



Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the configuration steps for provisioning Interalia's SBX system to successfully interoperate with Avaya Communication Server 1000E R7.5 using a SIP connection. Interalia's SBX 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 Intermedia's SBX solution with Avaya Communication Server 1000E R7.5, Avaya Aura® System Manager R6.1 and Avaya Aura® Session Manager R6.1 using a SIP connection through a third party SIP Gateway provided by an AudioCodes MP-118 /8FXS/3AC Gateway. The AudioCodes MediaPack 118 is an analog to SIP VoIP Gateway that provides superior voice technology for connecting legacy telephones, fax machines and PBX systems with SIP-based telephony networks and integration with new IP-based PBX systems. It is easily managed using a web interface. The SBX is an entry level system ideal for applications requiring up to 8 ports, one hour of total audio storage and needing only analog ports. The SBX is microprocessor-based voice application platforms that support multiple applications simultaneously on a port-by-port basis. Typical SBX applications include:

- Automatic Call Distribution (ACD) announcements
- Auto attendant
- Voicemail/IVR Failover
- Information Lines
- Music on hold (MOH)

The Intermedia system used for the test consists of an Intermedia SBX server connected to an AudioCodes MediaPack 118. The connection to the Avaya Communication Server 1000E was made via a SIP connection managed by Avaya Aura® Session Manager R6.1.

1.1. Applicable Systems

The interoperability compliance testing includes the following systems.

- Avaya Communication Server 1000E (CS1000E) R7.5, Avaya Aura® System Manager R6.1 and Avaya Aura® Session Manager R6.1
- Intermedia SBX (SBX)
- AudioCodes MP-118 /8FXS/3AC VoIP Gateway (MP118)

2. General Test Approach and Test Results

The test approach was to validate the correct operation of typical interactive voice response applications on the Interalia SBX such as ACD Announcements and Auto Attendant. The tests were all functional in nature and performance testing was not included. All the test cases passed successfully.

2.1. Interoperability Compliance Testing

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

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

Note: The compliance testing focused on testing using a SIP only connection to the CS1000E. The failover testing focused on verifying the ability of the SBX and MP118 systems to recover from disconnection such as power supply failure.

2.2. Test Results

Interalia SBX passed compliance testing with the following observations:

- [1] In an ACD environment on the CS1000E only RAN routes cannot be used to play announcements. SIP routes cannot be used to play announcements to callers holding in an ACD Q. However in a SIP Contact Centre environment messages would be heard in a Contact Centre queue.
- [2] In order to test fully the SBX applications over SIP, testing was performed using a series of group hunt lists associated with various SBX unique applications
- [3] No music on Hold could be heard when callers were placed on hold or transferred as the AudioCodes has no Music on Hold capability

2.3. Support

Technical support can be obtained for Interalia SBX 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 CS1000E R7.5 were used as the hosting PBX. The Intermedia SBX is connected to the hosting PBX using a SIP connection from the AudioCodes MediaPack 118 Gateway.

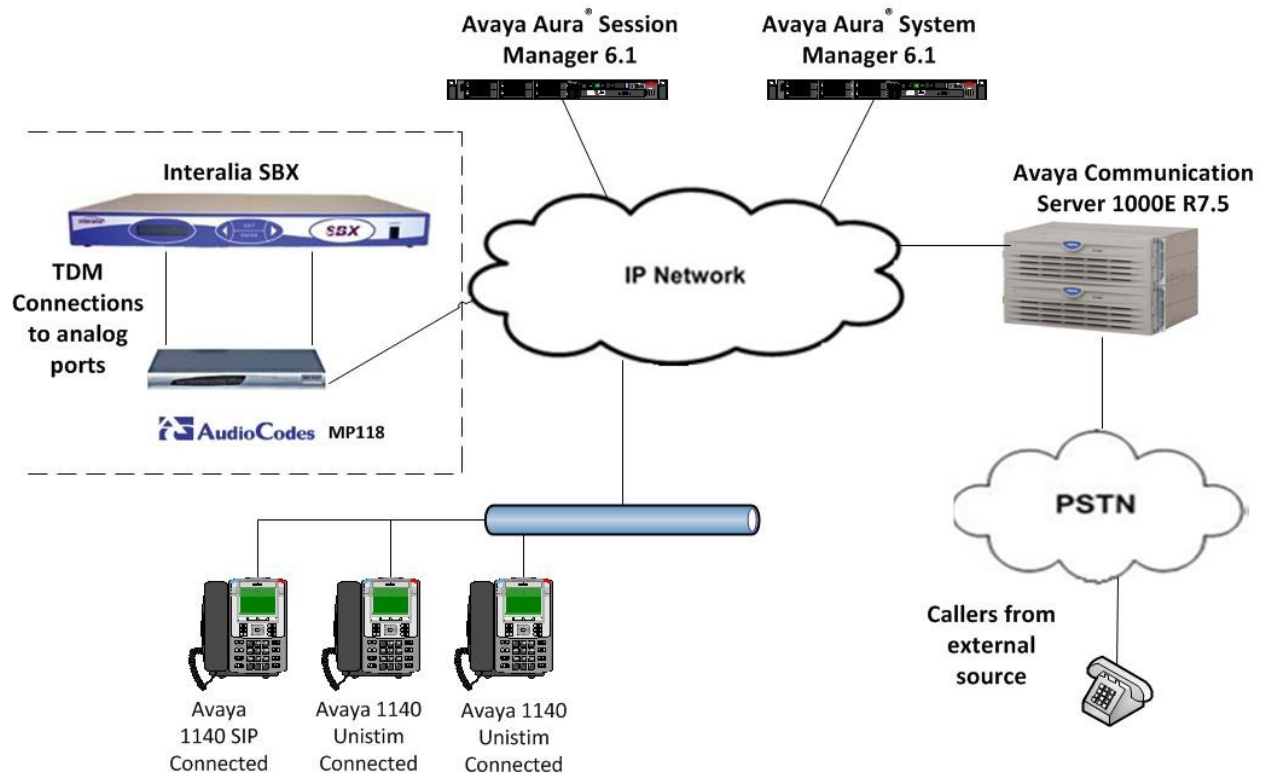


Figure 1: Network Topology and connectivity between Intermedia SBX and Avaya Communication Server 1000E R7.5

4. Equipment and Software Validated

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

| Equipment | Software Version |
|---------------------------------------|------------------------------------------------------------------------------------------------------|
| Avaya Communication Server 1000E CPPM | Avaya Communication Server 1000E R7.5 SP1 |
| Avaya S8800 Media Server | Avaya Aura® System Manager R6.1 SP0 |
| Avaya S8800 Media Server | Avaya Aura® Session Manager R6.1 SP0 |
| Avaya 1140E IP set | UNISTim V0625C8D SIP V04.00.04.00 SIP1140 |
| Interalia SBX | Firmware version: V1.35 Software: XMUCOM+ V7.25 Software : XMUCOM+ V8.0 (Windows 7 Compatible) |
| AudioCodes MP118/8FXS/3AC | MP-118 V5.60A.009.005 |

Table 1: Hardware and Software Version Numbers

Note: For a complete list of the patches installed on the CS1000E see **Appendix**.

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 the program “Reflections”. Configuration of the Avaya Aura® Session Manager was performed using a web GUI provided by the Avaya Aura® System Manager. The information provided in this section describes the configuration of the CS1000E for this solution. The information captured in this section shows the changes that are needed to add SIP trunks to an existing CS1000E system, adding group hunt lists, creating a dialling plan to the MP118 ports. However it does not show the complete setup of ACDQ’s 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 10**. The configuration operations described in this section can be summarized as follows:

- Configuring SIP route and SIP D-Channel on CS1000E
- Adding SIP trunks
- Creating a Coordinated Dialing Plan to access the SBX
- Creating Group Hunt Lists for the various SBX applications
- Creating a Pilot DN to call into the Group Hunt Lists
- Creating the MP118 as a SIP endpoint in Session Manager
- Creating a Routing Pattern in Session manager for the MP118 endpoint

Note: The configuration of the PRI interface to the PSTN is outside the scope of these Application Notes.

5.1. Creating a SIP route and D-Channel

To create a new D-Channel on the CS1000E overlay 17 is used. Use the **new** command in **LD 17** to create a virtual D-Channel for a SIP connection. Subsets of these commands are listed below.

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 x | Create new D-Channel |
| CTYP | DCIP | Card type is IP D-Channel |
| USR | ISDL | Integrated Services Signalling Link |
| IFC | SL1 | D-Channel interface type |

To create a new route on the CS1000E overlay 16 is used. 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. Subsets of these commands are listed below.

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

To create new trunks overlay 14 is used in the CS1000E. Use the **new** command in **LD 14** to add virtual SIP trunks to the new route created in **section 5.1**. Note that a number of trunks can be created subject to SIP trunk licenses available. Subsets of these commands are listed below.

LD 14

| Prompt | Response | Description |
|--------|-----------|------------------------------------|
| > | LD 14 | Enter Overlay 14 |
| REQ | NEW | 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 Dialling Plan

There are a number of ways one can setup a dialling plan to call the ports on the MP118. For the compliance testing a coordinated dialling 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**. Subsets of these commands are listed below.

Note: The route that you enter is the SIP route that was created in **section 5.1**

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 | xx | Route list Index number |
| ENTR | 0 | First entry for the RLI |
| ROUT | 20 | Enter the SIP route number |

To create a **CDP** overlay 87 is used. 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 above. Subsets of these commands are listed below.

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 dialling plan |
| TYPE | DSC | Distance Steering code |
| RLI | xx | Route list index Number |

5.4. Creating new Group Hunt Lists

If there are a number of channels on the SBX associated with the same service on the SBX then these channels will be added to a hunt group on the CS1000E in order to access them using a single route point or number. Depending on the nature of the service on the SBX the associated channel will be a part of a specific hunt group. On the SBX there are eight 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.

To create a group hunt list overlay 18 is used. Use the **new** command in **LD 18** to create a new group hunt list. Subsets of these commands are listed below.

LD 18

| Prompt | Response | Description |
|--------|----------|-----------------------------------------------|
| > | LD 18 | Enter Overlay 18 |
| REQ | NEW | Create new |
| TYPE | GHT | Group Hunt |
| LSNO | 1-8190 | Group Hunt List Number |
| CUST | 0 | Customer Number as defined in LD15 |
| SIZE | 1-96 | Amount of entries in the GHT |
| STOR | x y | x is the entry number and y is the CDP number |

5.5. Creating a Pilot DN

Create a Pilot DN (**PLDN**) which is the number called associated with the hunt group. To create a pilot DN overlay 57 is used on the CS1000E. Use the **chg** command in **LD 57**. Subsets of these commands are listed below.

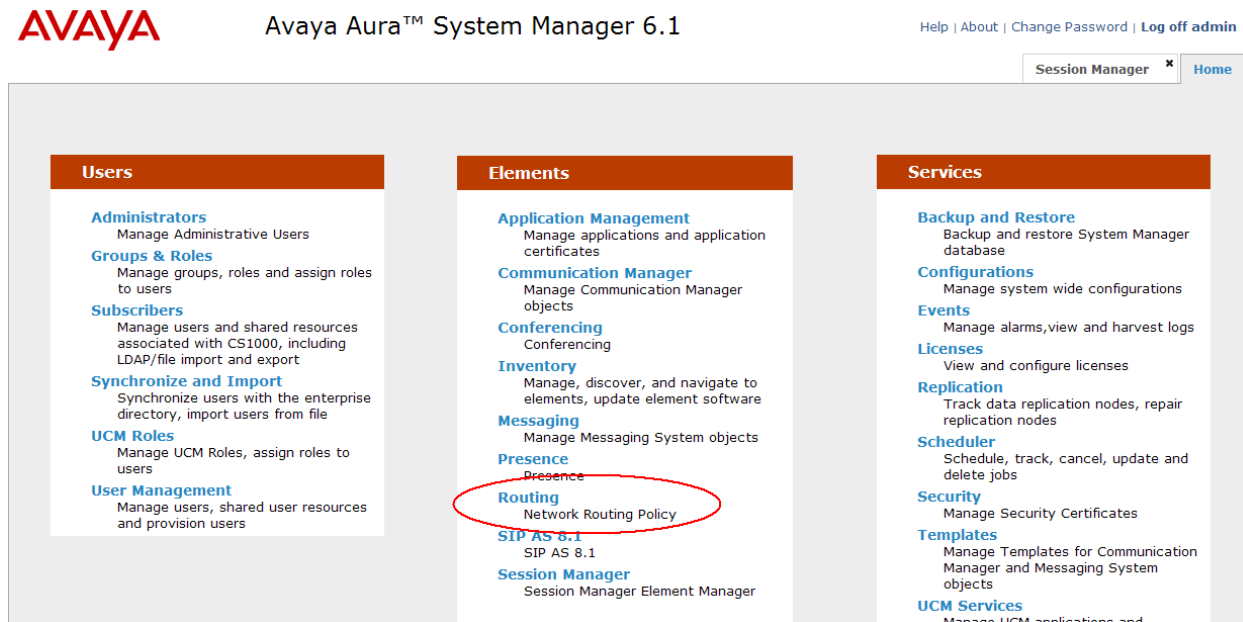
LD 57

| Prompt | Response | Description |
|--------|----------|-----------------------------------|
| > | LD 57 | Enter Overlay 57 |
| REQ | CHG | Change existing |
| TYPE | FFC | Flexible Feature Codes Data Block |
| CODE | PLDN | Pilot DN (Group hunt access DN) |
| USE | GPHT | Use is Group Hunt |
| LSNO | x | list number created section 5.4 |
| HTYP | LIN/RRB | Linear or Round Robin |

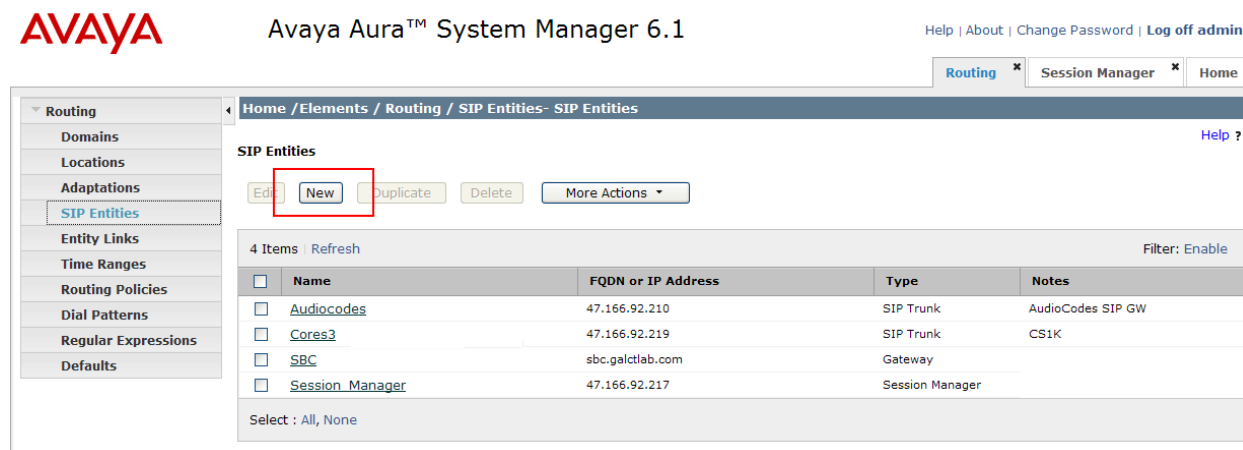
Note: If a Pilot DN does not already exist for another group hunt the new command in LD 57 may need to be used.

5.6. Creating the AudioCodes MP118 as a SIP Endpoint in Avaya Aura® Session Manager

Upon logging into System Manager click **Routing** as highlighted below.



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



A new page appears where the necessary information is filled out according to the specifics of the MP118 being added. The important information is highlighted in the screen below. Once the correct information is entered the **Commit** button is clicked.

| | | |
|---------------------|--------------------|------------------------------------|
| Domains | SIP Entity Details | <div>Commit<div>Help ?</div></div> |
| Locations | | |
| Adaptations | | |
| SIP Entities | | |
| Entity Links | | |
| Time Ranges | | |
| Routing Policies | | |
| Dial Patterns | | |
| Regular Expressions | | |
| Defaults | | |

General

* Name: Audiocodes

* FQDN or IP Address: 47.166.92.210

Type: SIP Trunk

Notes: AudioCodes SIP GW

Adaptation:

Location:

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV:

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: egress

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

5.7. Creating a routing pattern for the MP118 SIP endpoint to route calls

Create an **Entity Link** by clicking **New** as highlighted below.

Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Session Manager x Home

Home / Elements / Routing / Entity Links- Entity Links

Entity Links

Edit New Duplicate Delete More Actions

2 Items Refresh Filter: Enable

| <input type="checkbox"/> | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Trusted | Notes |
|--------------------------|--------|-----------------|----------|------|--------------|------|-------------------------------------|-------|
| <input type="checkbox"/> | ACodes | Session_Manager | TCP | 5060 | Audiocodes | 5060 | <input checked="" type="checkbox"/> | |
| <input type="checkbox"/> | coRes3 | Session_Manager | TCP | 5060 | Cores3 | 5060 | <input checked="" type="checkbox"/> | |

Select : All, None

The Protocol type and port number is filled in and **Commit** is clicked.

Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Session Manager x Home

Home / Elements / Routing / Entity Links- Entity Links

Entity Links

Commit Cancel

1 Item Refresh Filter: Enable

| Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Trusted | Notes |
|----------|-------------------|----------|--------|--------------|--------|-------------------------------------|-------|
| * ACodes | * Session_Manager | TCP | * 5060 | * Audiocodes | * 5060 | <input checked="" type="checkbox"/> | |

* Input Required

Commit Cancel

Create a **Routing Policy** using the **New** button highlighted below.

Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Session Manager x Home

Home / Elements / Routing / Routing Policies- Routing Policies

Routing Policies

Edit New Duplicate Delete More Actions

2 Items Refresh Filter: Enable

| <input type="checkbox"/> | Name | Disabled | Destination | Notes |
|--------------------------|---------------|--------------------------|-------------|-------|
| <input type="checkbox"/> | to Audiocodes | <input type="checkbox"/> | Audiocodes | |
| <input type="checkbox"/> | to Cores3 | <input type="checkbox"/> | Cores3 | |

Select : All, None

Select a **Name** and a **SIP Entity** and click **Commit** highlighted below.

Routing Policy Details

General

* **Name:** to Audiocodes

Disabled: ☐

Notes:

SIP Entity as Destination

| Name | FQDN or IP Address | Type | Notes |
|------------|--------------------|-----------|-------------------|
| Audiocodes | 47.166.92.210 | SIP Trunk | AudioCodes SIP GW |

Create a **Dial Pattern** by clicking **New** highlighted below.

AVAYA Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing x **Session Manager** x **Home**

Home / Elements / Routing / Dial Patterns- Dial Patterns

Dial Patterns

Edit **New** Duplicate Delete More Actions

3 Items Refresh Filter: Enable

| Pattern | Min | Max | Emergency Call | SIP Domain | Notes |
|---------|-----|-----|--------------------------|------------|-------|
| 2602 | 4 | 4 | <input type="checkbox"/> | -ALL- | |
| 322 | 3 | 4 | <input type="checkbox"/> | -ALL- | |
| 5 | 4 | 4 | <input type="checkbox"/> | -ALL- | |

Select : All, None

The digits that are to be routed to the MP118 are entered against **Pattern** along with the **Min** and **Max** length of these digits and **Commit** is pressed to save the configuration. In the below example 322 is entered so all numbers from 3220 to 3229 will be routed to the MP118 IP address over SIP.

Dial Pattern Details

General

* **Pattern:** 322

* **Min:** 3

* **Max:** 4

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

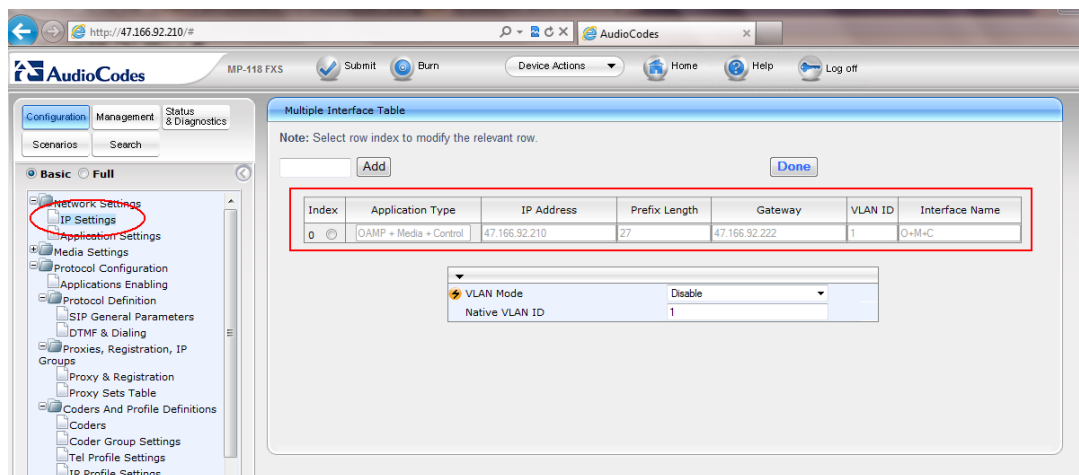
6. Configuring the AudioCodes MP118 VoIP Gateway

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

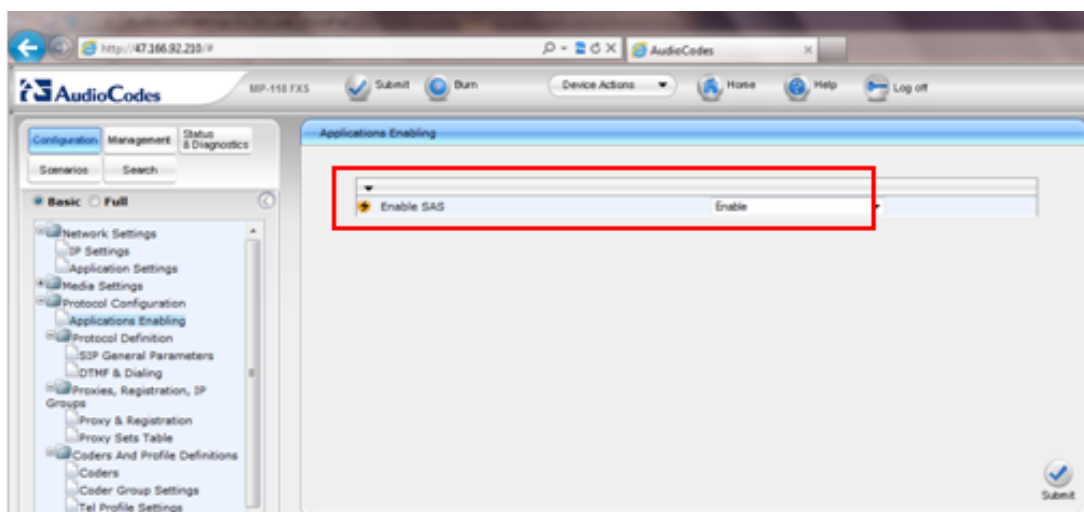
Note: As the SBX performs call transfers the call is placed on hold in the MP118 hence no music on hold is heard.

6.1. IP Settings on the AudioCodes MP118

Under the setting **IP Settings - IP Address and Gateway** of the MP118 are entered into the **IP settings** as illustrated below.



Under the setting **Applications Enabling** - the SAS has been enabled.

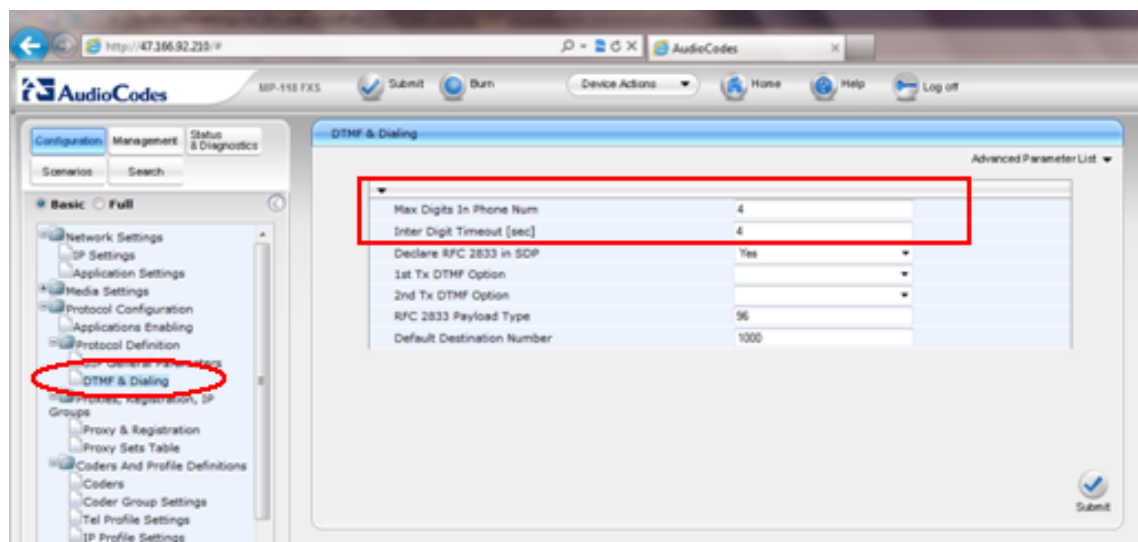


6.2. AudioCodes MP118 Protocol Configuration

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

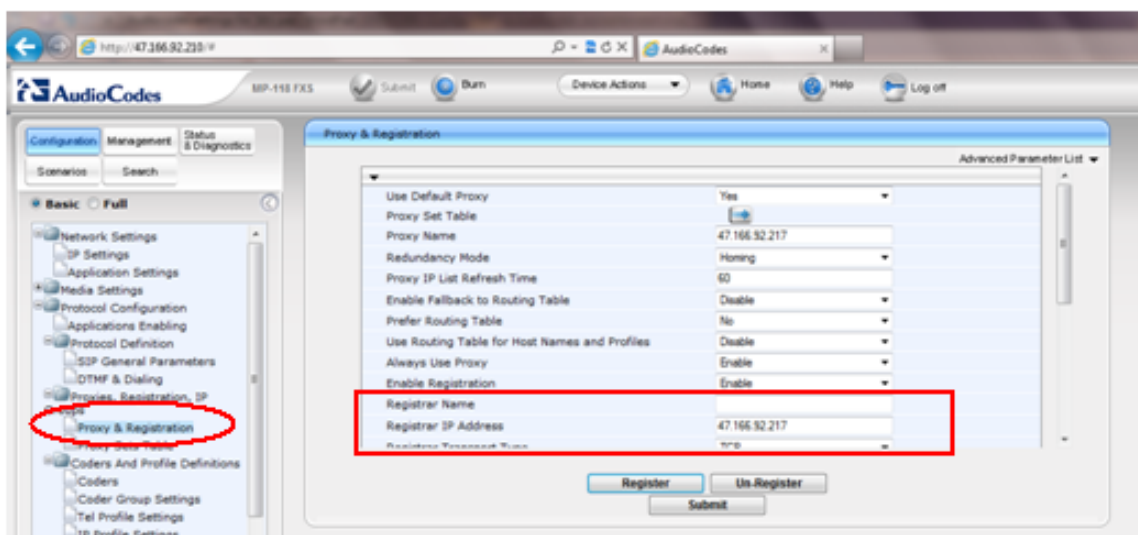
6.2.1. Protocol Definition

Under the setting **DTMF and Dialling** select the correct number of digits for the dial plan.



6.2.2. Groups

Under the setting **Proxy and Registration** the **Registrar IP Address** is the IP address of the Session Manager the MP118 is registering to. This is entered as highlighted below.



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

The screenshot shows the 'Proxy & Registration' configuration page in the AudioCodes MP-118 FXS web interface. The 'Registrar Transport Type' is set to 'TCP'. Other settings include Registration Time (1800), 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 (47.166.92.217), Gateway Registration Name, Subscription Mode (Per Endpoint), User Name, Password (Default_Passwd), and Chonce (Default_Chonce). Buttons for Register, Un-Register, and Submit are at the bottom.

| Parameter | Value |
|----------------------------------|----------------|
| Registrar Transport Type | TCP |
| Registration Time | 1800 |
| 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 | 47.166.92.217 |
| Gateway Registration Name | |
| Subscription Mode | Per Endpoint |
| User Name | |
| Password | Default_Passwd |
| Chonce | Default_Chonce |

Under the setting **Proxy Set Table** enter the Session Manager IP address for the **Proxy Address** and enter the transport type.

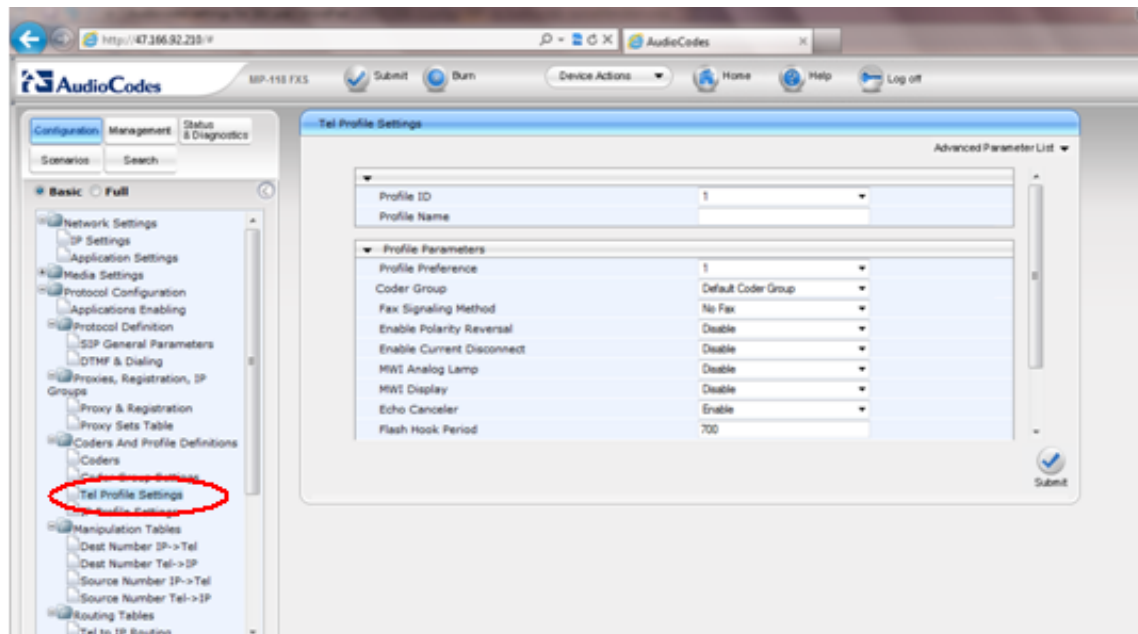
The screenshot shows the 'Proxy Sets Table' configuration page in the AudioCodes MP-118 FXS web interface. The 'Proxy Set ID' is 0. The table has two columns: 'Proxy Address' and 'Transport Type'. The first row is filled with '47.166.92.217' and 'TCP'. Below the table is a checkbox for 'Enable Proxy Keep Alive' which is currently disabled. A 'Submit' button is at the bottom right.

| Proxy Set ID | Proxy Address | Transport Type |
|--------------|---------------|----------------|
| 0 | 47.166.92.217 | TCP |
| 1 | | |
| 2 | | |
| 3 | | |
| 4 | | |
| 5 | | |

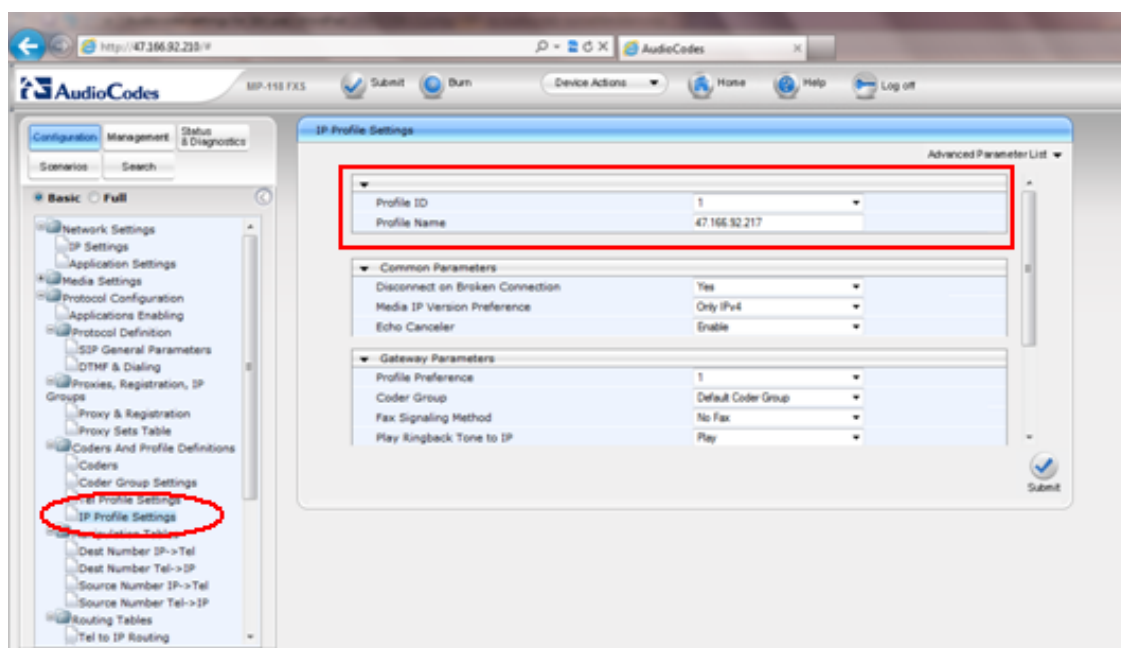
Enable Proxy Keep Alive: ☐ Disable

6.2.3. Coders and Profile Definitions

Under the setting **Tel Profile Settings** setup **Profile ID 1** as shown below this number is required for future settings.



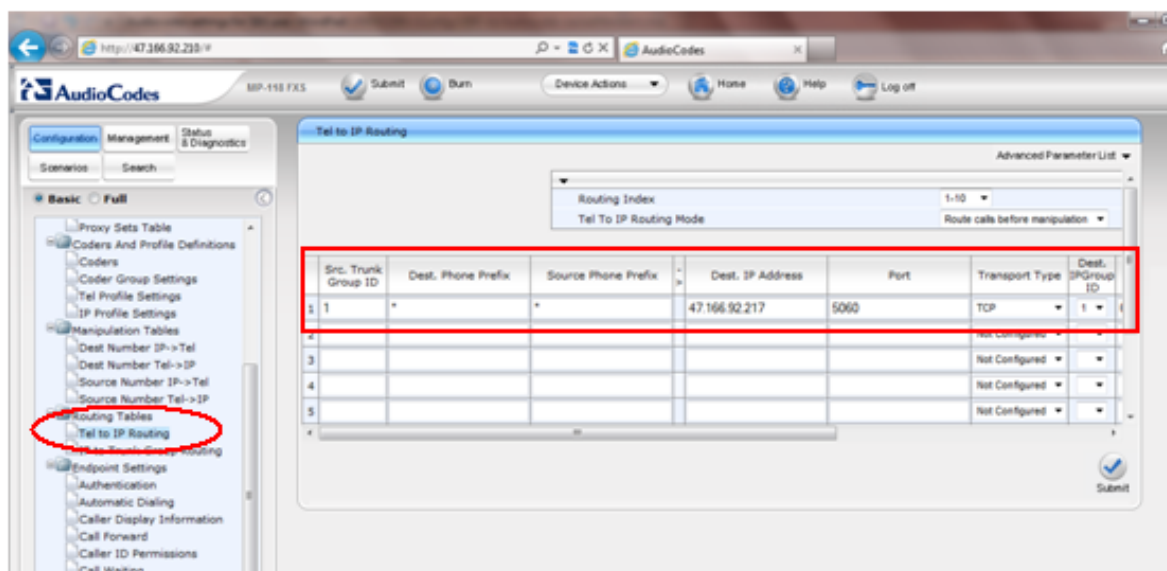
Under the setting **IP Profile Settings**, note the **Profile ID** that was configured above.



6.3. AudioCodes MP118 Routing table

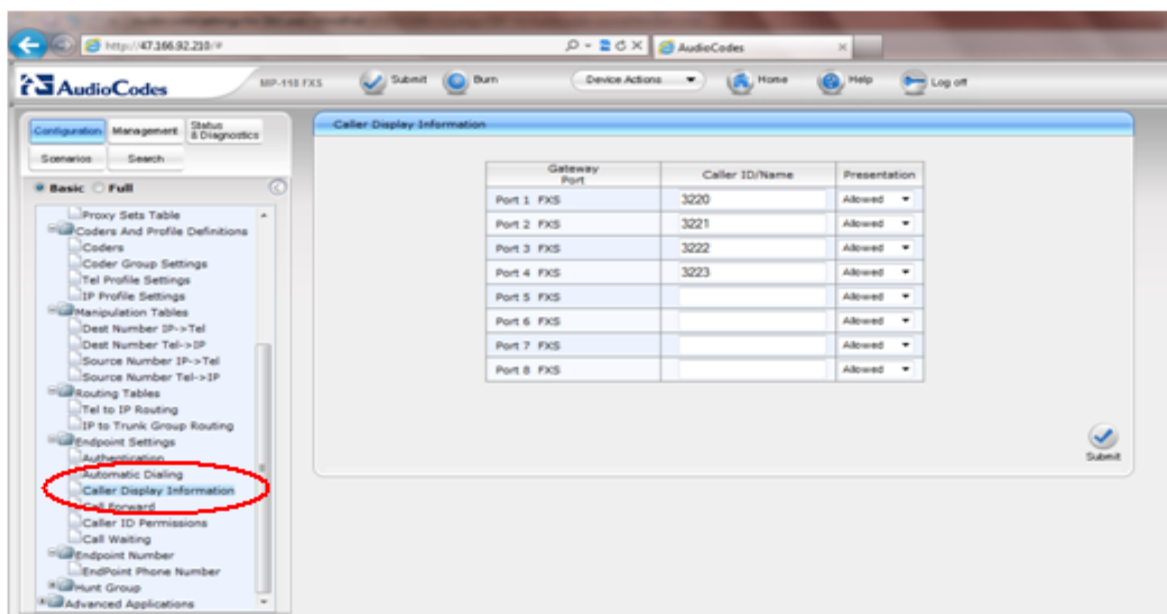
Under the setting **Tel to IP Routing** all calls are routed to the Session Manager.

Note: “*” is entered in the destination and source for the prefix that denotes **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 is 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.

Note: The **Tel Profile ID** which was created in **section 6.2.3**

The screenshot shows the AudioCodes MP118 configuration web interface. The left sidebar contains a tree view of configuration options, with 'Endpoint Phone Number' highlighted. The main area displays the 'Endpoint Phone Number Table' with the following data:

| | Channel(s) | Phone Number | Hunt Group ID | Tel Profile ID |
|---|------------|--------------|---------------|----------------|
| 1 | 1 | 3220 | | 1 |
| 2 | 2 | 3221 | | 1 |
| 3 | 3 | 3222 | | 1 |
| 4 | 4 | 3223 | | 1 |
| 5 | 5 | 1403555414 | | 0 |
| 6 | 6 | 1403555415 | | 0 |
| 7 | 7 | 1403555416 | | 0 |
| 8 | 8 | 1403555417 | | 0 |

Below the table are buttons for 'Register', 'Un-Register', and 'Submit'.

7. Configure the Interalia SBX

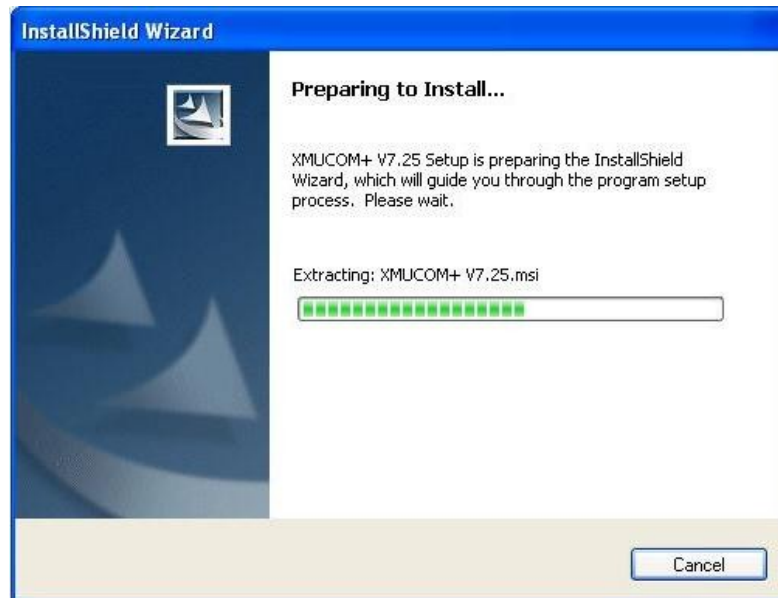
The following section documents the necessary steps taken to configure the SBX.

7.1. Installing Interalia SBX

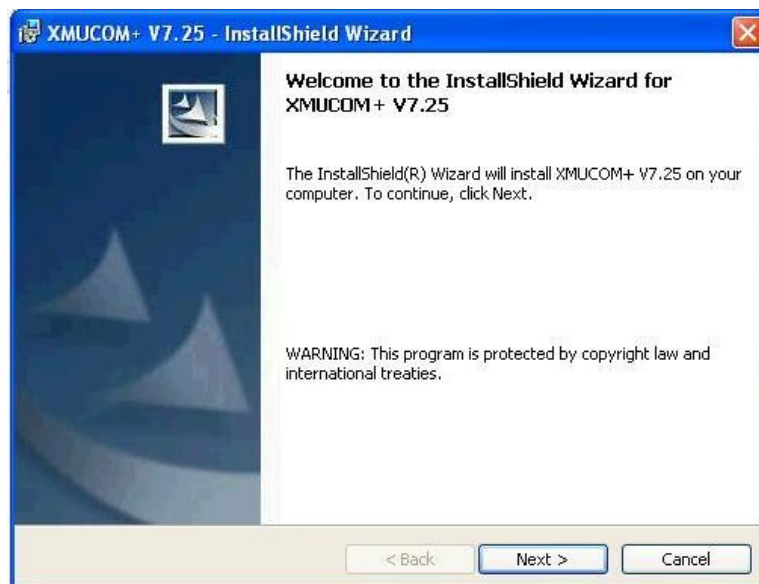
The installation of the SBX is from a CD containing the software. After placing the CD into a PC it automatically starts to the following screen. Click on **Install XMUCOM+(XMU+/SBX)** highlighted.



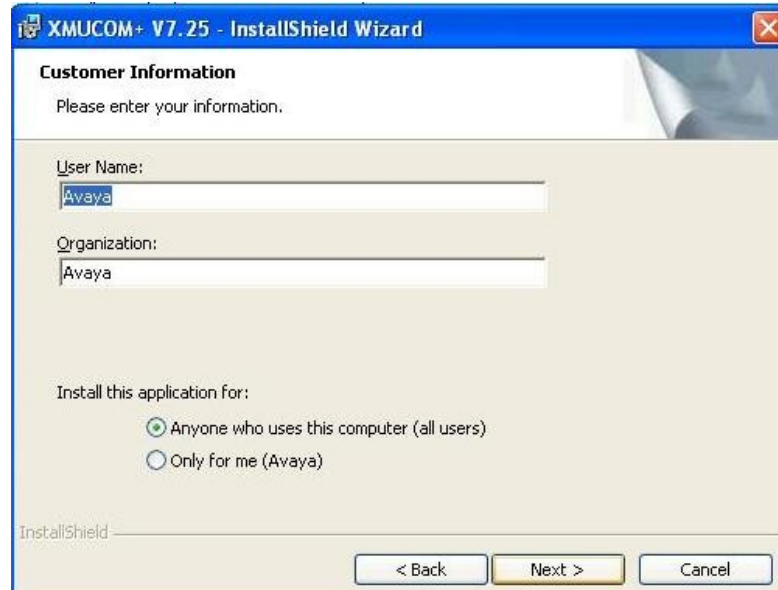
A **Preparing to Install** screen appears below.



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



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



The screenshot shows the 'Customer Information' screen of the XMUCOM+ V7.25 - InstallShield Wizard. The window title is 'XMUCOM+ V7.25 - InstallShield Wizard'. The main heading is 'Customer Information' with the instruction 'Please enter your information.' Below this, there are two text input fields: 'User Name:' with 'Avaya' entered, and 'Organization:' with 'Avaya' entered. Underneath, there is a section 'Install this application for:' with two radio button options: 'Anyone who uses this computer (all users)' (which is selected) and 'Only for me (Avaya)'. At the bottom left, the 'InstallShield' logo is visible. At the bottom right, there are three buttons: '< Back', 'Next >' (highlighted in yellow), and 'Cancel'.

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

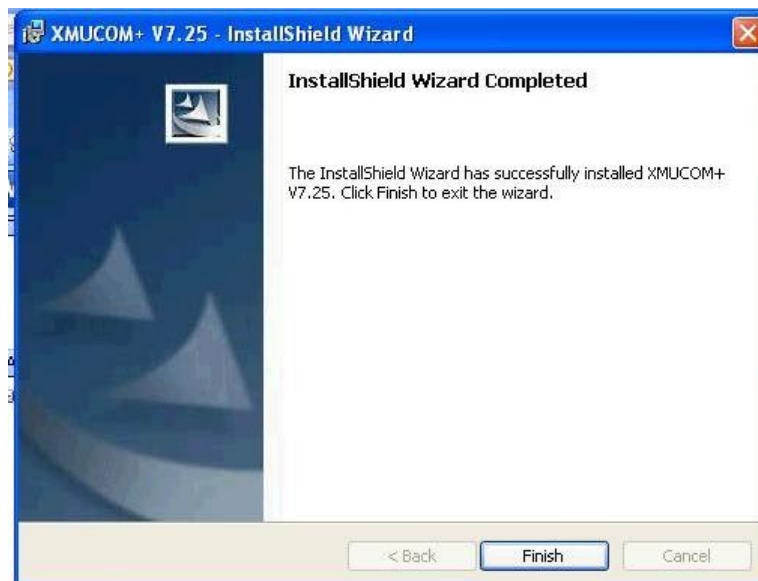


The screenshot shows the 'Destination Folder' screen of the XMUCOM+ V7.25 - InstallShield Wizard. The window title is 'XMUCOM+ V7.25 - InstallShield Wizard'. The main heading is 'Destination Folder' with the instruction 'Click Next to install to this folder, or click Change to install to a different folder.' Below this, there is a folder icon and the text 'Install XMUCOM+ V7.25 to: C:\Program Files\'. To the right of this text is a 'Change...' button. At the bottom left, the 'InstallShield' logo is visible. At the bottom right, there are three buttons: '< Back', 'Next >' (highlighted in yellow), and 'Cancel'.

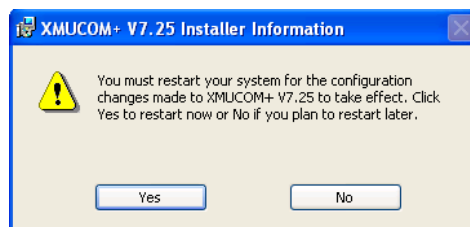
Review the selected settings and click **Install**.



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

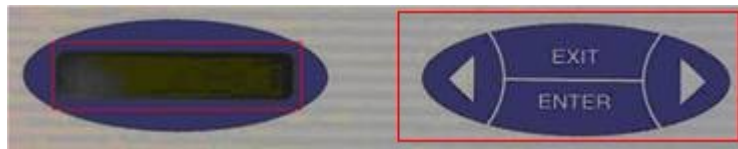


Restart the PC to allow configuration changes to take effect. Choose the **Yes** button to restart.



7.2. Setting the IP address on the Intermedia SBX

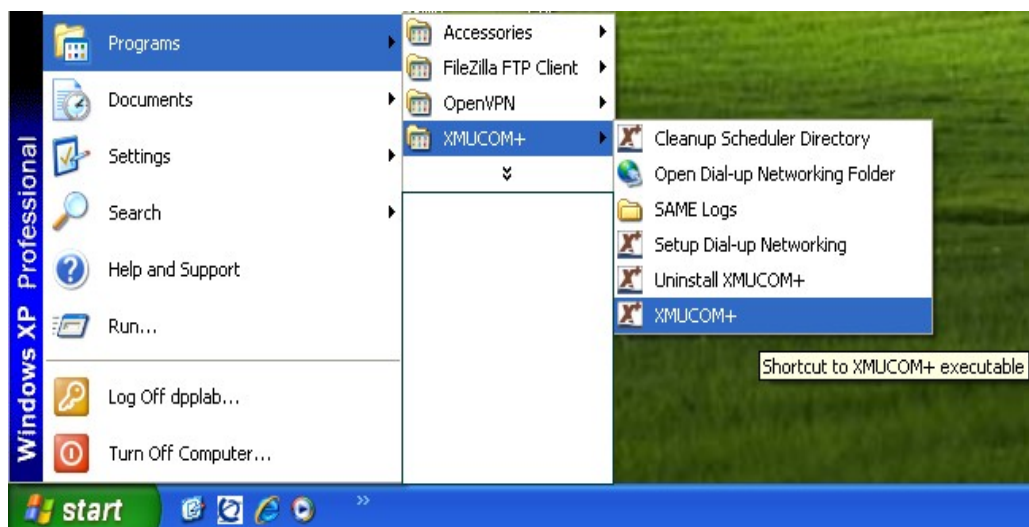
The IP address of the SBX is set from the menu on the front of the box by navigating through the menu using the left and right arrows and clicking **Enter** for each section that needs changing as highlighted below.



To set the IP address navigate to **Main System Menu** click **Enter** and navigate to **System Communications** press **Enter** and navigate to **Communications Ethernet** and press **Enter** then navigate to each of **Ethernet IP address** - **Ethernet IP mask** – **Gateway** to fill in the information.

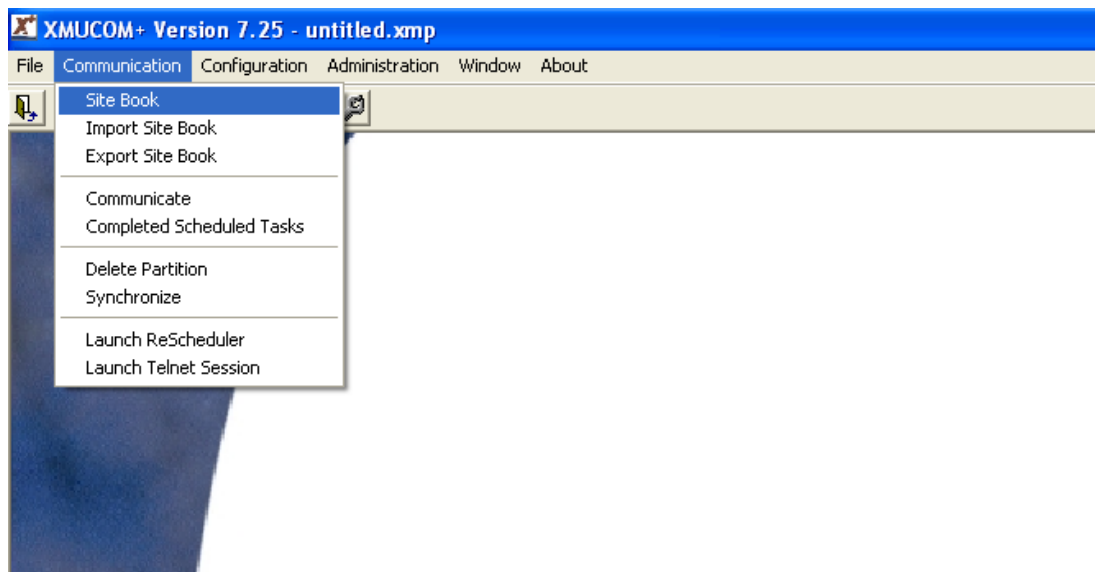
7.3. Initial Configuration of the Intermedia SBX to create a new site

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

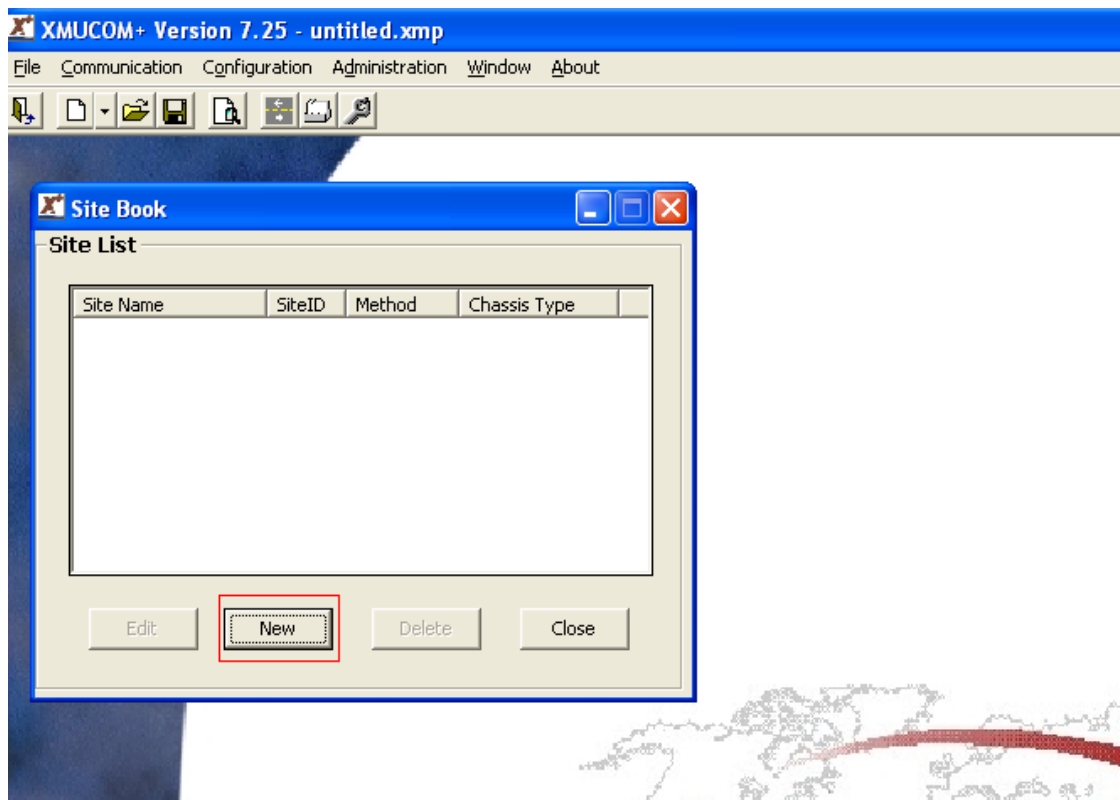


Note: The same program is used to configure the SBX and the XMU+ and the same screen options appear for both.

To create a Site Book on the Intermedia SBX choose the **Communication** Menu and select **Site Book** from the drop down menu

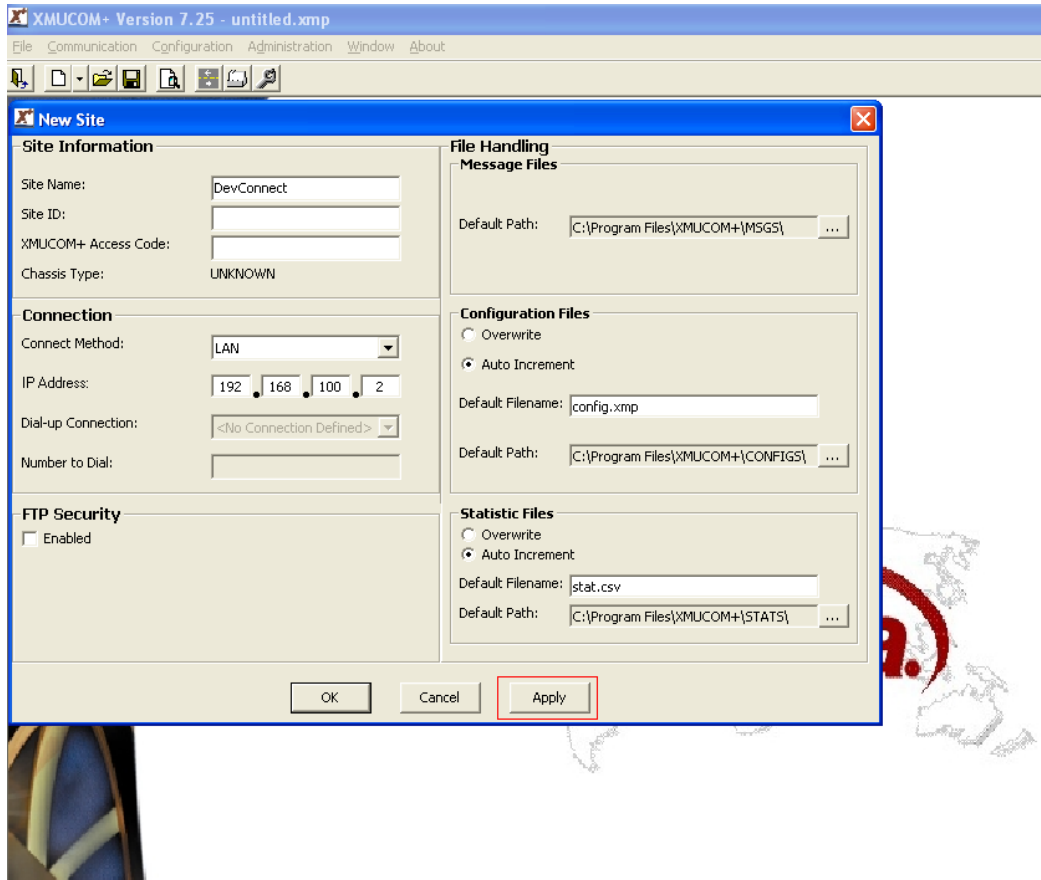


Click on the **New** Button

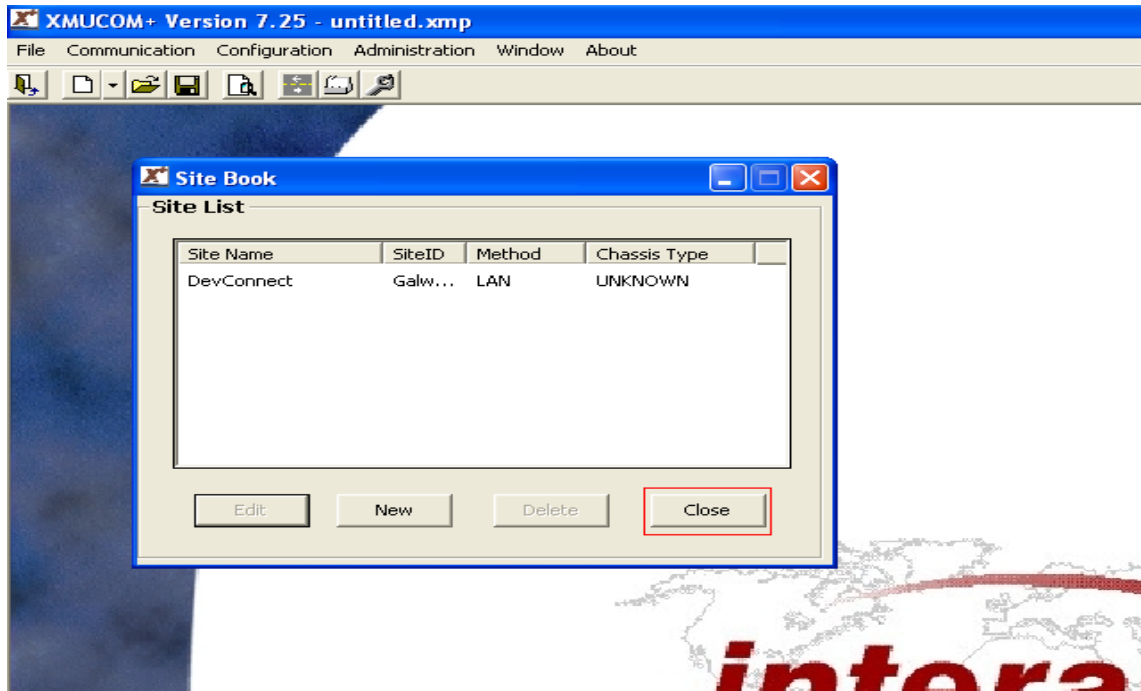


Enter the following information then select **Apply** followed by **OK**

- **Site Name**
- **Connect Method** (LAN was used for Compliance testing)
- **IP Address**



Click the **Close** button

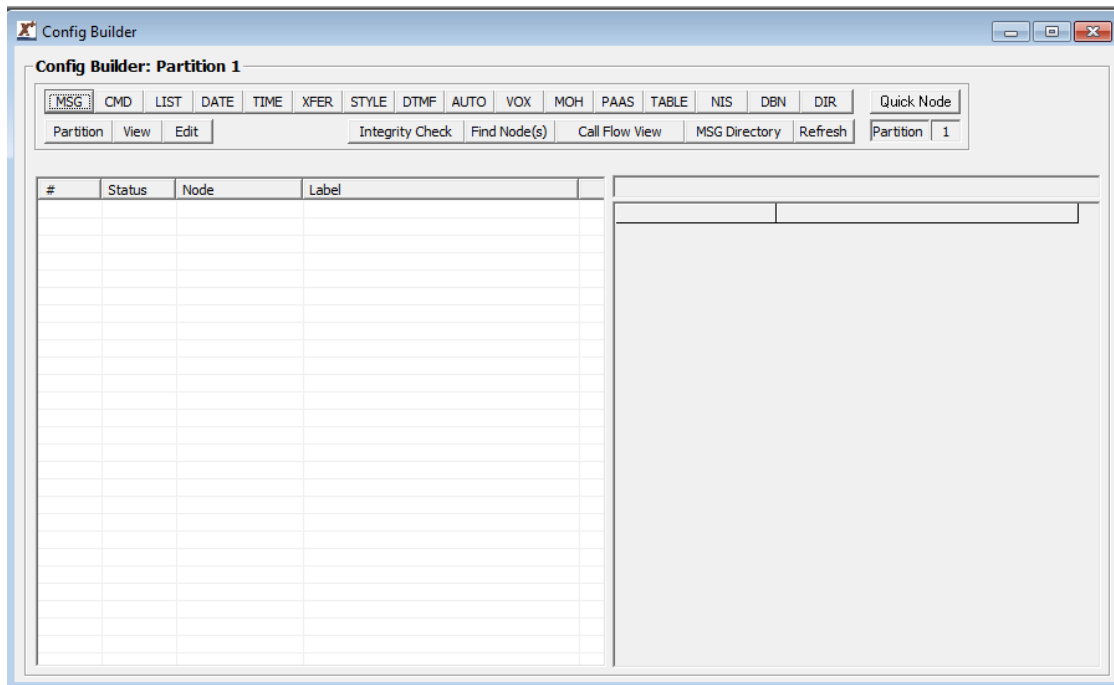


7.4. To configure the Ran announcements on the SBX

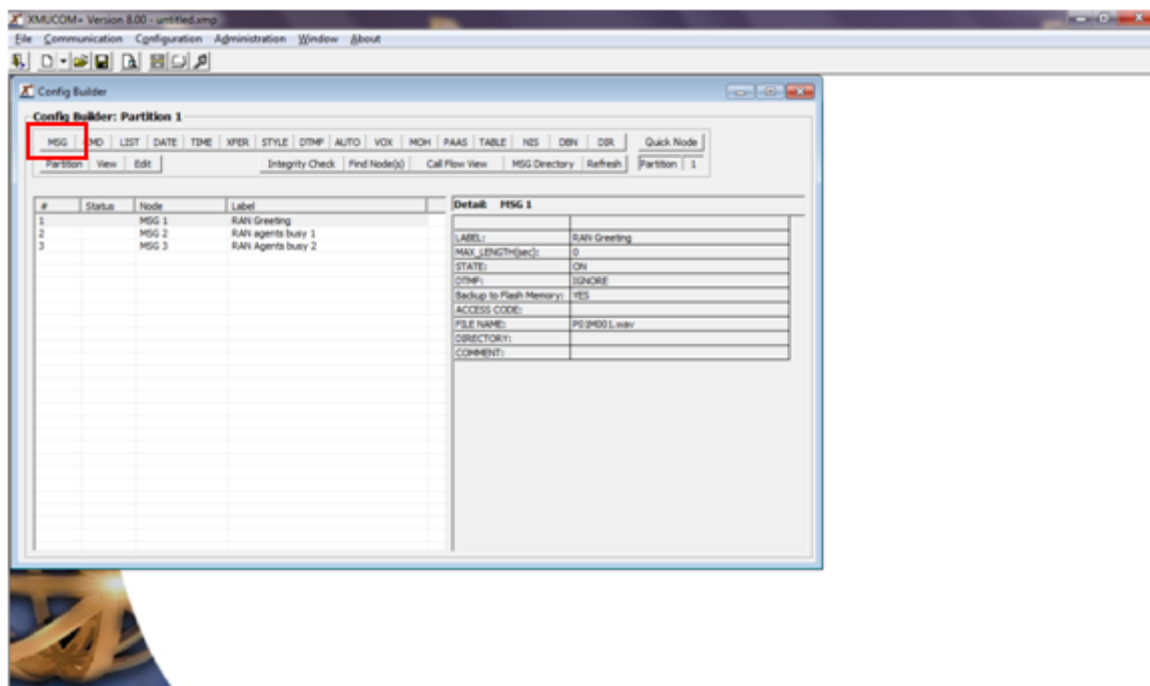
Select **File** → **New** → **SBX Configuration**.



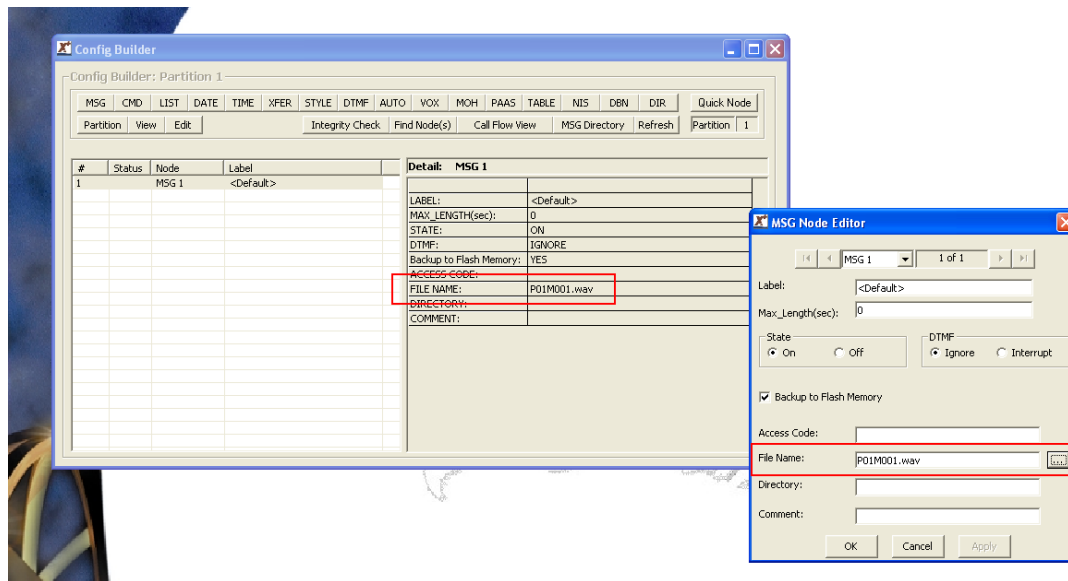
The software then opens with the following blank **Config Builder**.



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



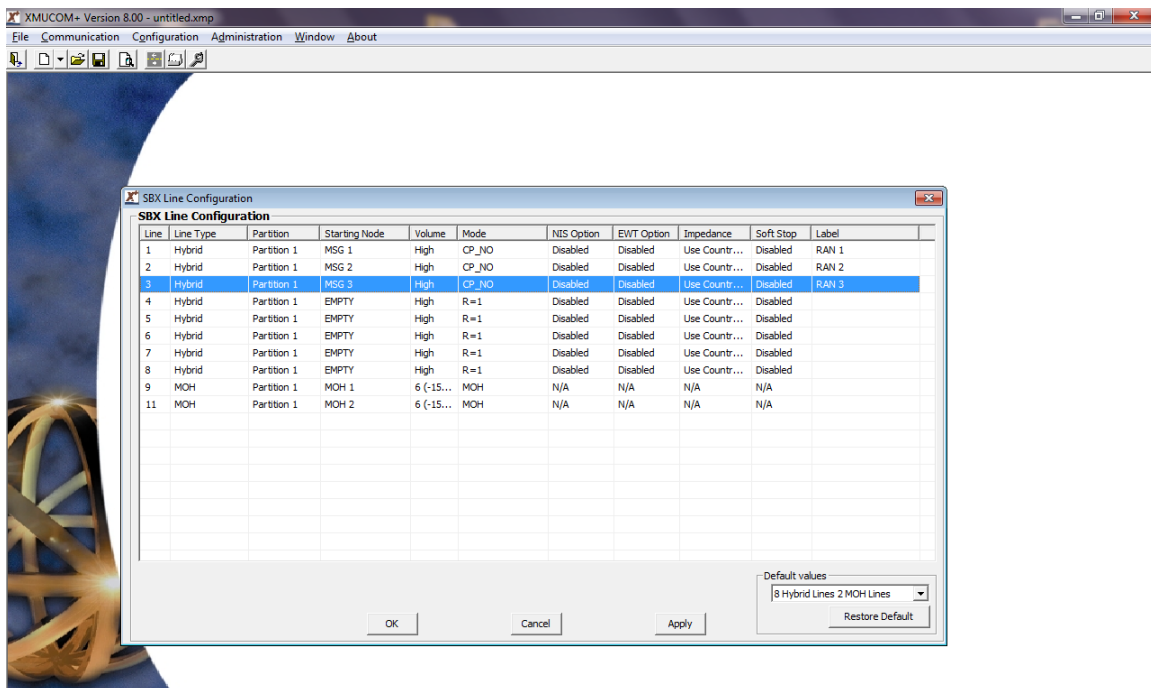
To add specific announcement messages double click on **FILE NAME** highlighted below. This opens the **MSG Node Editor** window. Beside the **File Name** browse to saved announcements in WAV format.



To assign these messages to the SBX ports enter the SBX line configuration as shown below.



The SBX 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 dialling the corresponding extensions on the MP118.



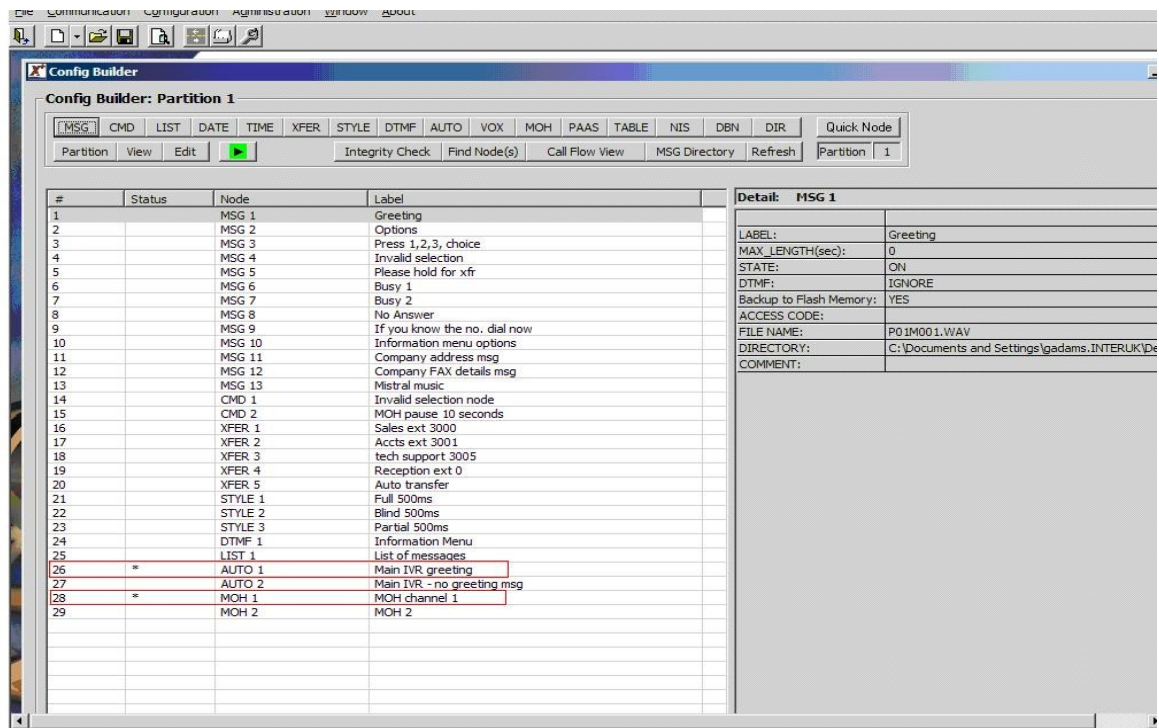
7.5. Configuring the Auto Attendant on the Intermedia SBX

Open the XMUCOM+ program in the installed 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.

| # | 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 |

| Detail: MSG 1 | |
|-------------------------|---------------------------------------------|
| LABEL: | Greeting |
| MAX_LENGTH(sec): | 0 |
| STATE: | ON |
| DTMF: | IGNORE |
| Backup to Flash Memory: | YES |
| ACCESS CODE: | |
| FILE NAME: | P01M001.WAV |
| DIRECTORY: | C:\Documents and Settings\gadams\INTERUK\De |
| COMMENT: | |

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. It displays set **Prompts**, **Actions** and **Exceptions**.

The screenshot shows the 'AUTO Node Editor' window. At the top, there's a navigation bar with 'AUTO 1' selected and '1 of 2' pages. The 'Label' field is set to 'Main IVR greeting'.

Prompts Section:

| Order | Command/Node | Label |
|-------|--------------|---------------------|
| 1 | MSG 1 | Greeting |
| 2 | MSG 9 | If you know the ... |
| 3 | MSG 2 | Options |

Buttons: Add, Insert, Remove, Edit, Up, Down, Quick Assign.

Actions Section:

| Entry | Command/Node | Label |
|-------------|--------------|-------------------|
| 1-1 | XFER 1 | Sales ext 3000 |
| 2-2 | XFER 2 | Accts ext 3001 |
| 3-3 | XFER 3 | tech support 3005 |
| 4-4 | DTMF 1 | Information Menu |
| 3000-3010 | XFER 5 | Auto transfer |
| 62000-63000 | XFER 5 | Auto transfer |

Buttons: Add, Insert, Remove, Edit, Up, Down, Quick Assign.

Exceptions Section:

| | Command/Node | Label |
|---------|--------------|-----------------------|
| Timeout | XFER 4 | Reception ext 0 |
| Invalid | CMD 1 | Invalid selection ... |
| Abort | XFER 4 | Reception ext 0 |
| * | NOOP | |
| # | NOOP | |

Buttons: Edit, Default.

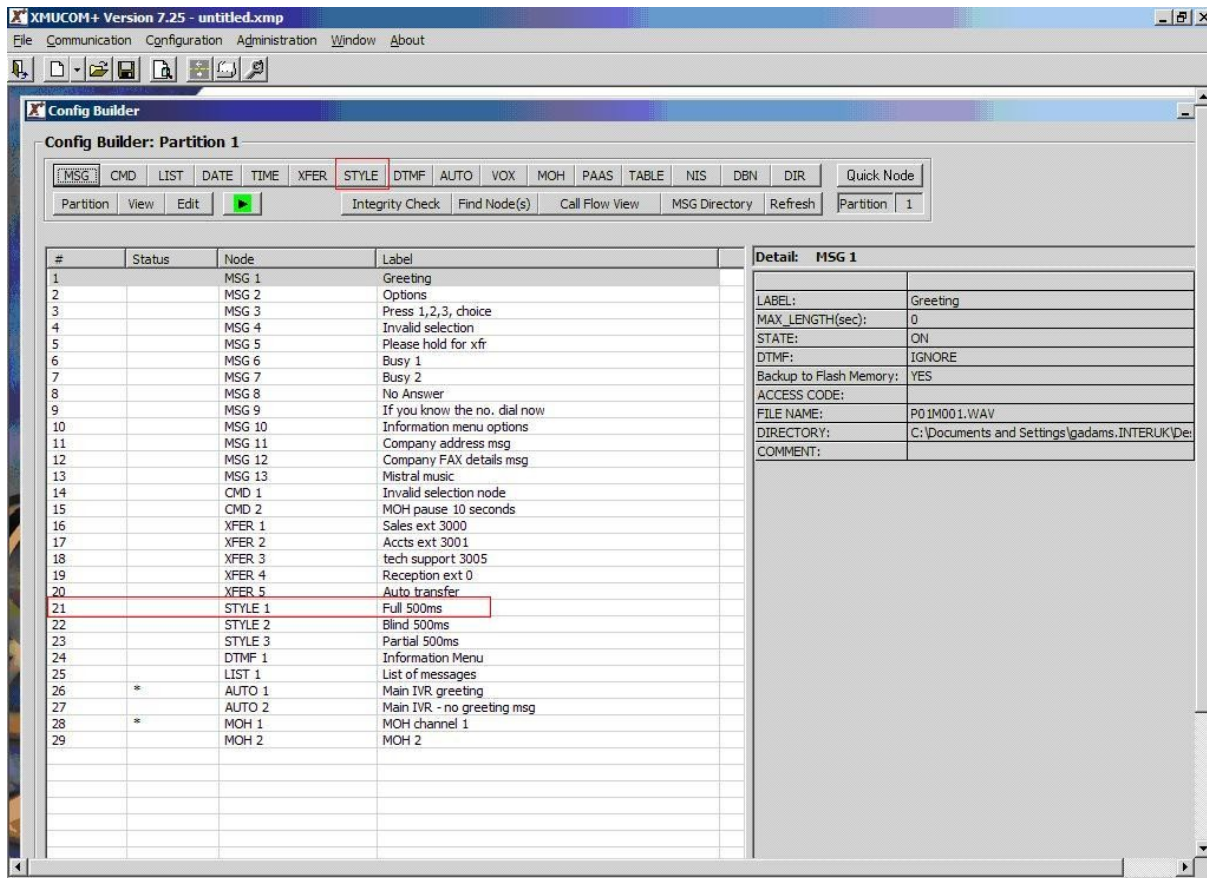
Configuration Fields:

- Number of Digits: 5
- Terminating Digit: No
- Retry Limit: 1
- Selection Time Out(sec): 5
- Digit Time Out(sec): 2

Buttons: OK, Cancel, Apply.

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 SBX 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 Supervised).



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 **500ms** in order to perform the transfer / retrieval of a call.

STYLE Node Editor

STYLE 1 1 of 3

Label: Full 500ms

Supervised

☐ BLIND ☐ PARTIAL ☒ FULL

☐ DIALCONTINUE ☐ ENHANCED

Hook Flash (msec): 500

Transfer: !

Disconnect:

Threshold: Low

Retrieve: !,!

Busy Retry (sec): 4

Num Retries: 3

Ring Limit: 8

Busy On (msec): 500

Busy Off (msec): 500

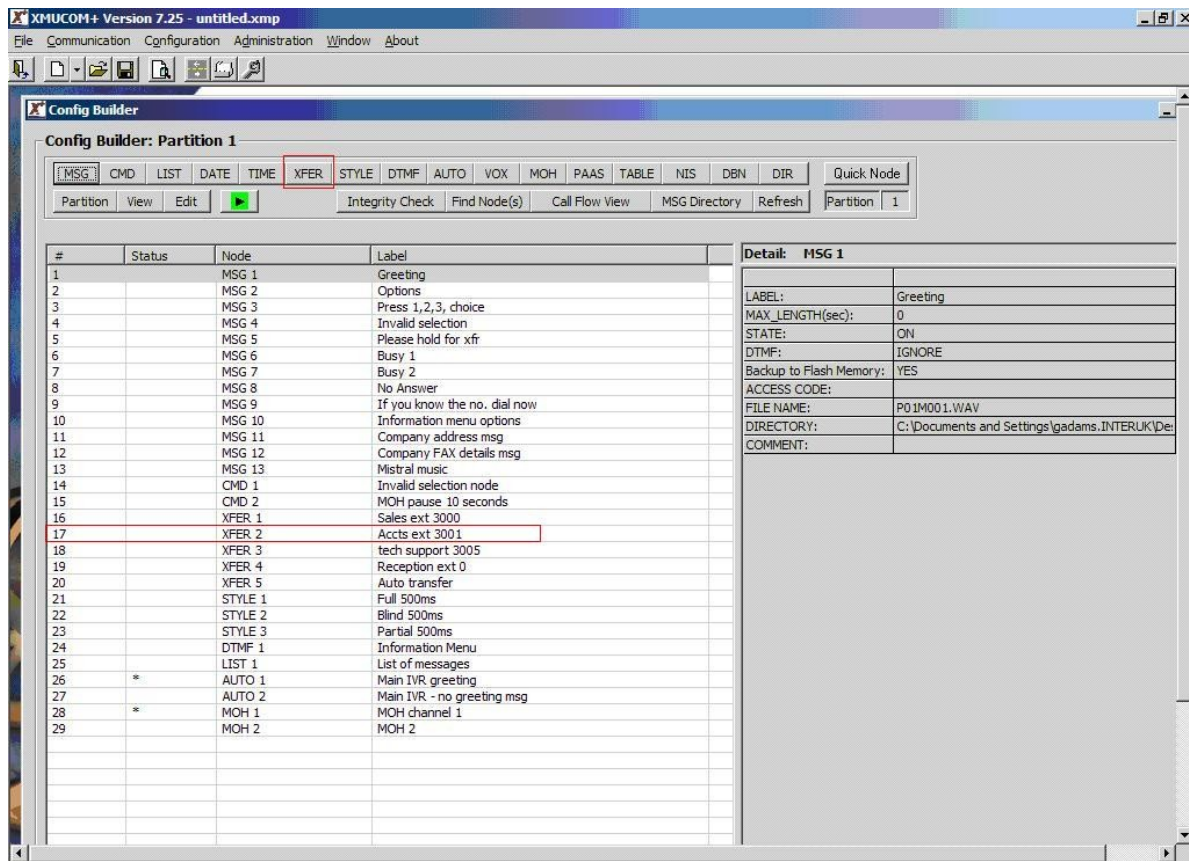
Ring On (msec): 1000

Ring Off (msec): 3000

OK Cancel Apply

Alter the Hook Flash values according to the call progress tones set for country option.

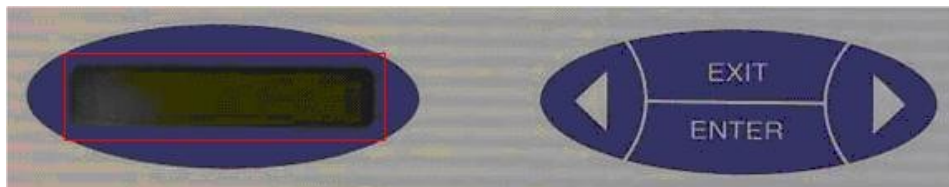
During the monitoring of a call, it is possible to inform the caller of their progress. In the main configuration builder window you can edit calls transfers i.e. Xfer 2. Click on the **XFER** node as highlighted in the diagram below.



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.

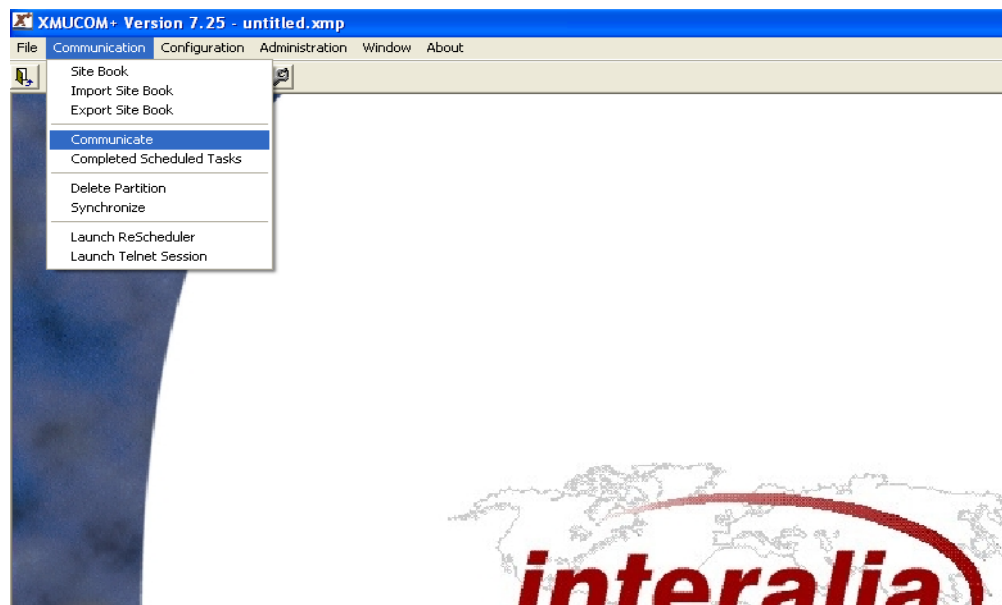
| | Command/Node | Label |
|-----------|--------------|---------------------|
| Hold | MSG 5 | Please hold for xfr |
| MOH | NOOP | |
| Busy 1 | MSG 6 | Busy 1 |
| Busy 2 | MSG 7 | Busy 2 |
| No Answer | MSG 8 | No Answer |
| Answer | BEEP | |

Note: Whilst carrying out the fully supervised transfers, the status mode on the front of the SBX display will change to indicate the different call states e.g. B, b, B etc or R, r, R.

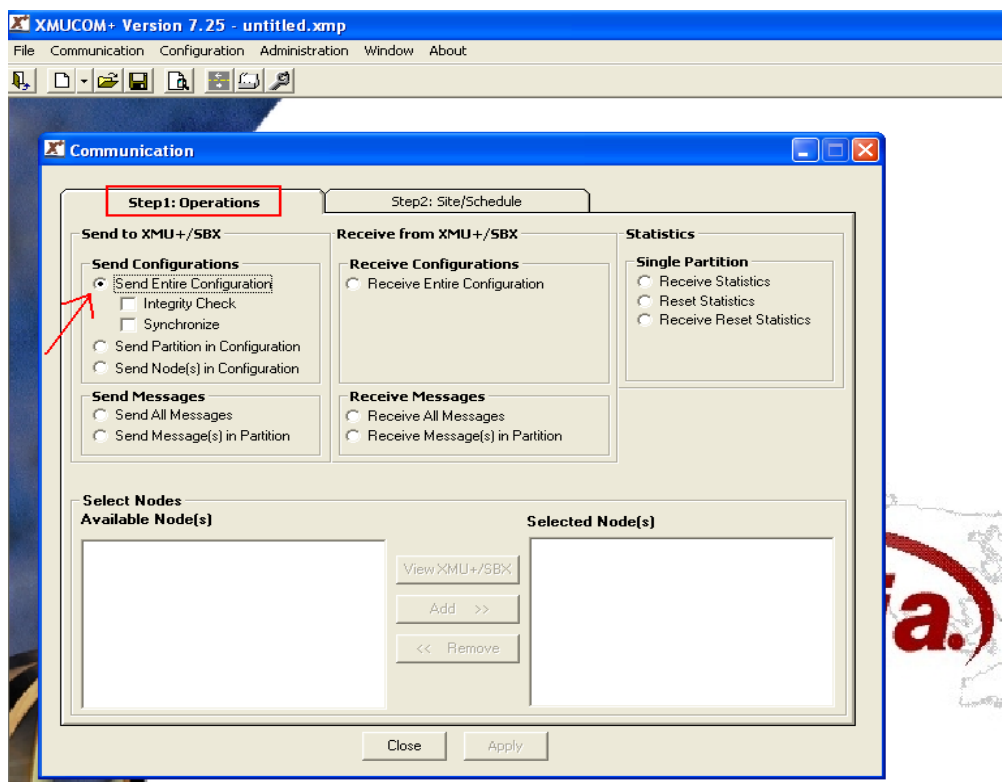


7.6. Downloading configuration to the SBX

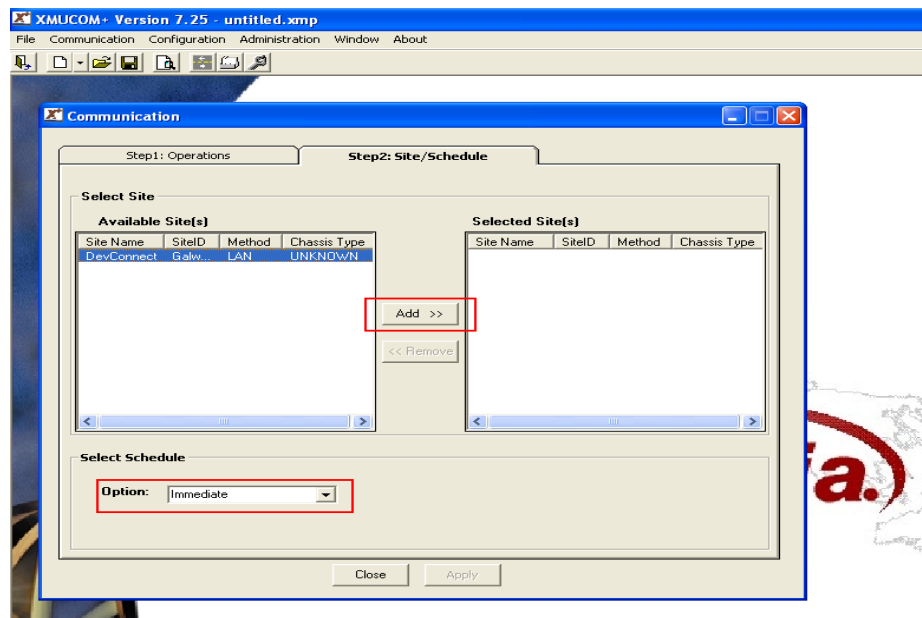
Choose the **Communicate** from the **Communication** Menu.



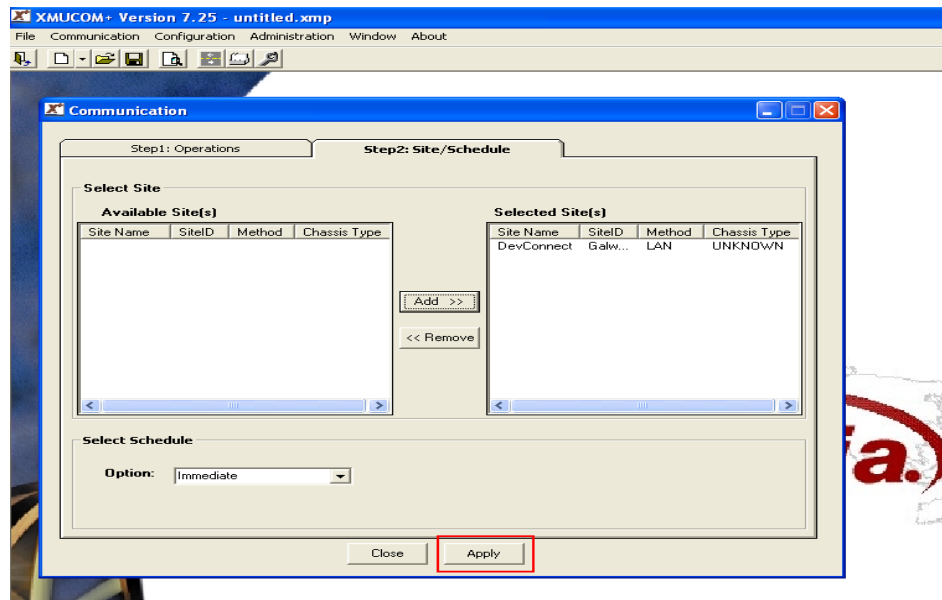
In the **Communication** Window, select Tab **Step 1: Operations** check the **Send Entire Configuration** Button.



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 you are downloading. XMUCOM+ will display a response dialog box while it is sending the messages, and notifies when all messages are sent.

8. Verification Steps

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

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

Check the status of the D-Channel setup in **Section 5.1** by running the command **STAT DCH** in overlay 96 as shown below.

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 |

Example shown below shows the D-Channel 66 shows it is operational and established.

| | | | | |
|----------------|-----|------|------|----------------|
| DCH 066 : OPER | EST | ACTV | AUTO | DES : VTRK_DCH |
|----------------|-----|------|------|----------------|

8.2. Status of the SIP trunks on Avaya Communication Server 1000E

Using overlay 32 stat the SIP trunks to ensure they all appear **idle**.

LD 32

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

Example shown 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 | | |
| 04 = UNIT 04 = IDLE | (ISL TRK) | (TIE IP IMM /IMM) |
| D-CH 66 EST ACTV | | |

8.3. 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.



Avaya Aura™ System Manager 6.1

Session Manager Dashboard
This page provides the overall status and health summary of each administered Session Manager.

Session Manager Instances

Service State Shutdown System As of 10:18 AM

1 Item Refresh Show ALL

| <input type="checkbox"/> | Session Manager | Type | Alarms | Tests Pass | Security Module | Service State | Entity Monitoring |
|--------------------------|---------------------------------|------|----------|------------|-----------------|--------------------|-------------------|
| <input type="checkbox"/> | Session Manager | Core | 31/13/24 | ✓ | Up | Accept New Service | 0/2 |

Select : All, None

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.

SIP Entity Link Monitoring Status Summary
This page provides a summary of Session Manager SIP entity link monitoring status.

Entity Link Status for All Session Manager Instances

Run Monitor

1 Item Refresh

| <input type="checkbox"/> | Session Manager Name | Entity Links Down/Total | Entity Links Partially Down | SIP Entities - Monitoring Not Started | SIP Entities - Not Monitored |
|--------------------------|---------------------------------|-------------------------|-----------------------------|---------------------------------------|------------------------------|
| <input type="checkbox"/> | Session Manager | 0/2 | 0 | 0 | 0 |

Select : All, None

All Monitored SIP Entities

Run Monitor

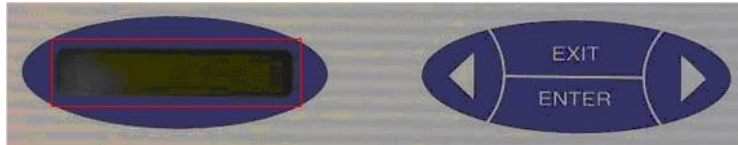
2 Items Refresh Show ALL Filter: Enable

| <input type="checkbox"/> | SIP Entity Name |
|--------------------------|----------------------------|
| <input type="checkbox"/> | AudioCodes |
| <input type="checkbox"/> | Cores3 |

Select : All, None

8.4. Verify SBX Status

The Status window, as highlighted below, on the display on the front of the SBX can be used to verify the communication of the SBX. 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.



9. Conclusion

These Application Notes describe the configuration steps required for Intermedia SBX to successfully interoperate with Avaya Communication Server 1000E using SIP trunks through the use of third party VoIP Gateway from AudioCodes. All functionality and serviceability test cases were completed successfully.

10. Additional References

This section references the Avaya and Intermedia product documentation that are relevant to these Application Notes. Product documentation for Avaya products may be found at <http://support.avaya.com>

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

The Intermedia documentation can be found at the following location:

[1] <http://www.intermedia.com/Products/XMU/XMU-Overview>

[2] <http://www.intermedia.com/Products/SBX/SBX-Overview>

Appendix

VERSION 4121

RELEASE 7

ISSUE 50 Q +

DepList 1: core Issue: 01 (created: 2011-03-15 10:26:33 (est))

IN-SERVICE PEPS

| PAT# | CR # | PATCH REF # | NAME | DATE | FILENAME | SPECINS |
|------|------------|-------------|----------|------------|--------------|---------|
| 000 | wi00688505 | ISS1:1OF1 | p30595_1 | 14/06/2011 | p30595_1.cpl | NO |
| 001 | wi00835294 | ISS1:1OF1 | p30565_1 | 14/06/2011 | p30565_1.cpl | NO |
| 002 | wi00832106 | ISS1:1OF1 | p30550_1 | 14/06/2011 | p30550_1.cpl | NO |
| 003 | wi00837618 | ISS1:1OF1 | p30594_1 | 14/06/2011 | p30594_1.cpl | NO |
| 004 | wi00852365 | ISS1:1OF1 | p30707_1 | 14/06/2011 | p30707_1.cpl | NO |
| 005 | wi00843623 | ISS1:1OF1 | p30731_1 | 14/06/2011 | p30731_1.cpl | YES |
| 006 | wi00839255 | ISS1:1OF1 | p30591_1 | 14/06/2011 | p30591_1.cpl | NO |
| 007 | wi00832626 | ISS2:1OF1 | p30560_2 | 14/06/2011 | p30560_2.cpl | NO |
| 008 | wi00857566 | ISS1:1OF1 | p30766_1 | 14/06/2011 | p30766_1.cpl | NO |
| 009 | wi00841980 | ISS1:1OF1 | p30618_1 | 14/06/2011 | p30618_1.cpl | NO |
| 010 | wi00837461 | ISS1:1OF1 | p30597_1 | 14/06/2011 | p30597_1.cpl | NO |
| 011 | wi00839821 | ISS1:1OF1 | p30619_1 | 14/06/2011 | p30619_1.cpl | NO |
| 012 | wi00842409 | ISS1:1OF1 | p30621_1 | 14/06/2011 | p30621_1.cpl | NO |
| 013 | wi00838073 | ISS1:1OF1 | p30588_1 | 14/06/2011 | p30588_1.cpl | NO |
| 014 | wi00850521 | ISS1:1OF1 | p30709_1 | 14/06/2011 | p30709_1.cpl | YES |
| 015 | wi00860722 | ISS1:1OF1 | p30784_1 | 14/06/2011 | p30784_1.cpl | YES |
| 016 | wi00839134 | ISS1:1OF1 | p30698_1 | 14/06/2011 | p30698_1.cpl | YES |
| 017 | wi00836981 | ISS1:1OF1 | p30613_1 | 14/06/2011 | p30613_1.cpl | NO |

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