Avaya Solution & Interoperability Test Lab

Avaya one-X Quick Edition Interoperability with Cisco Integrated Services Router (ISR) SIP Gateway - Issue 1.0

Abstract

Avaya one-X Quick Edition R3.1 provides the ability to use a SIP Gateway to provide access to the PSTN. These Application Notes detail the process to use the Cisco Integrated Services Router (ISR) as the SIP Gateway used by the Avaya one-X Quick Edition. Analog telephones are administered as stations. ISDN PRI and Analog tip and ring lines are administered to provide PSTN access.

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1. Introduction

Avaya one-X Quick Edition release 3.1, introduces a native SIP Registrar service. Avaya Quick Edition users can now register 3rd party SIP gateway services directly with Avaya Quick Edition. This provides Avaya Quick Edition users the ability to add new service capabilities to Avaya Quick Edition which compliments the SIP gateway services natively available through Avaya Quick Edition's line of SIP gateway products (G11 FXO Analog gateway; G20 ISDN/BRI Gateway; and the A10 FXS Analog Terminal Adapter).

These Application Notes specifically addresses the capability to leverage Avaya Quick Edition's native SIP Registrar services to attach (register) analog FXS and FXO port services provided through a Cisco Integrated Services Router (ISR) equipped with FXS, FXO, and ISDN PRI ports. The configuration of both the Avaya Quick Edition and Cisco ISR are covered in the sample configuration.

1.1. Network Topology

Figure 1 provides an overview of the network architecture used in the sample configuration of these Application Notes.

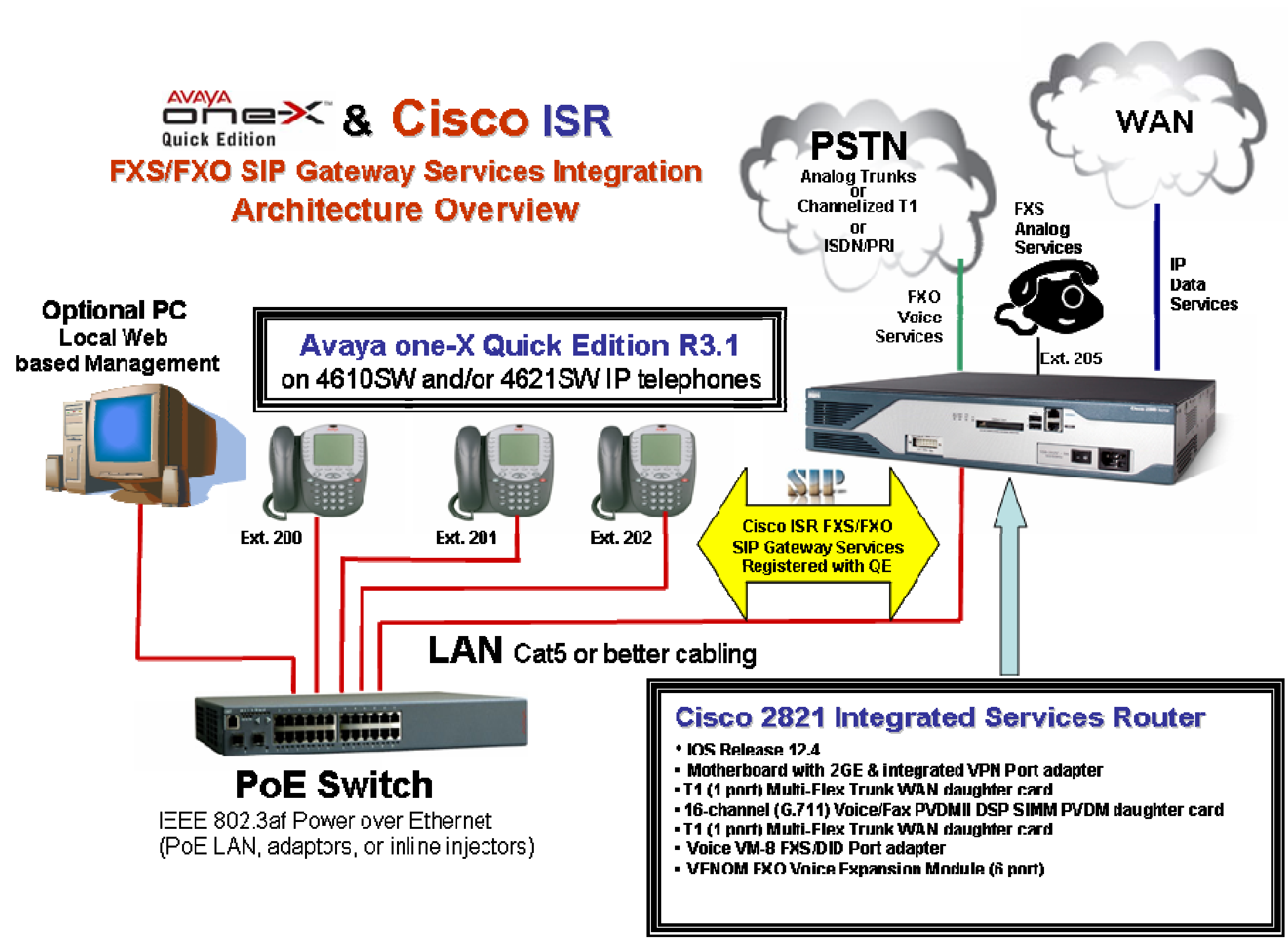


Figure 1: Avaya one-X Quick Edition solution with Cisco ISR

1.2. Services

These Application Notes cover three different categories of SIP Gateway services facilitated via the Cisco ISR:

- **FXS Services:** Analog devices, such as analog telephones, attached to the Cisco ISR. These analog devices are configured as extension numbers like the Avaya one-X Quick Edition IP Telephones and are able to make and receive calls. The ports where the analog devices (telephones) plug into the Cisco ISR are called FXS (Foreign eXchange Subscriber) ports. Details on how to configure Avaya one-X Quick Edition for these services are provided in Section 4.2 of this document.
- **FXO Analog Services:** Analog loop start telephone lines attached to the Cisco ISR. These analog lines from the local Public Switched Telephone Network (PSTN) provide the ability to make and receive calls. The ports where the telephone lines are plugged into the Cisco ISR are called FXO (Foreign eXchange Office) ports. Details on how to configure Avaya one-X Quick Edition for these services are provided in Section 4.3 of this document.
- **FXO ISDN PRI Services:** These Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) lines provide multiple channels (B channels) in digital format that perform the same function as analog telephone lines with digital quality. The PRI also includes one channel for signaling (D channel). Details on how to configure Avaya one-X Quick Edition for these services are provided in Section 4.4 of this document.

2. Equipment and Software Validated

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

Equipment	Software
Avaya one-X Quick Edition	R3.1
Cisco 2821 Integrated Services Router	IOS 12.4

Appendix A contains detailed configuration information for the Cisco ISR.

3. Configuring DHCP and DNS

The Avaya one-X Quick Edition capability to self-configure without a Dynamic Host Configuration Protocol (DHCP) server or a Domain Name System (DNS) server can not be used with this configuration.

IP addresses must be assigned to the Avaya one-X Quick Edition Telephones either statically or with DHCP. The Cisco ISR can be used as a DHCP server as described in Section 3.1.

An external DNS server is required and must be included in the Cisco ISR configuration. The Cisco ISR configuration requires the DNS name **qe.avaya** be resolved by the DNS server into the IP address of one of the Avaya one-X Quick Edition IP Telephones.

3.1. Configuring DHCP on Cisco ISR

This is an example of configuring DHCP on the Cisco ISR used in this configuration. DHCP must be used for the Avaya one-X Quick Edition to use the Cisco ISR as a SIP Gateway.

```
ip dhcp pool Quick
   network 20.1.100.0 255.255.255.0
   default-router 20.1.100.21
   dns-server 192.45.130.100
```

3.2. Configuring qe.avaya on DNS

The DNS Server (dns-server in the DHCP configuration of Section 3.1) should provide the IP address of one of the Avaya one-X Quick Edition IP Telephones as the address for qe.avaya.

4. Configuring Avaya one-X Quick Edition

Configure the Avaya one-X Quick Edition system following the instructions on support.avaya.com [1]. Test the system by placing calls between the Avaya one-X Quick Edition IP Telephones.

Attach a PC to the PC port of one of the Avaya one-X Quick Edition IP Telephones and ping **qe.avaya**. This will demonstrate that the DHCP and DNS configuration is correct and provide the IP address needed to open the Avaya one-X Quick Edition Web Administration application.

The following instructions are based on the address qe.avaya being resolved as 20.1.100.76.

Section 4.1 provides some basic data gathering that will be helpful in the configuration.
Section 4.2 provides instructions for administering telephone (FXS) capabilities.
Section 4.3 provides instructions for administering telephone line (FXO) capabilities.
Section 4.4 provides instructions for administering ISDN PRI capabilities.
Section 5 provides the parallel instructions for administering the Cisco ISR.

4.1. Collect Initial Data on Avaya one-X Quick Edition Configuration

Open a web browser to https://20.1.100.76.

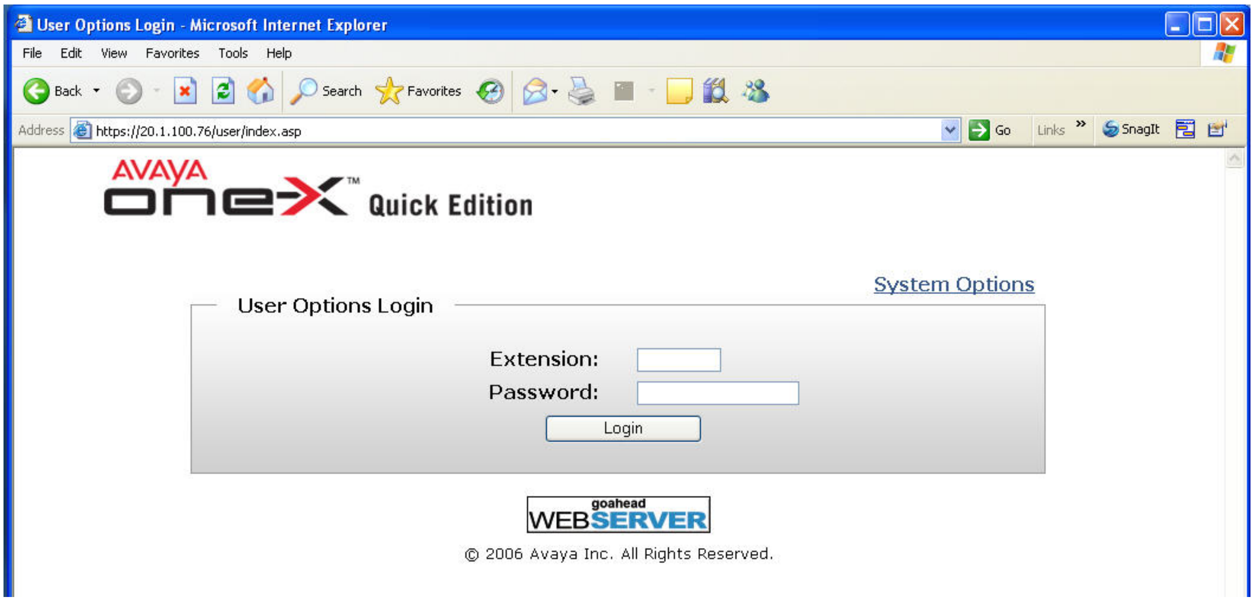


Figure 2: Avaya one-X Quick Edition Login Screen

Select **System Options**. When the screen refreshes enter the System Password and Click on **Login.**

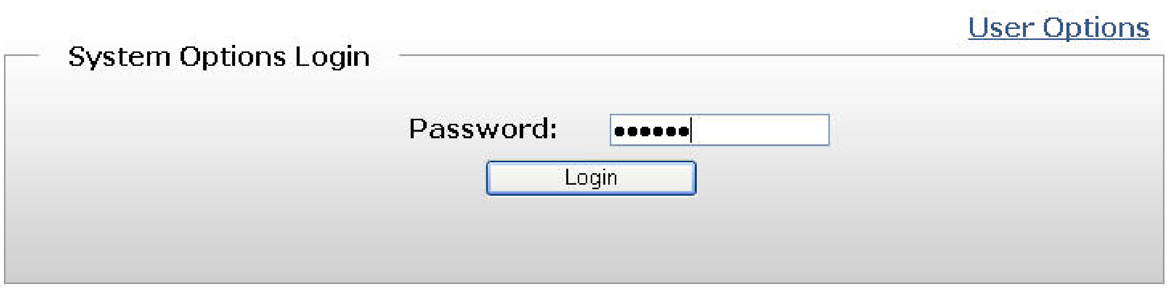
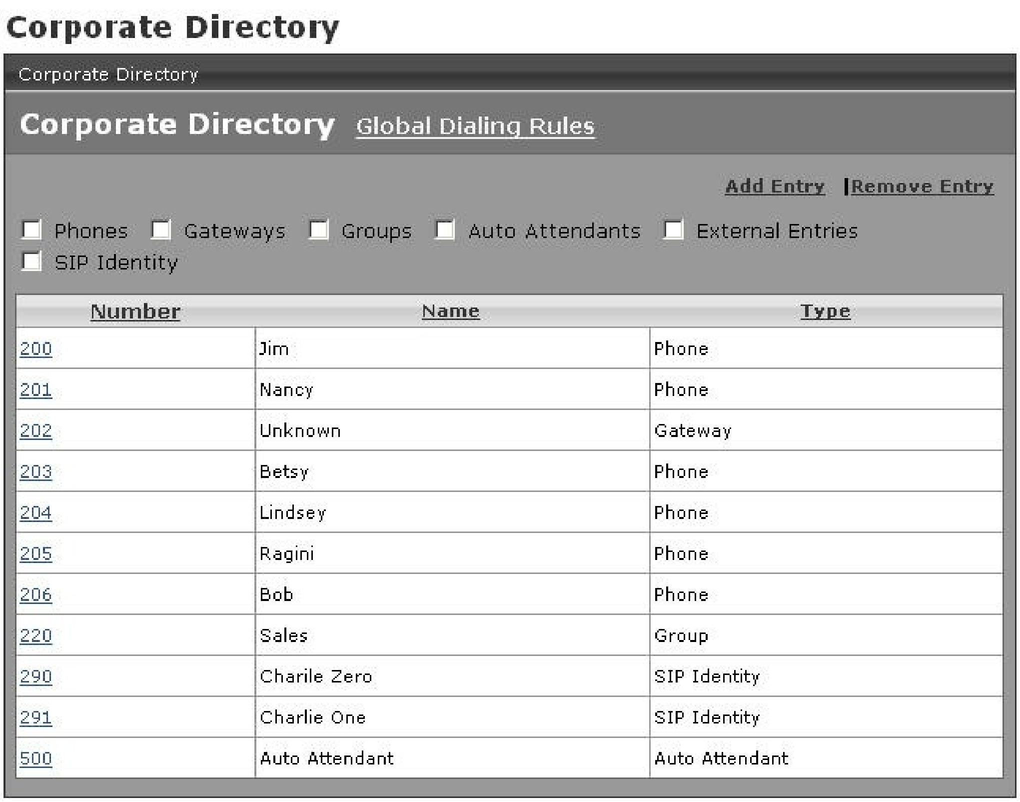


Figure 3: Avaya one-X Quick Edition System Options Login

Gather the information from the local Avaya one-X Quick Edition Corporate Directory and Dialing Configuration. From the System Options menu on the left of the web page select **Corporate Directory**.

Figure 4: Avaya one-X Quick Edition System Options Menu



Number	Name	Type
200	Jim	Phone
201	Nancy	Phone
202	Unknown	Gateway
203	Betsy	Phone
204	Lindsey	Phone
205	Ragini	Phone
206	Bob	Phone
220	Sales	Group
290	Charile Zero	SIP Identity
291	Charlie One	SIP Identity
500	Auto Attendant	Auto Attendant

Figure 5: Avaya one-X Quick Edition Corporate Directory

From the System Options menu on the left of the web page select **Dialing Configuration**.

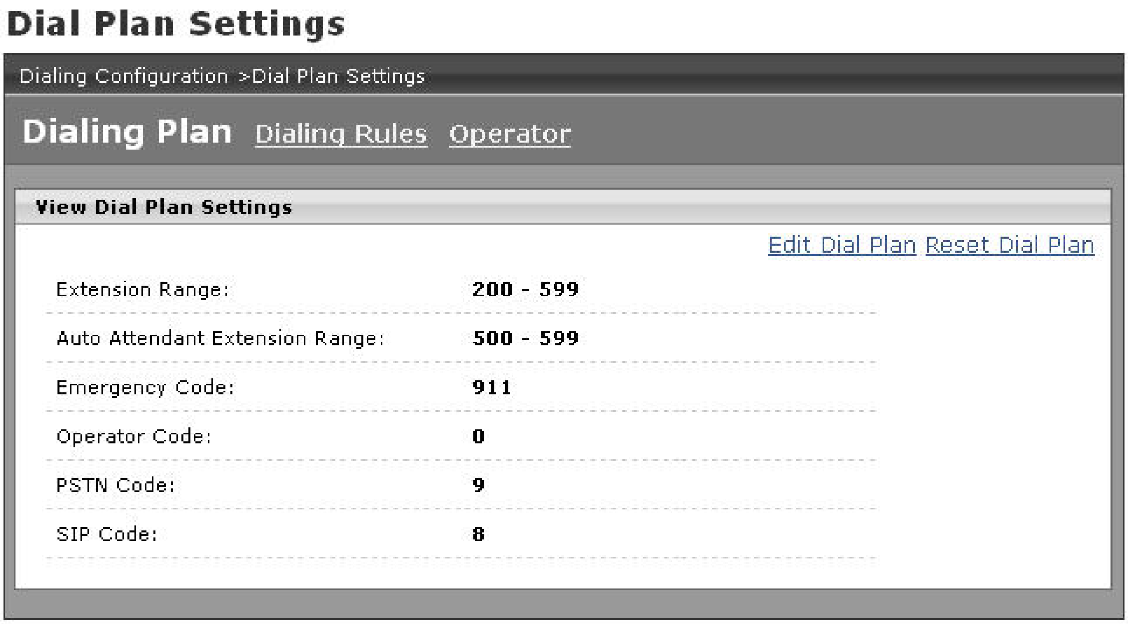


Figure 6: Avaya one-X Quick Edition Dial Plan

Avaya one-X Quick Edition provides extensive flexibility in administering the extensions and the dial plan. These application notes are written using the number assignments in Figure 5 and the Dial Plan in Figure 6. While following these application notes it will be useful to have a print-out of the local Avaya one-X Quick Edition Corporate Directory and Dial Plan.

4.2. Configuring Cisco ISR FXO Analog Stations on Avaya one-X Quick Edition

The Avaya one-X Quick Edition acts as a SIP Proxy/Registrar and the Cisco ISR needs to register a SIP Identity for each telephone plugged into the Cisco ISR FXSFXO ports.

From the System Options menu on the left of the web page select **SIP Proxy**.

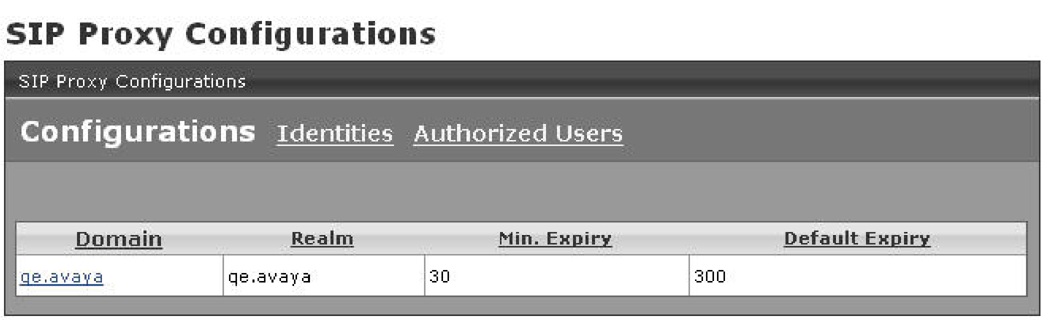


Figure 7: Avaya one-X Quick Edition SIP Proxy Menu

Select **Identities** from the menu on the right side of the screen.
Select **Add** from the menu on the far right side of the screen.

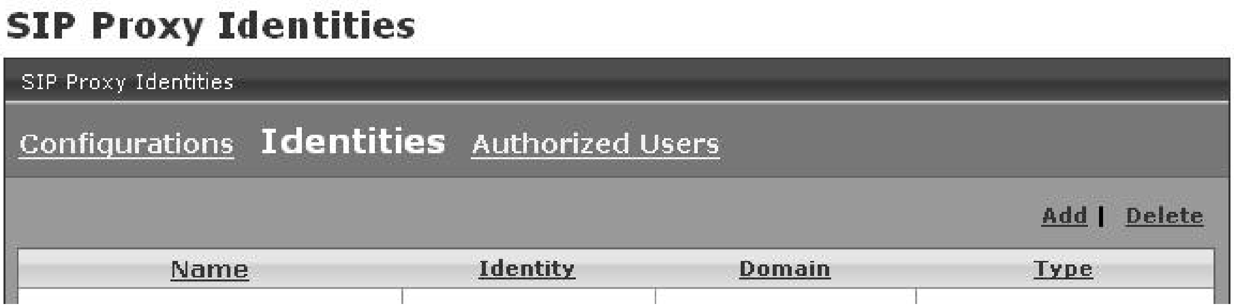


Figure 8: Avaya one-X Quick Edition Identities Screen

For each FXS port enabled on the Cisco ISR where a telephone will be plugged in add a Subscriber Identity. From the pull down menu select the Type **Subscriber**. Enter a *Name* in the Name field. Enter an extension from the dial plan (Figure 6) for this station in the Identity field. Click **Submit**.

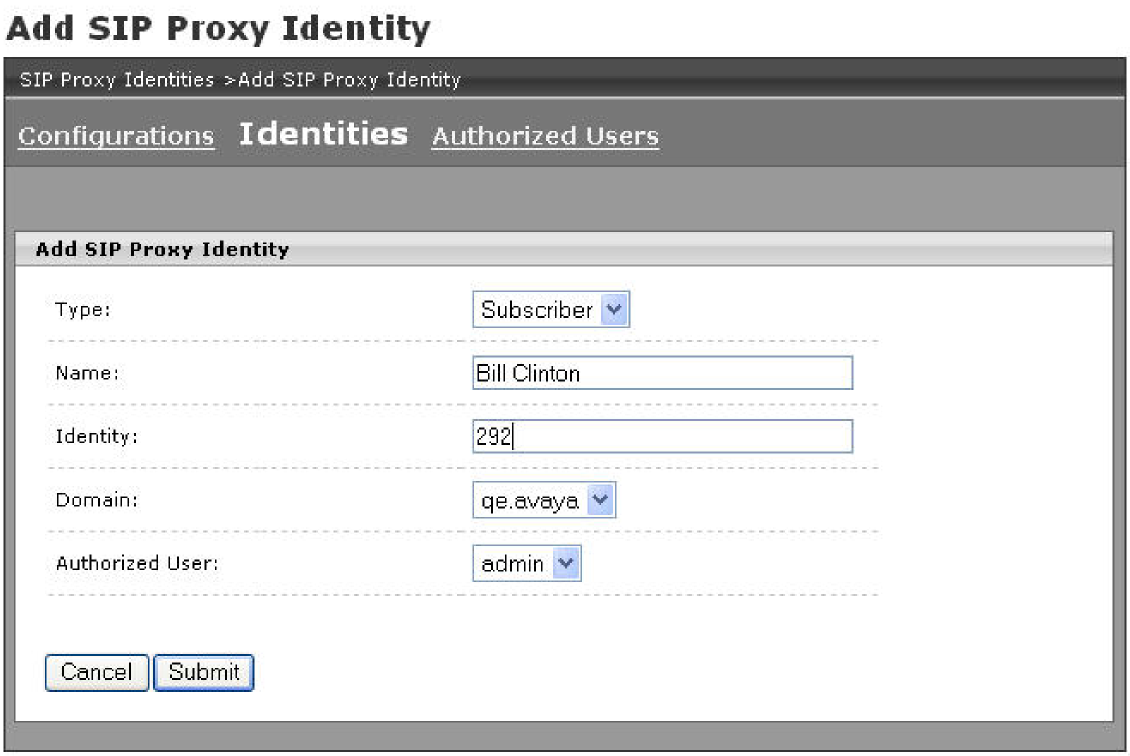


Figure 9: Add a Subscriber (FXS)

4.3. Configuring Cisco ISR FXO Telephone Lines on Avaya one-X Quick Edition

Typically the PSTN provider will provide several lines each identified by a particular telephone number. Each of these lines is plugged into an FXO port on the Cisco ISR. The Avaya one-X Quick Edition acts as a SIP Proxy/Registrar and the Cisco ISR needs to register a SIP Identity for only one of the FXO ports. The configuration of the Cisco ISR in Section 5 will provide a method of selecting which telephone line to use when a Avaya one-X Quick Edition station makes an outward (9+) call and a method of directing inward calls to a particular station. Using only one Proxy Identity minimizes the resources needed on the Avaya one-X Quick Edition IP Telephone receiving registration messages (qe.avaya).

4.3.1. Configuring the Primary Proxy Identity for FXO Trunks

From the System Options menu on the left of the web page select **SIP Proxy**.

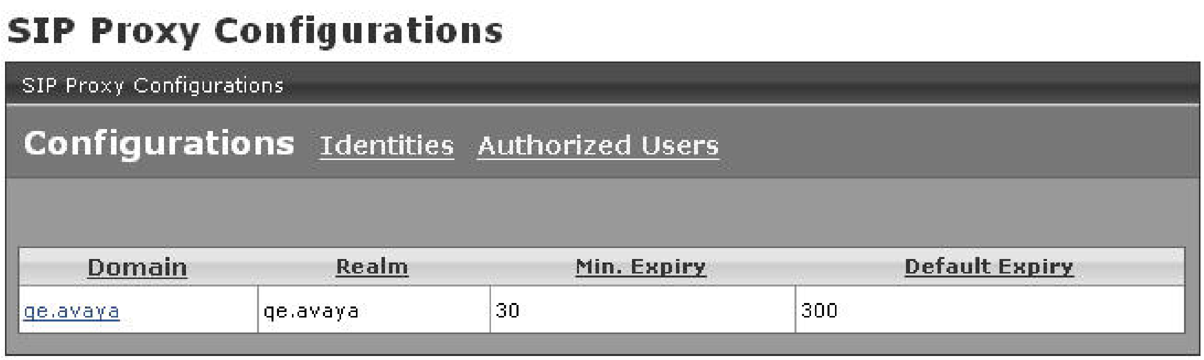


Figure 10: Avaya one-X Quick Edition SIP Proxy Menu

Select **Identities** from the menu on the right side of the screen.
Select **Add** from the menu on the far right side of the screen.

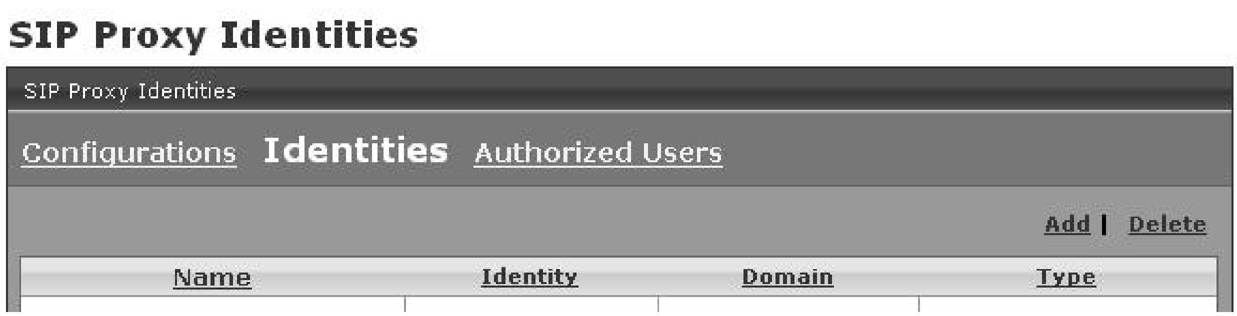


Figure 11: Avaya one-X Quick Edition Identities Screen

Select **Add** from the menu on the far right side of the Web Browser screen.

Select **Trunk** from the pull down menu for Type. Allow the screen to refresh. Enter a *Name* in the Name field. Enter a *number* in the Identity field. A useful value is to have the *number* be the E.164 number associated with a trunk although the *number* does not have to be an E.164 number. Select **Create as Registerer** from the Primary Registerer drop-down list. Select **Global** as the Outgoing Extension. Select **Auto Attendant** as the **Incoming Extension**. Select **Submit**. The choice of **Global** as the outgoing extension makes all of the telephone lines available to all of the Avaya one-X Quick Edition Telephones. The choice of **Incoming Extension** will control how Avaya one-X Quick Edition handles in coming calls. However, it will not have an effect on incoming call that have been directly mapped to an Avaya one-X Quick Edition extension on the FXO port of the Cisco ISR.

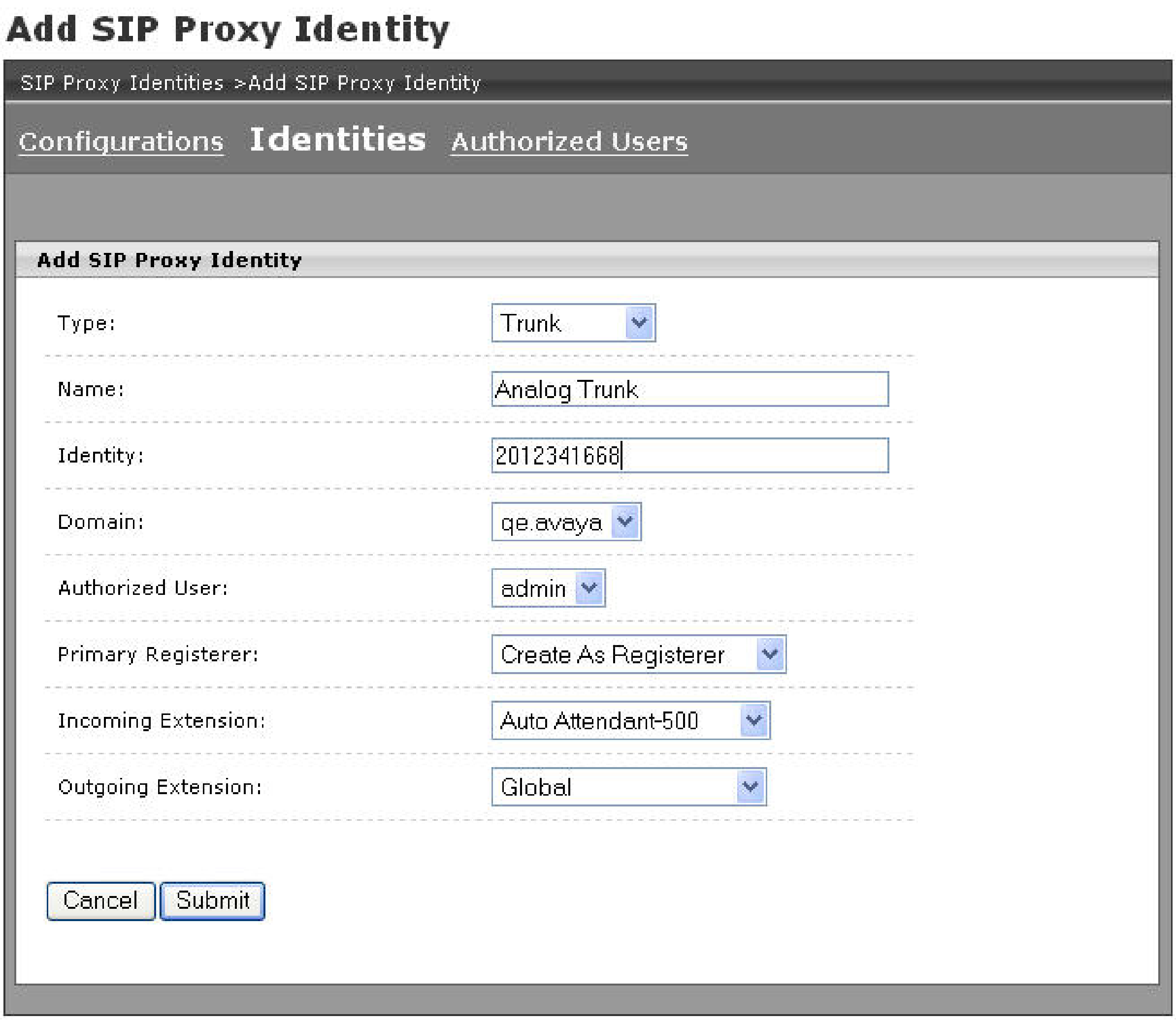


Figure 12: Create a Primary Registerer for FXO ports

4.3.2. Add Additional Trunks

Repeat the process in Section 4.3.1 for each FXO port. Instead of selecting **Create as Registerer** from the Primary Registerer pull down list, select the identity created in section 4.3.1.

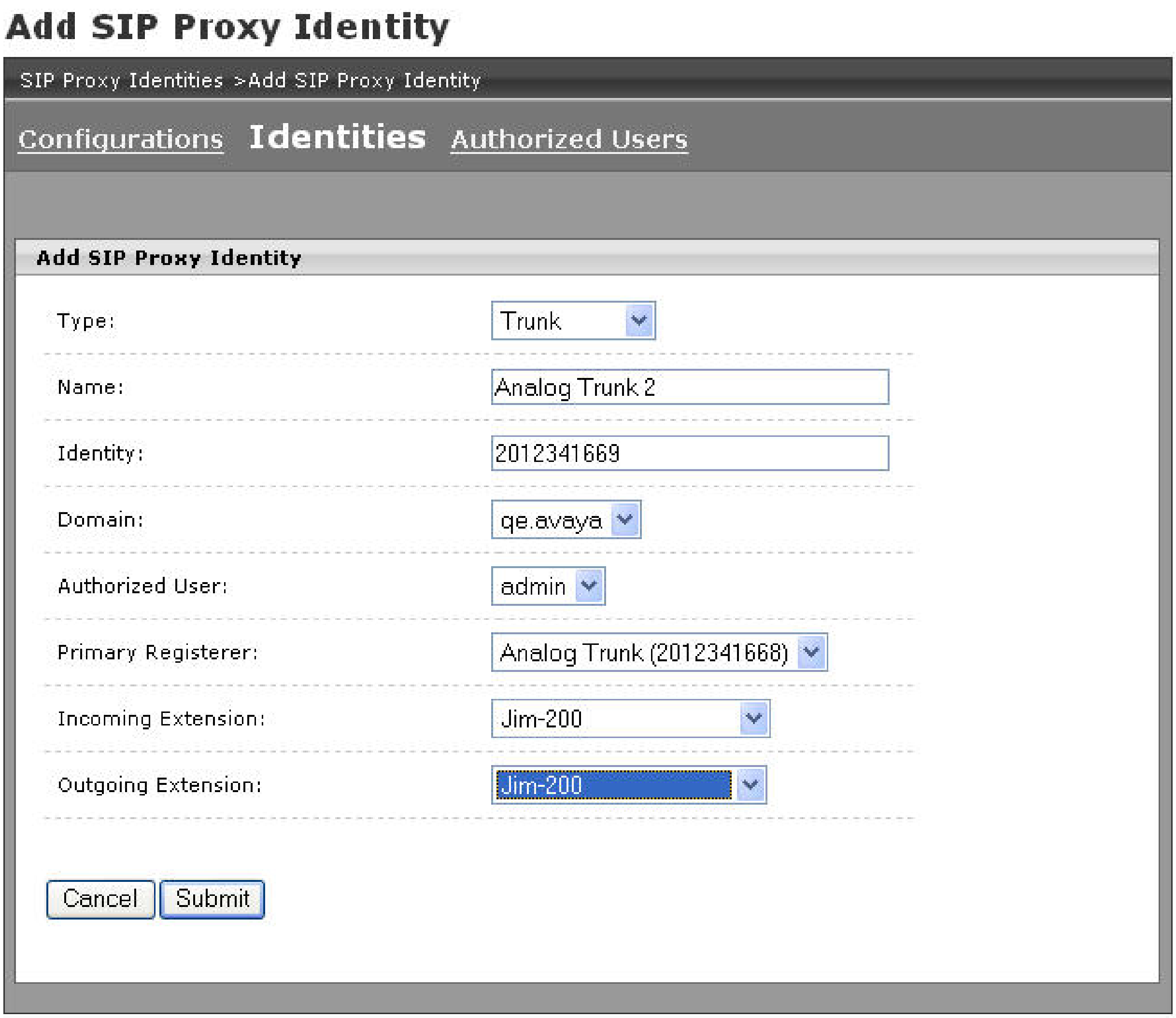


Figure 13: Create additional FXO ports

Selecting a particular station as the incoming extension will direct inward calls on that PSTN line to that station. Selecting a particular station as the outgoing extension will not effectively lock that PSTN line to that station for outgoing calls. See the dial-peer discussion in Section 5.3.

4.4. Configuring Trunks for ISDN PRI

4.4.1. Configuring Cisco ISR ISDN PRI Trunk Services on Avaya one-X Quick Edition

In the sample configuration of this Application Notes, the PSTN provider of the PRI has assigned the following telephone numbers:

2012342200
2012342201
2012342202
2012342203

The **Called Number Field** for inward ISDN PRI calls will contain one of these numbers. The numbers in the **Called Number Field** are a feature of the service from the ISDN PRI provider. The provider could provide only the extension with the leading digits stripped or they could provide a longer or shorter string as a function of the PSTN dialing plan. This information will be needed before the Avaya one-X Quick Edition SIP Proxy Identity can be populated.

For calls to the Avaya one-X Quick Edition stations the Cisco ISR will insert the Called Number in the To: field of the SIP INVITE. The Avaya one-X Quick Edition will then map the Called Number from the To: field to the appropriate Avaya one-X Quick Edition station. These Application Notes show how Avaya one-X Quick Edition can be administered to map the Called Number to a specific station, a group of stations, or the auto attendant

For calls from the Avaya one-X Quick Edition stations Avaya one-X Quick Edition inserts the dialed (9+) number in the To: field of the SIP INVITE (without the 9). The Cisco ISR will insert the dialed number from the To: field in the ISDN Called Number and the number in the From: field of the SIP INVITE in the ISDN Calling Number Field for outbound calls.

4.4.2. Configure Primary Proxy Identity for ISDN PRI

The Cisco ISR requires only one SIP Proxy Identity as a **Primary Registerer** for the PRI. The SIP Proxy Identity of the **Primary Registerer** in these Application Note is 2012342200. This number has been selected as the Main Number for this location. Inward calls to this number will be answered by the Auto Attendant. If an Avaya one-X Quick Edition station does not have a PSTN number assigned to it the Main number will be used for outward calls.

Select **Add** from the menu on the far right side of the screen. Select **Trunk** from the pull down menu at Type. Allow the screen to refresh. Enter a *Name* in the Name field. Enter a *number* in the Identity field (2012342200). Select **Create as Registerer** from the Primary Registerer pull-down menu. Select **Global** as the outgoing extension and **Auto Attendant** as the incoming extension.

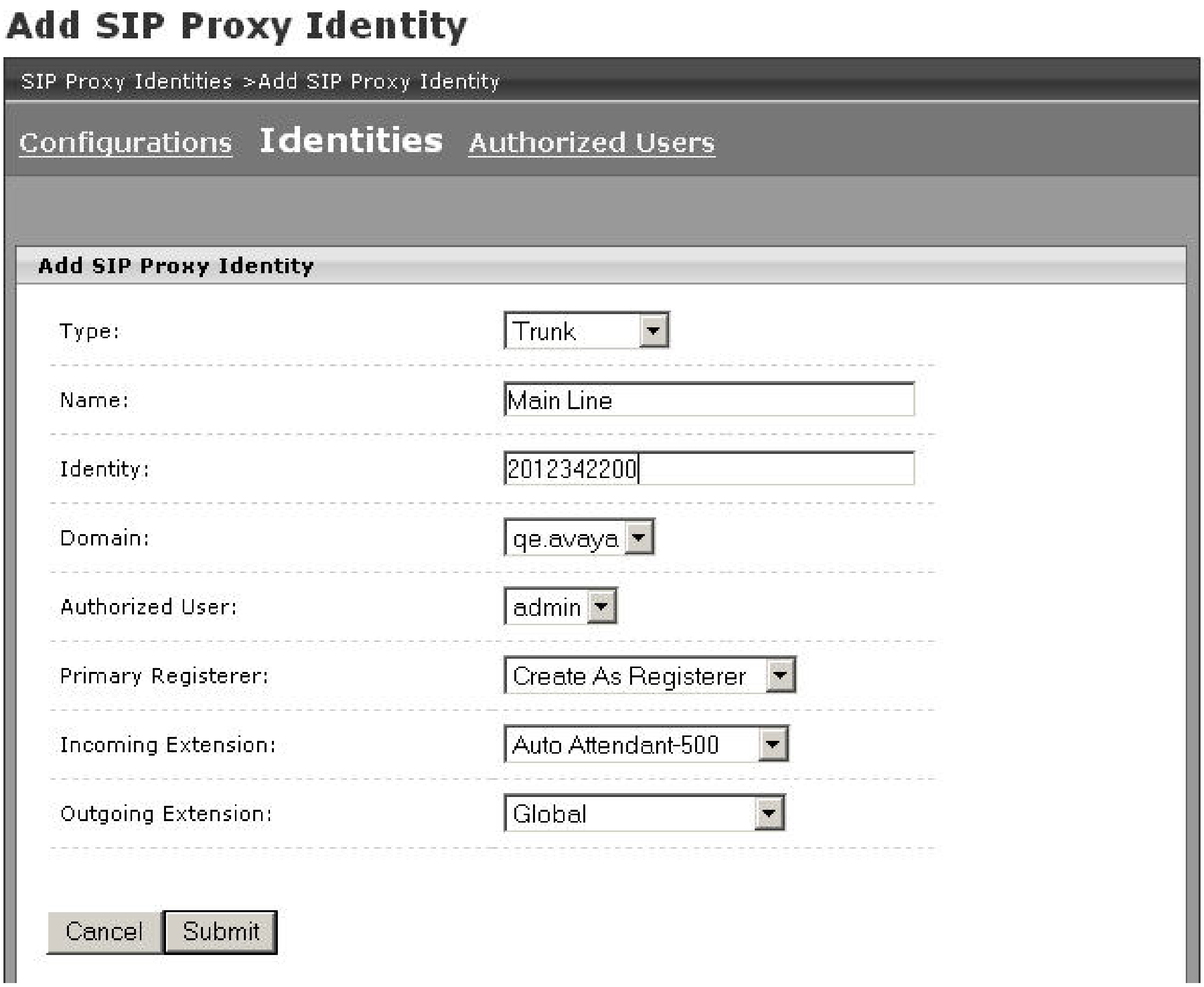


Figure 14: Create a Primary Registerer for PRI

4.4.3. Configure Individual Numbers to Stations for ISDN PRI

For several Avaya one-X Quick Edition stations a specific number is assigned so that inward calls to that number ring that specific station and outward calls from that station provide that specific number as the ISDN Calling Number.

For each of the PSTN numbers used exclusively with one station add a SIP Proxy Identity:

Select **Add** from the menu on the far right side of the screen. Select **Trunk** from the pull down menu at Type. Allow the screen to refresh. Enter a *Name* in the Name field. Enter a *number* in the Identity field (2012342201). Select the number created as the Primary Registerer (2012342200) from the Registerer pull-down menu... Select the station (x201) as both the outgoing and incoming extension. Repeat for 2012342202 (x202).

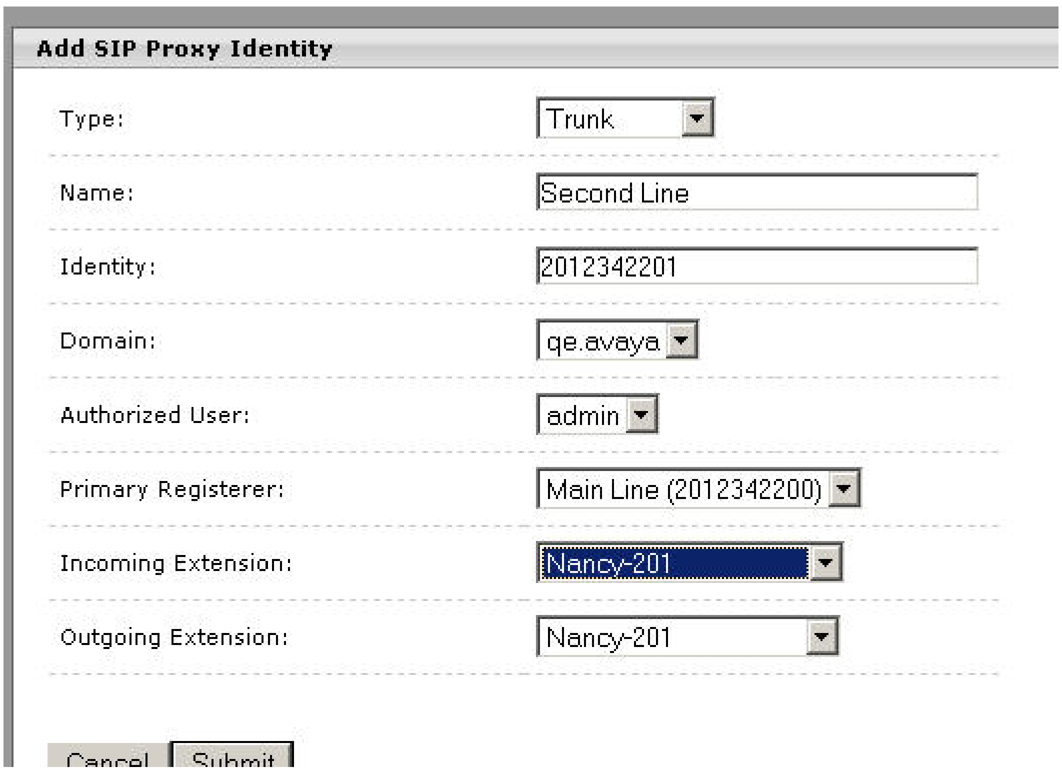


Figure 15: Add a SIP Proxy Identity with Exclusive Line Association

Calls to 2012342201 will ring on extension 201. Calls to 2012342202 will ring on extension 202.

4.4.4. Configure a Group Number Proxy for ISDN PRI

Select **Add** from the menu on the far right side of the screen. Select **Trunk** from the pull down menu at Type. Allow the screen to refresh. Enter a *Name* in the Name field. Enter a *number* in the Identity field (2012342203). Select **Main Line (2012342200)** from thePrimary Register pull-down menu. Select the sales group (x220) as both the outgoing and incoming extension.

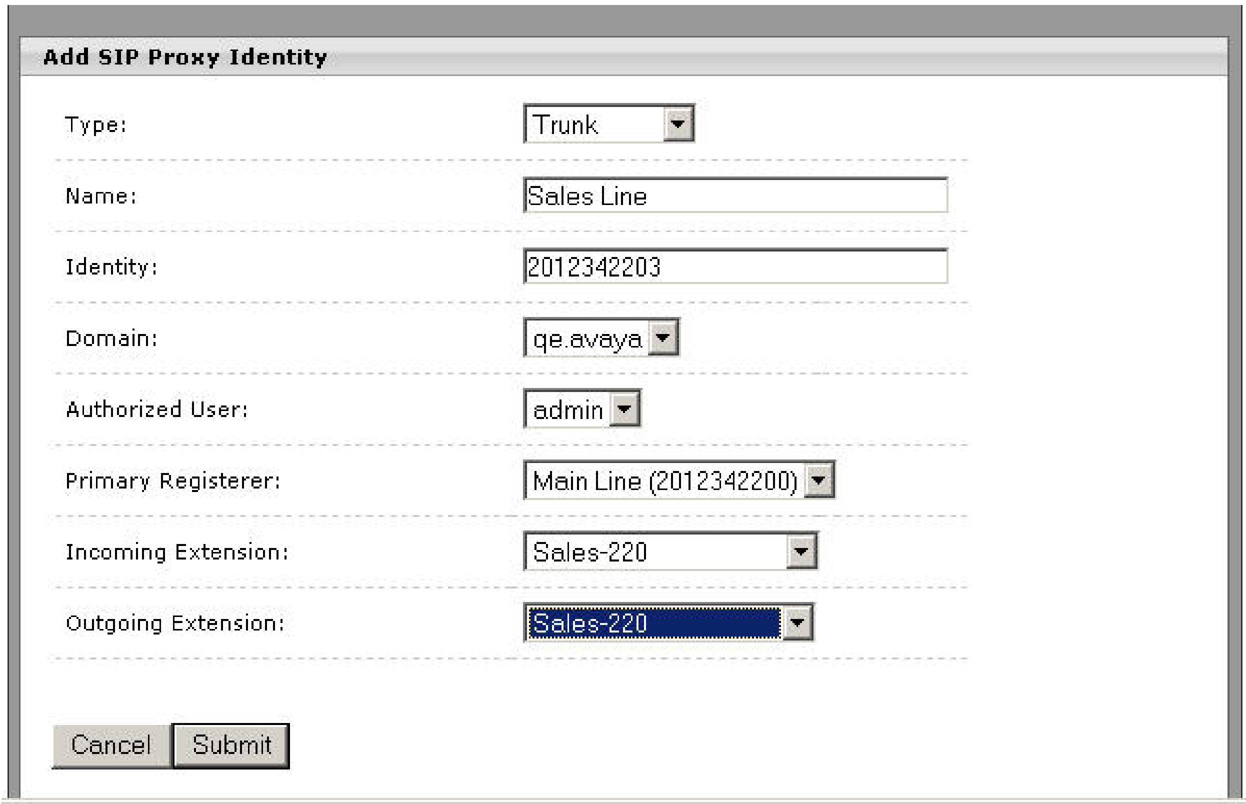


Figure 16: Create a SIP Proxy Identity for the Sales Line Group

Calls to 2012342203 will ring on the group number 220. Calls from any station in the group number 220 will put 2012342203 in the ISDN Called Number field.

5. Configuring the Cisco ISR

The following commands are specific to the Cisco 2821 used in the sample configuration. Refer to the Cisco documentation for the particular router being configured for additional details (Section 9).

5.1. Configuring the Cisco ISR for SIP Gateway Services

Log into the Cisco ISR and enter **`config t`** at the command line to enter the configuration mode.

Set the clock to GMT. SIP requires the use of GMT.

```
Clock timezone GMT 0
```

Populate the name-server with the DNS server IP address.

```
ip name-server 192.45.130.100
```

Bind SIP control and media to the interface where the Avaya one-X Quick Edition is connected.

```
voice service voip
 sip
  bind control source-interface GigabitEthernet0/0
  bind media source-interface GigabitEthernet0/0
  no call service stop
```

Create a voice class for the codec selected and include g711ulaw and g729r8. You must include both g711 and g729r8 because calls to the Avaya one-X Quick Edition Auto Attendant or Voice Mail system use g729r8. Do not use g729r8b.

```
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
```

Create the SIP User Agent (UA) in the Cisco ISR. The SIP UA defines how the Gateway software in the Cisco ISR will register with the SIP Proxy in the Avaya one-X Quick Edition telephones. It is essential that the form of address used in the **registrar** and the **sip-server** command is **dns:qe.avaya.** The Cisco ISR allows other forms (e.g. ipv4:20.1.100.76) but using these forms will result in SIP messages that Avaya one-X Quick Edition will not interpret correctly.

```
sip-ua
 retry invite 3
 retry response 3
 retry bye 3
 retry cancel 3
 retry register 10
 retry subscribe 10
 timers trying 1000
 registrar dns:qe.avaya expires 3600
 sip-server dns:qe.avaya
```

5.1.1. VoIP Dial-Peer

The Cisco ISR command to route VoIP calls to the Avaya one-X Quick Edition is dial-peer voice *number* voip. Calls into the Cisco ISR on FXO or PRI trunks will be routed to the Avaya one-X Quick Edition SIP Proxy by this command.

Create a dial-peer to route calls to the Avaya one-X Quick Edition telephones as VoIP calls.

dial-peer voice 2 voip	
description route to QE	
destination-pattern [25].T	Numbers that start with 2 or 5
voice-class codec 1	Voice codec class created previously
session protocol sipv2	SIP
session target sip-server	Sip-server was defined in the sip-ua as qe.avaya
session transport udp	QE uses UDP
dtmf-relay rtp-nte	QE uses RFC2833 for DTMF

The choice of the **destination-pattern [25].T** is particular to the sample configuration. The pattern means telephone numbers of any length that start with 2 or 5 should be routed to the Avaya one-X Quick Edition SIP Proxy. A configuration that used extensions that started with 7 rather than 2 would use **destination-pattern [75].T**. Cisco IOS Command documentation should be consulted for the full syntax of the destination pattern.

5.2. Telephone Stations (FXS)

For each telephone plugged into the Cisco ISR (FXS port) populate the **`station-id number`**. The **`station-id number`** must match the identity created for the subscriber in Section 4.2. The Cisco ISR will populate the From: field in the SIP INVITE to Avaya one-X Quick Edition with the station-id number.

```
voice-port 2/0/0
 station-id number 290
 caller-id enable

voice-port 2/0/1
 station-id number 291
 caller-id enable
```

Create a dial-peer for each telephone to route calls to the stations' extension to the port associated with that extension. This is the reverse of the VoIP ***dial-peer*** command explained in Section 5.1.1 which allows the Cisco ISR to route VoIP calls to the Avaya one-X Quick Edition.

Calls from Avaya one-X Quick Edition stations will be routed to the FXS port by matching the *number* in the destination-pattern command to the dialed telephone number.

```
dial-peer voice 290 pots
 destination-pattern 290
 port 2/0/0

dial-peer voice 291 pots
 destination-pattern 291
  port 2/0/1
```

The choice of using the same number in the dial-peer command and the destination-pattern is not essential, but the destination-pattern *number* must match the station-id *number* and the identity created in Section 4.2 for this telephone.

5.3. Telephone Lines (FXO)

This configuration uses the Private-Line Automatic Ringdown (PLAR) feature of the Cisco ISR to route calls to the Avaya one-X Quick Edition telephones.

```
connection plar number
```

When the Cisco ISR detects ringing on the FXO port it will transmit a SIP INVITE to *number@qe.avaya*. Avaya one-X Quick Edition uses the *number* to determine the map between the FXO port and the Avaya one-X Quick Edition extension. The *number* should be the **Incoming Extension** populated in Section 4.3. The SIP Proxy Identity will be overridden by the *number* in the **`connection plar command`**. The number can be an extension, a group number, or the auto attendant.

5.3.1. Route to the Auto Attendant

For an FXO port used for direct inward calls to a particular extension, populate the *number* field of the connection plar command with the extension *number*.

```
voice-port 2/0/19
 connection plar 2012341668205
 caller-id enable
```

The **`connection plar 2012341668205`** command directs the Cisco ISR to populate the SIP *To:* field with 2012341668205@qe.avaya. The Avaya one-X Quick Edition administration in Section 4.3.1 results in these calls being answered by the Auto Attendant.

The Cisco ISR will populate the SIP *From:* field with the Caller ID information from the PSTN. If the PSTN line does not support Caller ID use the station-id number command to populate the *From:* field.

```
voice-port 2/0/19
 connection plar 2012341668
 station-id number 2012341668
```

Either Caller ID must be available or the station-id must be populated. If neither of these is available the Avaya one-X Quick Edition will reject the incoming calls on this PSTN line.

5.3.2. Route to a Station Number

For an FXO port used for direct inward calls to a group extension e.g., sales, populate the *number* field of the `connection plar` command with the group *number*.

```
voice-port 2/0/20
 connection plar 2012341669220
 caller-id enable
```

The **`connection plar 2012341669220`** command directs the Cisco ISR to populate the SIP *To:* field with 2012341669220@qe.avaya. The administration in Section 4.3.2 enables incoming calls on this PSTN line to ring at the extensions associated with the group extension 201220 (sales).

5.3.3. Other routing possibilities

More than one FXO port can be routed to the same identity created in Section 4.3. Populate the `connection plar` *`number`*used for the FXO port with the *number* field of the identity created in Section 4.3. `connection plar` command with the Auto Attendant *number*.

```
voice-port 2/0/18
 connection plar 2012341668500
 caller-id enable
```

The `connection plar 500` command directs the Cisco ISR to populate the SIP *To:* field with 500@qe.avaya. The result is that the Auto Attendant will answer incoming calls on this PSTN line.

5.3.4. Dial-Peer

For calls from the Avaya one-X Quick Edition stations to the PSTN lines attached to the FXO ports create a dial-peer for each FXO port to dial outward the number dialed at the Avaya one-X Quick Edition telephone. The destination pattern .(period)T describes any sequence of digits in the SIP *To:* field. On the Avaya one-X Quick Edition telephone dialing 95551212# creates a SIP INVITE to 5551212. The result of **.T** destination pattern is that the call will be dialed on the first available line.

```
dial-peer voice 9 pots
 destination-pattern .T
 port 2/0/20
 no register e164

dial-peer voice 8 pots
 destination-pattern .T
 port 2/0/19
 no register e164
```

The `no register e164` command means that the Cisco ISR should not attempt to register these dial-peers with the Avaya one-X Quick Edition.

5.3.5. Dial-Peer for SIP Proxy Identity

The Avaya one-X Quick Edition only accepts registrations for all numeric identities. Since Avaya one-X Quick Edition must have at least one registration to match the SIP Proxy Identity created in Section 4.3.1 another dial-peer must be created.

```
dial-peer voice 201234166855500 pots
 destination-pattern 201234166855500
 port 2/0/19
```

This means that a call from a Avaya one-X Quick Edition station to 9+55500 will receive a second dial tone on the FXO port 2/0/19.

5.3.6. Using Dial-Peer to Reserve an FXO Trunk

Sometimes it is desirable to reserve a trunk for emergency use or the use of a particular individual in the enterprise. The dial-peers created in this document for outward calls all use the same destination pattern (.T). The Cisco ISR receives an outward call and uses the first available FXO trunk to dial that call. By changing the destination pattern additional digits can be required.

```
dial-peer voice 8 pots
 destination-pattern 116.T
 port 2/0/19
```

Using this destination-pattern instead of the one in Section 5.3.4 requires the user to dial 9+116+telephone number. The Avaya one-X Quick Edition system will strip the 9, as it does with all outward calls. The Cisco ISR will strip the 116.

5.4. ISDN PRI

The configuration of the ISDN PRI on the Cisco ISR will depend on the particular Cisco board being used for the trunk interface and the type of equipment being used by the PRI provider. The Cisco documentation for configuring ISDN PRI should be consulted.

5.4.1. ISDN PRI General Configuration

The PRI provider should have provided the information necessary to complete these steps.

Select the ISDN switch type used by the PRI provider.

```
isdn switch-type primary-5ess
```

Define the controller (Cisco card) parameters appropriate for this PRI interface.

```
controller T1 0/0/0
 framing esf
 clock source internal
 linecode b8zs
 pri-group timeslots 1-2,24
```

The `pri-group` command defines what B channels will be used by the PRI provider. In this case, 1 and 2 are used. The second number is the D channel (24 with T1 or 30 with E1 PRI). Note that the timeslots used on the Cisco ISR and on the PRI provider's equipment must be aligned. Also note that there can be more telephone numbers assigned than timeslots. When there are no available timeslots, users will receive a busy tone or a network tone indicating the call can not be completed.

Define the D-channel signaling for the PRI controller.

```
interface Serial0/0/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 isdn outgoing display-ie
 no cdp enable
```

Note that the interface number 23 is the D-channel (24).

5.4.2. PRI Dial-peer for outward calls

The destination-pattern for valid PRI calls required by the PRI provider can be complex. The Cisco documentation may be needed to determine the best destination-pattern for the PRI provider. The pattern in the sample configuration matches any string (9+string) dialed at the Avaya one-X Quick Edition station. This may be adequate for most PRI providers.

```
dial-peer voice 116 pots
 destination-pattern .T
 direct-inward-dial
 port 0/0/0:23
 forward-digits all
 no register e164
```

5.4.3. PRI Dial-peer for SIP Proxy Identity

As in the FXO case, Section 5.3.5, it is necessary to have at least one registered SIP Identity with the Avaya one-X Quick Edition. In Section 4.4.1 the identity 2012342200 was created as the Primary Registerer for the PRI interface. The appropriate dial-peer commands are:

```
dial-peer voice 201 pots
 destination-pattern 2012342200
 port 0/0/0:23
```

5.5. Save Configuration

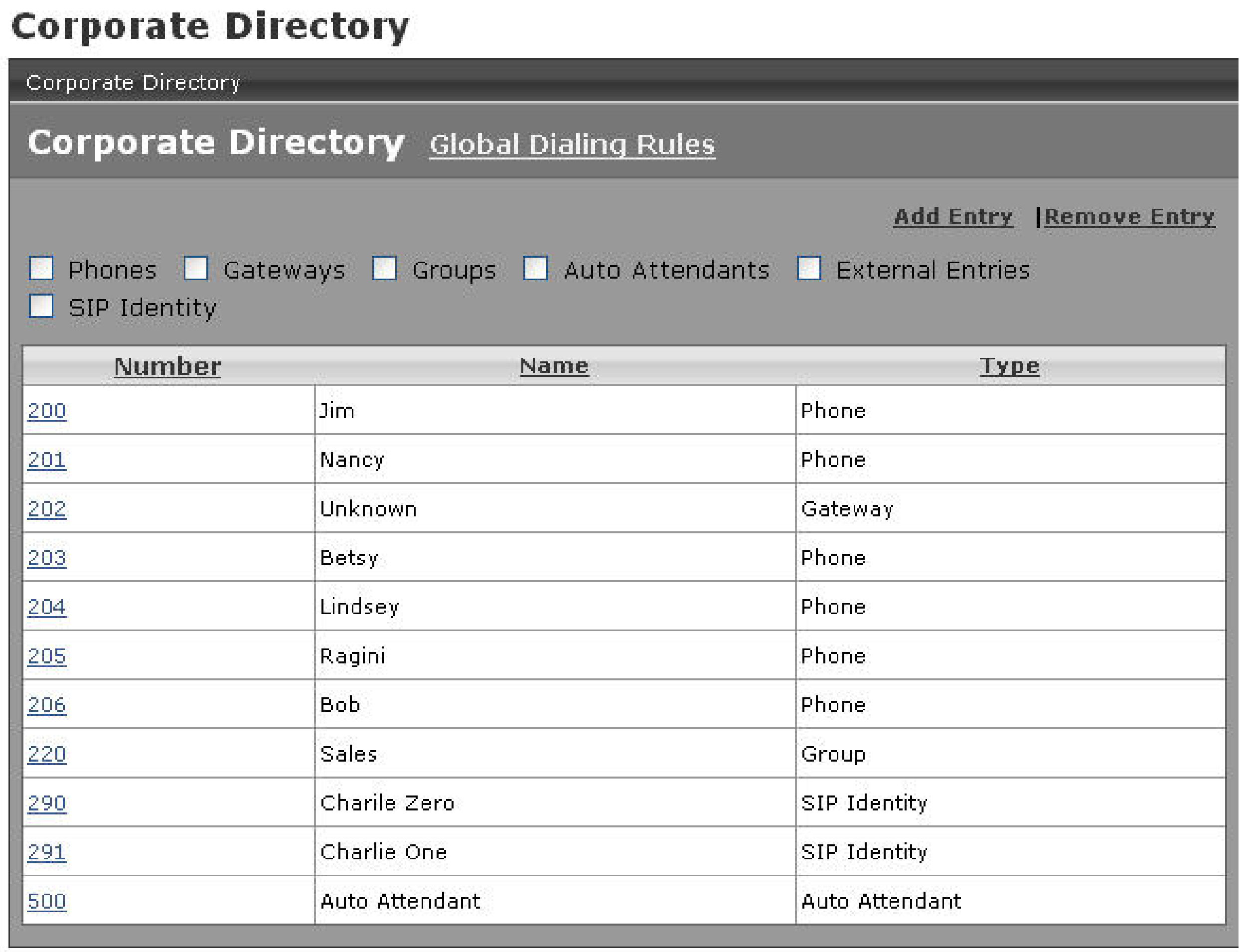
Exit the configuration mode and save the configuration on the Cisco ISR (write mem).

6. Verification Steps

Power cycle the Cisco ISR and the Avaya one-X Quick Edition telephones.

Open a web browser and log in to the Avaya one-X System Options as described in Section 4.

From the left side menu (Figure 4) select **Corporate Directory**. The Corporate Directory should show the FXS subscribers with **SIP Identity** in the **Type** column. It will not show the FXO trunks registered.



Number	Name	Type
200	Jim	Phone
201	Nancy	Phone
202	Unknown	Gateway
203	Betsy	Phone
204	Lindsey	Phone
205	Ragini	Phone
206	Bob	Phone
220	Sales	Group
290	Charile Zero	SIP Identity
291	Charlie One	SIP Identity
500	Auto Attendant	Auto Attendant

Figure 17: Corporate Directory

Log in to the Cisco ISR and enter the command

```
show sip-ua register status
```

The output should show the subscribers and the trunks as registered. Some of the other non-numeric destination patterns may also be displayed as unregistered. These can be ignored.

```
Line            peer              expires(sec)  registered
============    =============     ============  ===========
290             290               574           yes
291             291               575           yes
2012341668      2012341668        575           yes
2012342200      201               576           yes
```

Place a call to one of the subscriber stations from one of the Avaya one-X Quick Edition IP Telephones. Answer that call and verify that the voice path is connected. Place a call from one of the subscriber stations to one of the Avaya one-X Quick Edition IP Telephones. Answer that call and verify that the voice path is connected. This verifies that the subscriber stations are correctly administered.

From one of the Avaya one-X Quick Edition Telephones place a call to a number on the PSTN. Verify that the call completes and a voice path is connected.

From an outside line call the PSTN number associated with the Auto-Attendant. When the Auto-Attendant answers dial an extension and answer the call on the Avaya one-X Quick Edition Telephone.

From an outside line call the PSTN number associated with extension 205. Verify that the call rings at extension 205 but do not answer the call. Verify that the call goes to voicemail and that a message can be left and retrieved.

7. Troubleshooting

The Cisco ISR must have sufficient DSP resources for the calls that are being placed to the Avaya one-X Quick Edition telephones. More DSP resources are used for g729 calls than for g711 calls. The result is that a call may go through to an extension in g711 and then lose voice path when the call is transferred to the Auto Attendant. The calling party will hear only silence.

The command **`show voice dsp voice`** will display the current allocation of the DSP resources on the Cisco ISR.

```
----------------------------FLEX VOICE CARD 0 ------------------------------
                           *DSP VOICE CHANNELS*
DSP   DSP              DSPWARE CURR  BOOT                             PAK   TX/RX
TYPE  NUM CH CODEC     VERSION STATE STATE   RST AI VOICEPORT TS ABRT PACK COUNT
===== === == ======== ======= ===== ======= === == ========= == ==== ============
C5510 001 01 g729r8   4.4.801 busy  idle      0  0 2/0/19    52    0      355/467
C5510 001 02 g711ulaw 4.4.801 busy  idle      0  0 2/0/20    56    0         0/1
```

8. Conclusion

The Avaya one-X Quick Edition system can make use of the Cisco ISR for providing access to PSTN lines and analog stations. These Application Notes can be extended to additional Cisco PSTN devices such as E&M analog, channelized T1/E1 facilities and PRI, however, this testing is limited to the devices listed in this document.

9. Additional References

[1] Avaya one-X Quick Edition, Release 3.1.0 System Administration Guide

[2] Cisco, “Understanding Dial Peers and Call Legs on Cisco IOS Platforms”,
Document ID: 12164.

[3] Cisco, “Configuring Connection PLAR for VoIP Gateways”,
Document ID: 14368.

[4] Cisco, “ISDN Voice, Video and Data Call Switching with Router TDM Switching Features”,
Document ID: 64811.

Appendix A - Detailed Cisco ISR Configuration

Product	Description
CISCO2821	w/ AC PWR,2GE,4HWICs,3PVDM,1NME-X,2AIM,IP BASE,64F/2
MEM2800-64U256CF	64 to 256 MB CF Factory Upgrade for Cisco 2800 Series
EVM-HD-8FXS/DID	High density voice/fax extension module - 8 FXS/DID
EM-HDA-6FXO	6-port analog voice/fax expansion module
PVDM2-16	16-Channel Packet Voice/Fax DSP Module
VWIC-1MFT-T1	1-Port RJ-48 Multiflex Trunk - T1
FL-INTVVSRV-2821	Cisco 2821 Integrated VoiceVideo License: Gatekeeper IPIP GW

```
care2821#show diag
Slot 0:
C2821 Motherboard with 2GE and integrated VPN Port adapter, 2 ports
Port adapter is analyzed
Port adapter insertion time unknown
Onboard VPN: v2.2.0
EEPROM contents at hardware discovery:
PCB Serial Number        : FOC103422Y1
Hardware Revision        : 1.0
Top Assy. Part Number    : 800-26921-02
Board Revision           : A0
Deviation Number         : 0
Fab Version              : 03
RMA Test History         : 00
RMA Number               : 0-0-0-0
RMA History              : 00
Processor type           : 87
Hardware date code       : 20060827
Chassis Serial Number    : FTX1036A0US
```

```
Chassis MAC Address      : 0019.2f06.6e60
MAC Address block size   : 32
CLEI Code                : COM3D00BRA
Product (FRU) Number     : CISCO2821
Part Number              : 73-8853-04
Version Identifier       : V03
EEPROM format version 4
EEPROM contents (hex):
  0x00: 04 FF C1 8B 46 4F 43 31 30 33 34 32 32 59 31 40
  0x10: 03 E8 41 01 00 C0 46 03 20 00 69 29 02 42 41 30
  0x20: 88 00 00 00 00 02 03 03 00 81 00 00 00 00 04 00
  0x30: 09 87 83 01 32 1A 9B C2 8B 46 54 58 31 30 33 36
  0x40: 41 30 55 53 C3 06 00 19 2F 06 6E 60 43 00 20 C6
  0x50: 8A 43 4F 4D 33 44 30 30 42 52 41 CB 8F 43 49 53
  0x60: 43 4F 32 38 32 31 20 20 20 20 20 20 82 49 22 95
  0x70: 04 89 56 30 33 20 D9 02 40 C1 FF FF FF FF FF FF

PVDM Slot 0:
16-channel (G.711) Voice/Fax PVDMII DSP SIMM PVDM daughter card
Hardware Revision        : 3.2
Part Number              : 73-8538-04
Board Revision           : A0
Deviation Number         : 0
Fab Version              : 03
PCB Serial Number        : FOC09514RRV
RMA Test History         : 00
RMA Number               : 0-0-0-0
RMA History              : 00
Processor type           : 00
Product (FRU) Number     : PVDM2-16
Version Identifier       : NA
EEPROM format version 4
EEPROM contents (hex):
  0x00: 04 FF 40 03 EF 41 03 02 82 49 21 5A 04 42 41 30
  0x10: 88 00 00 00 00 02 03 C1 8B 46 4F 43 30 39 35 31
  0x20: 34 52 52 56 03 00 81 00 00 00 00 04 00 09 00 CB
  0x30: 88 50 56 44 4D 32 2D 31 36 89 4E 41 20 20 D9 02
  0x40: 40 C1 FF FF FF FF FF FF FF FF FF FF FF FF FF FF
  0x50: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
  0x60: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
  0x70: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF

WIC Slot 0:
T1 (1 port) Multi-Flex Trunk WAN daughter card
Hardware revision 1.0         Board revision B0
Serial number     28208605  Part number    800-03568-03
FRU Part Number   WIC/VIC-T1=

Test history      0x0         RMA number     00-00-00
Connector type    PCI
EEPROM format version 1
EEPROM contents (hex):
  0x20: 01 20 01 00 01 AE 6D DD 50 0D F0 03 00 00 00 00
  0x30: 58 00 00 00 02 06 27 00 FF FF FF FF FF FF FF FF

Slot 2:
Voice VM-8FXS/DID  Port adapter
```

```
Port adapter is analyzed
Port adapter insertion time unknown
EEPROM contents at hardware discovery:
Hardware Revision        : 1.0
Top Assy. Part Number    : 800-22992-03
Board Revision           : C0
Deviation Number         : 0
Fab Version              : 05
PCB Serial Number        : FOC10025GB5
RMA Test History         : 00
RMA Number               : 0-0-0-0
RMA History              : 00
Product (FRU) Number     : EVM-HD-8FXS/DID
Version Identifier       : V03
Longitudinal Calibration : 00 20 20 20 20 20 20 20
                           20
CLEI Code                : CNP5130AAC
EEPROM format version 4
EEPROM contents (hex):
  0x00: 04 FF 40 04 16 41 01 00 C0 46 03 20 00 59 D0 03
  0x10: 42 43 30 88 00 00 00 00 02 05 C1 8B 46 4F 43 31
  0x20: 30 30 32 35 47 42 35 03 00 81 00 00 00 00 04 00
  0x30: CB 8F 45 56 4D 2D 48 44 2D 38 46 58 53 2F 44 49
  0x40: 44 89 56 30 33 20 D9 02 40 C1 D3 09 00 20 20 20
  0x50: 20 20 20 20 20 C6 8A 43 4E 50 35 31 33 30 41 41
  0x60: 43 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
  0x70: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF

EM slot 0: None

EM Slot 1:
VENOM FXO Voice Expansion Module (6 port)
Hardware Revision        : 1.0
Top Assy. Part Number    : 800-24884-02
Board Revision           : A1
Deviation Number         : 0
Fab Version              : 01
PCB Serial Number        : FOC10084WHC
RMA Test History         : 00
RMA Number               : 0-0-0-0
RMA History              : 00
Product (FRU) Number     : EM-HDA-6FXO
Version Identifier       : V02
CLEI Code                : IPUCADUBAA
EEPROM format version 4
EEPROM contents (hex):
  0x00: 04 FF 40 04 18 41 01 00 C0 46 03 20 00 61 34 02
  0x10: 42 41 31 88 00 00 00 00 02 01 C1 8B 46 4F 43 31
  0x20: 30 30 38 34 57 48 43 03 00 81 00 00 00 00 04 00
  0x30: CB 8B 45 4D 2D 48 44 41 2D 36 46 58 4F 89 56 30
  0x40: 32 20 D9 02 40 C1 C6 8A 49 50 55 43 41 44 55 42
  0x50: 41 41 FF FF FF FF FF FF FF FF FF FF FF FF FF FF
  0x60: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
  0x70: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
```

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 Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com