Avaya Solution & Interoperability Test Lab

Application Notes for Packet One SIP Trunk System Version 3.1 Interoperability with Avaya Software Communication System Release 4.0 - Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Software Communication System Release 4.0 and Packet One SIP Trunk System Version 3.1. The Primary focus of testing is the system verification of SIP trunk interoperability which includes the call scenarios such as basic call, call forward no answer, call transfer (blind and consult) and conference. Calls were placed in both directions and involved various types of telephones.

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

This document provides a typical network deployment of Avaya Software Communication System (SCS) utilizing the Packet One SIP Trunk System Version 3.1 product offering. This document should serve as general guideline only, since it is not possible to document every possible variation of configuration.

The SCS system is configured as a SIP gateway endpoint on the Packet One network.

1.1. Interoperability Compliance Testing

The System verification testing of the SIP Trunk between the Avaya SCS Release 4.0 and Packet One SIP Trunk System Version 3.1 switch included:

- General call processing between systems including:
  - Codec negotiation (G.729 and G.711 u-law verification)
  - Hold/Retrieve on both ends
  - CLID displayed
  - Ring back tone
  - Speech path
  - Numbering plans
  - Advanced features (such as Call Park, Call Pick up, Conference)

- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward no answer, and conference). Call redirection is performed from both ends.
- DTMF on both directions.
- SIP Transport UDP.
- Voice Mail Server (hosted on Avaya SCS system).

1.2. Caveats

- Packet One system must enable G.711 u/a law to work with SCS features such as call park, conference and voicemail.
- Packet One strongly recommend to use codec G.729 for call park, voicemail and conference, as it consumes much less bandwidth as opposed to G.711.

1.3. Dependencies

Packet One provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

1.4. Support

For technical support on Packet One system, please contact Packet One technical support at:

- http://www.p1.com.my/support/support_overview.aspx

2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliant testing between the Software Communication System 4.0 and Packet One SIP Trunk System Version 3.1.

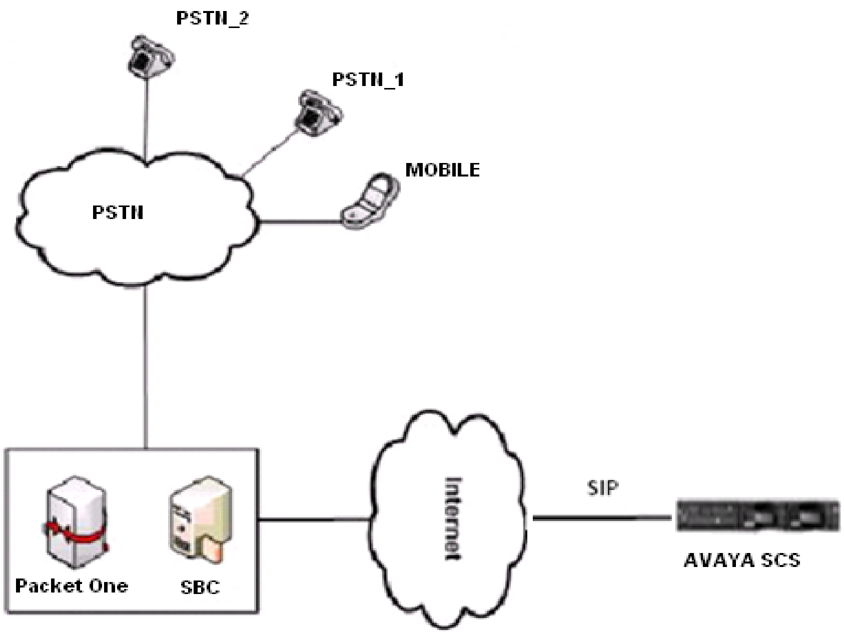

Figure 1 - Network diagram for Avaya - Packet One Setup

All test scenarios involving the establishment of calls will assume the following activities:

1. Calls will be checked for the correct call progress tones and cadences.
2. During the ringing state the ring back tone and destination ringing will be checked.
3. Calls will be checked in both hands-free and handset mode due to internal Avaya requirement.
4. Calls will be checked for speech path in both directions using spoken words to ensure clarity of speech.
5. The display(s) of the sets/clients involved will be checked for consistent and expected CLID (prefer to calling number) and redirection information both prior to answer and after call establishment.
6. The speech path and messaging system will be observed for timely End to End tone audio path generation and application responses.
7. The trace will be captured during the test cases execution for the monitoring of any errors.
8. Speech path and Caller ID are checked before and after the calls, which are put on/off hold from each end.

3. Equipment and Software Validated

The following table consists of hardware system requirement and software/Loadware version.

System	Software/Loadware Version
Avaya Software Communication System running on DELL OPTIPLEX 745	• Release 4.0, Load 4.2.1
Avaya phones	• 12x0: Version 01.02.02 • Polycom: Version 3.1.3 • SMC3456: Version 2.6 - RC14 build 53715
Packet One platform	• Version 3.1
Gateway	• N/A

4. Configure Avaya Software Communication System

4.1. Add a SIP server

This section describes the steps for adding a Server in SCS portal webpage. Enter the IP address of the SCS server in Web Browser to launch and login to the SCS web portal. Navigate to the **System** → **Server**, and then click on **Add Server** as shown in **Figure 2**.

Figure 2 – Add a Server

Input the target SCS Server information; **Hostname**, **IP Address** and **Password** for this Server as shown in **Figure 3**.

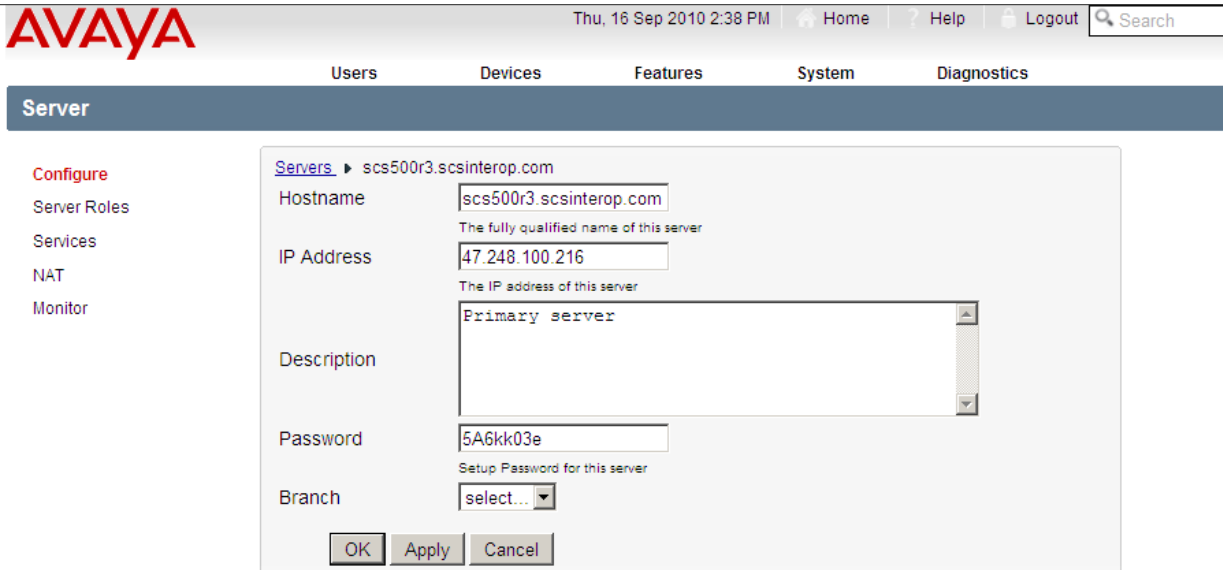

Figure 3 – SCS Server information.

4.2. Configure SCS Domain name

This section describes the steps on how to define a SIP domain name on the SCS Server. Domain Name attribute can be defined as an IP address or a fully qualified host name.
Go to **System** → **Domain**. Input domain name of the SIP server as shown in **Figure 4**.

Figure 4 – Define a domain name.

4.3. Check license on SIP server

This section describes the steps on how to check the license, which should be applied to the SCS system for user registration. Navigate to **System → Server → Licensing**.
New keycode should be generated and applied to SCS server as shown in **Figure 5**.

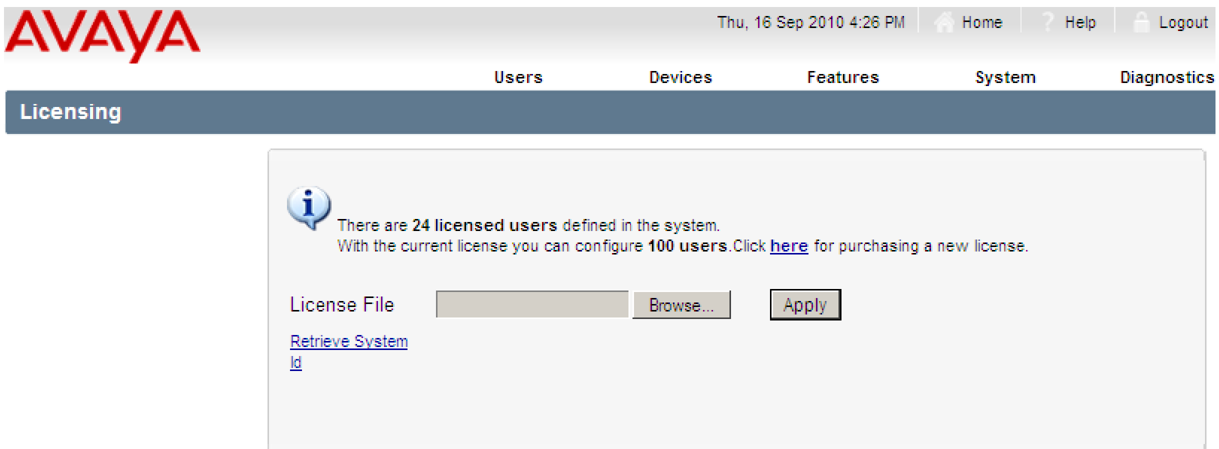

Figure 5 – Check SCS license

4.4. Create SCS user

This section describes how to create users on the SCS server. Go to the SCS server webpage, and click on **Users** menu tab. Then click on **Add New User** link as shown in **Figure 6**.

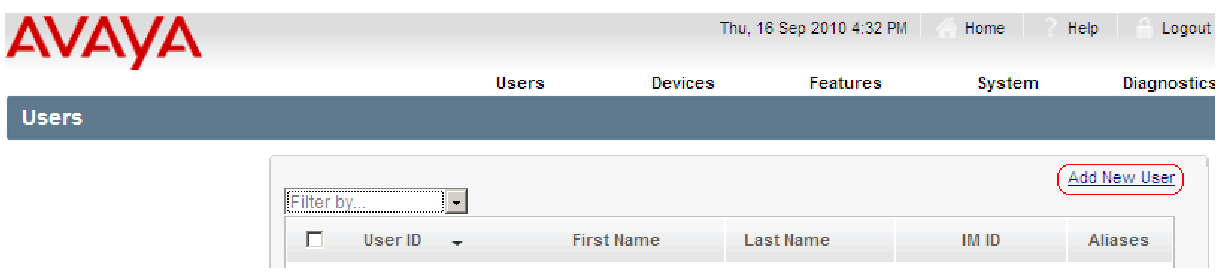

Figure 6 – Create SCS user page

Fill in the **User ID**, **Last**, **First** names, **PIN**, **SIP password** and **Group** as shown in **Figure 7**. Then click on **Apply** button to save the configuration information.

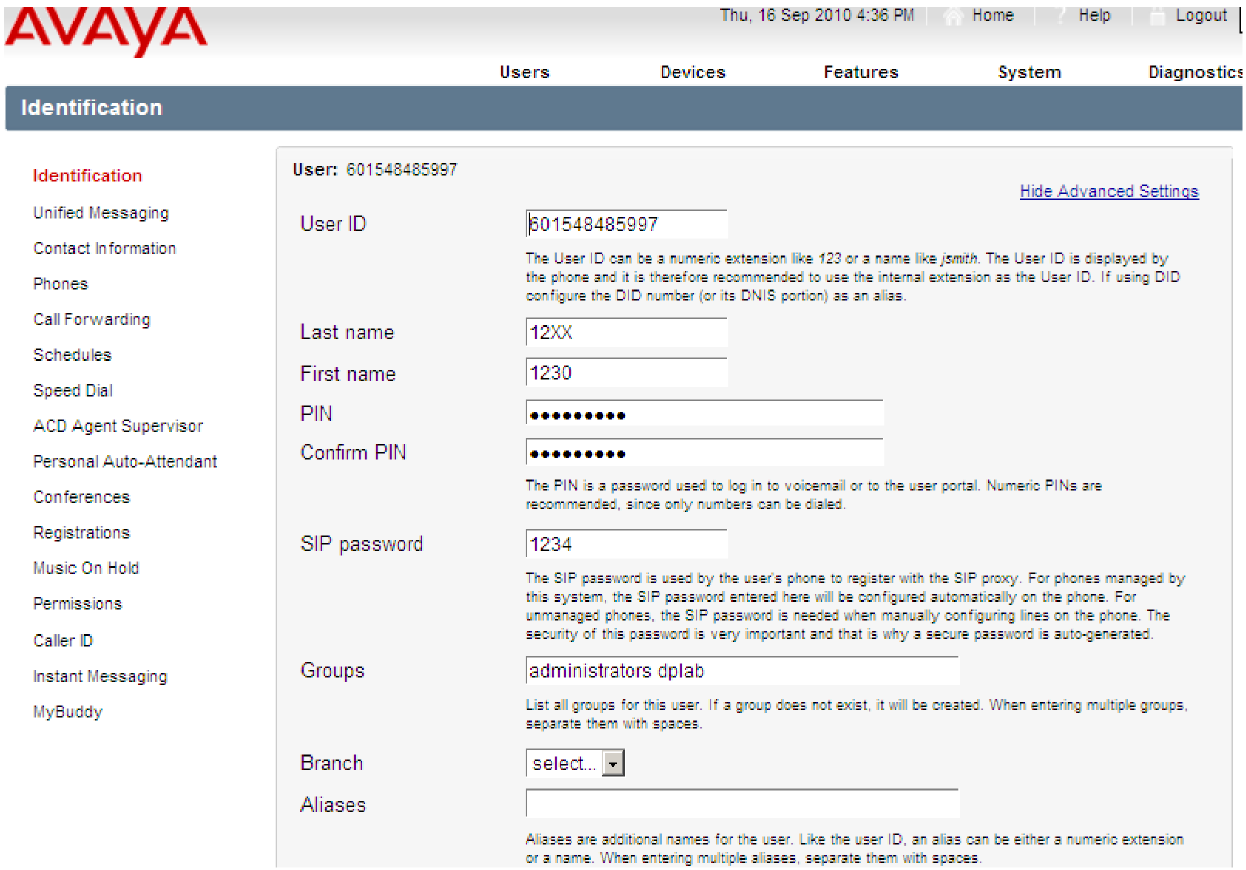

Figure 7 – Creating SCS user

In the Caller ID menu on the left column which is used to define the caller information. This will be sent to the Packet One Session Border Controller (SBC) for call set up.
Select the **Caller ID** menu, the Caller ID page will appear as shown in **Figure 8**. Fill in the Caller ID number and click on the **Apply** button.

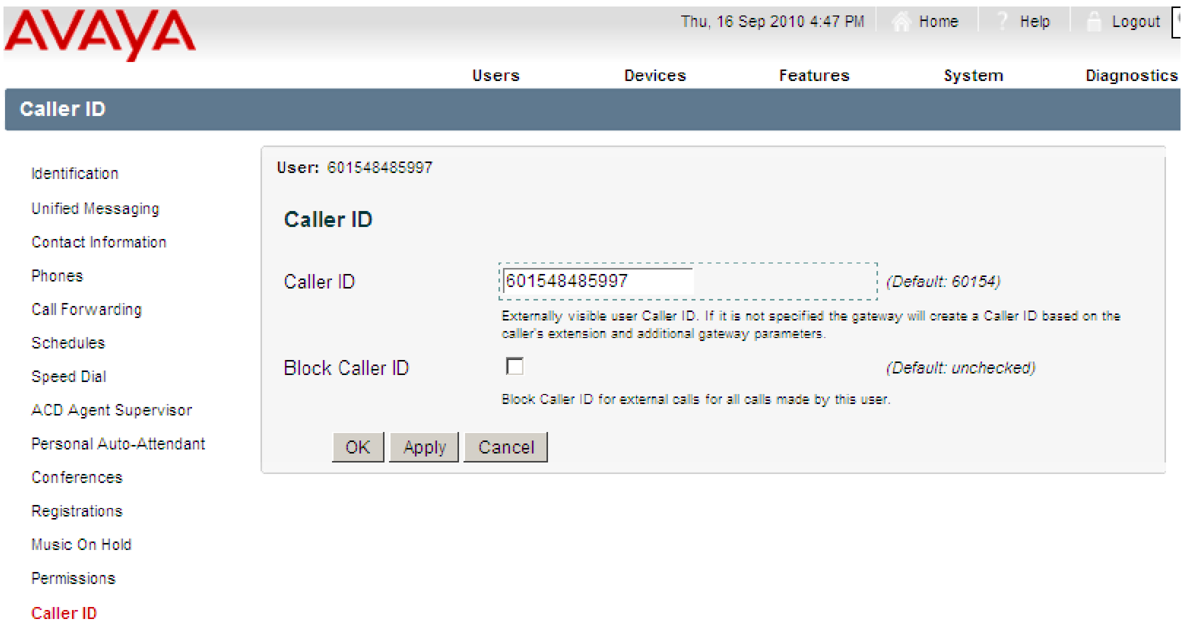

Figure 8 – Caller ID of SCS user

Go to **Phones** menu on the left column, which is used to assign Telephone set type to the target user and the User's Phones detail page will appear as shown in **Figure 9**. Click on the **Add Existing Phones** link, the user will be presented a list with available telephone type existing on the SCS system. Then select the telephone to assign to the target user.

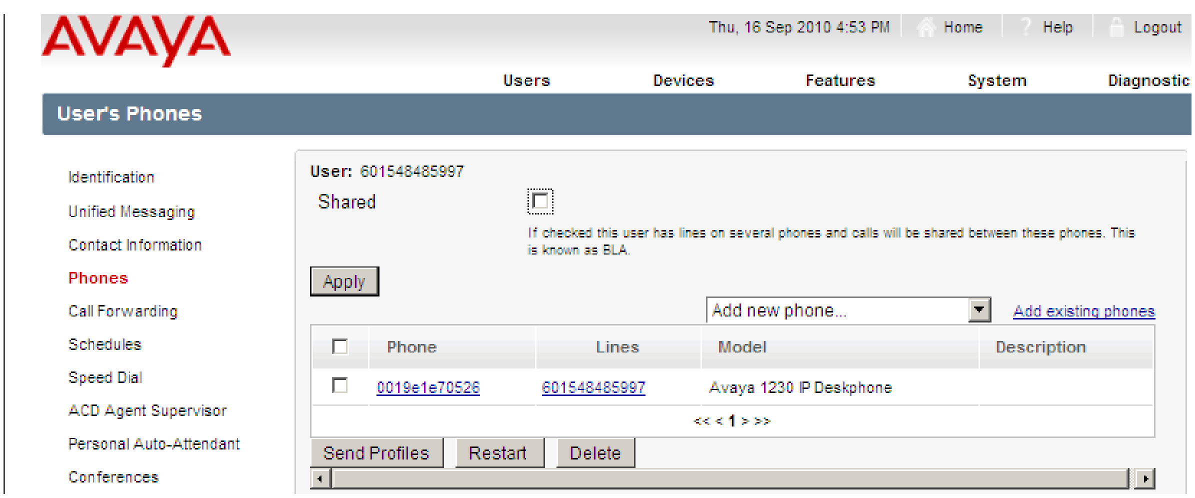

Figure 9 – Assign phone to SCS user

Check the check box of the assigned phone set and click on the **Send Profiles** button for updating the information of the user on the SCS system.

4.5. Configure SIP Trunk Gateway

This section describes how to configure the SIP Trunk Gateway on the SCS server.

- Access the web page of the SCS server.
- Click on **Devices → Gateways**, the Gateways page will appear (not shown).
- Click on the pull down menu **Add New Gateway**, choose from the list of gateway type, SIP trunk to add a gateway, the Gateway Details page will appear as shown in **Figure 10**.

Click on the **Enabled** checkbox and fill in the **Name**. At the **SBC Route** attribute, click on the pull down menu to choose the target route; sipXbridge-1. For IP Peering mode testing, enter the static IP address of ITSP's SBC at the **Address** attribute textbox as shown in **Figure 10**. Other fields are left at default values. Click on the **Apply** button to save the configuration changes.

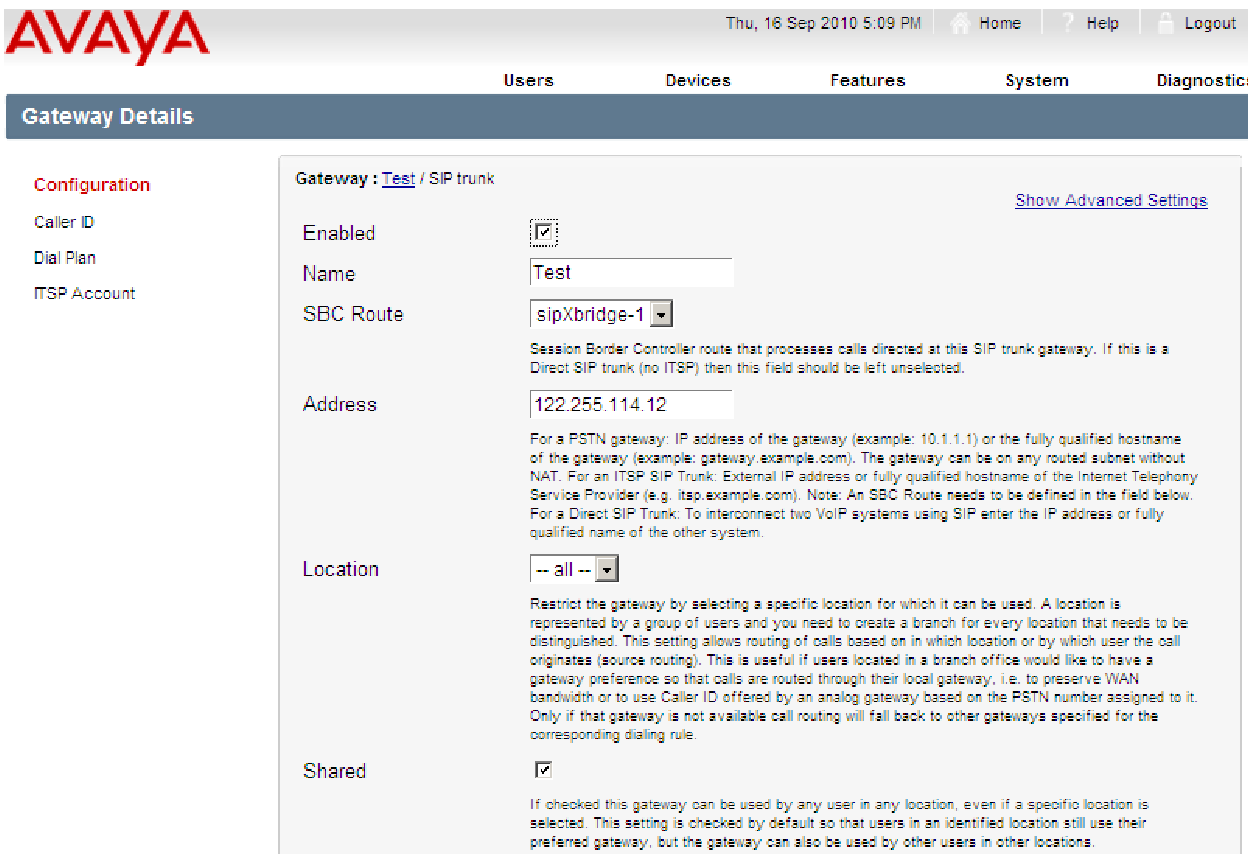

Figure 10 – SIP Trunk Gateway settings

For SIP Registration mode (dynamic registration), do as specified above steps for creating the SIP Trunk Gateway. Then go to **ITSP Account** menu on the left column, filling in the **Username**, **Authentication Username**, **Password** and **ITSP server address**. Others are left at default. Click on **Apply** button to save the configuration information as shown in **Figure 11**. Notes: Authentication Username and Password are provided by the ITSP for SCS to register to the ITSP's SBC.

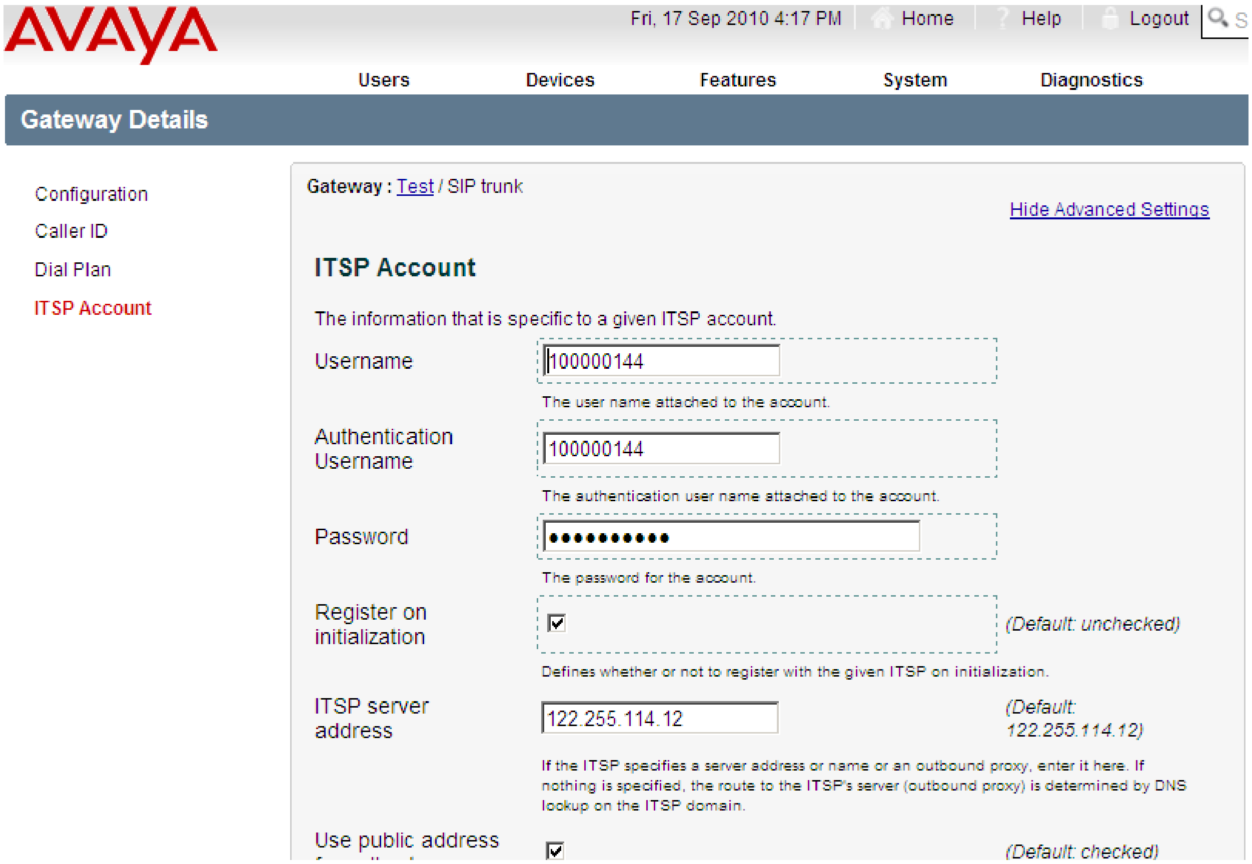

Figure 11 – SIP Registration

Under the **System** menu, select **Internet Calling**. Navigate to **NAT Traversal** menu on the left column. Disable the NAT Traversal by un-checking the **Enable NAT Traversal** check box. And un-check the **Server behind NAT** checkbox to disable this NAT Traversal configuration. See **Figure 12**.

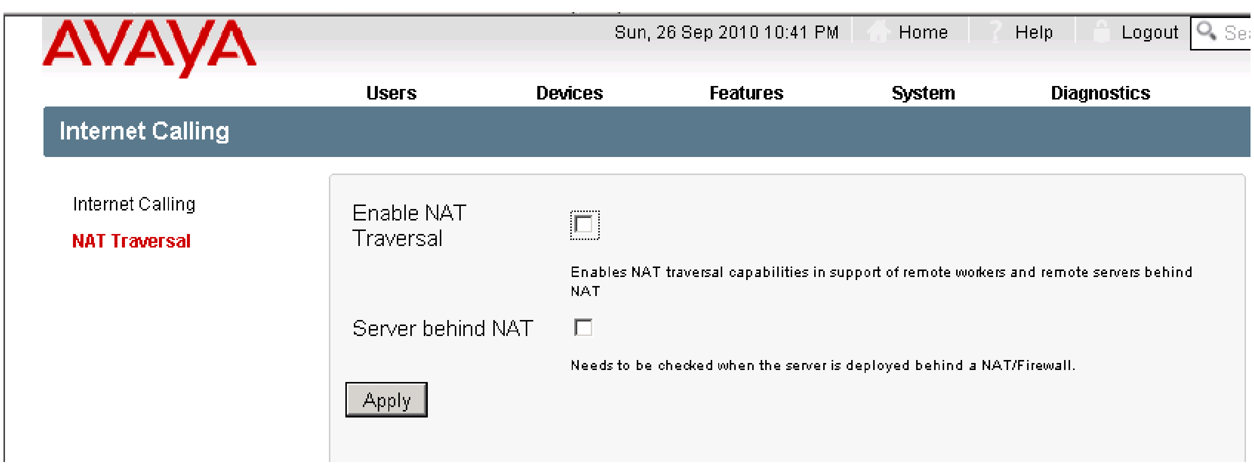

Figure 12: Disable NAT Traversal

4.6. Configure codec

This section describes how to configure the codec used by the SIPX Bridge and the SCS users when making the outbound calls.

- Access webpage of SCS server.
- Click on **Devices** → **SBC Routes**.
- Select sipXbridge-1, the **Edit SBC** page will appear.

Click on the **SIP** menu item on the left column. Then click on the **Show Advanced Settings** link for more details of the SIPX configuration as shown in **Figure 13**. Fill in the **Public port** the value of **5060** as shown. Empty the **Permitted Codecs** field to allow all codec to be used by the SIPX Bridge. Others are left at default values. Click on the **Apply** button to save the changes.

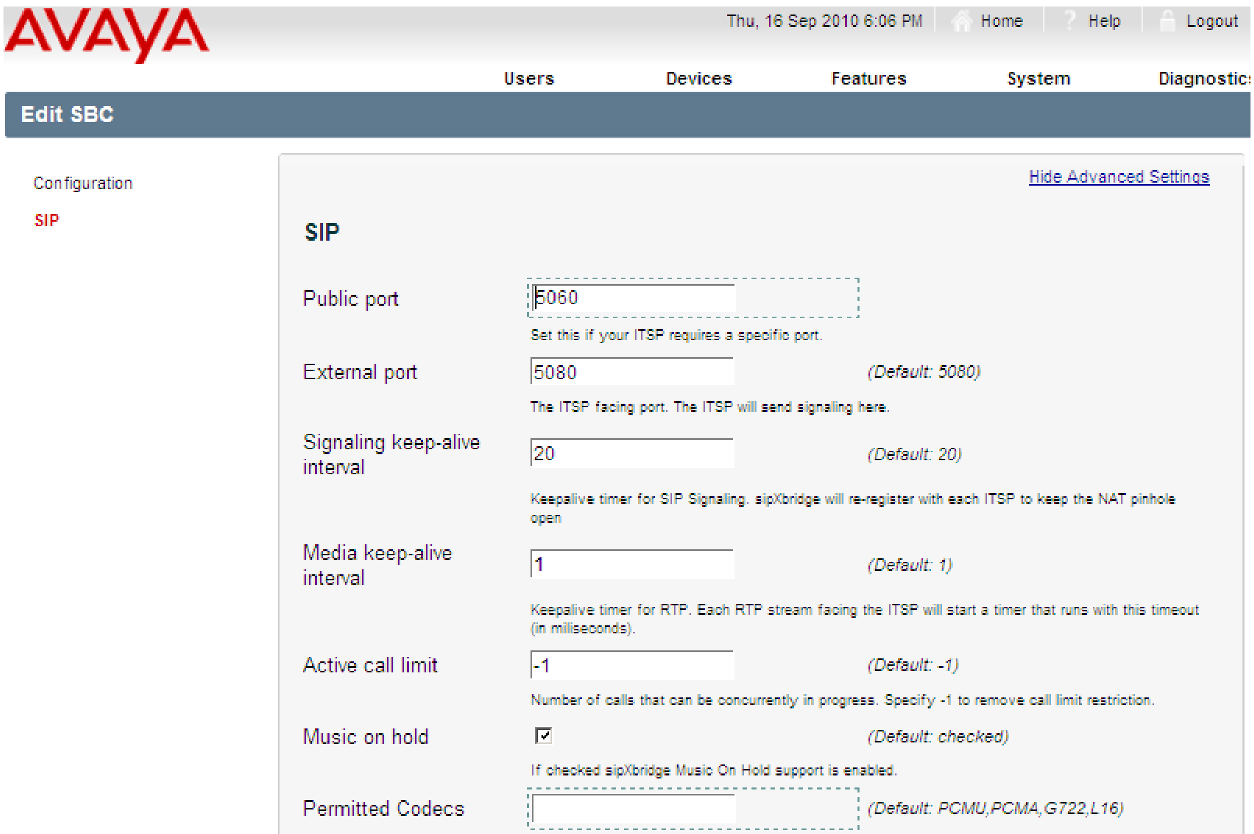

Figure 13 – SIPX configuration

To configure codec for the SCS users, go to the menu item **Devices → Phones**, click on the MAC address of the IP set which has been assigned the targeted Line. On the left menu column, click on the **Codec Preference** and select codec to be used as shown in **Figure 14**.
Click on the **Apply** button to save the changes. Click **Ok** button to get back to the Phones page. From here, click on the **Send Profiles** to update information on the SCS system (not shown).

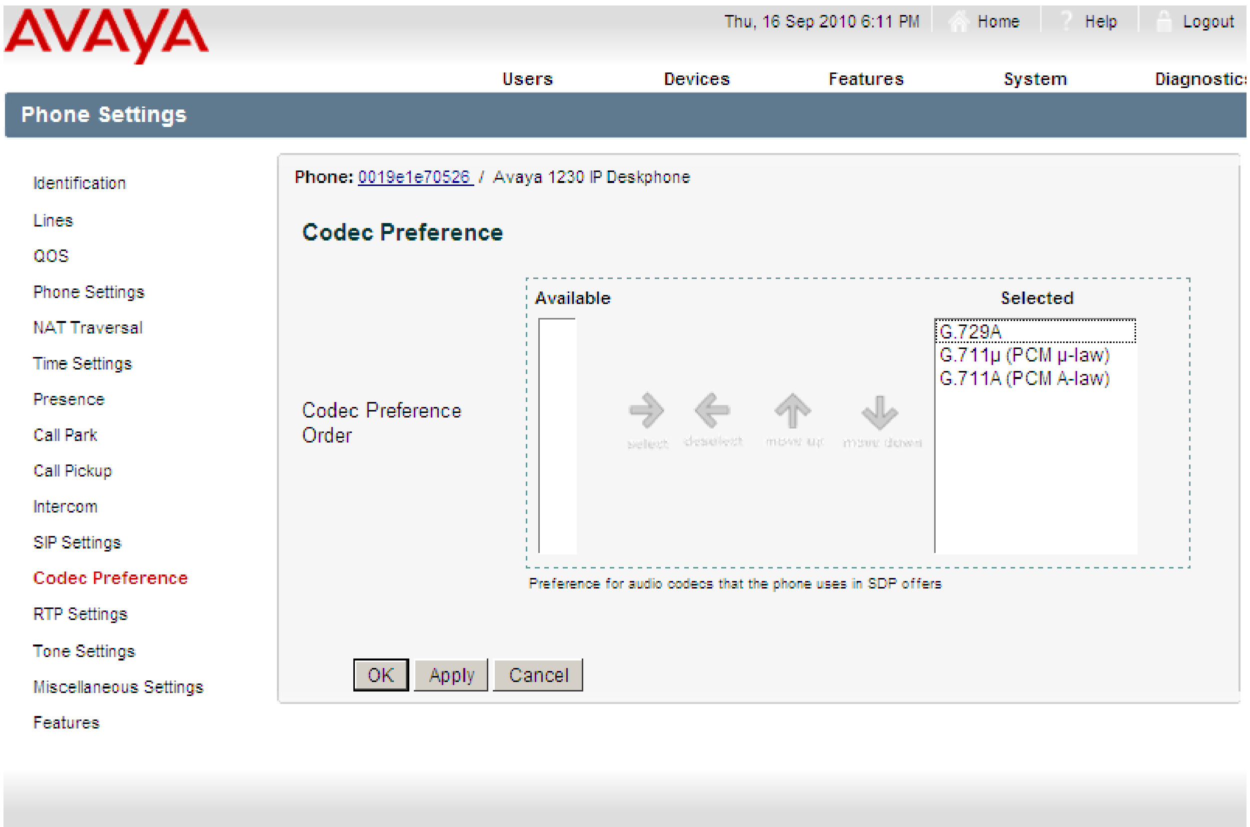

Figure 14 – Phone codec setting

4.7. Configure dial plans

This section describes how to configure a dialing plan on the SCS server.

- Access webpage of SCS server.
- Click on **System → Dial Plans**.
- From the pull down menu list of **Add New Rule**, choose the **Custom** rule as shown in **Figure 15**.

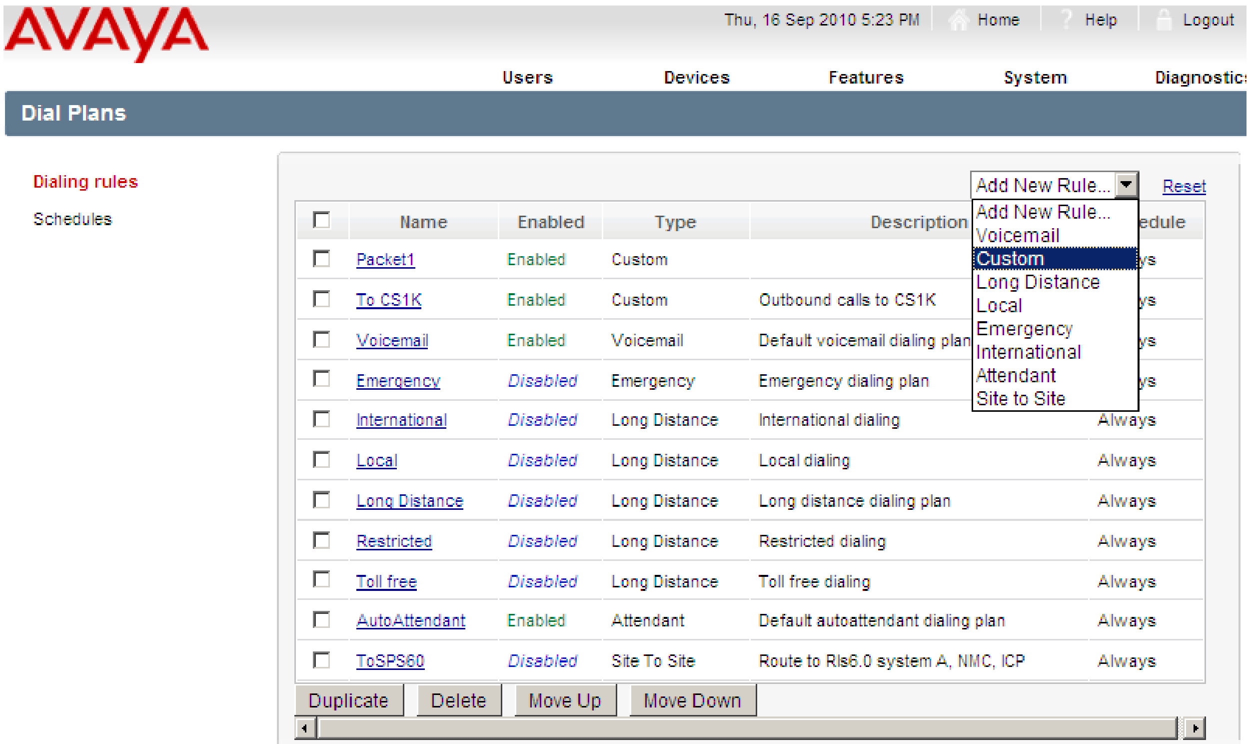

Figure 15 – Add a new rule for outbound calls

A detail custom dialing plan rule will appear as shown in **Figure 16**. Check on the **Enabled** check box. Fill in the name for the rule. Fill in the dialed numbers for making outbound calls to Public Service Telephone Number (PSTN) through the ITSP. The **Required Permissions** section is left blank in this case. Under the Resulting Call pull down menu list, choose **Entire Dialed Number** to forward all digits to the ITSP. Under the **Gateways** pull down menu list, choose the defined gateway as shown in **Figure 16**.

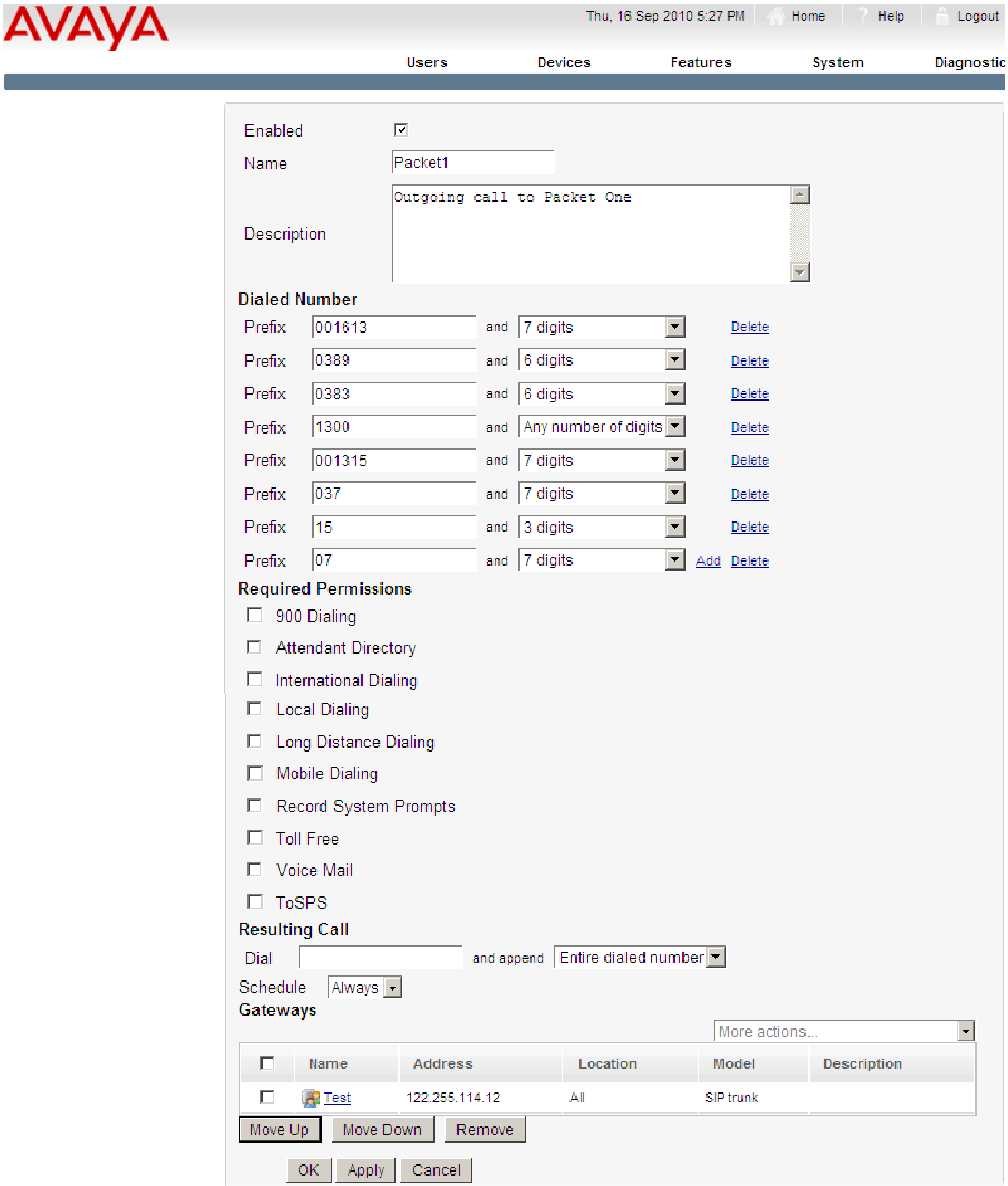

Figure 16 – Outbound calls rule

4.8. Configure Voice Mail dial plan on the SCS server

This section describes how to configure the voice mail dialing plans on the SCS server.

- Access webpage of SCS server.
- Click on **System → Dial Plans**

Select **Voicemail** under the **Add New Rule** pull down menu list. To configure Voicemail extension, see the details as shown in **Figure 17**. Click **Apply** button and then click on **Ok** button to save and exit the page respectively.

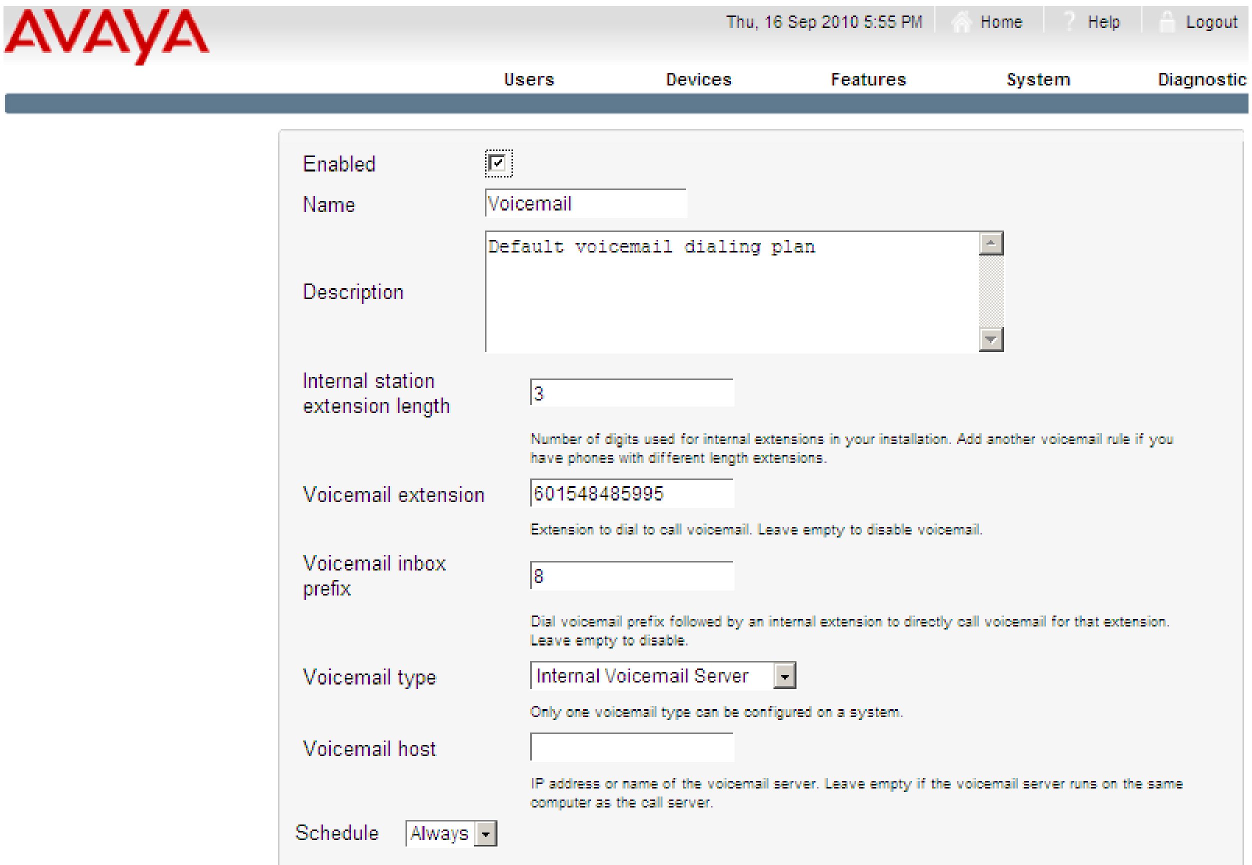

Figure 17 – Voicemail dial plan

To configure the permission for the target user who is associated with the newly created voicemail extension, navigate to **Users** menu and choose **Users**. A page of User list will appear (not shown). Click on the target user, a default Identification page will appear (not shown). On the left menu column, click on **Permissions** to go to Permission configuration details page. Make sure the **Voice Mail** check box is checked. Others are at default values.

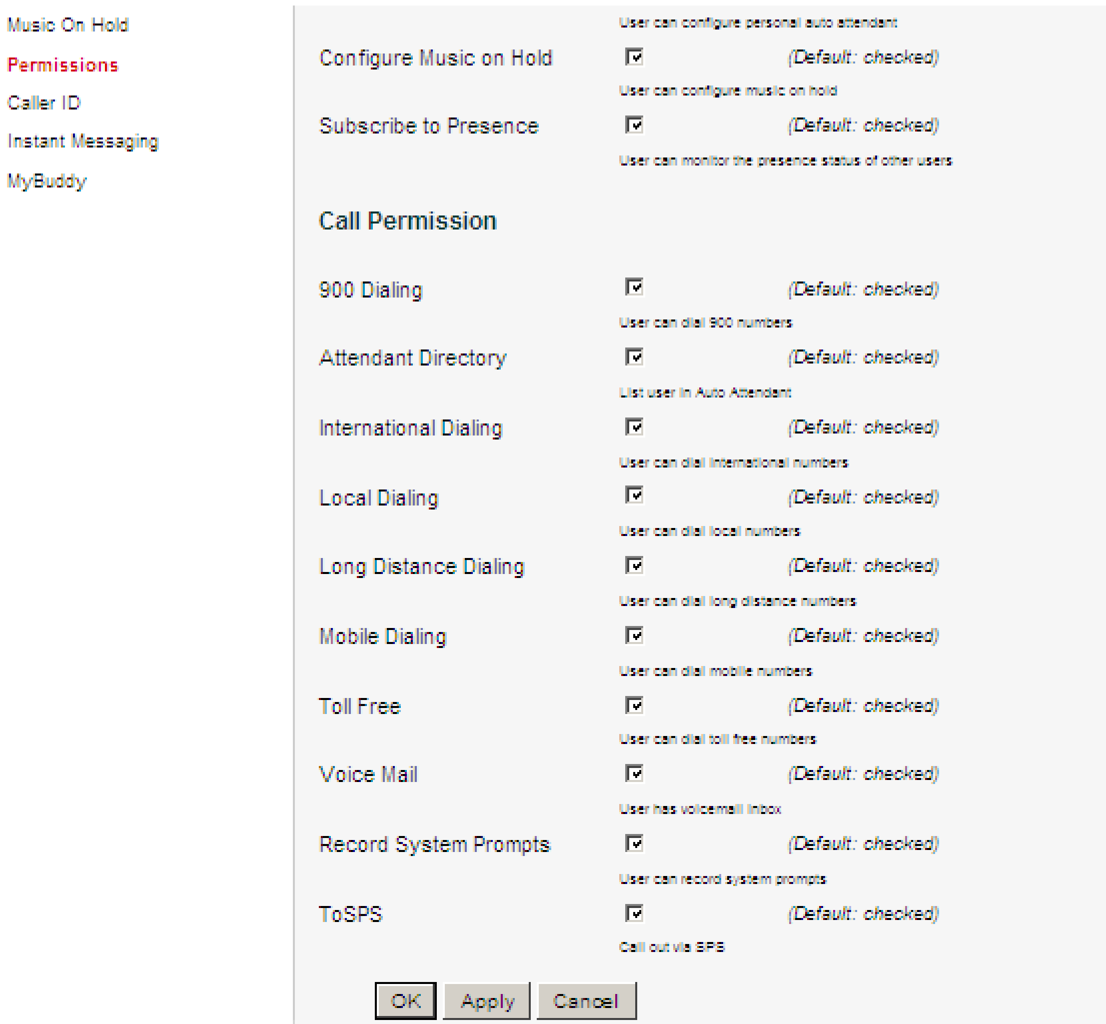

Figure 18 – Voice Mail permission for SCS user

4.9. Configure conference on SCS server

This section describes how to configure the conference on the SCS server.

- Access webpage of SCS server.
- Click on **Features → Conferencing**. The Conference Server page will appear (not shown).
- Click on the **Name** of the target server, the default Configuration page will appear (not shown).

On the left column menu, click on Conference menu item, the Conference Server page will appear (not shown). On the right upper corner of the Conference Server page, click on the **Add New Conference** link, the Conference detail page will appear as shown in **Figure 19**. Check the **Enabled** checkbox and fill in the conference **Name** and **Extension** of the associated owner of the conference. Click on the **Assign Owner** button to assign the Conference Owner. Other are left at defaults. Click on **Apply** button to save the change and **Ok** button to exit to the Conference Server page respectively.

Note: To assign a Conference owner, click on the **Assign Owner** button, an Add User to Conference page will appear (not shown). Fill in the user ID in the user box and click on the Search button to find the target user. Check on the target user checkbox and click Select button to assign that user to own the newly created conference.

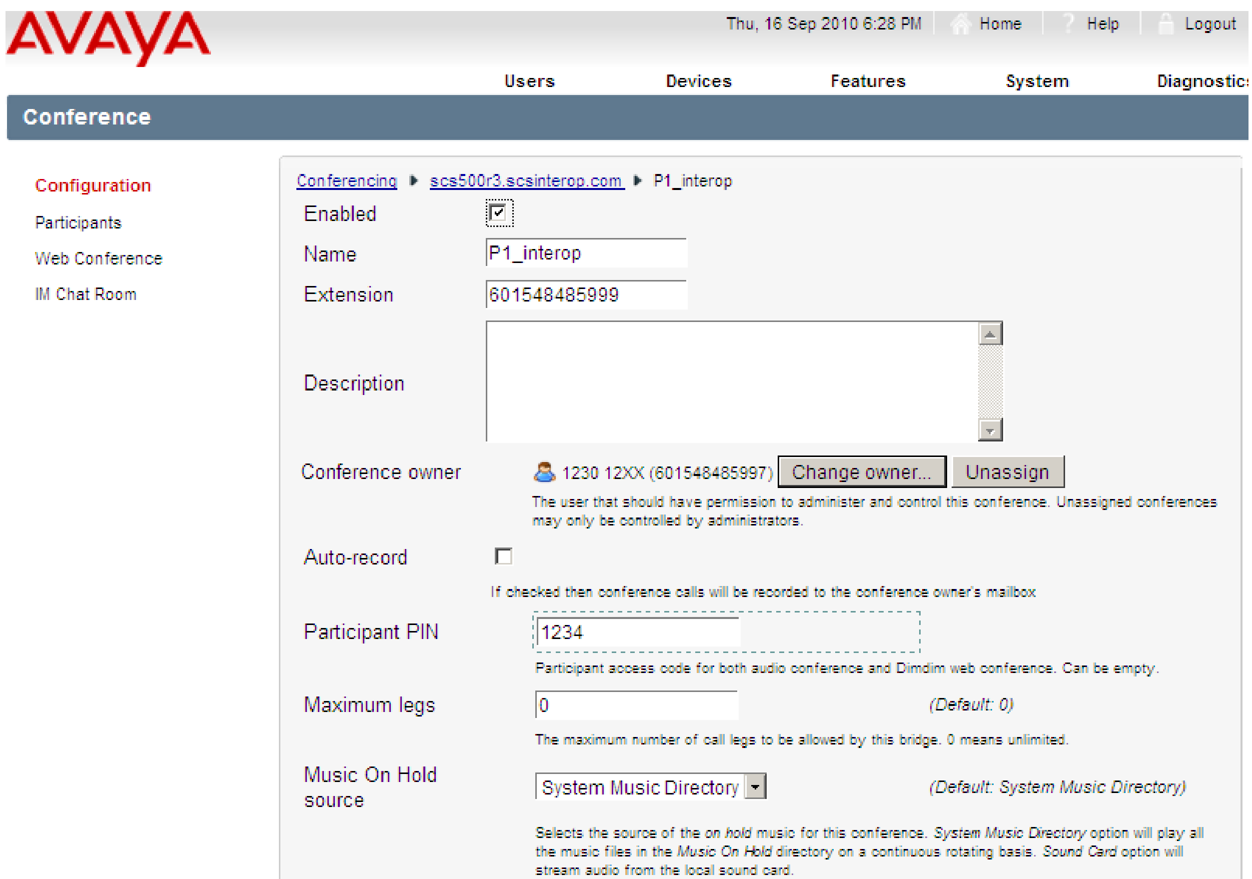

Figure 19 – Conference configuration

5. Packet One System configuration

Pack One is responsible for the set up and configuration details on the Packet One SIP Trunk System Version 3.1.

6. General Test Approach and Test Results

The focus of this interoperability compliant testing was to verify the SIP trunk connectivity between the Packet One SIP Trunk System Version 3.1 and Avaya Software Communication System Release 4.0. The compliance testing was performed on 2 methods of connectivity between Packet One system and Avaya SCS; IP peer (using static IP address) and SIP registration (using SIP account and authentication password on the ITSP).

6.1. General Test Approach

The general test approach was to have Packet One system connected to Avaya Software Communication System via SIP trunk using 2 methods of communications; Gateway using IP address (IP peer mode) and SIP registration (SIP account and password authentication). The SIP trunk communication should be established between Avaya SCS and Packet One system. Calls can be made from end to end, i.e. PSTN phone can call through created route from Packet One system to SCS SIP phones via SIP trunk. The main objectives were to verify the SIP trunk features:

- Basic call from PSTN phone to SCS SIP phones.
- Perform basic call operation: DTMF transmission, voicemail with MWI notification, hold/ un-hold.
- Redirect call between users/clients/endpoints: blind/consultative transfers, call forward no answer.
- Perform codec negotiation.
- Perform conferencing.
- PSTN numbering plans.

6.2. Test Results

The objectives outlined in **Section 6.1** were verified and met. The following observations were made during the compliance testing:

- Call waiting is not applicable to the Polycom set. Thus, call waiting test case were performed only on 12xx sets.
- Cannot make the outbound call to the PSTN number with Calling Number Restriction. Configure the Blocked Caller ID on the SCS user and make an outgoing call to the PSTN number. The call is declined at the Packet One system. The issue was fixed by Packet One team.
- Fail to blind transfer to an external number, no voice path. After receiving an incoming call from the PSTN_1 to the SCS user, the SCS user performs blind transfer to the PSTN_2. The call is transferred properly but there is no voice path between the PSTN_1 and the PSTN_2. This issue on the Packet One system and being investigated by the Packet One team.

- Fail to call forward no answer to an external user, no voice path. The call scenarios us as follow: The PSTN_1 calls the SCS user, the SCS user does not answer the call. The call will be forwarded to the PSTN_2. PSTN_2 answers the call but there is no voice path between PSTN_1 and PSTN_2. This issue is on the Packet One system and being investigated by Packet One team.
- Can not establish the conference with 3-way speech path to a PSTN number. The issue was fixed by the Packet One team.

7. Verification Steps

This section includes steps to verify the configuration by:

- Verifying that calls are established with two-way voice path when making a call from one SCS user to another local SCS user.
- Verify that calls are established with two-way voice path when making calls from PSTN phones to the SIP phones on the SCS server through the Packet One system via configured SIP trunk.
- Check the SIP messages and the RTPs which are sent back and forth between SCS user, the Avaya SCS server and the Packet One system.

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 6.1**, with some exceptions outlined in **Section 6.2**. The outstanding issues are on the Packet One system and being investigated by Packet One team. Some of these issues are considered as exceptions. The Packet One SIP Trunk System Version 3.1 is considered compliant with Avaya Software Communication System Release 4.0.

9. Additional References

Product documentation for Avaya products may be found at:
http://support.avaya.com/css/Products/P0634/Administration & System Programming

[1] *Configuring_User_Profiles_Task_Based_Guide, Release 3.0, Document Number NN40010-507*
[2] *End_User_Task_Based_Guide, Release 3.0, Document Number NN40010-520*

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.