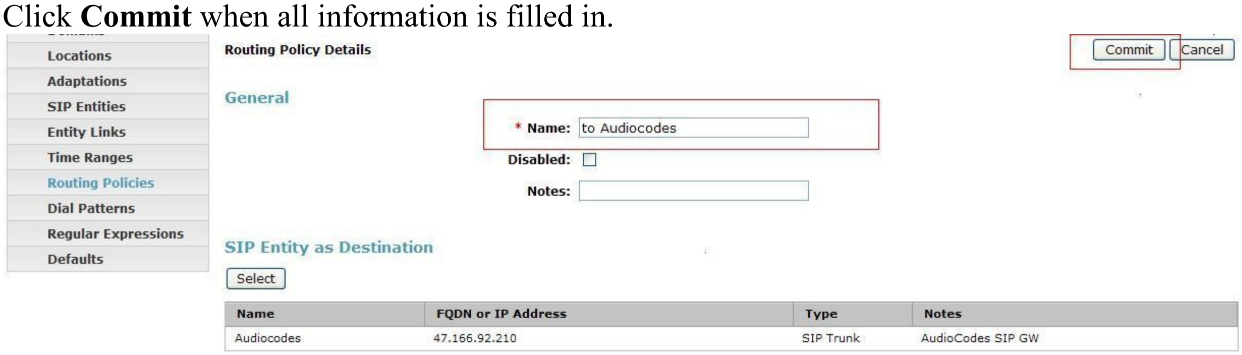
In the **Routing** page navigate to **Routing Policies** and then click **New** to create a Routing Policy, as highlighted below.



Enter the following information below.

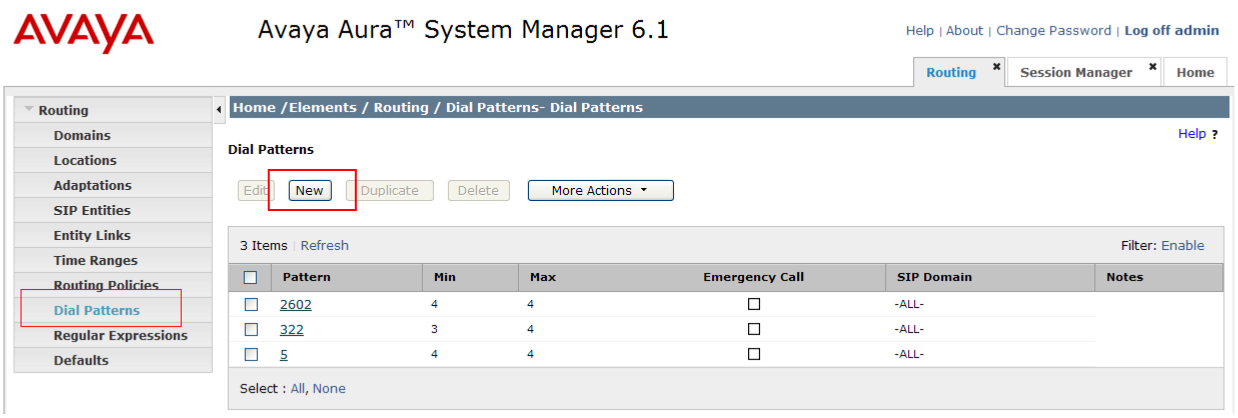
- **Name** Enter a Name for the Routing Policy

Click **Commit** when all information is filled in.



5.9. Create a Dial Pattern

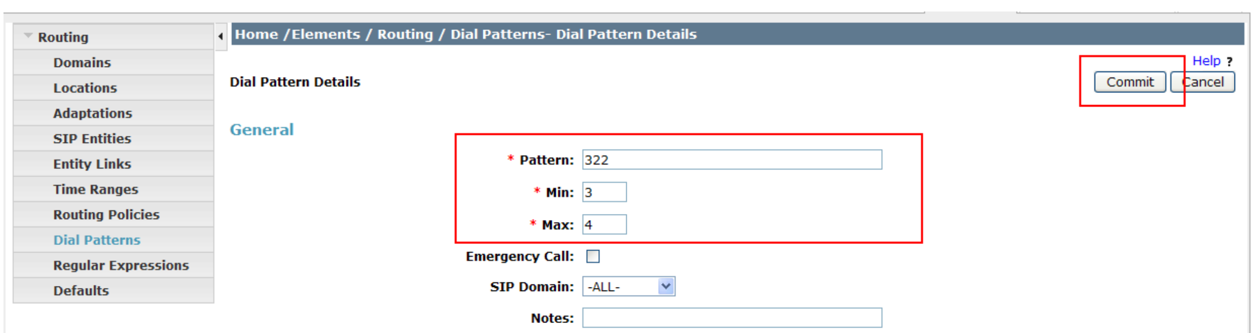
In the **Routing** page navigate to **Dial Patterns** and then click **New** to create a Dial Pattern, as highlighted below.



The digits that are to be routed to the MP118 are entered against **Pattern.** Enter the following below:

- **Pattern** **322**
- **Min** **3** i.e. min number of digits in the Dial Pattern
- **Max** **4** i.e. max number of digits in the Dial Pattern

Click **Commit** when all information is filled in. In the example below all numbers from 322 to 3229 will be routed to the MP118 IP address over SIP as the **Pattern** is set to **322**.



6. Configuring the AudioCodes MP118 VoIP Gateway

This section describes the procedures for configuring the MP118. These procedures assume the MP118 has been assigned an IP address.

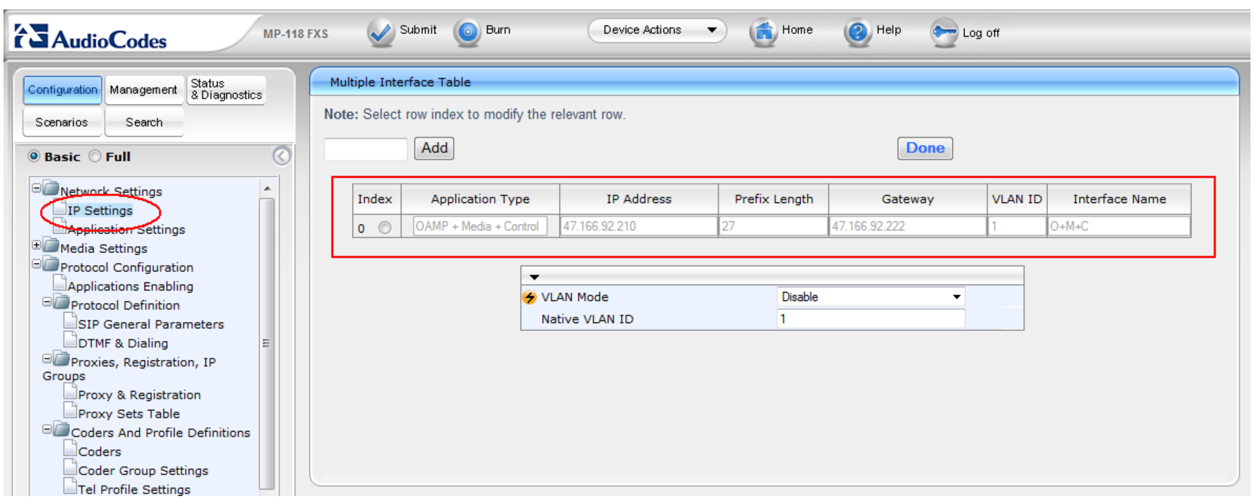
Configure the MP118 to act as a SIP Gateway bridge between the XMU+ and the CS1000E via Session Manager. Only Basic configuration settings are required in order to set up the MP118. The XMU+ is connected to the RJ11 ports on the rear of the MP118 and all functionality of the XMU+ is available.

Note: As the XMU+ performs call transfers the call is placed on hold in the AudioCodes MP118 hence no Music On Hold is heard.

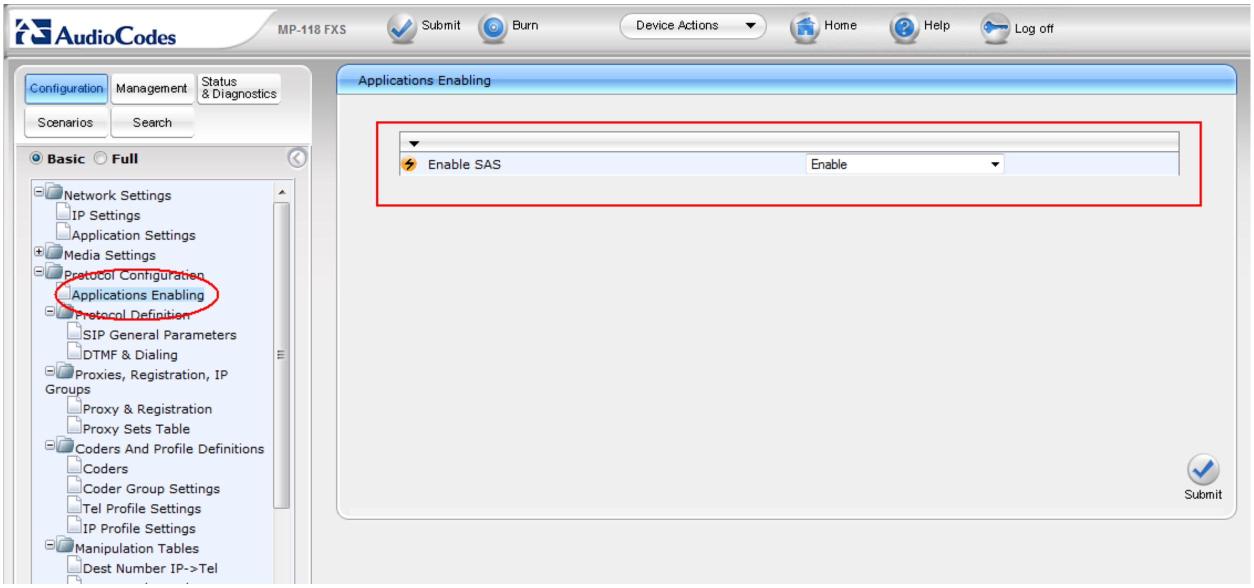
6.1. IP Settings on the AudioCodes MP118

When connected to the MP118 choose the **Configuration** tab and navigate to **IP Settings** and enter the following:

- **IP Address** IP address of the AudioCodes MP118
- **Gateway** IP address of the Gateway that the MP118 resides



Navigate to **Applications Enabling** and enable **SAS.**

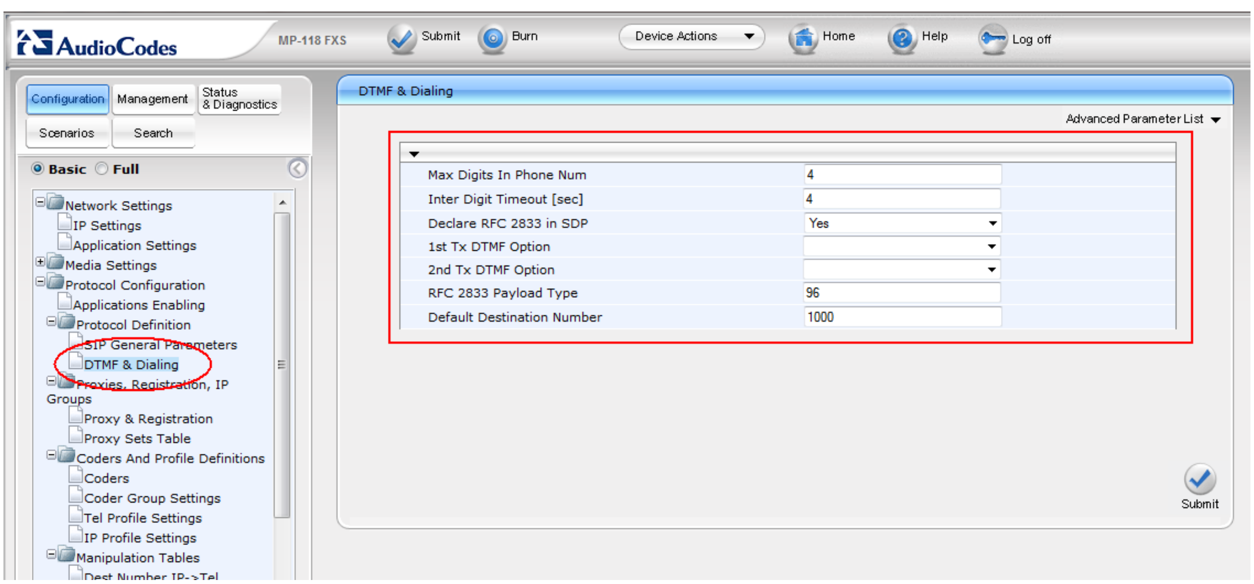


6.2. AudioCodes MP118 Protocol Configuration

This section describes setting the MP118 to talk with the Session Manager.

6.2.1. Protocol Definition

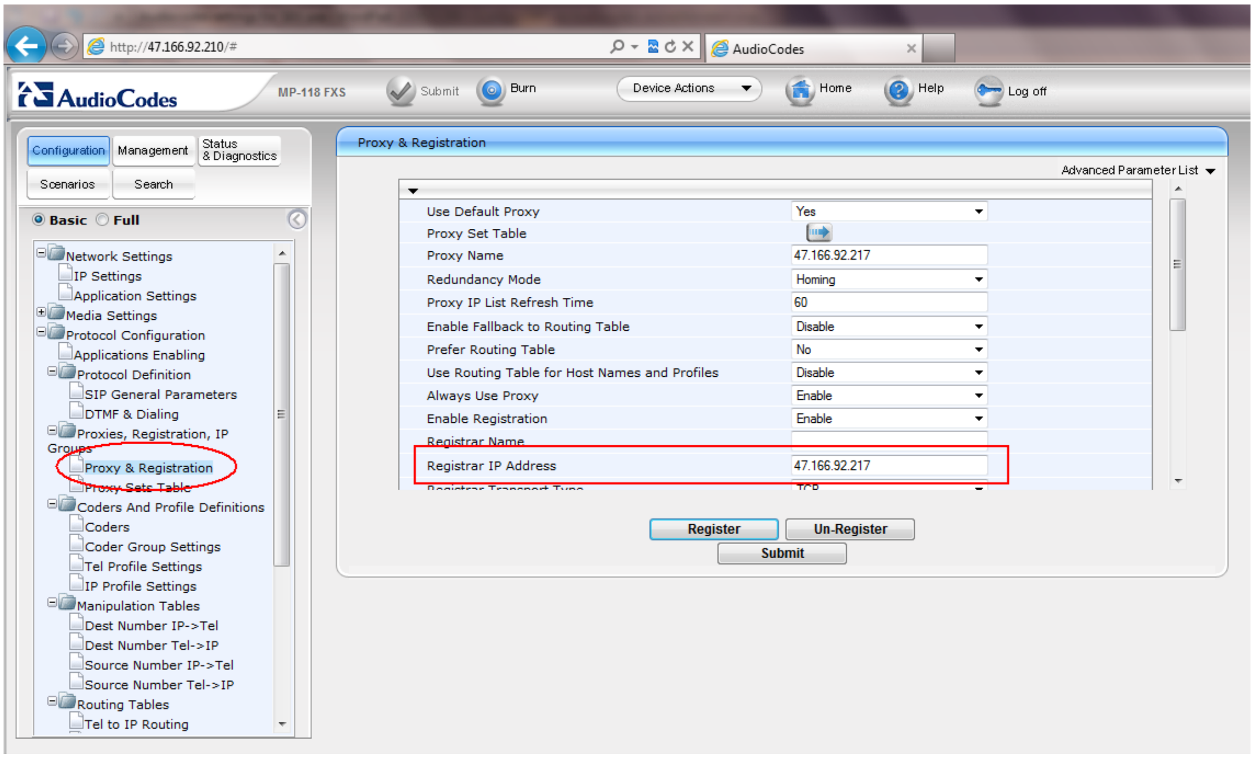
Navigate to **DTMF & Dialing** and enter the correct number of digits for the dial plan.



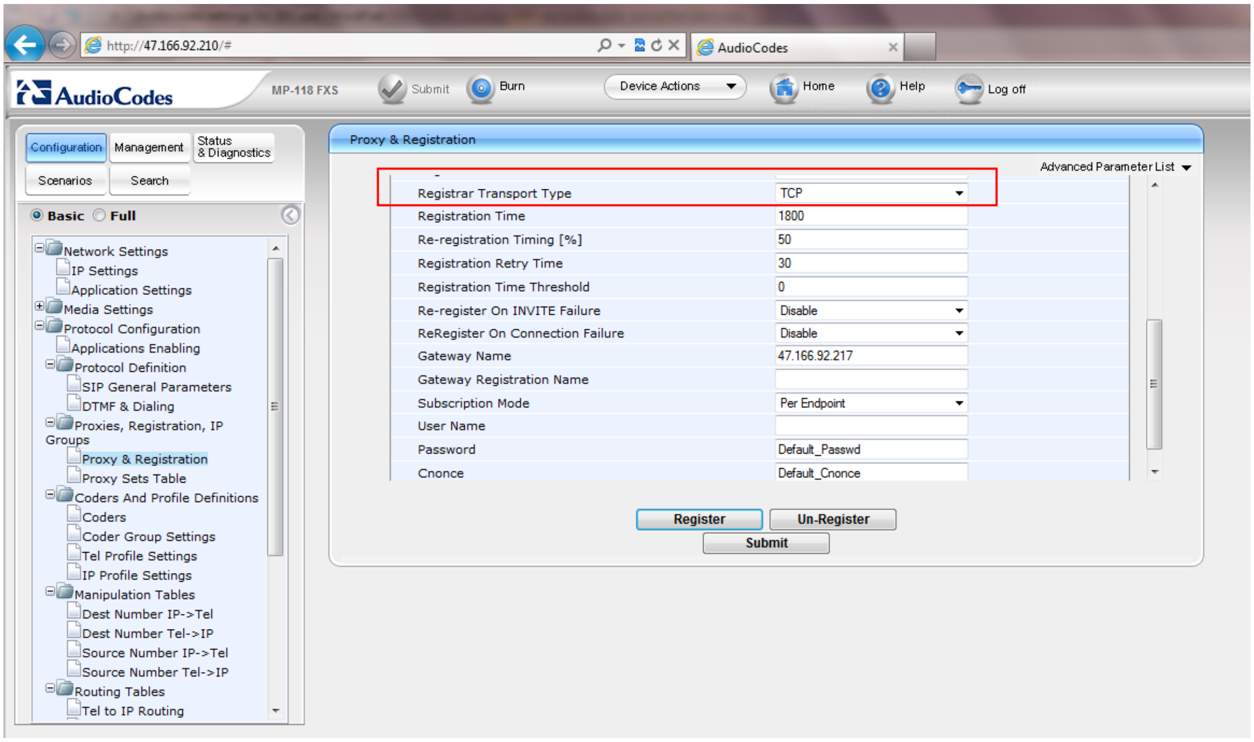
6.2.2. Groups

Navigate to **Proxy & Registration** and enter the following.

- **Registrar IP Address** Enter the IP address of the Session Manager the MP118 is registering to.
- **Registrar Transport Type** TCP

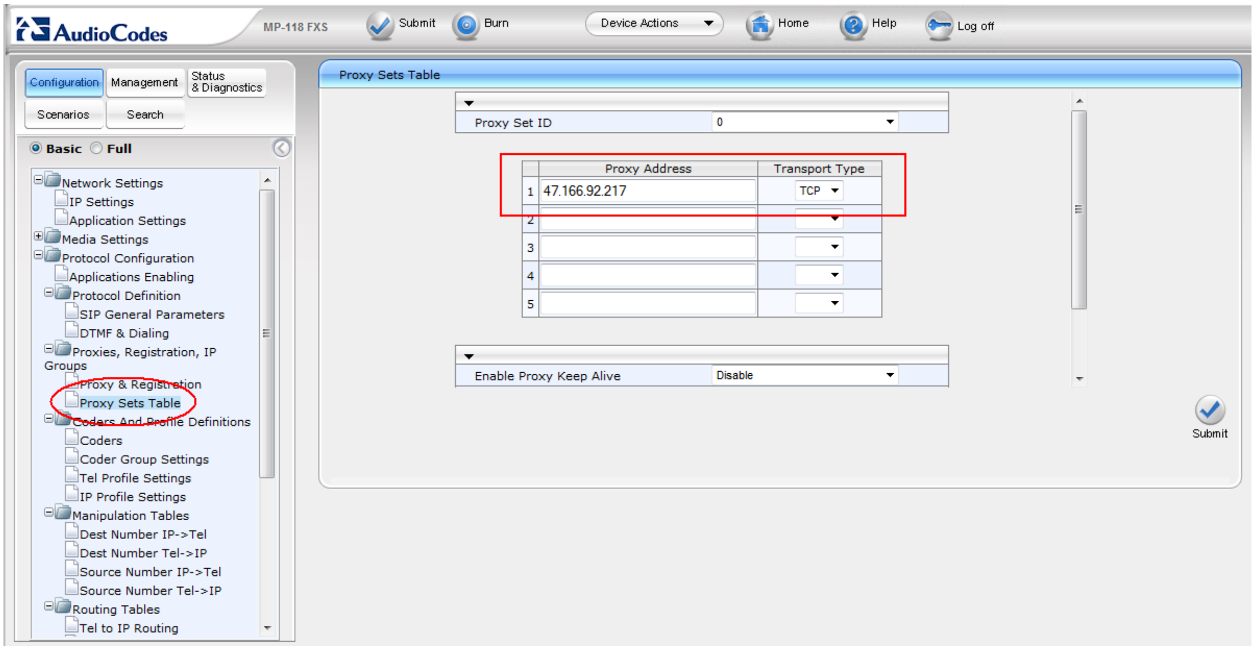


Continue to fill in the **Registrar Transport Type** as shown below **TCP** is selected.



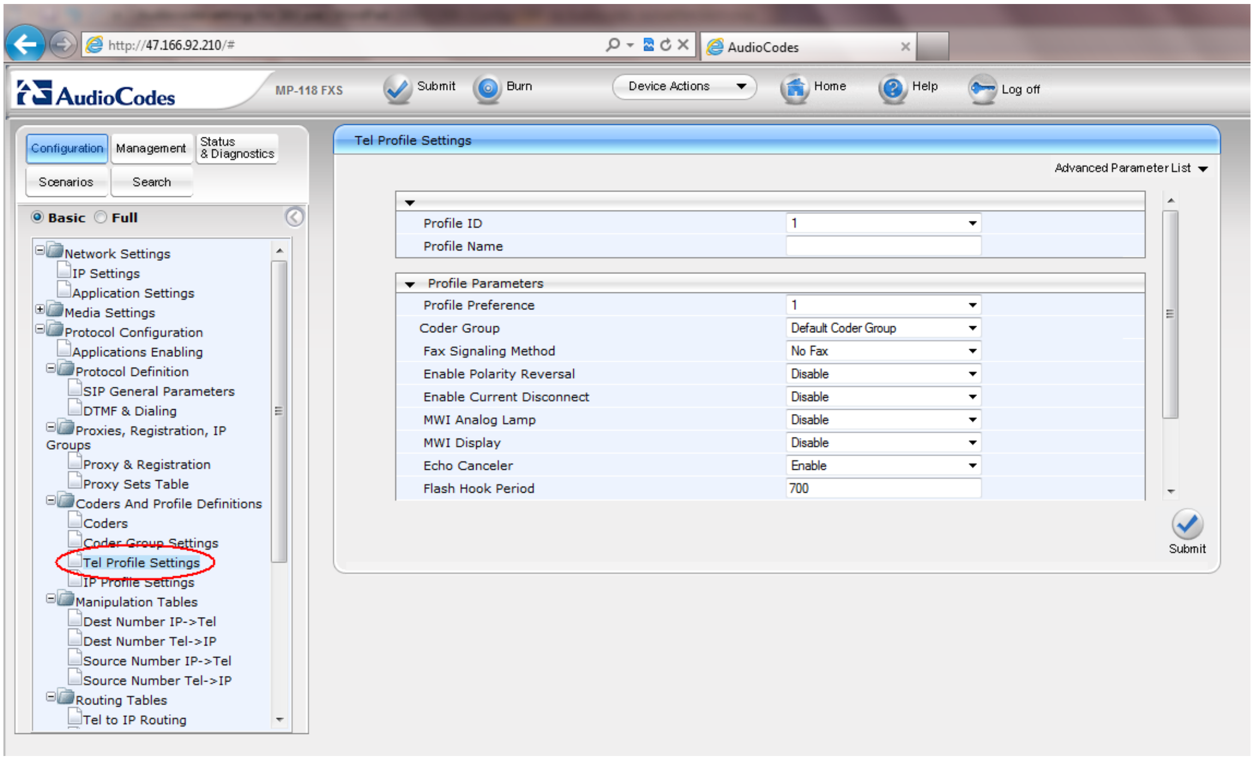
Navigate to **Proxy Set Table** and enter the following.

- **Proxy Address** Session Manager IP address
- **Transport Type** **TCP**

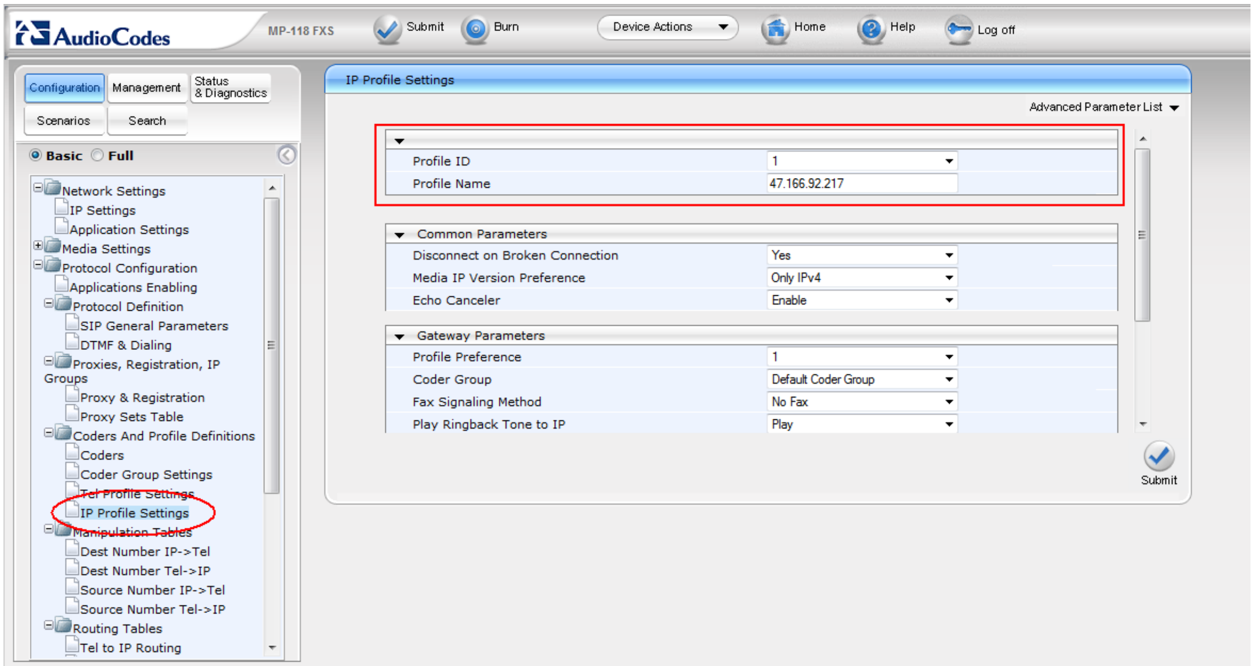


6.2.3. Coders and Profile Definitions

Navigate to **Tel Profile Settings** and enter **1** for **Profile ID**. This number will be used for later configurations.



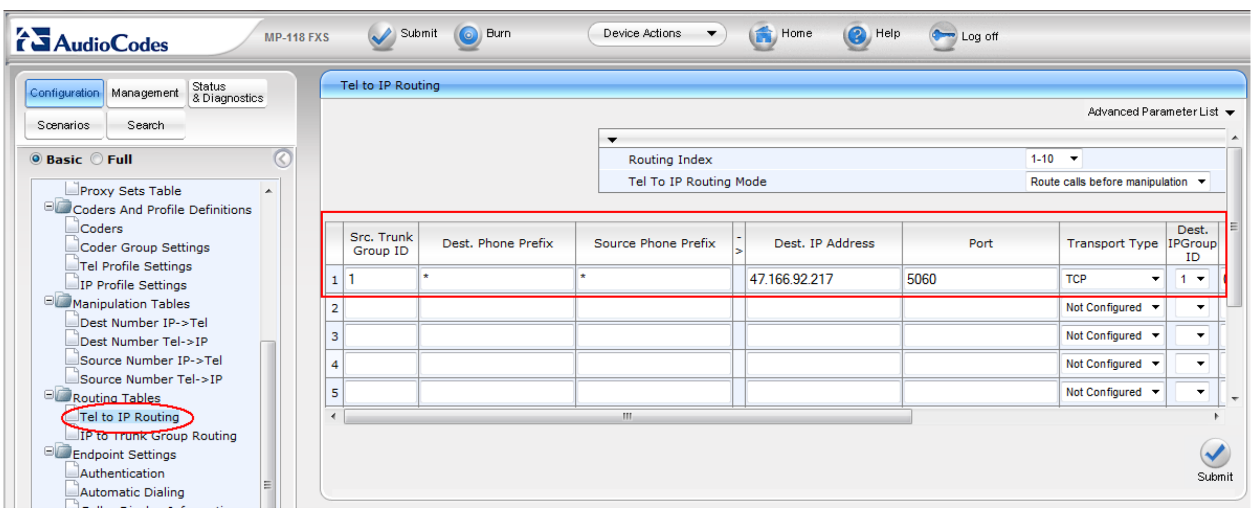
Navigate to **IP Profile Settings**, and select the **Profile ID** that was configured in the previous screenshot.



6.3. AudioCodes MP118 Routing table

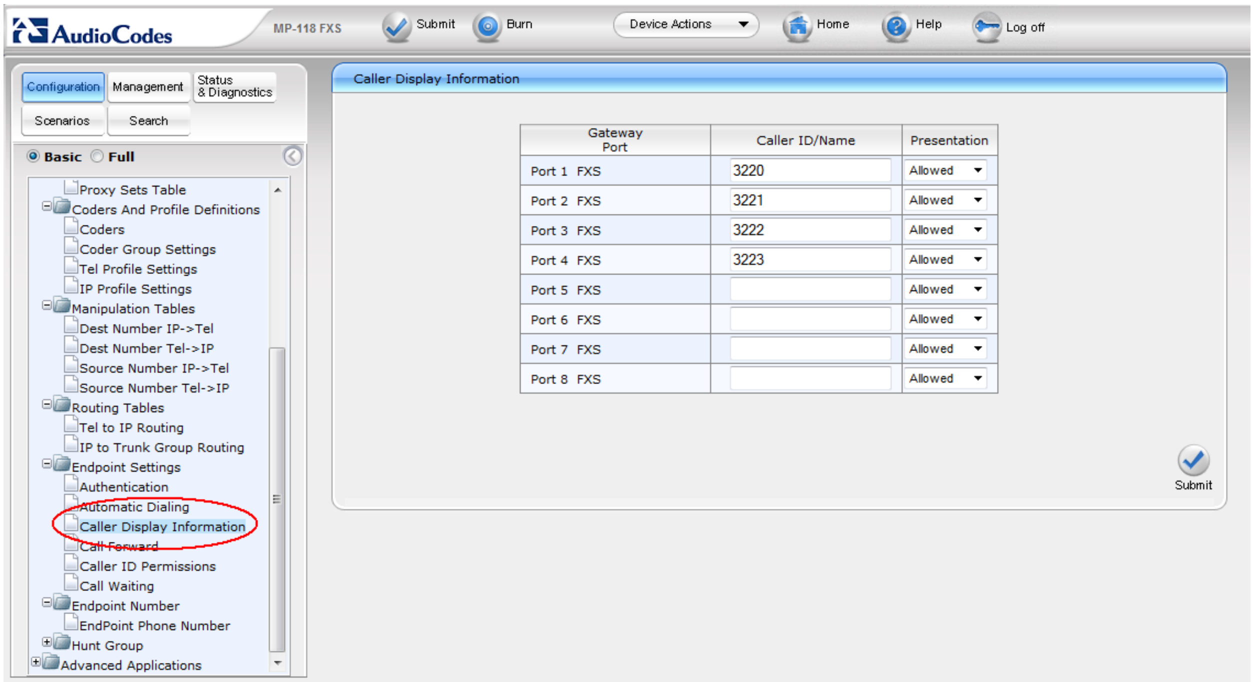
Under the setting **Tel to IP Routing** all calls are routed to the Session Manager.

Note: "*" is entered in the destination and source for the prefix that donates **all**.



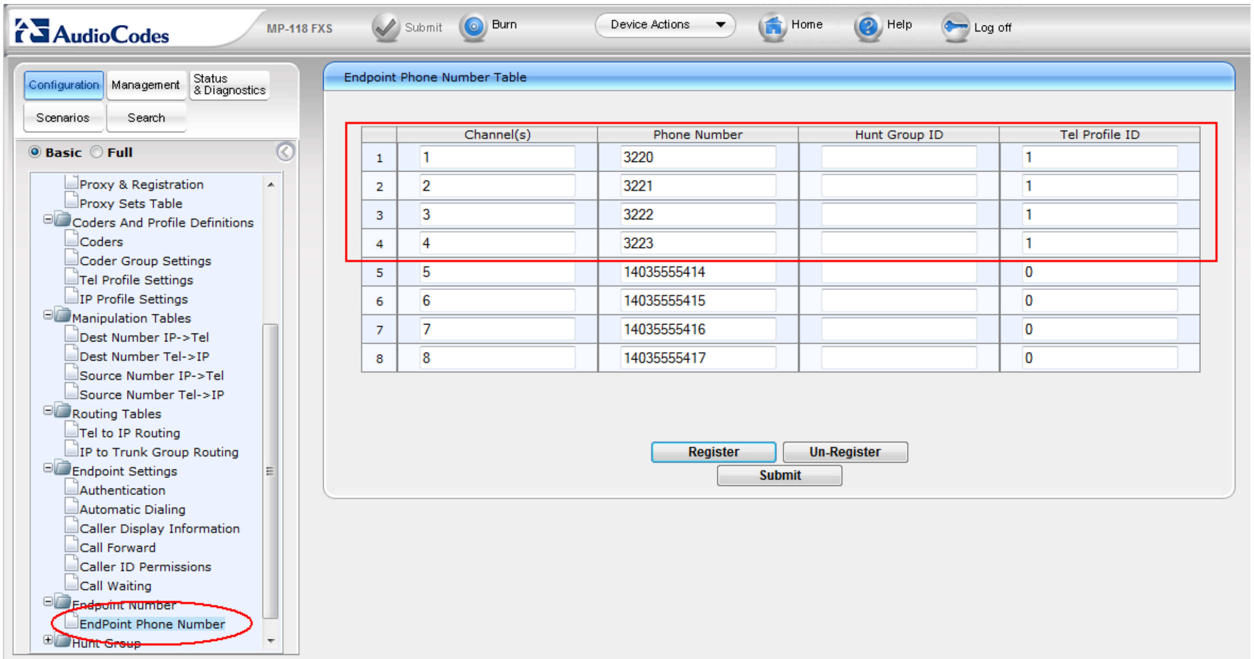
6.4. AudioCodes MP118 Endpoint Settings

Under the setting **Caller Display Information**, the CLID that should be displayed for extension numbers created on specific ports are entered.



6.5. AudioCodes MP118 Endpoint Number

Under the heading **Endpoint Phone Number**, the extension numbers are assigned to the MP118 analog ports. The example below shows the first four FXS extensions being configured. The **Tel Profile ID** value was created in **Section 6.2.3.**

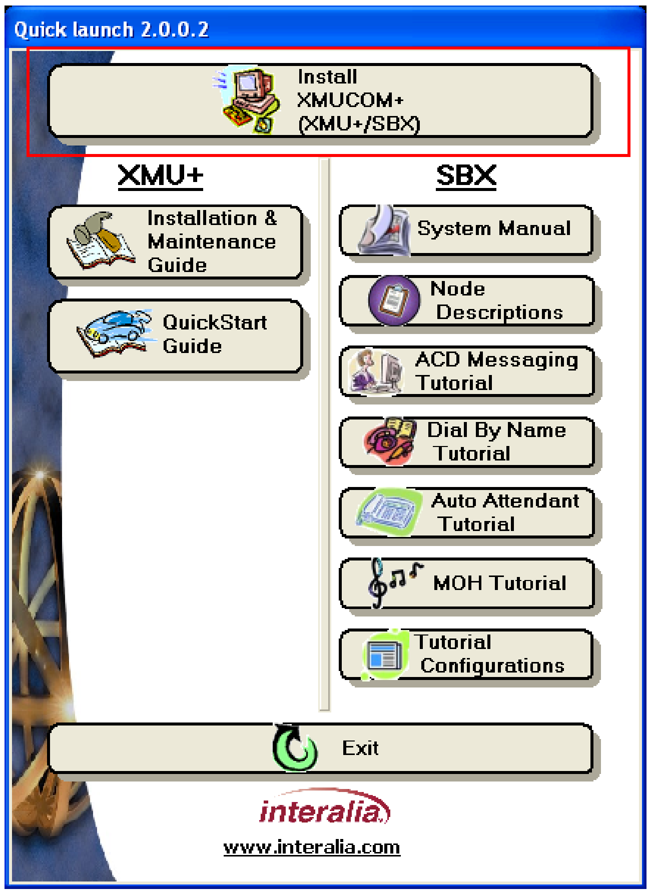


6.6. Configure Interalia XMU+

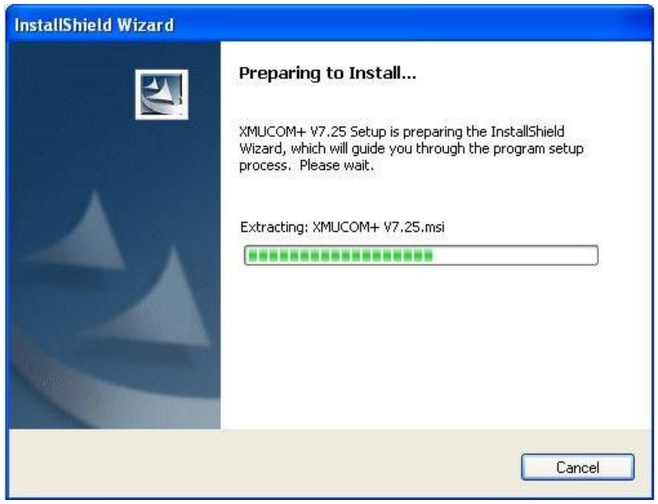
The following section documents the necessary steps taken to configure the XMU+.

6.6.1. Installing Interalia XMU+

The installation of the XMU+ is from a CD containing the software. After placing the CD into a PC it automatically starts to the following screen. Click on **Install XMUCOM+ (XMU+/SBX)** highlighted.

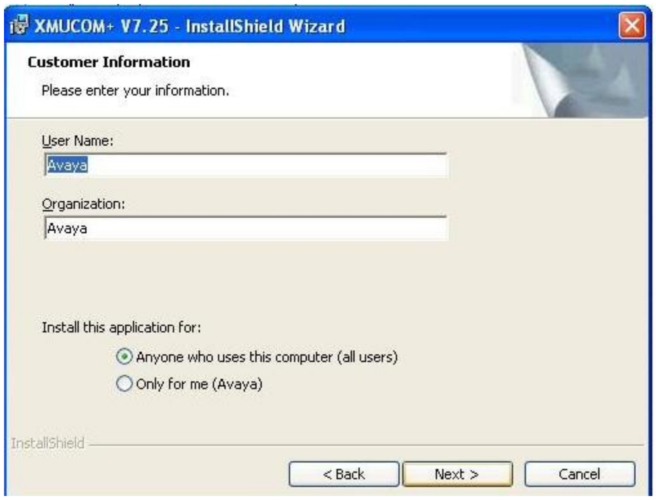


A **Preparing to Install** screen appears below.

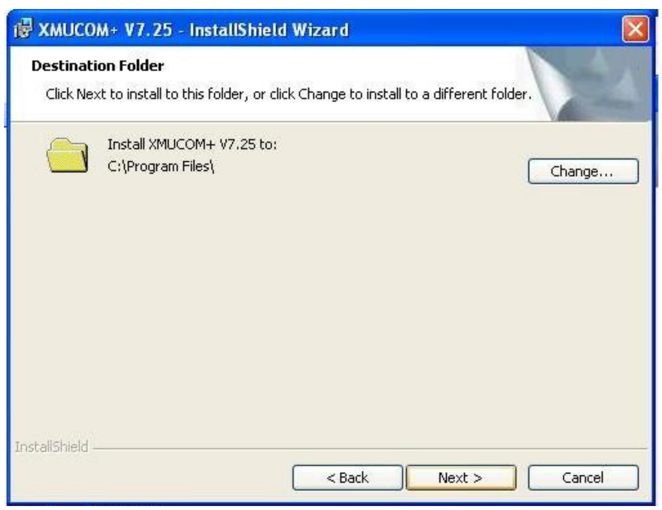


A welcome screen appears. Click **Next** to continue with the install.

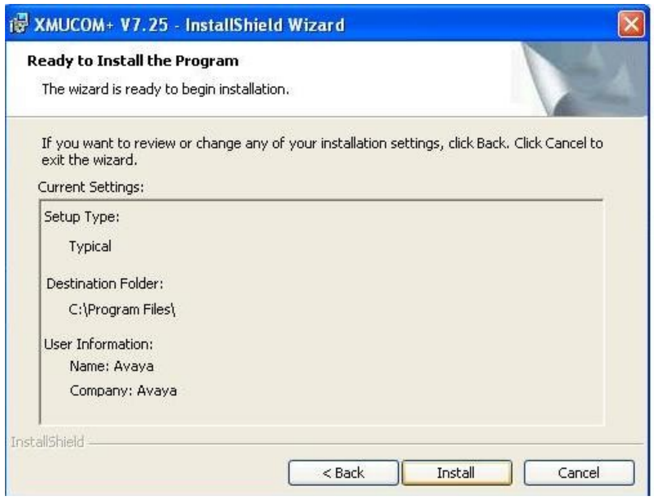
On the **Customer Information** screen, enter **User Name** and **Organization** and click the **Next** button.



Choose the default destination on the **Destination Folder** screen and the **Next** button.

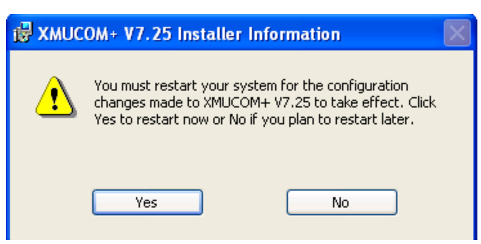


Review the selected settings and click **Install**.



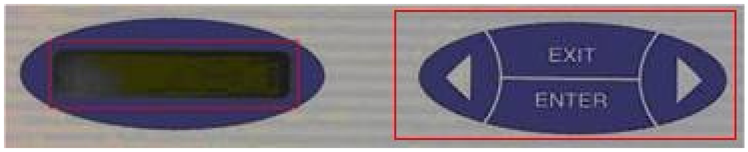
A screen appears to indicate the successful install of the product. Click **Finish** to complete it.

Restart the PC to allow configuration changes to take effect. Choose the **Yes** button to restart.



6.6.2. Setting the IP Information on Interalia XMU+

The IP information of the XMU+ is set from the menu on the front of the box by navigating through the menu using the left and right arrows and clicking **Enter** for each section that needs changing as highlighted below.

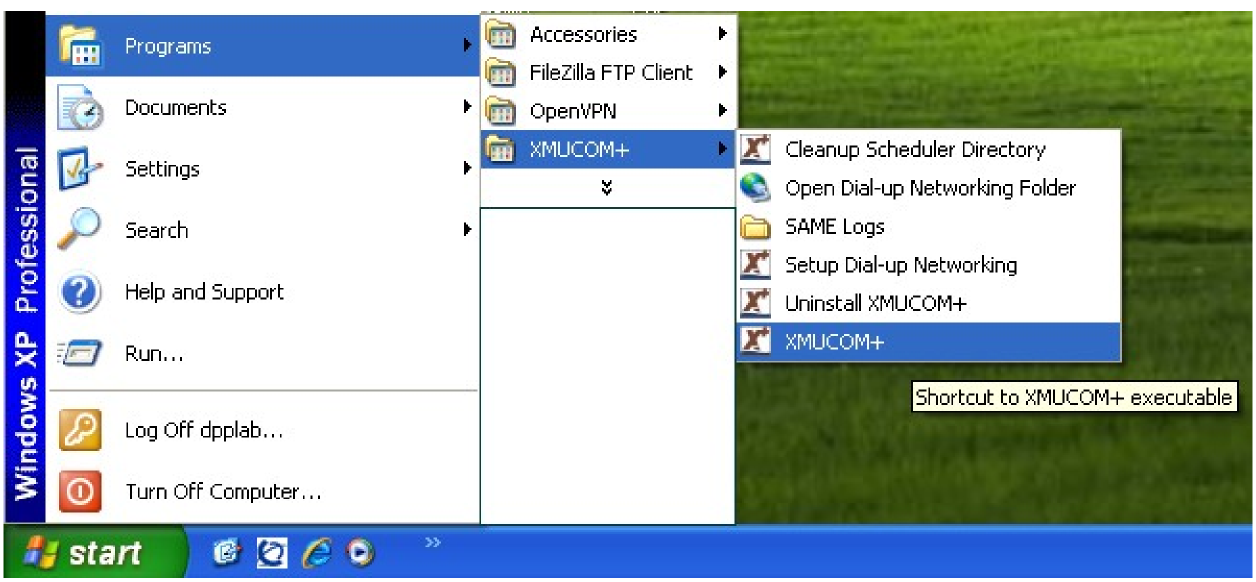


To set the IP information, carry out the following steps:

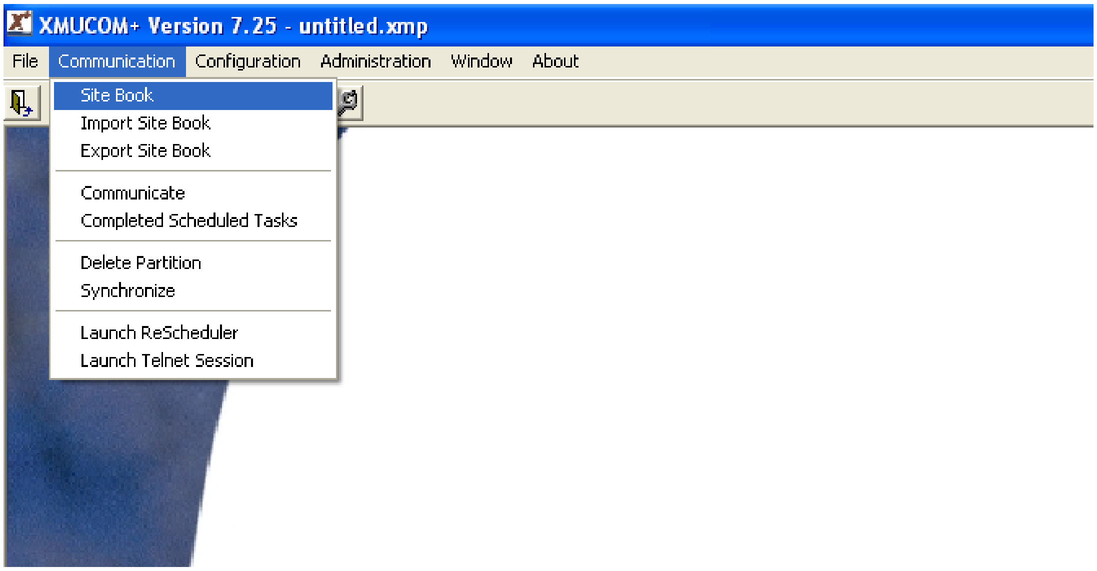
- Navigate to **Main System Menu** and press **Enter**
- Navigate to **System Communications** and press **Enter**
- Navigate to **Communications Ethernet** and press **Enter**
- Navigate to **Ethernet IP address**, enter the IP address and press **Enter**
- Navigate to **Ethernet IP mask**, enter the Subnet Mask and press **Enter**
- Navigate to **Gateway** enter the Default Gateway and press **Enter**

6.6.3. Create a Site Book on the Interalia XMU+

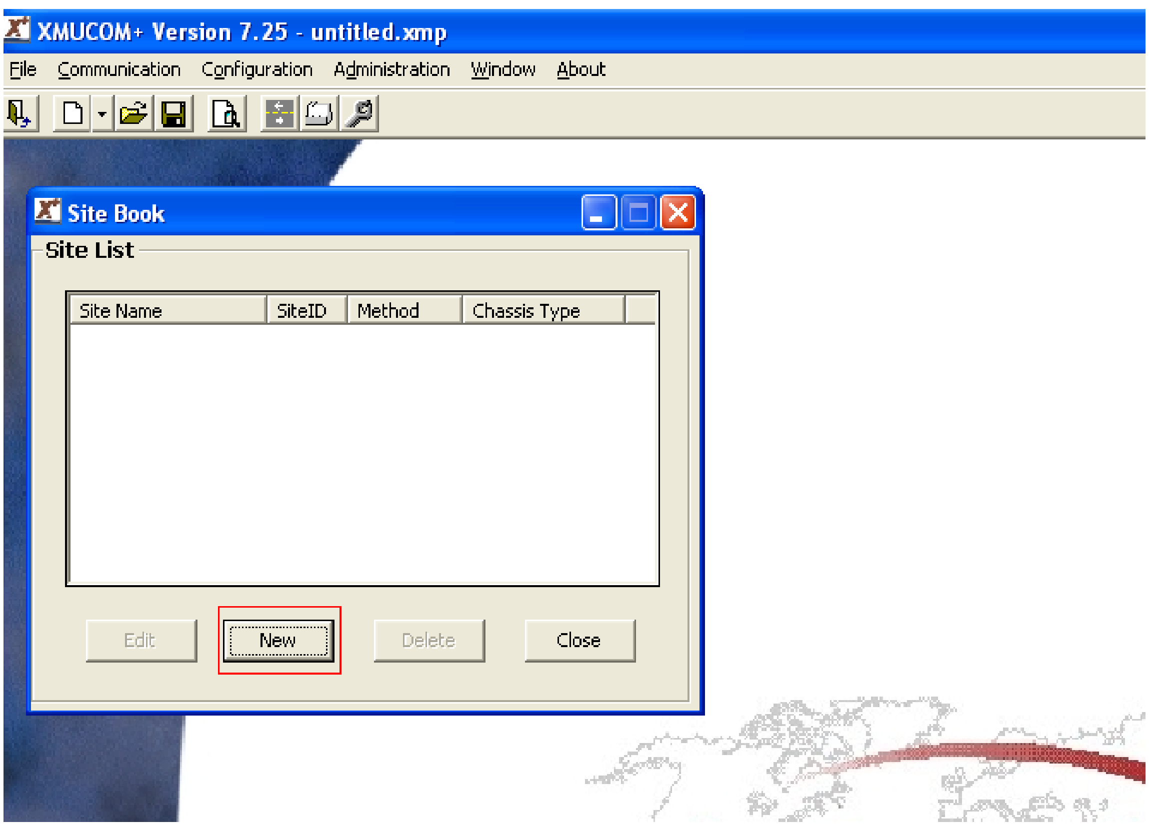
Once installed the XMUCOM+ program can be run from a shortcut on the desktop or by selecting the program as shown below.



Choose the **Communication** menu and select **Site Book** from the drop down menu.



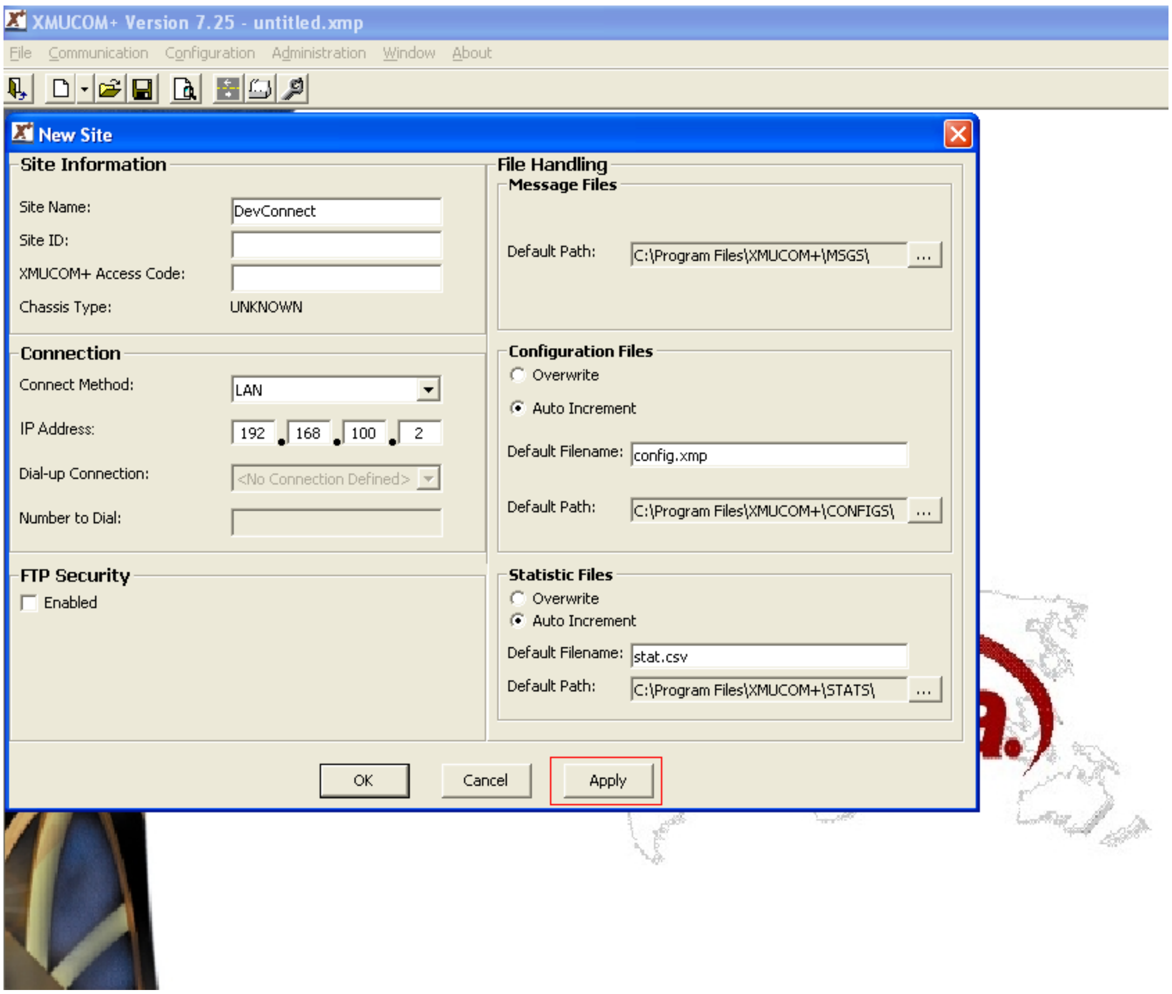
Click on the **New** button.



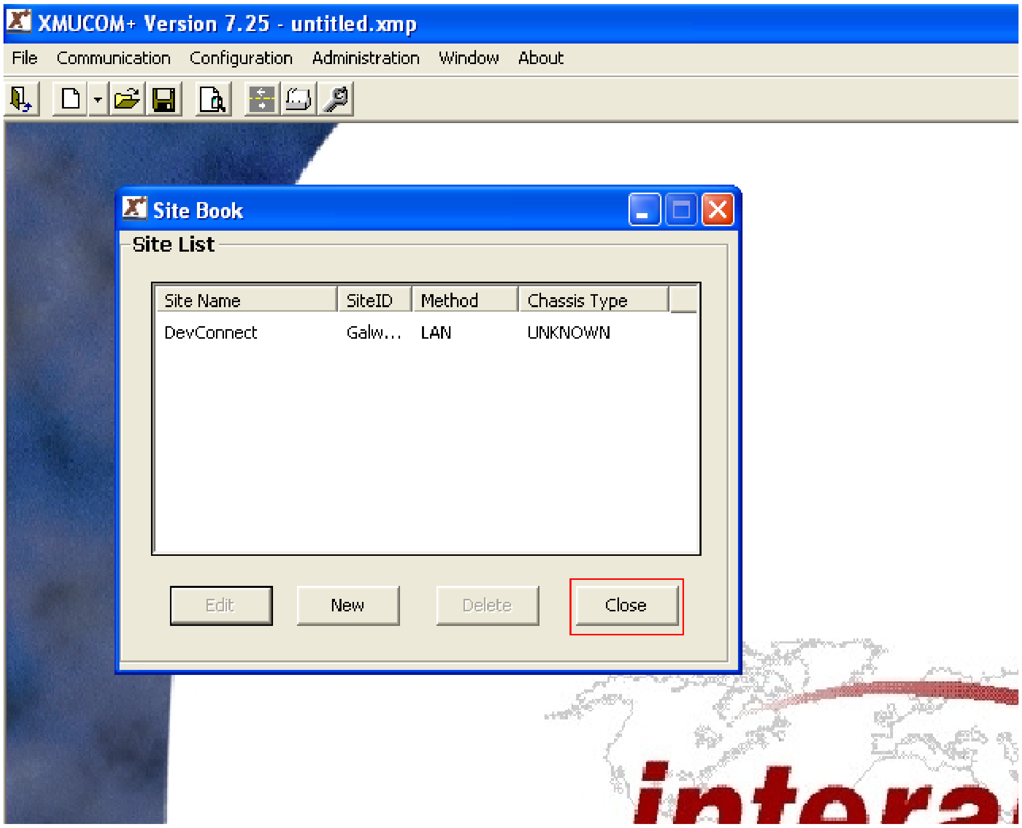
Enter the following.

- **Site Name** Enter a Name for the Site Name
- **Connect Method** LAN was used for Compliance testing
- **IP Address** Enter the IP Address of the XMU+

The **Site ID** and **XMUCOM+ Access Code** can be left blank. Select **Apply** followed by **OK** when all information is entered.



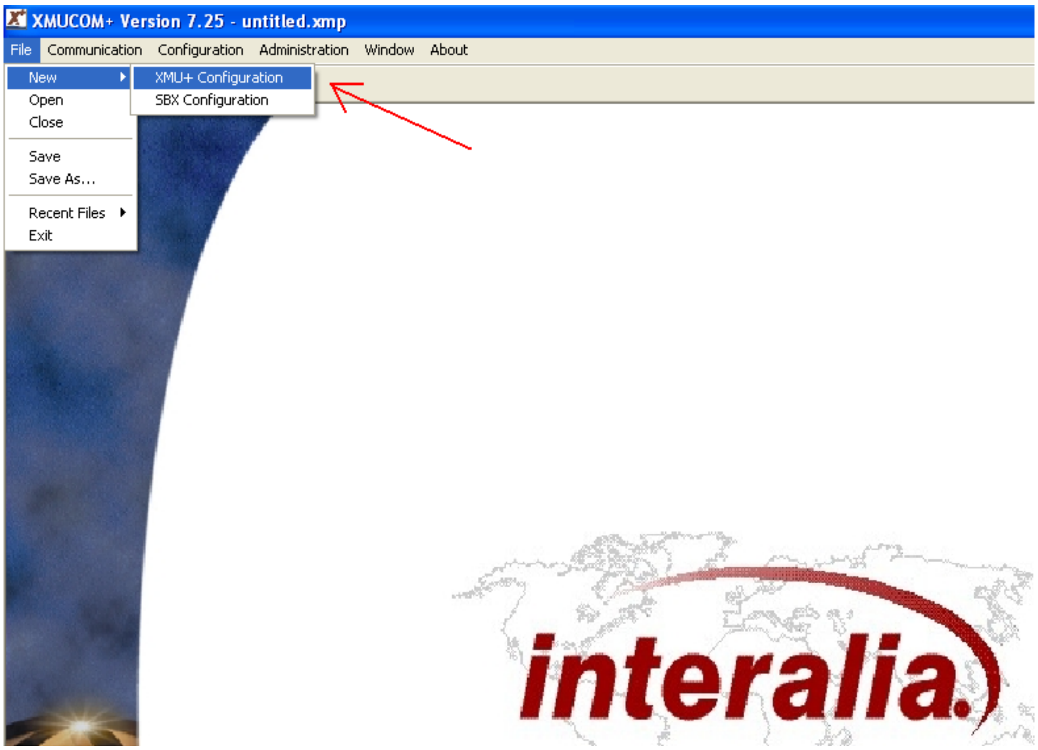
Click the **Close** button. Once the **Site Book** is created it is now possible to continue with the rest of the XMU+ configuration.



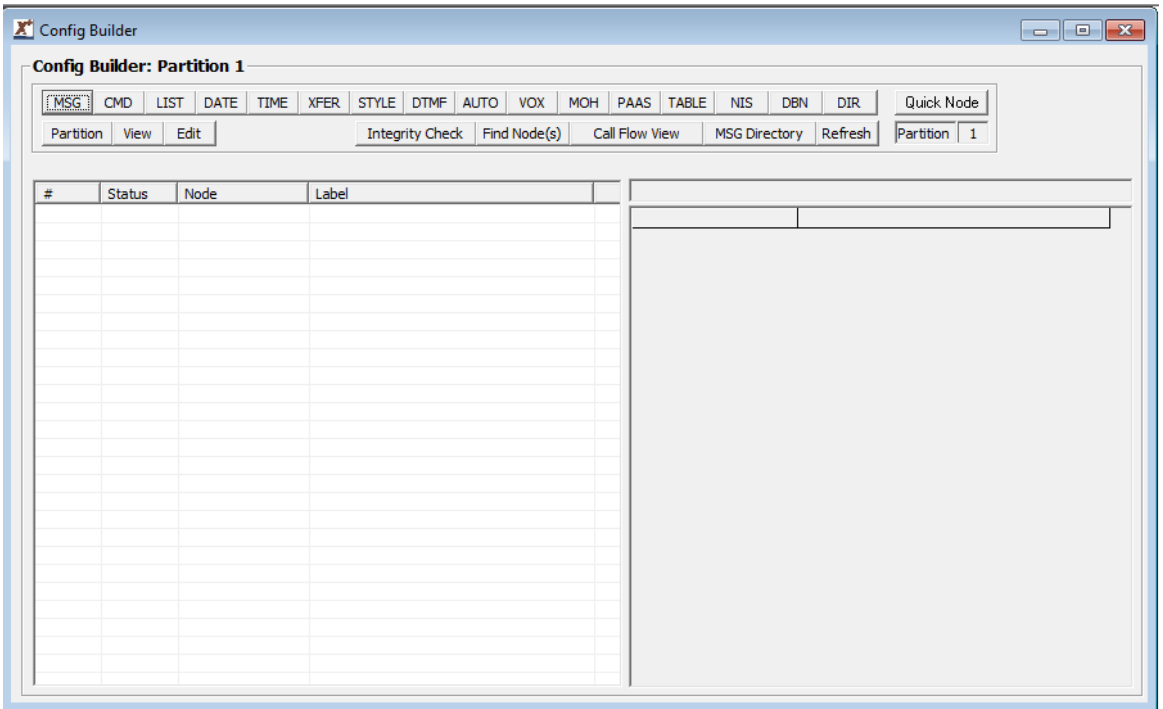
6.6.4. Configuring the Interalia XMU+ for RAN Announcements

Select **File → New → XMU+ Configuration**.

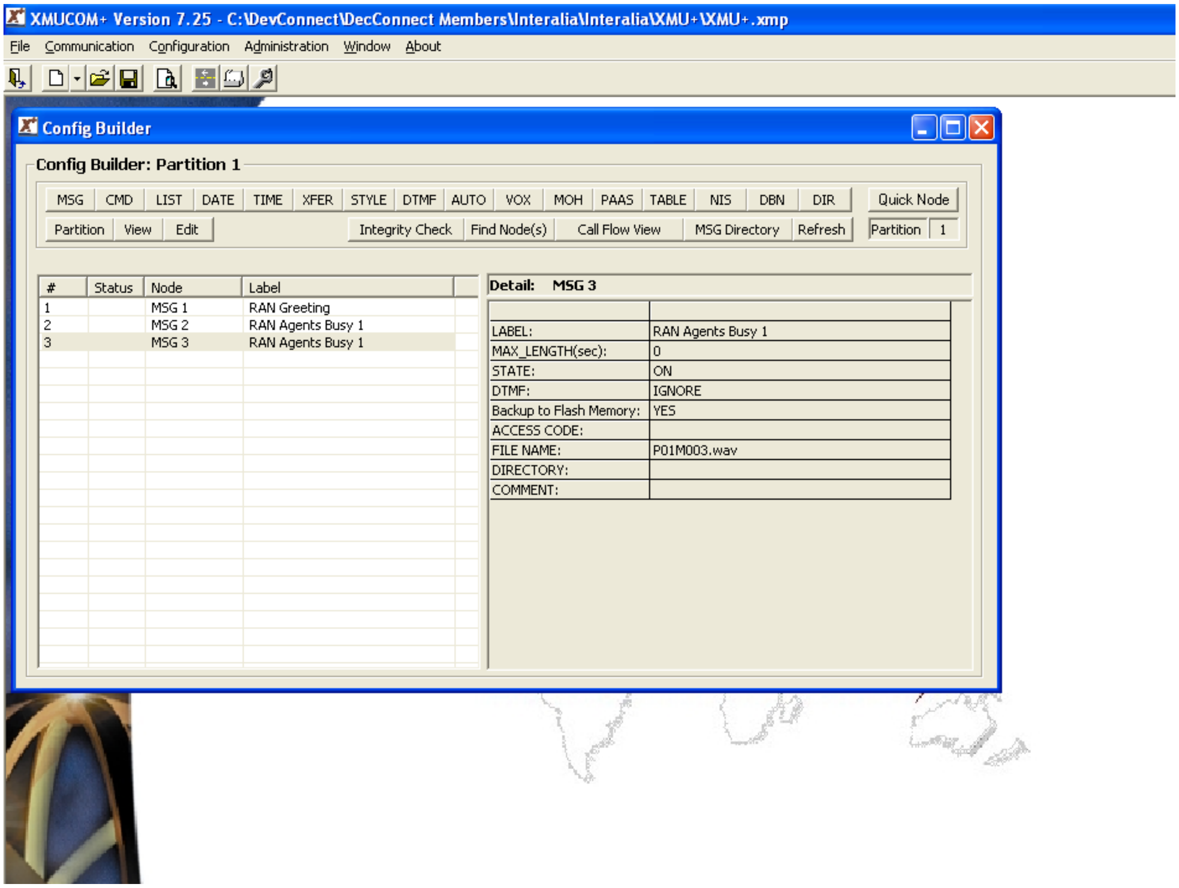
Note: The same program is used to configure the SBX and the XMU+ and the same screen options appear for both.



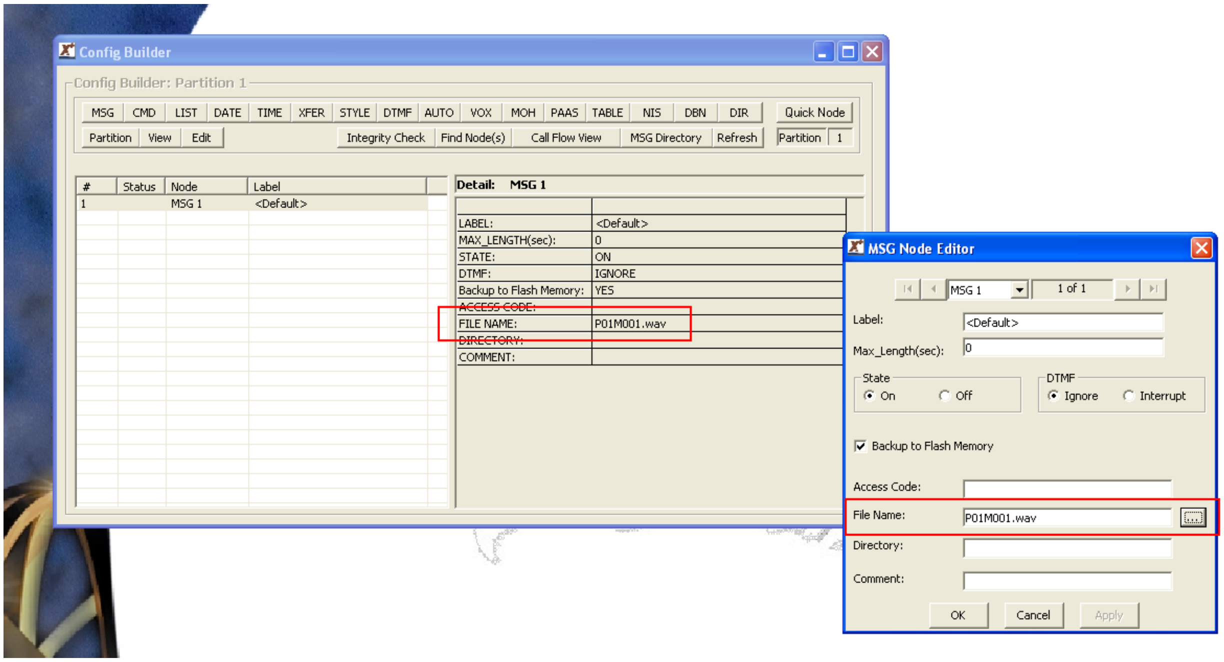
The software then opens with the following blank **Config Builder**.



To create 3 simple messages as RAN announcements for an initial greeting and two waiting in queue announcements, click on the **MSG** button as highlighted below.

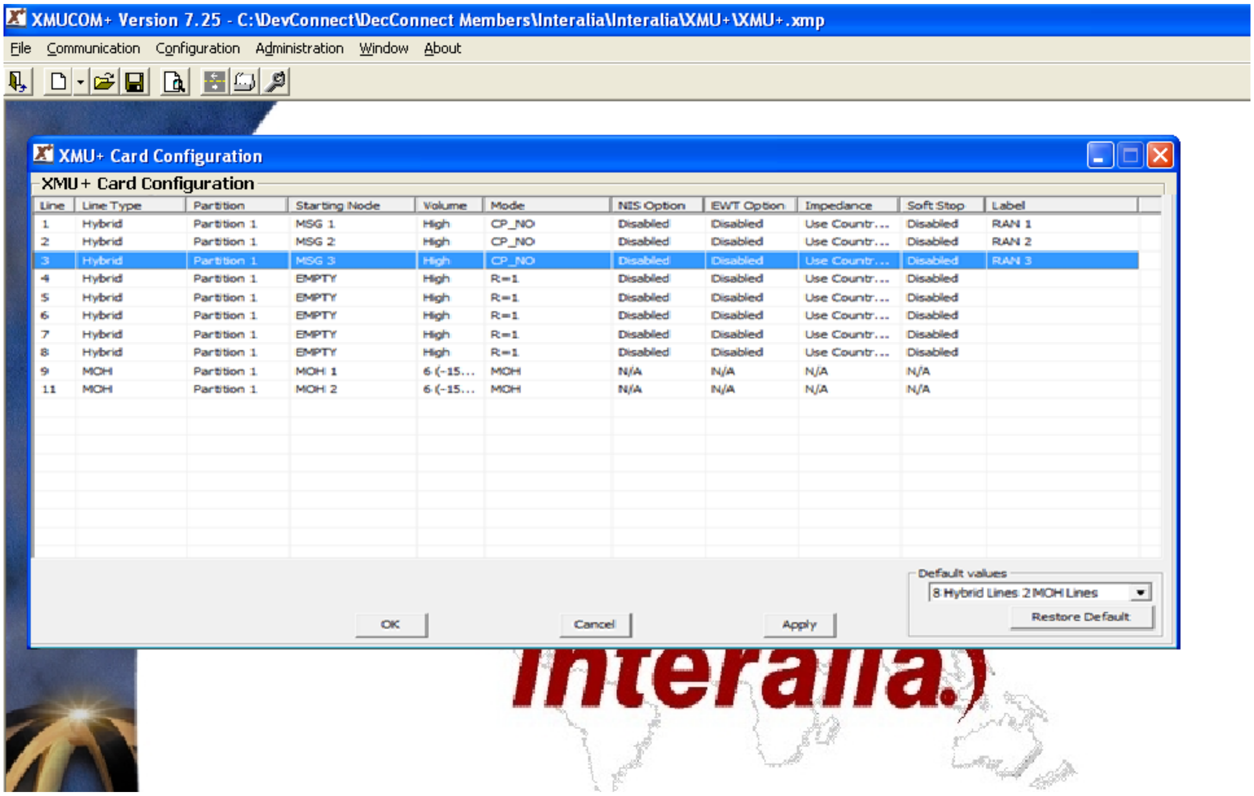


To add specific announcement messages double click on **FILE NAME** highlighted below. This opens the **MSG Node Editor** window. Beside the **File Name** browse to saved announcements in WAV format.



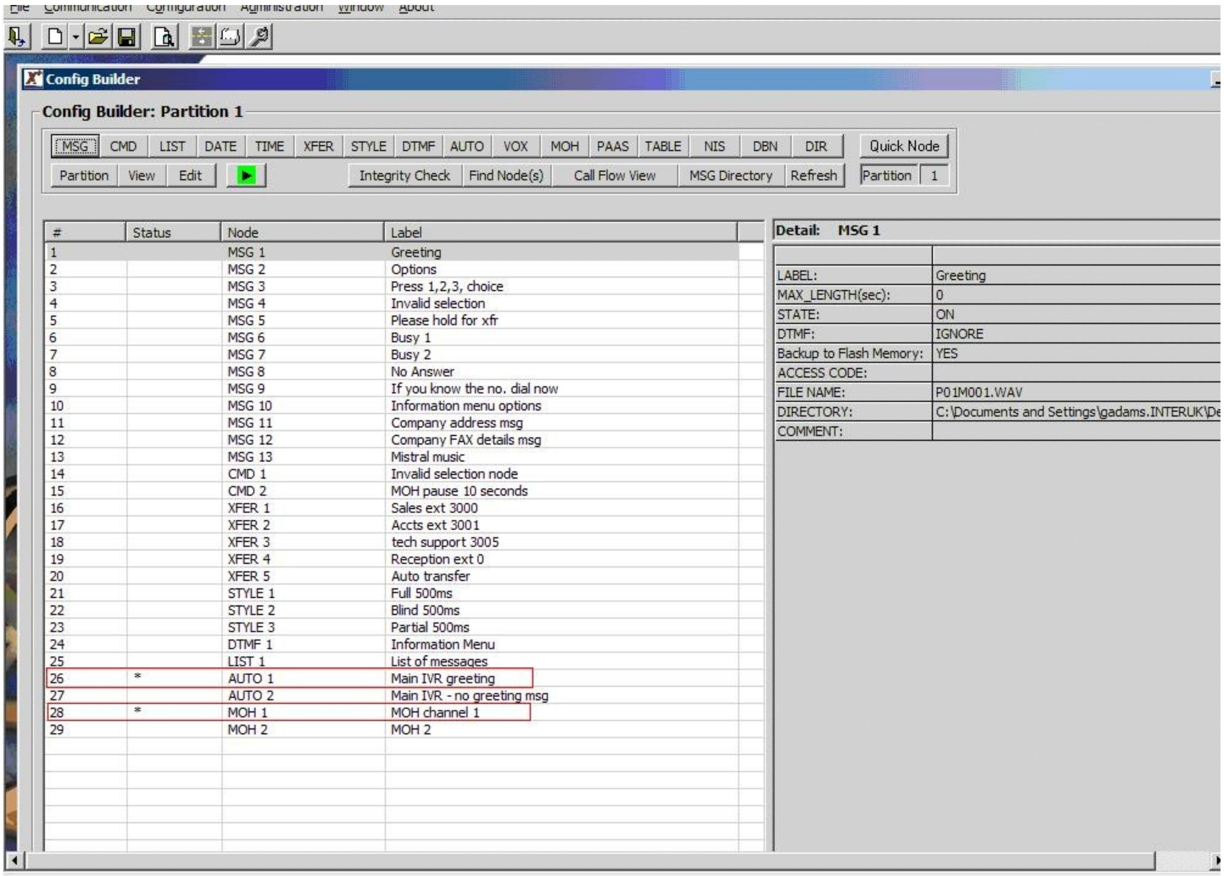
To assign these messages to the XMU+ ports, enter the XMU+ line configuration as shown below.

The XMU+ card set up shows only the first 3 ports in use as the RAN announcements that were previously setup. These can be heard from the PBX by dialing the corresponding extensions on the MP118.

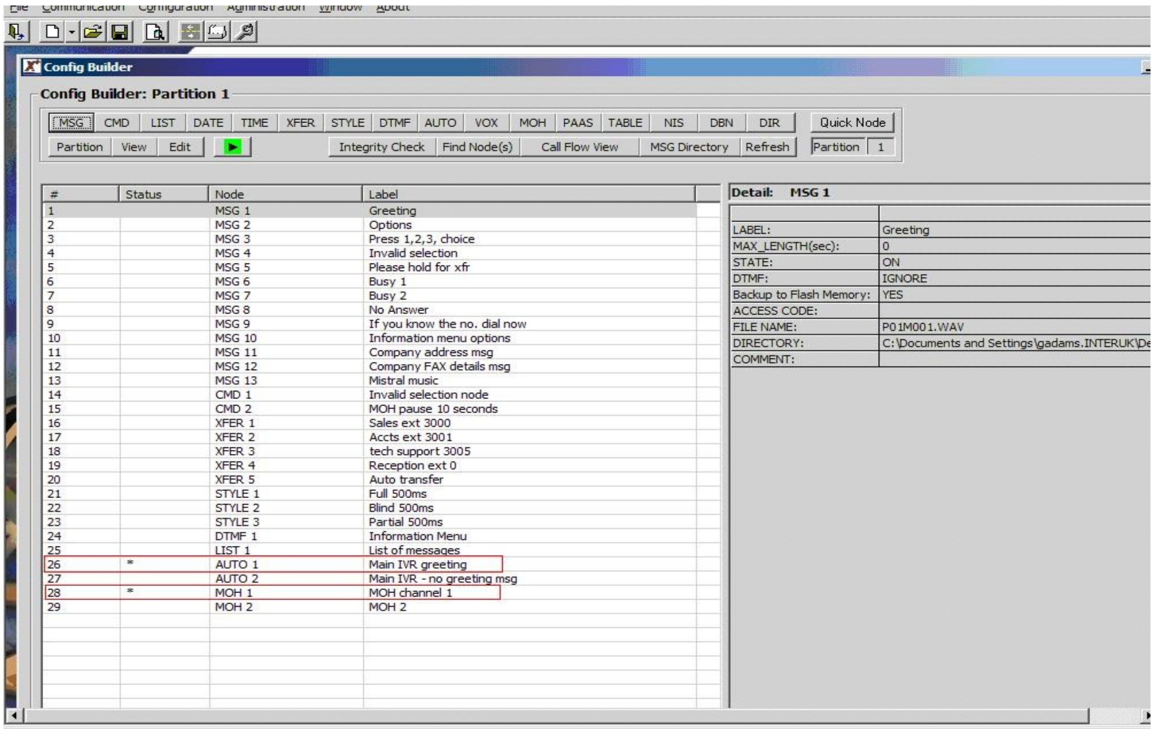


6.6.5. Configuring the Auto Attendant on the Interalia XMU+

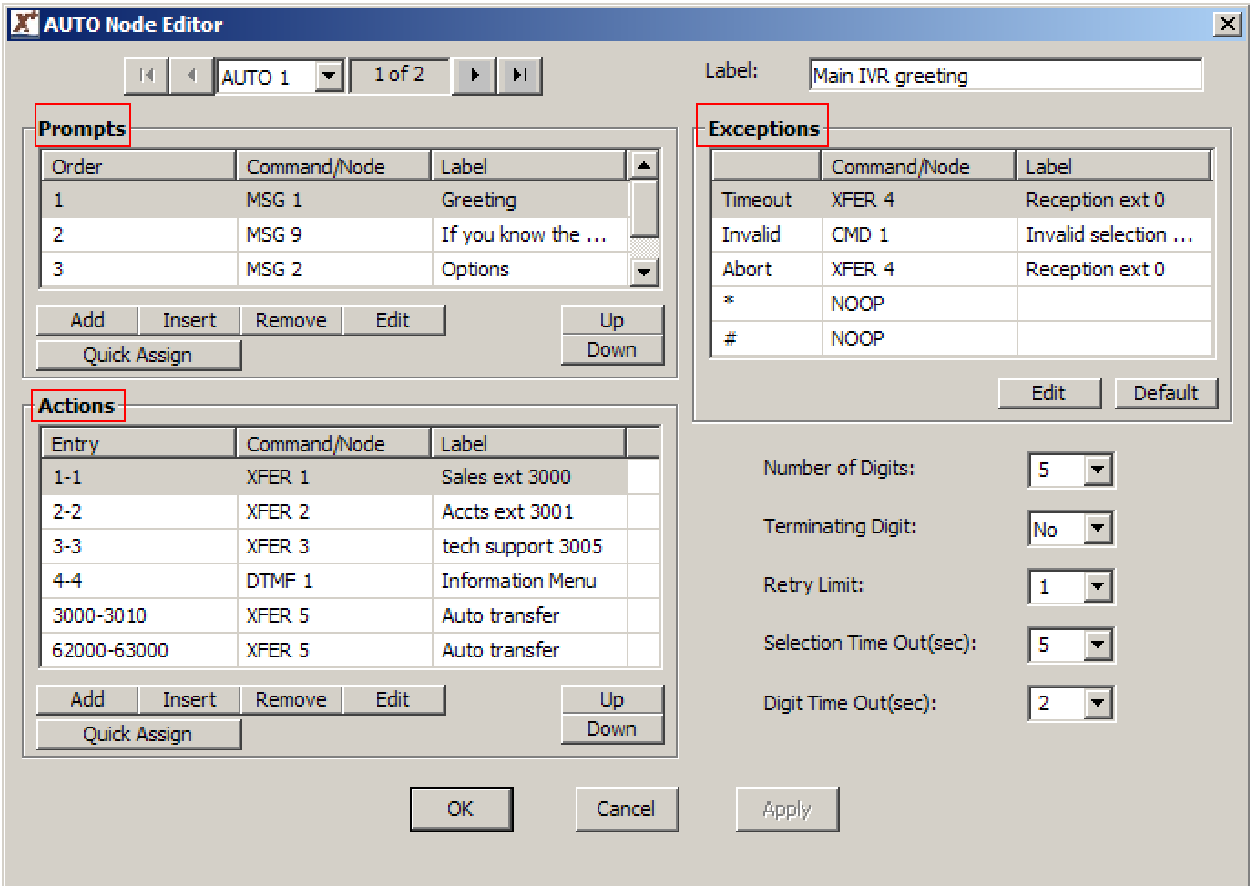
Open the XMUCOM+ program from the desktop shortcut as before. The **Config Builder** form is displayed which is the main IVR\Auto-Attendant configuration screen. One can populate this form with music and message files as well as IVR greetings. A completed form is displayed below.



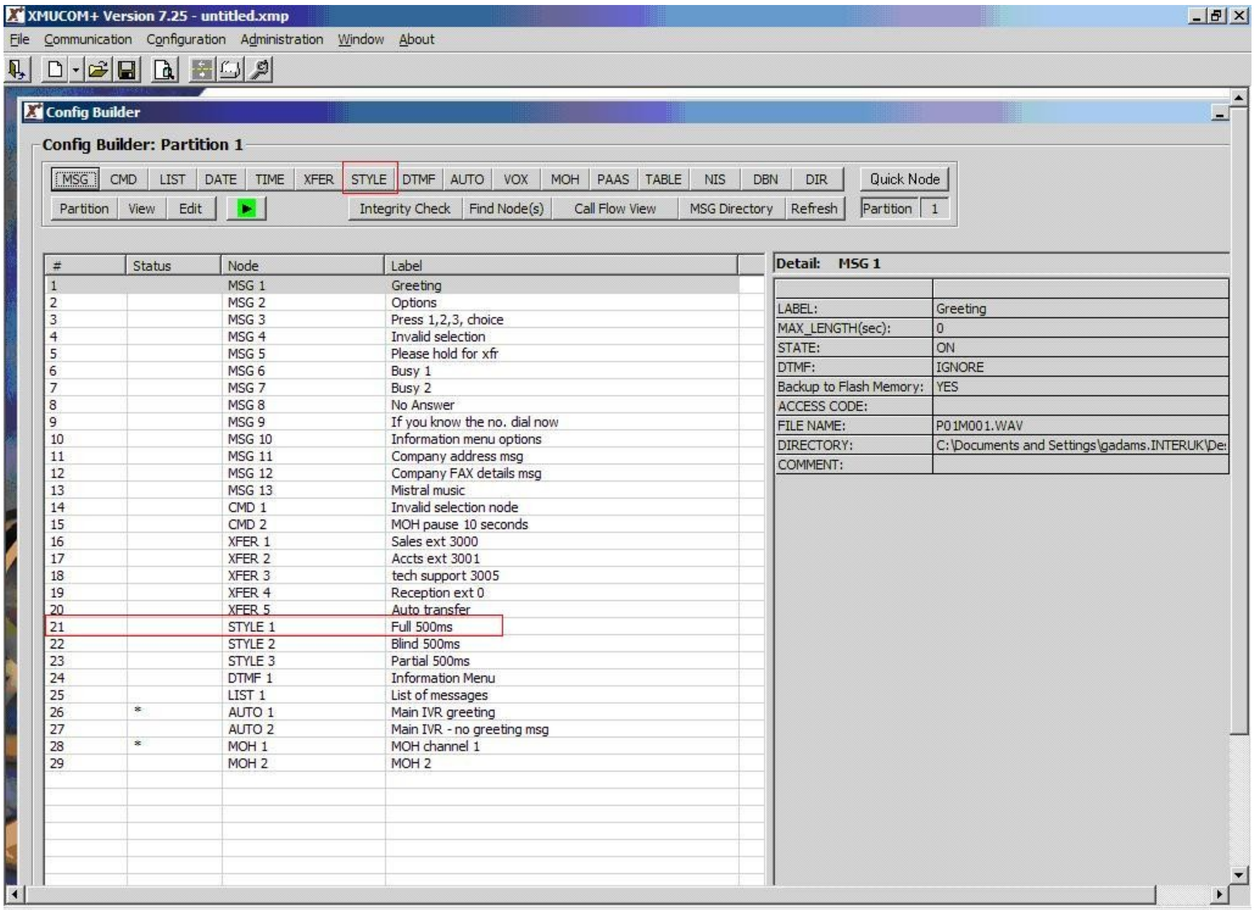
Note the highlighted entries in the screen above with an * as their **Status**. Once the configuration is built, the line card needs to be set to **AUTO 1** (shown on the next screen below) as this is the starting node and is shown by the * against its status in the screen shot above.



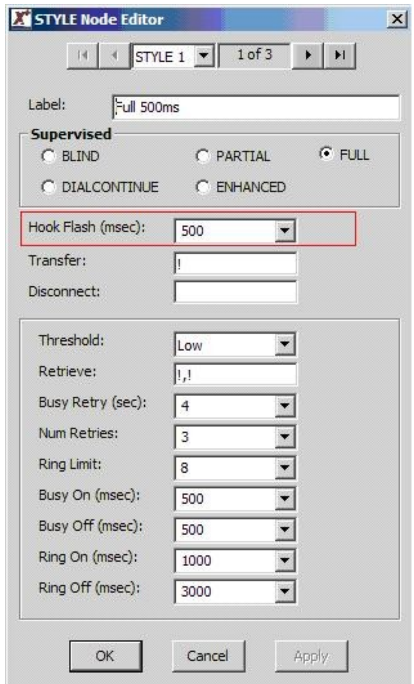
Double-click the entry for the **Node** displaying **AUTO 1** from the **Config Builder** screen above to open it for editing, and the **AUTO Node Editor** screen appears as shown below. The screen displays set **Prompts**, **Actions** and **Exceptions**.
Below are examples that were used in testing.



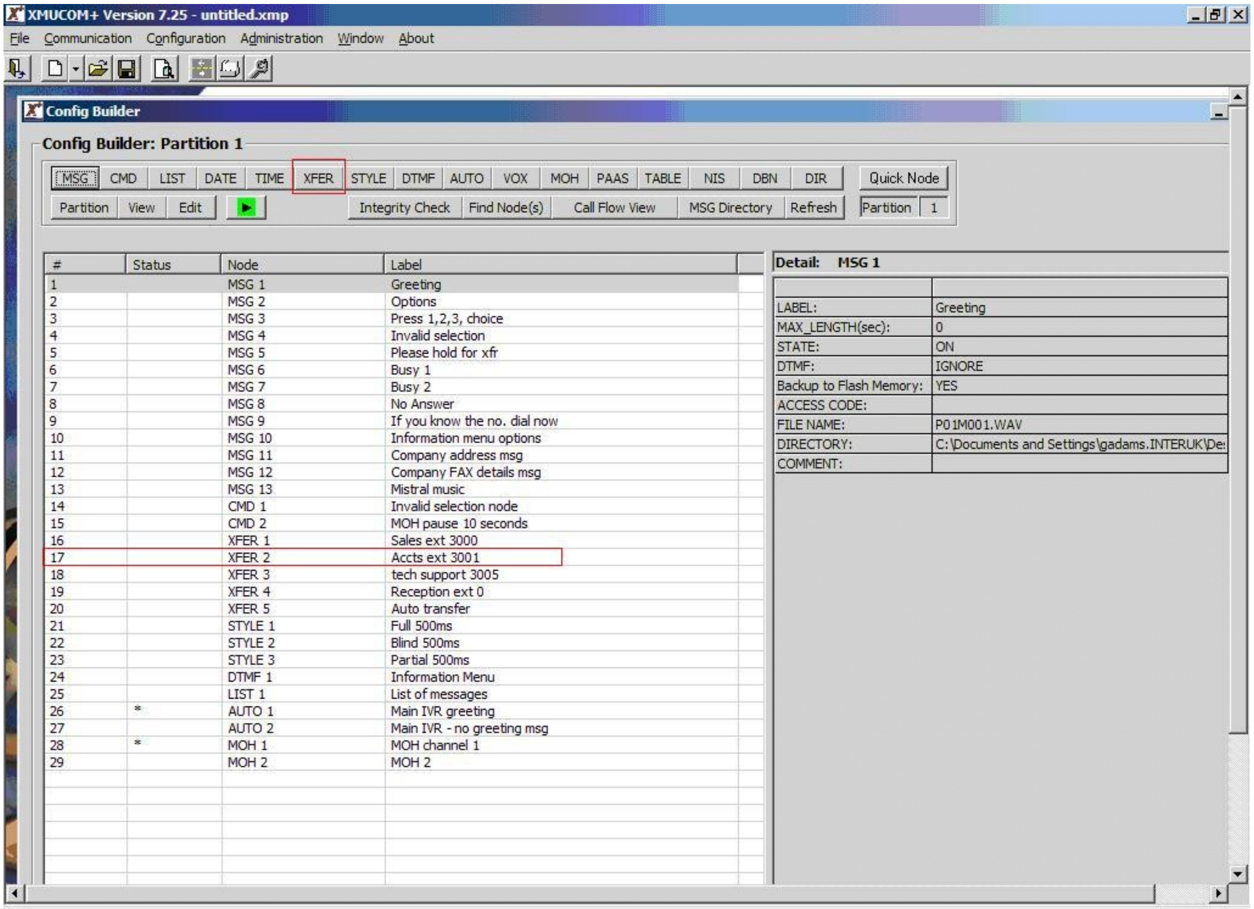
Incoming calls are directed to the Table which plays message 1, 2, or 3 depending on the time of day, calls then go to the Auto Node where a known extension can be dialed, listen to a recorded bulletin message or get transferred to a queue. In order to manage a call, the XMU+ will monitor for a busy or no answer indication, this is achieved by setting the Style node. Create a Style node through the **STYLE** tab as highlighted below. In the configuration builder, note the three transfer styles that require setting (Blind, Partial or Full).



Double click the **STYLE 1** entry from the **Config Builder** screen which is highlighted in the screen above as **Line 21**. The **STYLE 1** entry below will appear so the settings can be reviewed and edited. The transfer style shows the **Hook Flash (msec)** being set as **500** in order to perform the transfer / retrieval of a call.

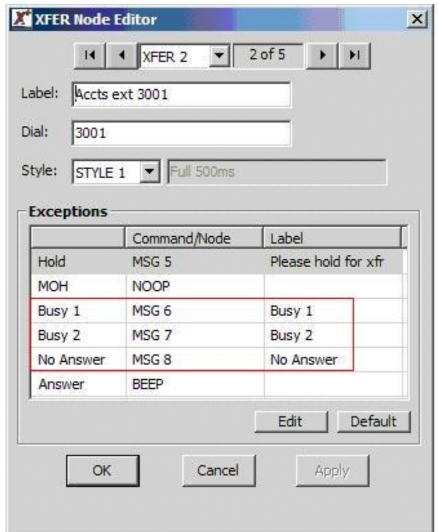


Alter the Hook Flash values according to the call progress tones set for country option. During the monitoring of a call, it is possible to inform the caller of their progress. In the main **Config Builder** window you can edit calls transfers i.e. **XFER 2**. Click on the **XFER** node as highlighted in the diagram below.



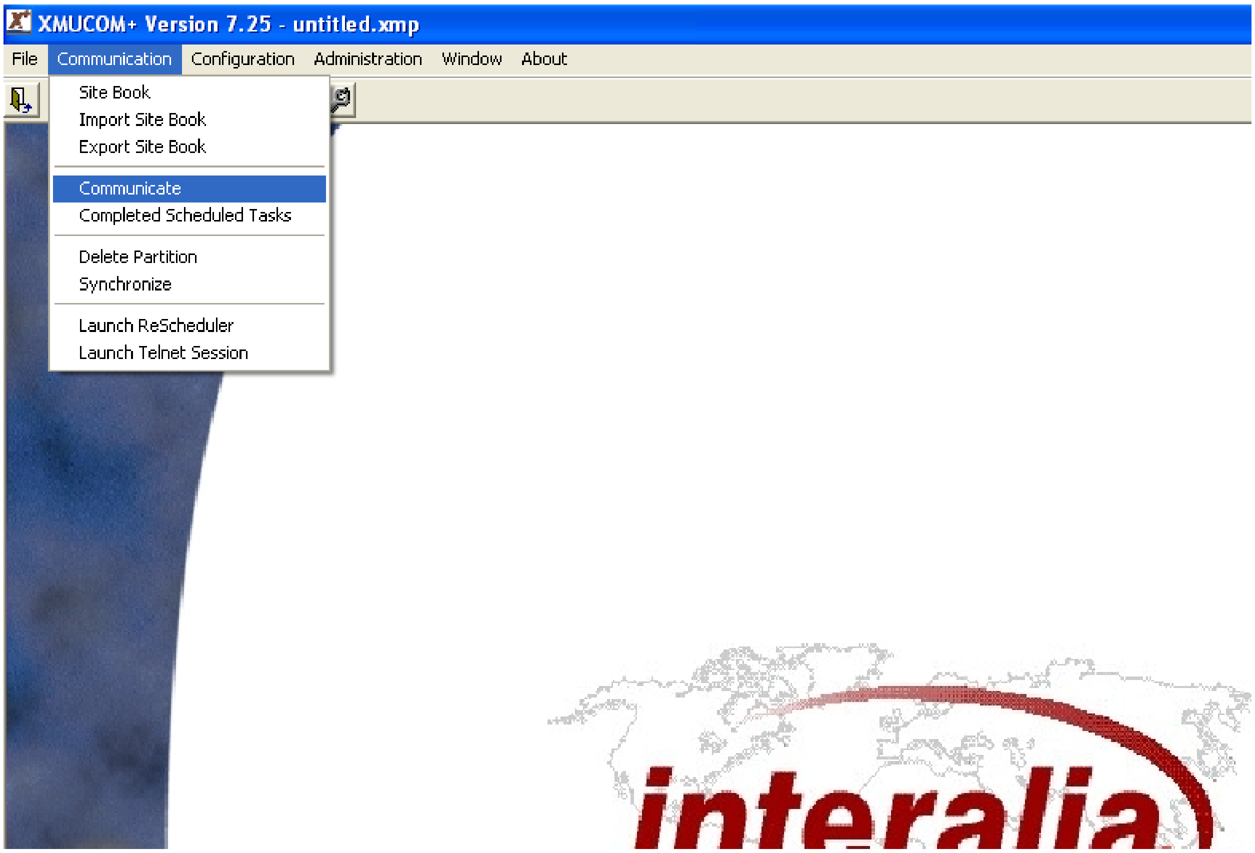
#	Status	Node	Label
1		MSG 1	Greeting
2		MSG 2	Options
3		MSG 3	Press 1,2,3, choice
4		MSG 4	Invalid selection
5		MSG 5	Please hold for xfr
6		MSG 6	Busy 1
7		MSG 7	Busy 2
8		MSG 8	No Answer
9		MSG 9	If you know the no. dial now
10		MSG 10	Information menu options
11		MSG 11	Company address msg
12		MSG 12	Company FAX details msg
13		MSG 13	Mistral music
14		CMD 1	Invalid selection node
15		CMD 2	MOH pause 10 seconds
16		XFER 1	Sales ext 3000
17		XFER 2	Accts ext 3001
18		XFER 3	tech support 3005
19		XFER 4	Reception ext 0
20		XFER 5	Auto transfer
21		STYLE 1	Full 500ms
22		STYLE 2	Blind 500ms
23		STYLE 3	Partial 500ms
24		DTMF 1	Information Menu
25		LIST 1	List of messages
26	*	AUTO 1	Main IVR greeting
27		AUTO 2	Main IVR - no greeting msg
28	*	MOH 1	MOH channel 1
29		MOH 2	MOH 2

By double clicking the **XFER 2** entry from the **Config Builder** screen highlighted in the screen above, the **XFER 2** entry below will appear so the settings can be reviewed and edited. If the extension is busy, messages 6 and 7 (**MSG 6** and **MSG 7**) will play according to the Xfer Style (Num Retries). If there is no answer, then the **No Answer** message (**MSG 8**) is set to play.

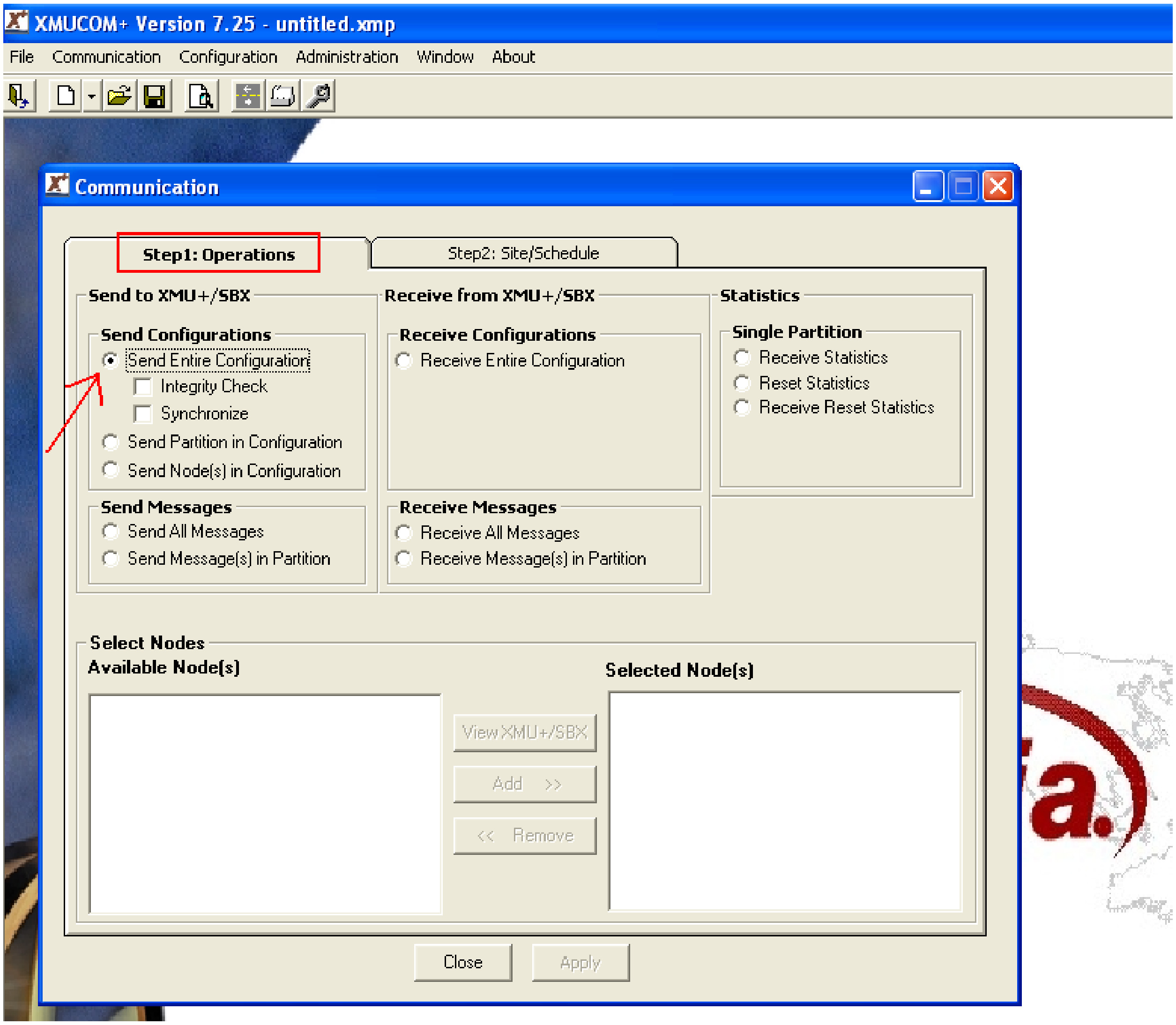


6.6.6. Downloading Configuration to the Interalia XMU+

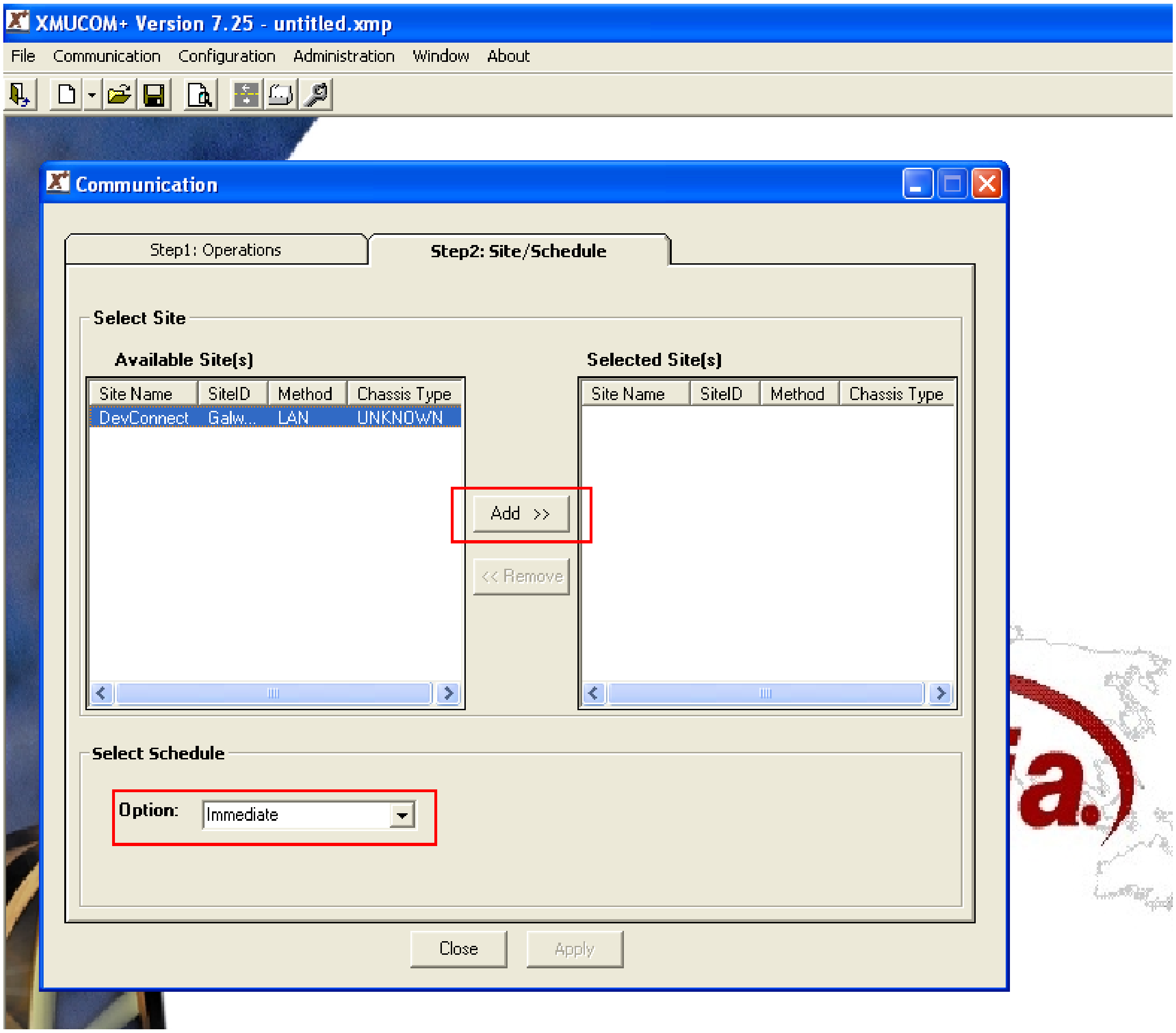
Once all the relevant information is configured it must be downloaded to the XMU+. To download the configuration to the XMU+, complete the following steps. Choose the **Communicate** from the **Communication** menu.



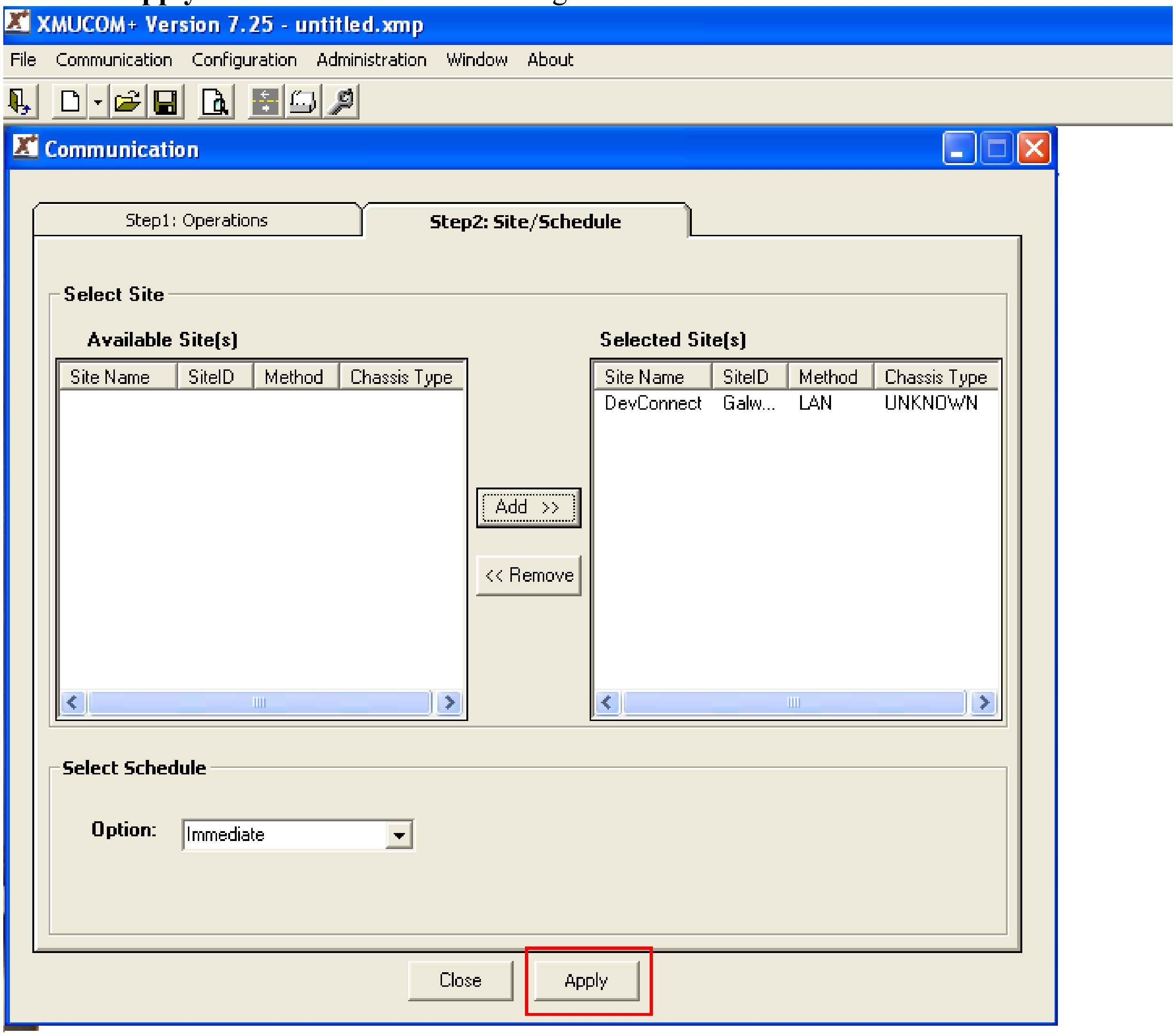
In the **Communication** Window, select tab **Step 1: Operations** and check the **Send Entire Configuration** button.



Select tab **Step2: Site/Schedule** and choose the site followed by clicking the **Add** >> button. **Select Schedule Option** can be set to **Immediate**.



Click the **Apply** button to download the configuration to the XMU+.



The download process could take some time depending on the speed of the selected communication connection and the number of messages being downloaded. XMUCOM+ will display a response dialog box while sending the messages, and notifies when all messages are sent.

Note: If this is the first time downloading to the XMU+, the Chassis Type will be unknown and a warning window will appear. Click the **OK** button to continue with the download.

7. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the CS1000E, XMU+ using a SIP connection from the MP118.

7.1. Verify Avaya Communication Server 1000E

The following steps can ensure that the communication between the CS1000E and the XMU+ is functioning correctly.

7.1.1. Status of D-Channel on Avaya Communication Server 1000E

Check the status of the D-channel setup in **Section 5.1.1** by running the command **STAT DCH** in overlay 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	Check status of all D-Channels
DCH 066	**OPER EST**	**DES : VTRK_DCH**

7.1.2. Status of the SIP Trunks on Avaya Communication Server 1000E

Using overlay 32, check the status of the SIP trunks to ensure all appear **idle.**

LD 32

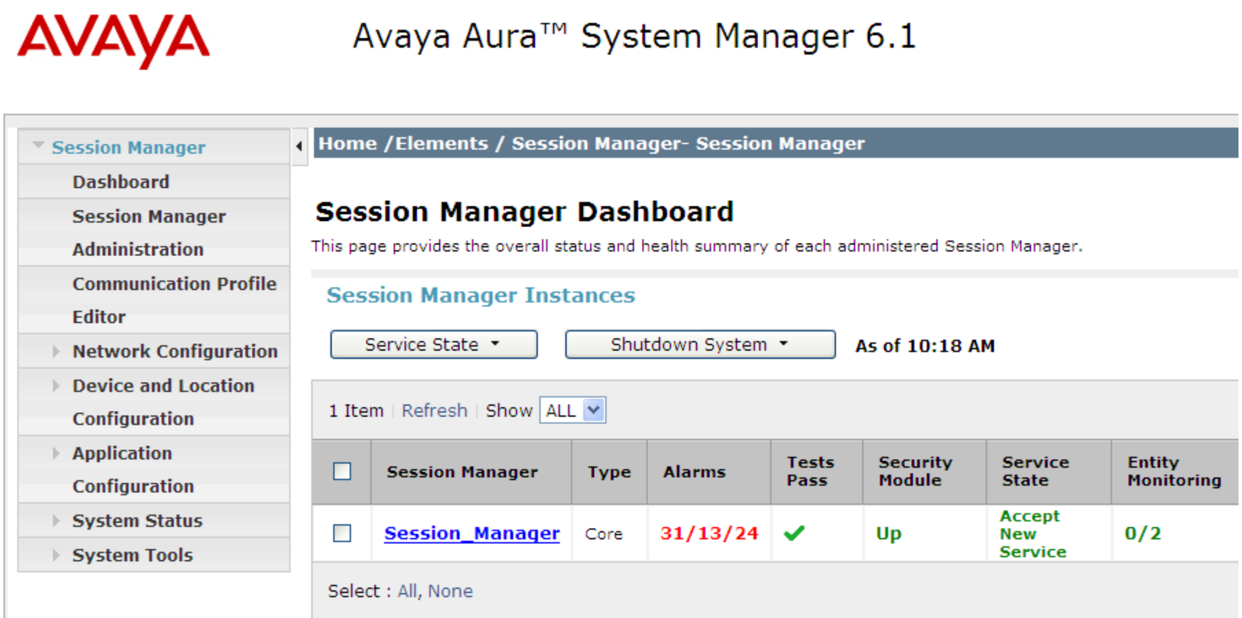
Prompt	Response	Description
>	**LD 32**	Enter Overlay 32
.	STAT [L S C]	Check status of Loop Shelf Card

The example below shows the first four SIP trunks are idle and ready to accept calls.

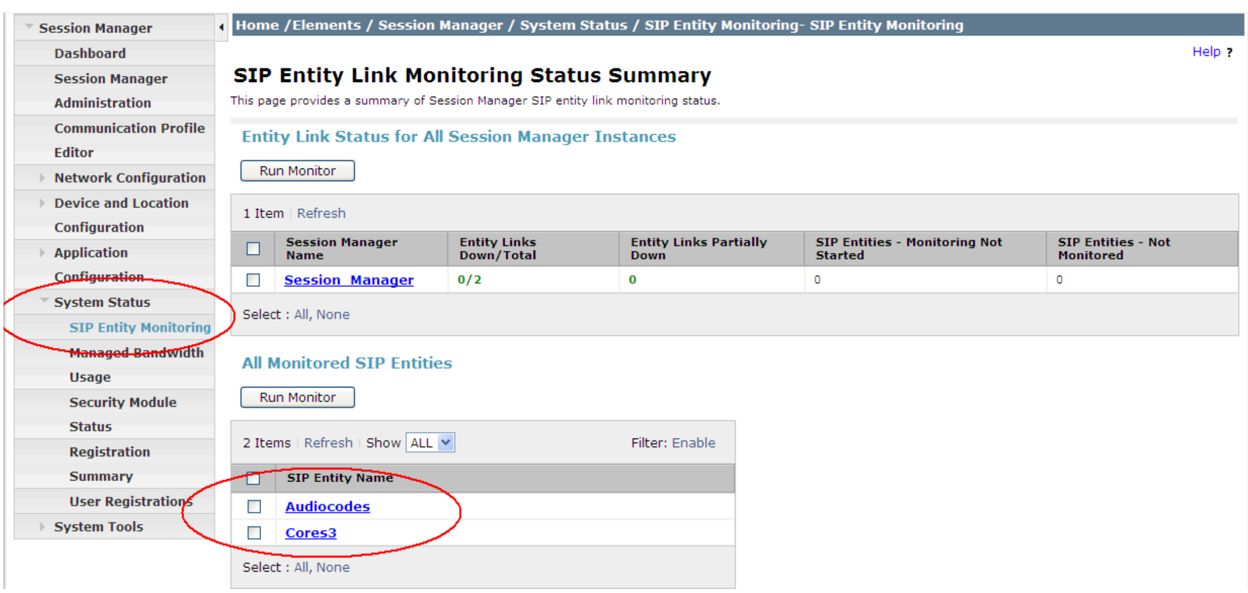
```
>ld 32
.stat 96 0 3
00 = UNIT 00 = IDLE            (ISL TRK)(TIE IP  IMM /IMM )
    D-CH 66  EST  ACTV

01 = UNIT 01 = IDLE            (ISL TRK)(TIE IP  IMM /IMM )
    D-CH 66  EST  ACTV

02 = UNIT 02 = IDLE            (ISL TRK)(TIE IP  IMM /IMM )
    D-CH 66  EST  ACTV

03 = UNIT 03 = IDLE            (ISL TRK)(TIE IP  IMM /IMM )
    D-CH 66  EST  ACTV
```

7.1.3. Status of the Session Manager Connections to the AudioCodes MP118 and the Avaya Communication Server 1000E

Check to see if the Session Manager is running and able to accept call by clicking on **Session Manager** from the main menu. The status should show as below.

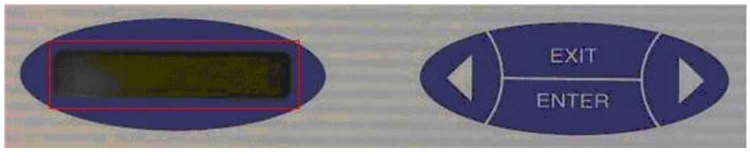


Check the status of the connections to the MP118 and the CS1000E from the Session Manager by clicking on **System Status** and **SIP Entity Monitoring**; see examples of two entities highlighted below. Click on each **SIP Entity Name** to check the status of each connection.



7.2. Verify Interalia XMU+ Status

The **Status** window, as highlighted below, on the display on the front of the XMU+ can be used to verify the communication of the XMU+. It is accessible by pressing the right arrow to enter the Status window. This shows the call flow as calls are made and received by the interactive voice response system.



8. Conclusion

These Application Notes describe the configuration steps required for Interalia XMU+ to successfully interoperate with Avaya Communication Server 1000E using SIP trunks through the use of the AudioCodes MP118. All functionality and serviceability test cases were completed successfully.

9. Additional References

This section references the Avaya and Interalia product documentation that are relevant to these Application Notes.

Product documentation for Avaya products may be found at http://support.avaya.com

[1] Input Reference Administration Avaya Communication Server 1000, Release 7.5; Document No. NN43001-611_05.02, Dec 2010

The Interalia documentation can be found at the following location:
http://www.interalia.com/Products/XMU/XMU-Overview

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.