



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring SIP Trunking between McLeodUSA Communications SIP Trunking Solution with an Avaya IP Telephony Solution – 1.0**

### **Abstract**

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the McLeodUSA SIP Trunking solution and an Avaya IP telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya SIP, H.323, digital and analog endpoints.

McLeodUSA is a domestic service provider offering affordable, flexible and reliable integrated voice and data services to small and medium-sized business in nearly 500 cities throughout the Midwest, Rocky Mountain, Southwest and Northwest regions.

McLeodUSA is a member of the Avaya DevConnect Service Provider program. 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 steps to configure Session Initiation Protocol (SIP) trunking between the McLeodUSA SIP Trunking solution and an Avaya IP telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya SIP, H.323, digital and analog endpoints.

McLeodUSA is a domestic service provider offering affordable, flexible and reliable integrated voice and data services to small and medium-sized business in nearly 500 cities throughout the Midwest, Rocky Mountain, Southwest and Northwest regions.

SIP Trunking from McLeodUSA combines local and long-distance voice service, secure data networking and broadband Internet access on one performance-guaranteed connection.

Customers using this Avaya IP telephony solution with McLeodUSA SIP Trunking solution are able to place and receive PSTN calls via a dedicated broadband Internet connection using the SIP protocol. This converged network solution is an alternative to more traditional PSTN trunks such as T1 or ISDN PRI.

McLeodUSA can connect directly to an IP phone system as well as an external router. Customers looking for a turnkey access solution can select a Managed Service option, where McLeodUSA provides and manages the router that interfaces with customer equipment.

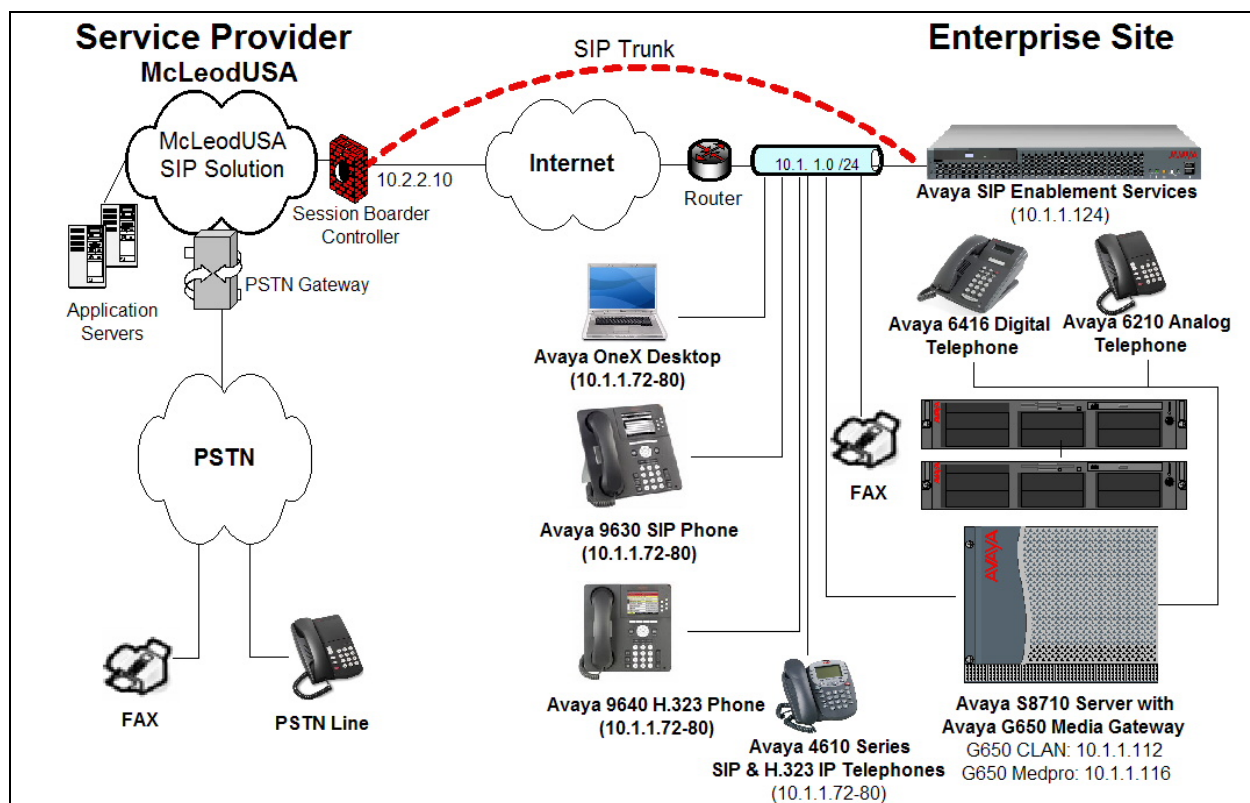
McLeodUSA's SIP Trunking solution offers the following capabilities:

- Outbound domestic calling to local and long distance services
- Outbound international calling
- Incoming Direct Inward Dial (DID) service
- Dynamically reroute inbound calls from one location to another location on the McLeodUSA MPLS network, without long distance charges
- Call restriction, by area code, by number, or by call type

**Figure 1** illustrates an example Avaya IP telephony solution connected to McLeodUSA's SIP Trunking solution. This is the configuration used during the DevConnect compliance testing process. Please refer to **Section 6** for the features tested with this solution.

The Avaya IP telephony solution used to create a simulated customer site contained:

- Avaya S8710 Server with an Avaya G650 Media Gateway. The S8710 served as the host processor for Avaya Communication Manager.
- Avaya SIP Enablement Services (SES) software operating on an Avaya S8500B server platform.
- Avaya 9630 IP telephone (configured for the SIP protocol)
- Avaya 9640 IP telephone (configured for the H.323 protocol)
- Avaya 4610 IP telephone (configured to use either the SIP or H.323 protocol).
- Avaya 6416 digital and 6210 analog phones.
- Avaya OneX Desktop SIP Softphone



**Figure 1: Avaya IP Telephony Network using McLeodUSA SIP Trunking Service**

## 1.1 Call Flows

To better understand how calls are routed between the PSTN and the enterprise site shown in **Figure 1** using SIP trunks, two call flows are described in this section. The first call scenario illustrated in **Figure 2** is a PSTN call to the enterprise site terminating on a typical analog telephone supported by Avaya Communication Manager.

1. A user on the PSTN dials a McLeodUSA provided DID number assigned to an Avaya Communication Manager telephone at the enterprise site. The PSTN routes the call to the McLeodUSA network (as the local service provider). McLeodUSA then routes the DID number to the assigned customer.

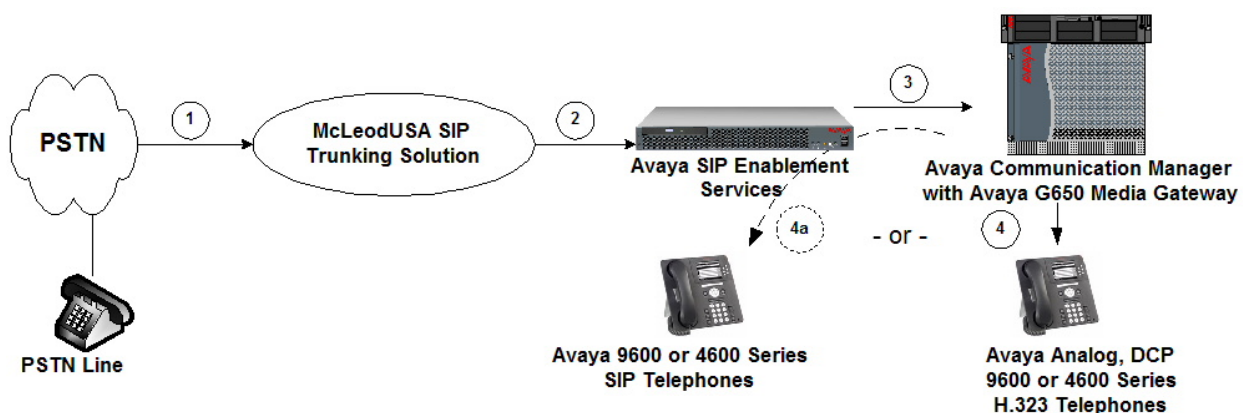
2. Based on the DID number, McLeodUSA offers the call to Avaya SES using SIP signaling messages sent over the converged access facility. Note that the assignment of the DID number and the address of the Avaya SES server was previously established during the ordering and provisioning of the service.

3. Avaya SES routes the call to the Avaya S8710 Server running Avaya Communication Manager over a SIP trunk between the elements.

4. Avaya Communication Manager terminates the call to the directly connected non SIP Phone, (i.e. Analog, Digital, or H.323 IP Phones), as shown in step 4.

- or -

4a. Inbound calls destined for a SIP extension at the enterprise are routed to Avaya Communication Manager which then transmits the appropriate SIP signaling via Avaya SES to the SIP telephone (as shown by the 4a arrow.)



**Figure 2: Incoming PSTN Calls to Avaya Communication Manager**

Appendix A illustrates an example of a SIP INVITE message sent by McLeodUSA for an incoming DID call.

The second call scenario illustrated in **Figure 3** is an outgoing call from an Avaya telephone at the enterprise site to the PSTN via the SIP trunk to McLeodUSA.

1. An Avaya H.323, analog or digital telephone served by Avaya Communication Manager originates a call to a user on the PSTN.

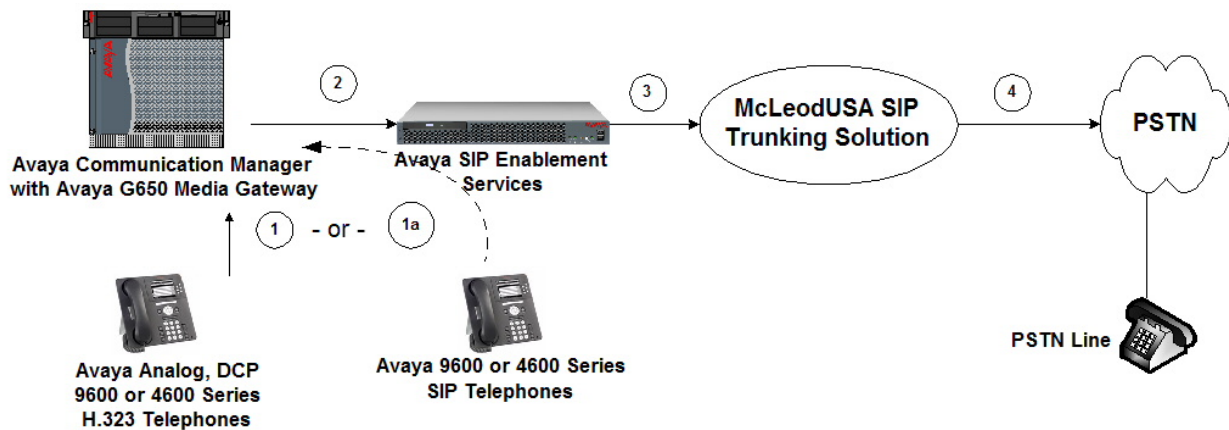
- or -

1a. An Avaya SIP telephone originates a call that is routed via Avaya SES (as shown by the 1a arrow) to Avaya Communication Manager.

2. The call request is handled by Avaya Communication Manager where origination treatment such as class of service restrictions and automatic route selection is performed. Avaya Communication Manager selects the SIP trunk and sends the SIP signaling messages to Avaya SIP Enablement Services.

3. Avaya SIP Enablement Services routes the call to McLeodUSA.

4. McLeodUSA completes the call to the PSTN.



**Figure 3: Outgoing Calls from Avaya Communication Manager to the PSTN**

## 2. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Avaya S8710 Server with an Avaya G650 Media Gateway	Communication Manager 4.0.1 R014x.00.1.731.2 Update: 00.1.731.2-14300
Avaya SIP Enablement Services on S8500B Media Server	SES-4.0.0.0-033.6
Avaya 9640 IP Telephone	R1.5 – H.323
Avaya 9630 IP Telephone	R1.0.2.2 (20) – SIP
Avaya 4621 IP Telephone	R2.2.2 – SIP
Avaya one-X Desktop Edition SIP endpoint	R2.1 – SIP – Build 78
Avaya 4610 IP Telephone	R2.8.3 – H.323 –
Avaya 6416 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
McLeodUSA Global SIP Trunking Solution Components	
McLeodUSA Application Server	R14 SP1
McLeodUSA PSTN Gateway	V3.10.1.7.xp19
McLeodUSA Session Border Controller (CXC-50)	V3.3.1

**Table 1: Equipment and Software Tested**

The specific configuration above was used for the McLeodUSA compatibility testing. Note that this solution will be compatible with all other Avaya Server and Media Gateway platforms running similar versions of Avaya Communication Manager and Avaya SIP Enablement Services.

### 3. Configure Avaya Communication Manager

This section describes the steps for configuring a SIP trunk on Avaya Communication Manager. The SIP trunk is established between Avaya Communication Manager and Avaya SIP Enablement Services (SES) server. This trunk will carry the SIP signaling sent to the McLeodUSA SIP Trunking solution.

This SIP trunk also provides the trunking for SIP endpoint devices such as Avaya 4620 SIP telephone and Avaya SIP soft phone using Avaya Communication Manager in the recommended off-PBX stations (OPS) configuration. Avaya SIP telephones are configured as OPS stations on Avaya Communication Manager. OPS SIP stations register with Avaya SES but have calling privileges and features managed by Avaya Communication Manager. Avaya Communication Manager acts as a back-to-back SIP user agent when a SIP phone places or receives a call over a SIP trunk to a service provider.

Note the use of SIP endpoints is optional. The steps discussed in Sections 3.2 and 4.3 describing SIP endpoint administration may be omitted if SIP endpoints are not used. In the Avaya SIP architecture, the Avaya SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to McLeodUSA's SIP Trunking solution. There is no direct SIP signaling path between McLeodUSA and Avaya Communication Manager or Avaya SIP endpoints.

For incoming calls, the Avaya SES uses media server routing maps to direct the incoming SIP messages to the appropriate Avaya Communication Manager. Once the message arrives at the Avaya Communication Manager further incoming call treatment, such as incoming digit translations, class of service restrictions, etc. may be performed.

All outgoing calls to the PSTN are processed within Avaya Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Avaya Communication Manager selects a SIP trunk, the SIP signaling is routed to the Avaya SES. Within the Avaya SES, the outbound proxy setting directs the outbound SIP messages to the McLeodUSA gateway.

The dial plan for the configuration described in these Application Notes consists of 10-digit dialing for local and long-distance calls over the PSTN. In addition, Directory Assistance calls (411) and International calls (011+Country Code) were also supported. Avaya Communication Manager routes all calls using Automatic Route Selection (ARS), except for intra-switch calls.

Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The general installation of the S8710 server, G650 Media Gateway and circuit packs such as the CLAN is presumed to have been previously completed and is not discussed here.

## 3.1 Sip Trunk Configuration

### Step 1: Confirm Necessary Optional Features

Log into the Avaya Communication Manager's SAT interface and confirm that sufficient SIP trunk and Off PBX Telephone capacities are enabled. Use the **display system-parameters customer-options** command to determine these values as shown in **Figure 4**. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

display system-parameters customer-options		Page 1 of 10
OPTIONAL FEATURES		
G3 Version: V13		
Location: 1	RFA System ID (SID): 1	
Platform: 8	RFA Module ID (MID): 1	
		USED
	Platform Maximum Ports: 44000	229
	Maximum Stations: 36000	53
	Maximum XMOBILE Stations: 0	0
	Maximum Off-PBX Telephones - EC500: 10	0
	<b>Maximum Off-PBX Telephones - OPS: 36000</b>	<b>23</b>
	Maximum Off-PBX Telephones - SCCAN: 0	0
(NOTE: You must logoff & login to effect the permission changes.)		

**Figure 4: System-Parameters Customer-Options Form – Page 1**



On Page 2, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the McLeodUSA network, SIP endpoints and any other SIP trunks used. Note that each SIP OPS telephone on a call with McLeodUSA uses two SIP trunks for the duration of the call.

display system-parameters customer-options		Page 2 of 10
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	100	20
Maximum Concurrently Registered IP Stations:	100	1
Maximum Administered Remote Office Trunks:	0	0
Maximum Concurrently Registered Remote Office Stations:	0	0
Maximum Concurrently Registered IP eCons:	0	0
Max Concur Registered Unauthenticated H.323 Stations:	5	0
Maximum Video Capable H.323 Stations:	10	0
Maximum Video Capable IP Softphones:	10	0
<b>Maximum Administered SIP Trunks:</b>	<b>200</b>	<b>156</b>
Maximum Number of DS1 Boards with Echo Cancellation:	1	0
Maximum TN2501 VAL Boards:	1	1
Maximum G250/G350/G700 VAL Sources:	50	0
Maximum TN2602 Boards with 80 VoIP Channels:	2	1
Maximum TN2602 Boards with 320 VoIP Channels:	2	0
Maximum Number of Expanded Meet-me Conference Ports:	0	0
(NOTE: You must logoff & login to effect the permission changes.)		

**Figure 5: System-Parameters Customer-Options Form – Page 2**

## Step 2: Assign Node Names

In the **IP Node Names** form, assign the node name and IP address for the Avaya SIP Enablement Server at the enterprise site as shown in **Figure 6**. In this case “SES” and “10.1.1.124” are being used, respectively. The SES node name will be used throughout the other configuration screens of Avaya Communication Manager.

In this example “CLAN” and “10.1.1.112” are the name and IP address assigned to the TN799DP Control-Lan card. The CLAN entry was previously created during the installation of the system. Note, in smaller gateways such as an Avaya G350, the S8300 processor address (procr) is used as the SIP signaling interface instead of the CLAN interface.

```
change node-names ip
```

Page 1 of 1

Name		IP Address		IP NODE NAMES		Name		IP Address	
<b>CLAN</b>		<b>10</b>	<b>.1</b>	<b>.1</b>	<b>.112</b>			.	.
default		0	.0	.0	.0			.	.
ipsi		10	.1	.1	.109			.	.
medpro-hw11		10	.1	.1	.116			.	.
procr		10	.1	.1	.104			.	.
<b>SES</b>		<b>10</b>	<b>.1</b>	<b>.1</b>	<b>.124</b>			.	.
vall-tn2501ap		10	.1	.1	.122			.	.
windowPC		10	.1	.1	.101			.	.
		.	.	.	.			.	.
		.	.	.	.			.	.

( 8 of 8 administered node-names were displayed )  
 Use 'list node-names' command to see all the administered node-names  
 Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

**Figure 6: IP Nodes Names Form**

### Step 3: Define IP Codecs

Open the **IP Codec Set** form using the ip-codec value specified in the **IP Network Region** form (**Figure 7**) and enter the audio codec type to be used for calls routed over the SIP trunk. The settings of the **IP Codec Set** form are shown in **Figure 8**. Note that the **IP Codec Set** form may include multiple codecs listed in priority order to allow the codec for the call to be negotiated during call establishment. For McLeodUSA's service, G.729a is the preferred codec followed by G.711mu.

```
change ip-codec-set 1
```

Page 1 of 2

IP Codec Set

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
<b>1: G.729A</b>	<b>n</b>	<b>2</b>	<b>20</b>
<b>2: G.711MU</b>	<b>n</b>	<b>2</b>	<b>20</b>
3:			
4:			
5:			
6:			
7:			

**Figure 7: IP Codec Set Form**

#### Step 4: Define IP Network Region

The **IP Network Region** form specifies the parameters used by the SIP trunk group serving the Avaya SES server (used to reach McLeodUSA and any optional SIP endpoints). Note that these parameters also apply to any other elements (such as H.323 phones, MedPro cards, etc.) also assigned to this region. Use the **change ip-network-region** form command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on the Avaya SES. In this configuration, the domain name is *east.devcon.com*. This field is required for endpoints to call the public network.
- By default, **IP-IP Direct Audio** (both **Intra** and **Inter Region**) is enabled to allow audio traffic to be sent directly between endpoints without using media resources such as the TN2302AP IP Media Processor (MedPro) card. In the case of McLeodUSA, **IP-IP Direct Audio** will be supported and these parameters will be enabled.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this configuration, this codec set will apply to calls with McLeodUSA as well as any IP telephone (H.323 or SIP) within the enterprise.

In this case, the SIP trunk is assigned to the same IP network region as the G650 Media Gateway, CLAN and MedPro cards. If multiple network regions are used, Page 3 of each **IP Network Region** form must be used to specify the codec set for inter-region communications.

Note also that the **IP Network Region** form is used to set the QoS packet parameters that provides priority treatment for signaling and audio packets over other data traffic on McLeodUSA's Global SIP Trunking solution. These parameters may need to be aligned with the specific values provided by McLeodUSA.

change ip-network-region 1		Page 1 of 19
IP NETWORK REGION		
Region: 1		
Location: 1	<b>Authoritative Domain: east.devcon.com</b>	
Name: Default System All		
MEDIA PARAMETERS		
Codec Set: 1	<b>Intra-region IP-IP Direct Audio: yes</b>	
UDP Port Min: 2048	<b>Inter-region IP-IP Direct Audio: yes</b>	
UDP Port Max: 65531	IP Audio Hairpinning? y	
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46	RTCP Reporting Enabled? y	
Audio PHB Value: 46	RTCP MONITOR SERVER PARAMETERS	
Video PHB Value: 26	Use Default Server Parameters? y	
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5	AUDIO RESOURCE RESERVATION PARAMETERS	
H.323 IP ENDPOINTS		
H.323 Link Bounce Recovery? y	RSVP Enabled? n	
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

**Figure 8: IP Network Region Form**

## Step 5: Configure the Signaling Groups

For interoperability with McLeodUSA, a minimum of two signaling groups must be configured. One signaling group will be used for inbound and outbound PSTN calls to and from the McLeodUSA gateway while the second signaling group is required for calls involving OPS SIP stations – recall that SIP OPS stations register with the SES and therefore require this dedicated SIP trunk to leverage the calling privileges and features managed by Avaya Communication Manager. For the purposes of this document, these groups will be referred to as the “PSTN Signaling and Trunk group” and the “OPS Signaling and Trunk group”. The configuration steps below show how to configure both of these signaling groups.

Configure the **PSTN Signaling Group** form which will be used for PSTN calls shown in **Figure 9** as follows:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security). TLS is the only link protocol that is supported for SIP trunking with Avaya SIP Enablement Services.
- **Near-end Node Name** Specify the Avaya Control-Lan card (node name “CLAN”). This field value is taken from the **IP Node Names** form shown in **Figure 6**. For smaller media server platforms, the near (local) end of the SIP signaling group may be the S8300 server processor (procr) rather than the CLAN.
- **Far-end Node Name** Specify the Avaya SIP Enablement Services (node name “SES”). This field value is taken from the **IP Node Names** form shown in **Figure 6**.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- Enter the IP Network Region value assigned in the IP Network Region form (**Figure 7**). Note that if the **Far-end Network Region** field is different from the near-end network region, the preferred codec will be selected from the IP codec set assigned for the interregional connectivity for the pair of network regions. In this case, the same ip network region (Network Region 1) was used for local and PSTN calls; however, different network regions can be used in the field.
- Enter the domain name of the Avaya SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *englab1.mcleodusa.net*. For outbound PSTN calls to McLeodUSA, this field sets the domain in the Uniform Resource Identifier (URI) of the SIP “To” address in the outbound INVITE message. For inbound PSTN calls from McLeodUSA, the domain part of the incoming INVITE’s “From” header is compared to the **Far-end Domain** field of the configured signaling groups. The call is then routed to the signaling group that matches this field. Mis-configuring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- The **Direct IP-IP Audio Connections** field is set to ‘y’. McLeodUSA supports the Avaya **Direct IP-IP Audio** feature. This feature can be disabled if desired.
- The **DTMF over IP** field should remain set to the default value of *rtp-payload*. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833. [8]
- The default values for the other fields may be used.

add signaling-group 1 Page 1 of 1

SIGNALING GROUP

Group Number: 1

Group Type: sip

Transport Method: tls

Near-end Node Name: CLAN

Far-end Node Name: SES

Near-end Listen Port: 5061

Far-end Listen Port: 5061

Far-end Network Region: 1

Far-end Domain: englab1.mcleodusa.net

Bypass If IP Threshold Exceeded? n

DTMF over IP: rtp-payload

Direct IP-IP Audio Connections? y

IP Audio Hairpinning? n

Session Establishment Timer(min): 120

**Figure 9: PSTN Signaling Group Form**

Now, configure the *OPS Signaling Group* form following the same steps used for the signaling group above with one exception, set the **Far-end Domain** field to the local SES SIP domain, in this case the value used is *east.devcon.com* as shown in **Figure 10**:

add signaling-group 2 Page 1 of 1

SIGNALING GROUP

Group Number: 2

Group Type: sip

Transport Method: tls

Near-end Node Name: CLAN

Far-end Node Name: SES

Near-end Listen Port: 5061

Far-end Listen Port: 5061

Far-end Network Region: 1

Far-end Domain: east.devcon.com

Bypass If IP Threshold Exceeded? n

DTMF over IP: rtp-payload

Direct IP-IP Audio Connections? y

IP Audio Hairpinning? n

Session Establishment Timer(min): 120

**Figure 10: OPS Signaling Group Form**

## Step 6: Configure the Trunk Groups

As described above in step 5, two trunks must also be configured. One trunk will be paired with the PSTN signaling group and the other with the OPS signaling group.

Configure the *PSTN Trunk Group* form as shown in **Figure 11** using the **add trunk-group** command. In this case the trunk group number chosen is 1. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a mnemonic **Group Name**.
- Specify an available trunk access code (**TAC**).
- Set the **Service Type** field to *tie*.
- Specify the *PSTN* signaling group associated with this trunk group in the **Signaling Group** field as previously specified in **Figure 9**.
- Specify the **Number of Members** supported by this SIP trunk group.

Note that one trunk member is required for each call between a non-SIP endpoint and McLeodUSA. Calls involving a SIP endpoint and McLeodUSA will use two trunk members for the duration of the call.

add trunk-group 1		Page 1 of 21	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: y	
Group Name: McLeodUSA	PSTN Access	COR: 1	TN: 1 TAC: 101
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Signaling Group: 1	
		Number of Members: 10	

**Figure 11: PSTN Trunk Group Form – Page 1**

On Page 3 of the **Trunk Group** form:

- set the **Numbering Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

change trunk-group 1	Page 3 of 21
TRUNK FEATURES	
ACA Assignment? n	Measured: none
	Maintenance Tests? y
Numbering Format: public	
	Prepend '+' to Calling Number? n
	Replace Unavailable Numbers? n

**Figure 12: PSTN Trunk Group Form – Page 3**

On page 4 of the **Trunk Group** form:

- set the **Telephone Event Payload Type** to *101*. This is the value McLeodUSA recommends for their service.

change trunk-group 4	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling Number? n	
Send Transferring Party Information? n	
Service Provider Network Call Redirection? n	
Telephone Event Payload Type: 101	

**Figure 13: PSTN Trunk Group Form – Page 4**

Now, configure the *OPS Trunk Group* form as shown in **Figure 14** using the **add trunk-group** command. In this case the trunk group number chosen is 2. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a mnemonic **Group Name**.
- Specify an available trunk access code (TAC).
- Set the **Service Type** field to *tie*.
- Specify the *inbound* signaling group associated with this trunk group in the **Signaling Group** field as previously specified in **Figure 10**.
- Specify the **Number of Members** supported by this SIP trunk group.

Note that one trunk member is required for each call between a non-SIP endpoint and McLeodUSA. Calls involving a SIP endpoint and McLeodUSA will use two trunk members for the duration of the call.

```

add trunk-group 2                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 1                                     Group Type: sip          CDR Reports: y
  Group Name: McLeodUSA SIP In                       COR: 1          TN: 1          TAC: 102
    Direction: two-way                               Outgoing Display? n
    Dial Access? n                                   Night Service:
    Queue Length: 0
  Service Type: tie                                   Auth Code? n
                                               Signaling Group: 2
                                               Number of Members: 10

```

**Figure 14: OPS Trunk Group Form – Page 1**

On Page 3 of the **Trunk Group** form:

- Set the **Numbering Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

```

change trunk-group 2                                     Page 3 of 21
TRUNK FEATURES
    ACA Assignment? n                               Measured: none
                                               Maintenance Tests? y
                                               Numbering Format: public
                                               Prepend '+' to Calling Number? n
                                               Replace Unavailable Numbers? n

```

**Figure 15: PSTN Trunk Group Form – Page 3**



## Step 7: Configure Calling Party Number Information

Use the **change public-unknown-numbering** command to configure Avaya Communication Manager to send the full calling party number to the far-end.

In this case, all stations with a 5-digit extension beginning with 6 should send the calling party number 732-85x-xxxx when an outbound call uses SIP trunk group #1 (*this is the PSTN trunk group specified in Step 6*). This calling party number will be sent to the far-end in the SIP “From” header. Note that the number used here is just an example, in a real deployment you would use the DID, (Direct Inward Dial), numbers assigned from McLeodUSA.

**Figure 16** shows the use of the **change public-unknown-numbering** command to implement this rule.

change public-unknown-numbering 0										Page	1 of	2
NUMBERING - PUBLIC/UNKNOWN FORMAT												
Ext		Trk		CPN		Total		CPN		Total		
Len	Code	Grp(s)		Prefix		Len	Len	Code	Grp(s)	Prefix	CPN	CPN
5	6	1		73285		10						Len

**Figure 16: Numbering Public/Unknown Format Form**

## Step 8: Automatic Route Selection for Outbound Calls

In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to the McLeodUSA SIP Trunking solution to a PTSN destination.

Use the **change dialplan analysis** command to add **9** as a feature access code (**fac**).

change dialplan analysis							Page 1 of 12		
DIAL PLAN ANALYSIS TABLE									
							Percent Full: 3		
Dialed	Total	Call		Dialed	Total	Call	Dialed	Total	Call
String	Length	Type		String	Length	Type	String	Length	Type
1	3	dac							
7	4	ext							
8	4	ext							
9	1	fac							
*	3	fac							
#	3	fac							

**Figure 17: Change Dialplan Analysis Form**

Use the **change feature-access-codes** command to specify **9** as the access code for outside dialing.

change feature-access-codes		Page 1 of 7
FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:		
Abbreviated Dialing List2 Access Code:		
Abbreviated Dialing List3 Access Code:		
Abbreviated Dial - Prgm Group List Access Code:		
Announcement Access Code: *03		
Answer Back Access Code:		
Attendant Access Code:		
Auto Alternate Routing (AAR) Access Code:		
<b>Auto Route Selection (ARS) - Access Code 1: 9</b>		Access Code 2:
Automatic Callback Activation:		Deactivation:
Call Forwarding Activation Busy/DA: *10	All: *11	Deactivation: #10
Call Park Access Code:		
Call Pickup Access Code:		
CAS Remote Hold/Answer Hold-Unhold Access Code:		
CDR Account Code Access Code:		
Change COR Access Code:		
Change Coverage Access Code:		
Contact Closure Open Code:	Close Code:	
Contact Closure Pulse Code:		

**Figure 18: Feature Access Codes Form**

Use the **change ars analysis** command to configure the route pattern selection rule based upon the number dialed following the dialed digit “9”. In this sample configuration, the PSTN numbers dialed are all in the form 1AAANNXXXXX (A= Area Code, N=[2-9], X=[0-9]). If the area code (AAA) is 732, the call is to be routed to a route pattern containing the SIP trunk groups used for McLeodUSA. Note that further administration of ARS is beyond the scope of these Application Notes but discussed in References [1] and [2].

change ars analysis 173		Page 1 of 2
ARS DIGIT ANALYSIS TABLE		
Location: all		Percent Full: 3
Dialed	Total	Route
String	Min Max	Pattern
173	11 11	1
		fnpa
		n

**Figure 19: ARS Analysis Form**

Use the **change route-pattern** command to define the SIP trunk group included in the route pattern that ARS selects. In this configuration, route pattern 1 will be used to route calls to trunk group 1 (the SIP trunk created in **Step 6, Figure 11**).

change route-pattern 1														Page 1 of 3	
Pattern Number: 1 Pattern Name: To PSTN															
SCCAN? n Secure SIP? n															
Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC															
No Mrk Lmt List Del Digits QSIG Intw															
1:	1	0		1										n	user
2:														n	user
3:														n	user
4:														n	user
5:														n	user
6:														n	user
BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM No. Numbering LAR															
0 1 2 3 4 W Request Dgts Format Subaddress															
1:	y	y	y	y	y	n	n				rest				none
2:	y	y	y	y	y	n	n				rest				none
3:	y	y	y	y	y	n	n				rest				none
4:	y	y	y	y	y	n	n				rest				none
5:	y	y	y	y	y	n	n				rest				none
6:	y	y	y	y	y	n	n				rest				none

**Figure 20: Route Pattern Form**

Note that additional entries need to be added to the ARS table for additional numbering plan options such as 411 information or 911 emergency services.

### Step 9: Configure Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper extension(s).

The incoming digits sent in the INVITE message from McLeodUSA are manipulated as necessary to route calls to the proper extension on Avaya Communication Manager. Note that this step assumes that the McLeodUSA subscriber has already received their DID numbers from McLeodUSA.

In the examples used in these Application Notes, the incoming DID numbers provided by McLeodUSA do not have a direct correlation to the internal extensions assigned within Avaya Communication Manager. Thus all incoming called number digits are deleted and replaced by the assigned extension number.

To create a fully mapped extension number as shown in **Figure 21**:

- Open the **Incoming Call Handling Treatment** form for the *PSTN* SIP trunk group configured in **Figure 11**, in this case Trunk Group 1.
- For each extension assigned a DID number from McLeodUSA, enter **10** into the **Called Len** and **Del** fields, and the entire **10 digit DID number** into the **Called Number** field. Enter the desired Avaya Communication Manager extension number into the **Insert** field.

change inc-call-handling-trmt trunk-group 1					Page 1 of 30
INCOMING CALL HANDLING TREATMENT					
Service/ Feature	Called Len	Called Number	Del	Insert	
public-ntwrk	10	7133433751	10	60003	
public-ntwrk	10	7133433752	10	61004	
public-ntwrk	10	7133433754	10	60001	

**Figure 21: Incoming Call Handling Treatment – Full Extension Mapping**

If the customer's extension numbering aligns with the DID numbers (i.e., the final DID digits match the extension), it is not necessary to define an entry for each DID number. Assuming a PBX dial plan that used the 5 digit extensions 60000 thru 61999 and assuming McLeodUSA provided DID numbers of 732-626-0000 thru 9999, the incoming number translation would be done similar to **Figure 22**. Note that the Called Number entry in this case represents the common matching portion applicable to all incoming numbers. Thus 732626 matches all numbers in the assigned DID block from McLeodUSA.

change inc-call-handling-trmt trunk-group 2					Page 1 of 30
INCOMING CALL HANDLING TREATMENT					
Service/ Feature	Called Len	Called Number	Del	Insert	
<b>tie</b>	<b>10</b>	<b>732626</b>	<b>5</b>		

**Figure 22: Incoming Call Handling Treatment – Simple Extension Mapping**

### Step 10: Save Avaya Communication Manager Changes

Enter "save translation" to make the changes permanent.

## 3.2 96xx SIP Endpoint Configuration

This section describes the administration of 96xx SIP telephones and requires the preceding SIP Trunk configuration to have been completed. SIP telephones are optional and not required to use the McLeodUSA SIP Trunking solution.

### Step 1: Assign a Station

The first step in adding an off-PBX station (OPS) for Avaya SIP telephones registered with Avaya SIP Enablement Services is to assign a station as shown in **Figure 23**. This example uses an Avaya One X 9630 Deskphone.

Using the **add station** command from the SAT:

- Set the station **Type** to the value “9630”.
- Enter “X” in the **Port** field to indicate station administration without port hardware.
- Enter a **Name** for the station that will be displayed.
- The **Security Code** is left blank for SIP OPS extensions.

The remaining fields are configured per normal station administration that is beyond the scope of these Application Notes. Note that the Class of Restrictions (**COR**) and Class of Service (**COS**) will govern the features and call restrictions that apply to this station.

add station 60004		Page 1 of 5
STATION		
Extension: 60004	Lock Messages? n	BCC: 0
Type: 9630	Security Code:	TN: 1
Port: X	Coverage Path 1:	COR: 1
Name: 9630 SIP	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
Speakerphone: 2-way	Message Lamp Ext: 60004	
Display Language: english	Mute Button Enabled? y	
Survivable GK Node Name:	Button Modules: 0	
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
Customizable Labels? y		

**Figure 23: Station Administration – Page 1**

On Page 2 of the **Station** form,

- Set the **Restrict Last Appearance** value to ‘n’ on phones that have 3 or fewer call appearances to maintain proper SIP conference and transfer operation. Setting the **Restrict Last Appearance** value to ‘y’ reserves the last call appearance for outbound calls. Certain SIP conference and transfer features will not function properly if a third appearance is not available for incoming calls.

add station 60004		Page 2 of 5
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Auto Select Any Idle Appearance? n	
LWC Activation? y	Coverage Msg Retrieval? y	
LWC Log External Calls? n	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Redirect Notification? y	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? n	<b>Restrict Last Appearance? n</b>	
Active Station Ringing: single	EMU Login Allowed? n	
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed		
Multimedia Mode: enhanced		
MWI Served User Type: sip-adjunct	Display Client Redirection? n	
	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
	Multimedia Early Answer? n	
	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 60004	Always Use? n IP Audio Hairpinning? n	

**Figure 24: Station Administration – Page 2**

On Page 4 of the **Station** form, configure at least 3 call appearances under the Button Assignments section for the SIP telephone as shown in **Figure 25**.

add station 60004		Page 4 of 5
STATION		
SITE DATA		
Room:	Headset? n	
Jack:	Speaker? n	
Cable:	Mounting: d	
Floor:	Cord Length: 0	
Building:	Set Color:	
ABBREVIATED DIALING		
List1:	List2:	List3:
BUTTON ASSIGNMENTS		
1: call-appr	5:	
2: call-appr	6:	
3: call-appr	7:	
4:	8:	

**Figure 25: Station Administration – Page 4**

A similar number of call appearances should be configured on the SIP Telephone which is beyond the scope of these Application Notes. The parameters to administer call appearances (and many other settings) are described in Reference [6].

## Step 2: Configure Off-PBX Station Mapping

The second step of configuring an off-PBX station is to configure the **Off-PBX Telephone** form so that calls destined for a SIP telephone at the enterprise site are routed to Avaya SIP Enablement Services, which will then route the call to the SIP telephone.

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 26**:

- Specify the **Station Extension** of the SIP endpoint.
- Set the **Application** field to *OPS*.
- Set the **Phone Number** field to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured on Avaya SIP Enablement Services also match the extensions of the corresponding AWOH (Administration Without Hardware) stations on Avaya Communication Manager. However, this is not a requirement.
- Set the **Trunk Selection** field to 2, which is the number assigned to the *OPS* SIP trunk group used to route the call to the SIP station. This trunk group number was previously defined in **Figure 14**.
- Set the **Configuration Set** value. In these Application Notes, Configuration Set 1 uses the default values of the Configuration Set form.

change off-pbx-telephone station-mapping 60004					Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION					
Station Extension	Application	Dial Prefix	Phone Number	Trunk Selection	Configuration Set
60004	OPS	-	60004	2	1

**Figure 26: Stations with Off-PBX Telephone Integration – Page 1**

On Page 2, set the **Call Limit** field to the maximum number of calls that may be active simultaneously at the station. In this example, the call limit is set to '3', which corresponds to the number of call appearances configured on the station form. Accept the default values for the other fields.

change off-pbx-telephone station-mapping 60000					Page 2 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION					
Station Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	
60000	3	both	all	both	

**Figure 27: Stations with Off-PBX Telephone Integration – Page 2**

**Step 3: Repeat for each SIP Phone**

Repeat Steps 1 and 2 for each SIP phone to be added.

**Step 4: Save Avaya Communication Manager Changes**

Enter “save translation” to make the changes permanent.

### **3.3 Configuration of Non G.729a SIP Endpoints**

As discussed in **Section 3.1, Step 4**, the preferred codec for McLeodUSA’ SIP Trunking Solution is G.729a. However, the Avaya 46xx SIP phones do not support the G.729a codec. Instead, they support G.729b. Because of this G.729 codec incompatibility, the voice path could be disrupted when the phone is configured to Shuffle or use the Direct-IP feature. To avoid this, separate network regions can be configured for these phones which have inter region codec values established between the network region that interfaces McLeodUSA’ SIP Trunking devices and the network region that the phones reside in such that a non G.729 codec type is used. The details for this configuration are beyond the scope of this document, please refer to the Avaya Communication Manager Network Region Configuration Guide [9].



## 4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services (SES). Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. It is assumed that Avaya SIP Enablement Services software and the license file have already been installed on Avaya SIP Enablement Services. During the software installation, the **initial\_setup** script is run on the Linux shell of the server to specify the IP network properties of the server along with other parameters. For additional information on these installation tasks, refer to [4].

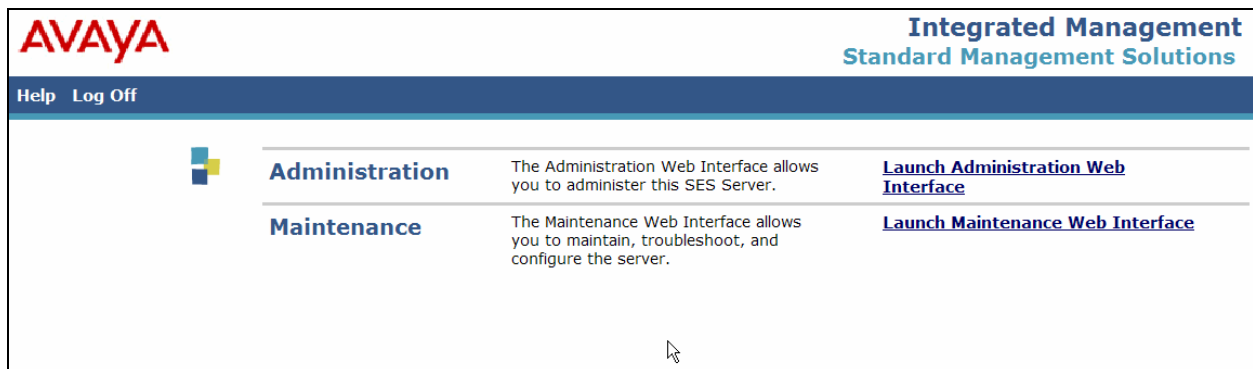
This section is divided into two parts: **Section 4.1** provides the steps necessary to configure a SIP trunk to McLeodUSA's SIP Trunking solution. **Section 4.2** provides the steps necessary to complete the administration for optional SIP endpoints.

### 4.1 SIP Trunking to McLeodUSA

#### Step 1: Log in to Avaya SIP Enablement Services

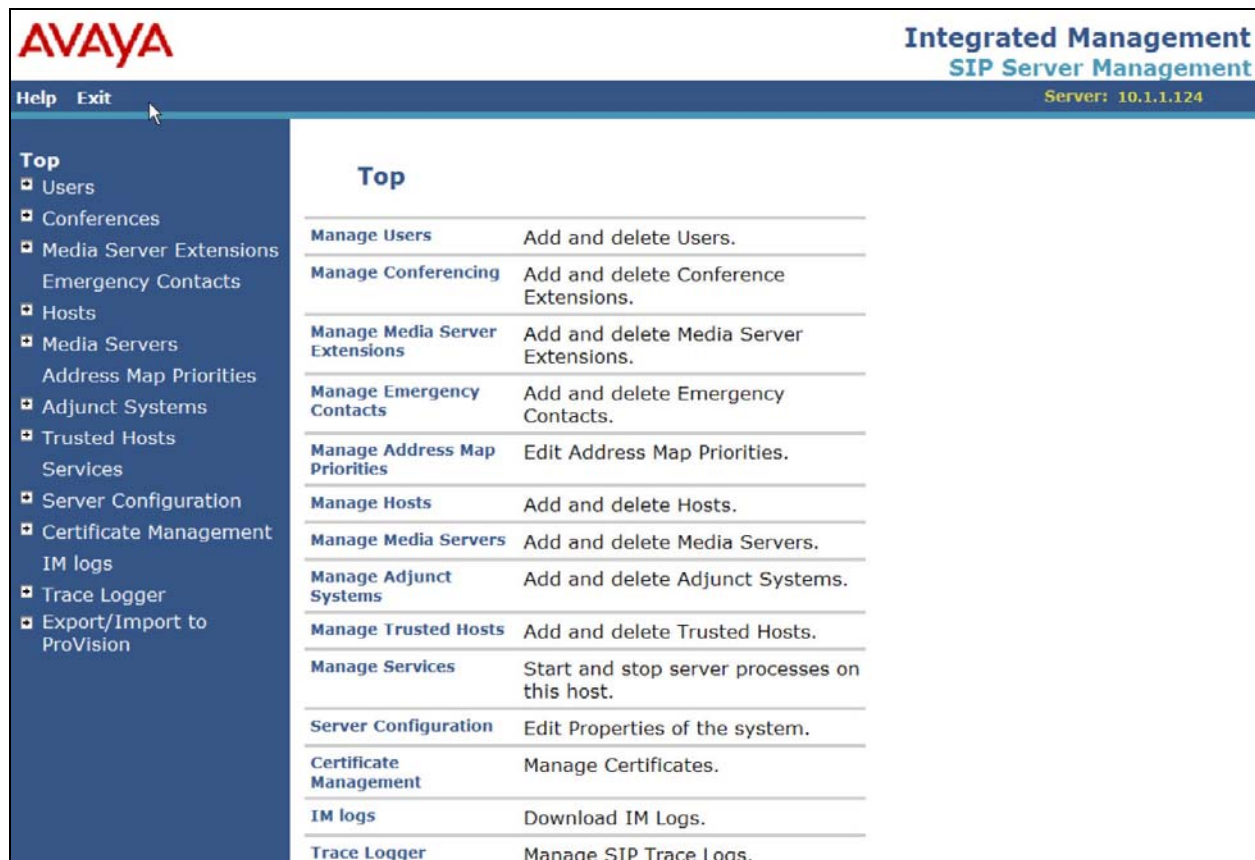
Access the SES Administration web interface, by entering **http://<ip-addr>/admin** as the URL in an Internet browser, where *<ip-addr>* is the IP address of Avaya SIP Enablement Services server.

Log in with the appropriate credentials and then select the *Launch Administration Web Interface* link from the main screen as shown in **Figure 28**.



**Figure 28 - Avaya SES Main Screen**

The SES administration home screen shown in **Figure 29** will be displayed.



**Figure 29: Avaya SES Administration Home Page**

## Step 2: Verify System Properties

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. This screen displays the SES version and the network properties entered via the **initial\_setup** script during the installation process.

In the **System Properties** screen,

- Verify the **SIP Domain** name assigned to Avaya SIP Enablement Services.
- Verify the **License Host** field. This is the host name, the fully qualified domain name, or the IP address of the SES that is running the WebLM application and has the associated license file installed. This entry should be set to **localhost** unless the WebLM server is not co-resident with this server.
- If changes were necessary in the **System Properties** screen, click the **Update** button.

AVAYA

Integrated Management  
SIP Server Management

Help Exit

Server: 10.1.1.124

Top

Users

Conferences

Media Server Extensions

Emergency Contacts

Hosts

Media Servers

Address Map Priorities

Adjunct Systems

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Admin Accounts

License

IM Log Settings

SNMP Configuration

Certificate Management

IM logs

Trace Logger

Export/Import to ProVision

View System Properties

SES\_Version

SES-4.0.0.0-033.6

System Configuration

simplex

Host Type

home/edge

SIP Domain\*

east.devcon.com

Note that the DNS domain is: east.devcon.com

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

License Host\*

localhost

Management System

Access Login

Management System

Access Password

DiffServ/TOS Parameters

Call Control PHB Value\*

46

802.1 Parameters

Priority Value\*

6

Network Properties

Local IP

10.1.1.124

Local Name

ses\_eh.east.devcon.com

Logical IP

10.1.1.124

Logical Name

ses\_eh.east.devcon.com

Gateway IP Address

10.1.1.97

Redundant Properties

Management Device

SAMP

Fields marked \* are required.

Update

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Figure 30: System Properties

### Step 3: Verify the Avaya SES Host Information

Verify the Avaya SES Host information using the **Edit Host** page. In these Application Notes the Avaya SES **Host Type** is a combined **home/edge**. This means that the both the McLeodUSA SIP Trunking Solution and Avaya Communication Manager are contacting the same SES. Display the **Edit Host** page (**Figure 31**) by following the **Hosts** link in the left navigation pane and then clicking on the **Edit** option under the **Commands** section of the **List Hosts** screen.

On the **Edit Host** screen:

- Verify that the IP address of this combined SES Home/Edge server is in the **Host IP Address** field.
- Do not modify the **DB Password** or **Profile Service Password** fields. If these fields are changed, exit the form without using the **Update** button. These values must match the values entered during the SES installation; incorrect changes may disable the SES.
- Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols**.
- Verify that **TLS** is selected as the **Link Protocol**.
- For **Outbound Proxy**, see **Step 6**.
- Default values for the remaining fields may be used.
- Click the **Update** button only if changes are necessary. Otherwise exit the **Edit Host** page by selecting the **Top** link on the left navigation bar.

The screenshot displays the 'Edit Host' configuration page within the Avaya Integrated Management SIP Server Management interface. The page title is 'Integrated Management SIP Server Management' with the server IP '10.1.1.124' shown. A left-hand navigation menu includes options like 'Top', 'Users', 'Conferences', 'Media Server Extensions', 'Emergency Contacts', 'Hosts', 'Media Servers', 'Adjunct Systems', 'Trusted Hosts', 'Services', 'Server Configuration', 'Certificate Management', 'IM logs', 'Trace Logger', and 'Export/Import to ProVision'. The main configuration area contains the following fields and settings:

- Host IP Address\***: 10.1.1.124
- DB Password\***: [Redacted]
- Profile Service Password\***: [Redacted]
- Host Type**: home/edge
- Parent**: none
- Listen Protocols**: ☒ UDP ☒ TCP ☒ TLS
- Link Protocols**: ☐ UDP ☐ TCP ☒ TLS
- Access Control Policy (Default)**: ☐ Allow All ☒ Deny All
- Emergency Contacts Policy**: ☒ Allow ☐ Deny
- Minimum Registration (seconds)**: 300
- Registration Expiration Timer (seconds)\***: 86400
- Line Reservation Timer (seconds)\***: 30
- Outbound Routing**: ☒ Internal ☒ External
- Allowed From**: [Empty field]
- OutboundProxy**: 10.2.2.10
- Port**: 5060
- Outbound Direct Domains**: [Empty list box]
- Default Ringer Volume\***: 5
- Default Ringer Cadence**: 2
- Default Receiver Volume\***: 5
- Default Speaker Volume\***: 5
- VMM Server Address**: [Empty field]
- VMM Server Port**: 5005
- VMM Report Period**: 5

A red circle highlights the **Update** button at the bottom of the form. A note at the bottom states: 'Fields marked \* are required.'

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**Figure 31: Edit Host**

#### Step 4: Add Avaya Communication Manager as Media Server

Under the **Media Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site. This will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager.

In the **Add Media Server** screen, enter the following information:

- A descriptive name in the **Media Server Interface** field (e.g., SP8710-SignalGroup1). Select IP address of the home SES server in the **Host** field as specified in **Figure 31**.
- Select **TLS** (Transport Link Security) for the **Link Type**. TLS provides encryption at the transport layer. TLS is the only link protocol that is supported for SIP trunking with Avaya Communication Manager.
- Enter the IP address of the Avaya S8710 Server CLAN board in the **SIP Trunk IP Address** field. (Note: This may be the IP address of the media server processor in smaller Avaya Communication Manager configurations such as an Avaya S8300 Server using an Avaya G350 Media Gateway.)
- After completing the **Add Media Server** screen, click on the **Add** button.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The left sidebar contains a navigation menu with options like Users, Conferences, Media Server Extensions, Emergency Contacts, Hosts, Media Servers, Adjunct Systems, Trusted Hosts, Services, and Server Configuration. The main content area is titled 'Add Media Server Interface' and contains the following fields and options:

- Media Server Interface Name\***: Text input field containing 'SP8710-SignalGroup1'.
- Host**: Dropdown menu showing '10.1.1.124'.
- SIP Trunk** section:
  - SIP Trunk Link Type**: Radio buttons for TCP and TLS (selected).
  - SIP Trunk IP Address\***: Text input field containing '10.1.1.112'.
- Media Server** section:
  - Media Server Admin Address**: Text input field.
  - Media Server Admin Login**: Text input field.
  - Media Server Admin Password**: Text input field.
  - Media Server Admin Password Confirm**: Text input field.
- SMS Connection Type**: Radio buttons for SSH and Telnet.
- A note at the bottom states: 'Fields marked \* are required.'
- The **Add** button is circled in red.

Figure 32: Add Media Server

## Step 5: Specify Address Maps to Media Servers

Incoming calls arriving at Avaya SIP Enablement Services are routed to the appropriate Avaya Communication Manager for termination services. This routing is specified in a Media Server Address Map configured on Avaya SIP Enablement Services.

This routing compares the Uniform Resource Identifier (URI) of an incoming INVITE message to the pattern configured in the Media Server Address Map, and if there is a match, the call is routed to the designated Avaya Communication Manager. The URI usually takes the form of *sip:user@domain*, where *domain* can be a domain name or an IP address. Patterns must be specific enough to uniquely route incoming calls to the proper destination if there are multiple Avaya Communication Manager systems supported by the Avaya SES server.

In these Application Notes, only incoming calls from the PSTN require a media server address map entry. Calls originated by Avaya SIP telephones configured as OPS are automatically routed to the proper Avaya Communication Manager by the assignment of an Avaya Media Server extension to that phone. Address map definitions for SIP endpoints not assigned a media server extension and connections to multiple service providers are beyond the scope of these Application Notes.

For McLeodUSA's SIP Trunking solution, the *user* portion of the SIP URI will contain the 10 digit value specified for the incoming direct inward dialed telephone number. An example of a SIP URI in an INVITE message received from McLeodUSA would be:

sip:7133433752@10.1.1.124;user=phone;npdi=yes

The user portion in this case is the 10 digit DID number "7133433752". The strategy used to define the media server address maps will be to create a pattern that matches the DID numbers assigned to the customer by McLeodUSA. The SES will forward the messages with matching patterns to the appropriate CLAN interface of the S8710 media server.

To configure a **Media Server Address Map**:

- Select **Media Servers** in the left pane of the Administration web interface. This will display the **List Media Servers** screen.
- Click on the **Map** link associated with the appropriate media server to display the **List Media Server Address Map** screen.
- Click on the **Add Map In New Group** link. The screen shown in **Figure 33** is displayed. The **Host** field displays the name of the media server that this map applies to.
- Enter a descriptive name in the **Name** field
- Enter the regular expression to be used for the pattern matching in the **Pattern** field. In this configuration, the DID numbers provided by McLeodUSA are 713-343-3751 thru 3754. (*Note: The full string is not visible in Figure 33 due to the text box size*) The pattern specification (without the double quotes) for DID numbers assigned is: `^sip:713343375[1234]`. This means that URIs beginning with "sip: 713343375" followed by the digits 1, 2, 3 and 4 will match the pattern and be routed to the interface defined for S8710-CLAN. Appendix B provides a detailed description of the syntax for address map patterns.



- Click the **Add** button once the form is completed.

**AVAYA** Integrated Management SIP Server Management  
Server: 10.1.1.124

Help Exit

**Top**

- Users
- Conferences
- Media Server Extensions
  - Emergency Contacts
- Hosts
- Media Servers
  - Address Map Priorities
- Adjunct Systems
- Trusted Hosts
- Services
- Server Configuration

**Add Media Server Address Map**

Name\* McLeodUSA713343

Pattern\* sip:713343375[1234]

Replace URI ☒

Fields marked \* are required.

**Add**

**Figure 33: Media Server Address Map**

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 34**.

**AVAYA** Integrated Management SIP Server Management  
Server: 10.1.1.124

Help Exit

**Top**

- Users
- Conferences
- Media Server Extensions
  - Emergency Contacts
- Hosts
  - List
  - Migrate Home/Edge
- Media Servers
  - List
  - Add
  - Address Map Priorities
- Adjunct Systems
- Trusted Hosts
- Services

**List Media Server Address Map**

Commands	Name	Commands	Contact
Edit Delete	McLeodUSA713343		
Edit Delete	sip:\${user}@10.1.1.112:5061;transport=tls		

Add Another Map Add Another Contact Delete Group

Add Map In New Group

**Figure 34: List Media Server Address Map**

Note that after the first **Media Server Address Map** is added, the **Media Server Contact** is created automatically. For the **Media Server Address Map** added in **Figure 33**, the following contact was created:

sips:\$(user)@10.1.1.112:5061;transport=tls

The contact specifies the IP address of the CLAN and the transport protocol used to send SIP signaling messages. The incoming DID number sent in the user part of the original request URI is substituted for \$(user).

#### **Step 6: Specify Outbound Proxy to McLeodUSA**

Outbound PSTN calls are directed by Avaya Communication Manager automatic route selection (ARS) according to the customer's network design guidelines. These guidelines determine what types of outgoing calls should be sent to McLeodUSA's SIP Trunking solution. The ARS routing decisions (for trunk group selection) will be customer specific and are beyond the scope of these notes.

Avaya SIP Enablement Services then uses the Outbound Proxy setting to route the call to McLeodUSA's SIP proxy. The Outbound Proxy is a setting which is used to route calls from the local SIP domain of the SES to a foreign domain. It can be thought of as a default route for all calls that are destined for end point outside the local SIP domain.

To configure the Outbound Proxy feature:

- Expand the **Hosts node** in the left pane of the Administration web interface and click on the **List** button. This will display the **List Hosts** screen.
- Click on the **Edit** button for the Host. This will display **Edit Host** screen as shown in **Figure 35**.
- Enter the IP address of the McLeodUSA SIP Proxy in the **OutboundProxy** field. For the testing described in this document, 10.2.2.10 was used.
- Enter the port number 5060 in the **Port** field. (5060 is the well known port number of the SIP protocol).
- Select the **UDP** option for the transport layer protocol.



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Help   Exit

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License

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SNMP Configuration

Certificate Management

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Trace Logger

Export/Import to ProVision

Edit Host

Host IP Address\*

10.1.1.124

DB Password\*

.....

Profile Service Password\*

.....

Host Type

home/edge

Parent

none

Listen Protocols

☒ UDP   ☒ TCP   ☒ TLS

Link Protocols

☐ UDP   ☐ TCP   ☒ TLS

Access Control Policy (Default)

☐ Allow All   ☒ Deny All

Emergency Contacts Policy

☒ Allow   ☐ Deny

Minimum Registration (seconds)

300

Registration Expiration Timer (seconds)\*

86400

Line Reservation Timer (seconds)\*

30

Outbound Routing

☒ Internal   ☒ External

Allowed From

OutboundProxy

10.2.2.10

Port

5060

☒ UDP   ☐ TCP   ☐ TLS

Outbound Direct Domains

Default Ringer Volume\*

5

Default Ringer Cadence

2

Default Receiver Volume\*

5

Default Speaker Volume\*

5

VMM Server Address

VMM Server Port

5005

VMM Report Period

5

Fields marked \* are required.

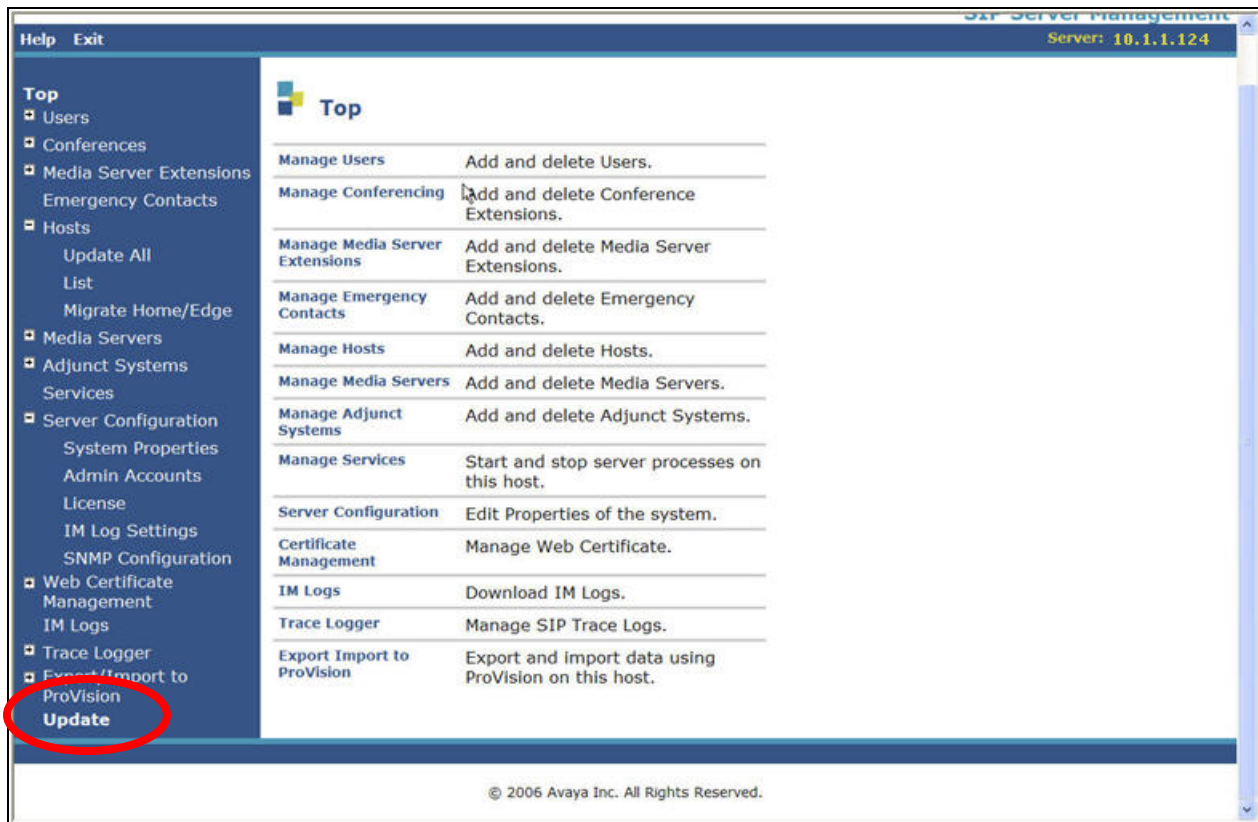
Update

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**Figure 35: Edit Host Screen**

### Step 7: Save the Changes

After making changes within Avaya SES, it is necessary to commit the database changes using the **Update** link that appears when changes are pending. Perform this step by clicking on the **Update** link found in the bottom of the blue navigation bar on the left side of any of the SES Administration screens as shown in **Figure 36**.



**Figure 36: Update Following SES Administrative Changes**

### Step 9: Specify the McLeodUSA SIP Proxy as a Trusted Host

The final step to complete the SIP trunk administration on Avaya SES is to designate the IP address of McLeodUSA SIP Gateway as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address.<sup>1</sup> If multiple SIP proxies are used, the IP address of each SIP proxy must be added as a trusted host.

<sup>1</sup> Note, if the trusted host step is not done, authentication challenges to incoming SIP messages (such as INVITES and BYEs) will be issued by the SES. This may cause call setup to fail, active calls to be disconnected after timeout periods, and/or SIP protocol errors.

To configure a trusted host:

- Telnet or SSH to the Avaya SES Linux shell using the administrative login and password.
- Enter the following trustedhost command at the Linux shell prompt:

```
trustedhost -a 10.2.2.10 -n 10.1.1.124 -c McLeodUSA
```

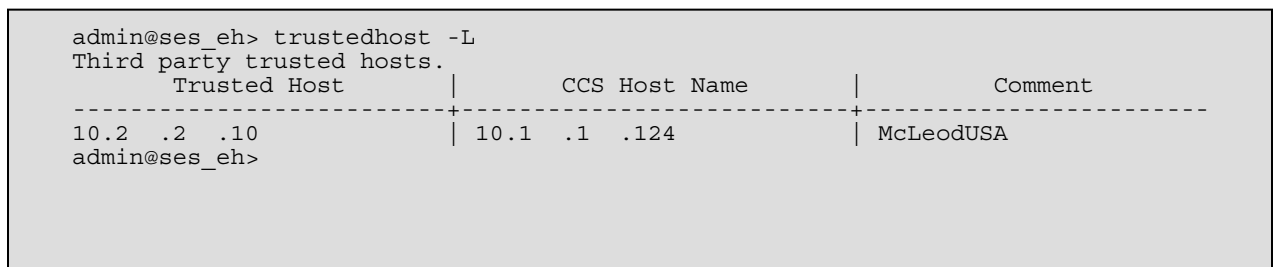
The `-a` argument specifies the address to be trusted; `-n` specifies the SES host name; `-c` adds a comment. Once the entry is made, a DNS lookup is automatically invoked to resolve the FQDN, (Fully Qualified Domain Name). The resolved address is held in an expanded table not visible to the user.

- Use the following trustedhost command to verify the entry is correct:  
`trustedhost -L`

**Figure 37** illustrates the results of the trustedhost commands.<sup>2</sup>

- Complete the trusted host configuration by returning to the main Avaya SES Administration web page and again clicking on the **Update** link as shown in **Figure 36**.

If the **Update** link is not visible, refresh the page by selecting **Top** from the left hand menu. Note this step is required even though the trusted host was configured via the Linux shell.



```
admin@ses_eh> trustedhost -L
Third party trusted hosts.
  Trusted Host      | CCS Host Name      | Comment
-----|-----|-----
10.2 .2 .10        | 10.1 .1 .124      | McLeodUSA
admin@ses_eh>
```

**Figure 37: Configuring a Trusted Host**

<sup>2</sup> For completeness, the `-d` argument allows the trust relationship to be deleted. For, example,

```
trustedhost -d 10.2.2.10 -n 10.1.1.124
```

removes the trust relationship added above.

## 4.3 Configuration for SIP Telephones

This section provides very basic instructions for completing the administration necessary to support the optional Avaya SIP telephones. Additional features such as the use of mnemonic addressing and instant messaging are also supported by Avaya SES but are beyond the scope of these Application Notes. Also, please refer back to **Section 3.3** for G.729a support considerations.

### Step 1: Add a SIP User

Create the SIP user record as follows:

- In Avaya SES administration, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- In the **Add User** screen, enter the extension of the SIP endpoint in the **Primary Handle** field.
- Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
- In the **Host** field, select the Avaya SIP Enablement Services server hosting the domain (10.1.1.124) for this user. Enter the **First Name** and **Last Name** of the user.
- To associate a media server extension with this user, select the **Add Media Server Extension** checkbox. Calls from this user will always be routed through Avaya Communication Manager over the SIP trunk for origination services.
- Press the **Add** button. This will cause a confirmation screen to appear.
- Press **Continue** on the confirmation screen.

The screenshot shows the 'Add User' form in the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, 'Integrated Management SIP Server Management', and 'Server: 10.1.1.124 4'. A left navigation pane lists various system components, with 'Users' expanded to show 'Add'. The main form area contains the following fields and options:

- Primary Handle\***: Text input with '60004' entered.
- User ID**: Empty text input.
- Password\***: Password input field with masked characters.
- Confirm Password\***: Password input field with masked characters.
- Host\***: Dropdown menu showing '10.1.1.124'.
- First Name\***: Text input with 'John' entered.
- Last Name\***: Text input with 'Doe' entered.
- Address 1**: Empty text input.
- Address 2**: Empty text input.
- Office**: Empty text input.
- City**: Empty text input.
- State**: Empty text input.
- Country**: Empty text input.
- Zip**: Empty text input.
- Add Media Server Extension**: A checkbox that is checked.

Below the fields, a note states: 'Fields marked \* are required.' At the bottom left of the form is an **Add** button.

**Figure 38: Add User**

## Step 2: Specify Corresponding Avaya Communication Manager Extension

The SIP phone handle must now be associated with the corresponding extension on Avaya Communication Manager. After adding the User above, the **Add Media Server Extension** screen will appear.

- In the **Add Media Server Extension** screen, enter the **Extension** configured on the media server, shown in **Figure 23**, for the OPS extension on Avaya Communication Manager previously defined in **Section 3.2**. Usually, the media server extension and the user extension are the same (recommended) but it is not required.
- Select the **Media Server** assigned to this extension.
- Click on the **Add** button.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo and the text 'Integrated Management SIP Server Management' with the server IP '10.1.1.124'. A navigation menu on the left lists various functions like Users, Conferences, Media Server Extensions, and Hosts. The main content area is titled 'Add Media Server Extension' and contains a form for adding a media server extension for user 60004. The form has two input fields: 'Extension' with the value '60004' and 'Media Server' with a dropdown menu showing 'SP8710-clan1a03'. Below the form is a red circle highlighting the 'Add' button.

**Figure 39: Add Media Server Extension**

## Step 3: Repeat for Each SIP User

Repeat Steps 1 and 2 for each SIP user.

## 5. McLeodUSA Services Configuration

The McLeodUSA SIP Trunking solution offers the growing number of business customers with IP-based phone systems, many different configurations that can be scaled based on specific voice, data and configuration needs.

Customers can choose to terminate service directly to their IP phone system, to an external router provided by the customer, or to an Integrated Access Device (IAD) with the standard and premium options provided by McLeodUSA. If the customer requires an Ethernet hand-off to their equipment, the McLeodUSA standard or premium equipment option is required.

The standard product solution is available in bandwidths ranging from 1.5 Mbps up to 12.0 Mbps in single DS1 increments. Solutions with McLeodUSA-provided equipment are limited to 6.0 Mbps. Each solution includes a minimum of 6 simultaneous call paths, with the ability to scale to the customer's specific business need in increments of 6 call paths. The standard configuration includes 30 call paths for each 1.5 Mbps of bandwidth, although up to 48 call paths for each 1.5 Mbps of bandwidth can be accommodated if "non-voice" bandwidth is not required.

For customers selecting the standard or premium equipment option, McLeodUSA also includes access to 3 FXS ports (per 1.5MB) on the IAD for customer use, typically for analog device support. Customers must identify all numbers that are associated with analog devices so that voice compression is disabled.

All telephone numbers associated with the McLeodUSA SIP Trunking solution must be in a McLeodUSA-served rate center within approximately 1,000 central offices in a 20-state footprint. This includes numbers that will be ported to the service or numbers requested as new in conjunction with the installation of service. Specific availability is available at [www.mcleodusa.com](http://www.mcleodusa.com)

The product solution is sold by McLeodUSA field sales executives and by other Direct and Indirect Sales channels such as Master Agents, their sub Agents, Lead Agents and through Referral Agents. Companies interested in becoming a McLeodUSA Business Partner may visit [www.mcleodusa.com/IndirectSales.do](http://www.mcleodusa.com/IndirectSales.do) for more information.

## 6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between McLeodUSA's SIP Trunking Solution and an Avaya IP Telephony Solution. This section covers the general test approach and the test results.

### 6.1. General Test Approach

A simulated enterprise site consisting of an Avaya IP telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available SIP Trunking solution provided by McLeodUSA. This allowed the enterprise site to use SIP trunking for PSTN calling.

The following features and functionality were covered during the SIP trunking interoperability compliance test:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by McLeodUSA.
- Outgoing calls from the enterprise site were completed via McLeodUSA to the PSTN destinations.
- Calls using SIP, H.323, digital and analog endpoints supported by the Avaya IP telephony solution.
- Various call types including: local, long distance, international, toll free, and directory assistance calls.
- Calls using G.729a and G.711mu codec.
- G.711 fax calls.
- DTMF tone transmission using RFC 2833 with successful Voice Mail/Vector navigation
- Telephone features such as hold, transfer, conference.
- Direct IP-to-IP media (also known as "shuffling") with SIP/H323 telephones.

## 6.2. Test Results

Interoperability testing of the sample configuration was completed with successful results.

The following minor issues described below were observed.

### 6.2.1 Call Forwarding Off-Net and EC500 features

#### **Issue Observed**

Call Forwarding OffNet and EC500 do not work when incoming SIP trunk call is then forwarded off-net via the SIP Trunk to McLeodUSA.

#### **Discussion/Workaround**

McLeodUSA authorizes calls into their network by examining the Calling Party Number (CPN) in the user portion of the From header and the Called Party Number in the user portion of the Request-URI of an incoming INVITE message. If neither the Calling Party Number nor the Called Party Number are DIDs residing on McLeodUSA SIP servers, the call will be rejected.

There are two features which can create this scenario.

The first is the OffNet Call Forwarding feature. The specific scenario occurs when a phone call is placed from a phone outside the Avaya Communication Manager SIP domain to a phone within the Avaya Communication Manager SIP domain. The phone within the Avaya Communication Manager SIP domain has Call Forwarding OffNet enabled such that it will forward the call back to McLeodUSA's SIP service for completion outside the Avaya Communication Manager SIP domain. To maintain the original identity or continuity of the call and to maintain backwards compatibility with RFC 2543 [13], the value of the From header is not changed to an approved McLeodUSA DID. Instead, it maintains the original caller's CPN.

The second feature is Extension to Cellular or EC500 (See [2] for a full description of the feature). As described above, if a phone call is placed from the PSTN to a phone in the Avaya Communication Manager SIP domain which has EC500 enabled and configured to call another phone outside the Avaya Communication Manager SIP domain, the From header is not changed to an approved McLeodUSA DID. Instead, it maintains the original caller's CPN.

Although Avaya does not modify the From header, Avaya Communication Manager does add a History-Info header to the INVITE request (as specified in RFC 4244 [12]) which does contain the CPN of the phone within the Avaya Communication Manager SIP domain that forwarded the call off-net. The purpose of the History-Info header is to provide "information as to how and why a call arrives at a specific application or user". In these OffNet Call Forwarding scenarios, it indicates that the call arrived at the McLeodUSA SIP servers because it was forwarded by a phone within the Avaya Communication Manager SIP domain. The CPN in the History-Info header contains a value that could be recognized by McLeodUSA, allowing the call to be successfully authorized instead of rejected.



However, McLeodUSA does not currently recognize the History-Info header and instead supports the Diversion header. The Diversion header is documented in the IETF Internet Draft draft-levy-sip-diversion-08. Both the Diversion header and the History-Info header were proposed to IETF to solve similar problems. Ultimately, the IETF selected the History-Info header mechanism and ratified it as a standard in RFC 4244 [12]. The Diversion header never became an IETF standard, and its Internet Draft expired on February 25<sup>th</sup>, 2005. For this reason, Avaya only supports the History-Info standard.

In the future, Avaya Communication Manager may support a feature that, when enabled, will allow the CPN in the From header to be overwritten locally. Although modifying the From header in this fashion technically violates SIP standards, it will allow better interoperability with McLeodUSA in the absence of mutual support for the History-Info header.

For Call Forwarding, McLeodUSA provides a workaround where the user can enable Call Forward on the McLeodUSA SIP servers using either feature codes or “Your Account” web interface. Please refer to **Appendix C** for detailed steps.

**Note:** In contrast to the call scenarios described above, if the original call is placed from a phone within the Avaya SIP Domain instead of from outside the Avaya SIP Domain, this issue does not exist because a McLeodUSA approved DID can be applied to as the CPN via the **Public Unknown Numbering** form.

## **6.2.2 MWI**

### **Issue Observed**

Communication Manager does not support MWI coming from ITSP's.

### **Discussion/Workaround**

When a message is left on McLeod's VOIP platform, MWIs are generated and sent to Communication Manager, Communication Manager rejects the MWIs. As a result the status of voice mail left on the network is not alerted to the users either via stutter tone or lamp lights on the phones.

The workaround is to enable Integrated Messaging for email notification of network voice mail.

## 7. Verification Steps

This section provides verification steps that may be performed in the field to verify that the SIP, H.323, digital and analog endpoints can place outbound and receive inbound PSTN calls through McLeodUSA.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
3. Verify that the user on the PSTN can terminate an active call by hanging up.
4. Verify that an endpoint at the enterprise site can terminate an active call by hanging up.

## 8. Support

For technical questions regarding a customer installation, please call the McLeodUSA toll-free number provided in the customer information packet. For general, non-technical questions, please contact McLeodUSA Customer Care at 1-800-593-1177.

## 9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya Communication Manager and Avaya SIP Enablement Services telephony solution to McLeodUSA's SIP Trunking solution. McLeodUSA's SIP Trunking solution is a Voice over IP solution for customers ranging from small businesses to large enterprises. SIP trunks use the Session Initiation Protocol (SIP) to connect private company networks to the public telephone network via converged IP access. It provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunk lines.

## 10. References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administrator Guide for Avaya Communication Manager*, February 2007, Issue 3.1, Document Number 03-300509.
- [2] *Feature Description and Implementation for Avaya Communication Manager*, February 2007, Issue 5, Document Number 555-245-205
- [3] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 4.0*, Feb 2007, Issue 10, Document Number 210-100-700.
- [4] *Installing and Administering SIP Enablement Services*, May 2007, Issue 4, Document Number 03-600768
- [5] *SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series, and S8700 series Media Server*, May 2007, Document Number 555-245-206.
- [6] *4600 Series IP Telephone LAN Administrator Guide*, October 2007, Document Number 555-233-507
- [7] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 1.5*, May 2007, Document Number 16-300698
- [8] *Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide Release 1.0*, May 2007, 16-601944
- [9] *Avaya Communication Manager Network Region Configuration Guide*, October 2005, 103244
- [10] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [11] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>
- [12] RFC 4244, *An Extension to the Session Initiation Protocol (SIP) for Request History Information*, <http://www.ietf.org/>
- [13] RFC 2544, *SIP: Session Initiation Protocol*, <http://www.ietf.org/>

## APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by McLeodUSA and the Avaya SIP network at the enterprise site. Customers may use these INVITE messages for comparison and troubleshooting purposes. Differences in these messages may indicate different configuration options selected.

### Sample SIP INVITE Message from McLeodUSA to Avaya SIP Enablement Services:

```
INVITE sip:7133433752@10.1.1.124:15020 SIP/2.0
From: "AVAYA INC C/O T"<sip:7328521637@englab1.mcleodusa.net>;tag=a242c740-13c4-47434a5b-1ec883cb-5271910c
To: "7133433752 7133433752"<sip:7133433752@10.1.1.124>
Call-ID: CXC-58-4a2eaea0-a242c740-13c4-47434a5b-1ec883cb-8f1885e@englab1.mcleodusa.net
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.2.2.10:5060;branch=z9hG4bK-32e-47434a5b-1ec883cb-6fd49a55
Supported: 100rel,timer
Accept: multipart/mixed,application/media_control+xml,application/sdp
Max-Forwards: 9
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
Min-SE: 60
Contact: <sip:7328521637@englab1.mcleodusa.net:5060;maddr=10.2.2.10;transport=UDP>
Content-Type: application/SDP
Content-Length: 283
```

```
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): 509740 1 IN IP4 10.2.2.10
Session Name (s): -
Connection Information (c): IN IP4 10.2.2.10
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 22010 RTP/AVP 18 0 8 101
Media Attribute (a):ptime:20
Media Attribute (a):fmtp:101 0-15
Media Attribute (a):fmtp:18 annexb=no
Media Attribute (a):rtptime:18 G729/8000
Media Attribute (a):rtptime:0 PCMU/8000
Media Attribute (a):rtptime:8 PCMA/8000
Media Attribute (a):rtptime:101 telephone-event/8000
Media Attribute (a):sendrecv
```

## Sample SIP INVITE Message from Avaya SIP Enablement Services to McLeodUSA:

INVITE sip:17328521637@englab1.mcleodusa.net SIP/2.0  
Call-ID: 80fa3796ff9fdc1a6c946da5f6500  
CSeq: 1 INVITE  
From: "9640 IP H.323" <sip:7133433752@east.devcon.com:5061>;tag=80fa3796ff9fdc1a5c946da5f6500  
Record-Route: <sip:10.1.1.124:5060;lr>,<sip:10.1.1.112:5061;lr;transport=tls>  
To: "17328521637" <sip:17328521637@englab1.mcleodusa.net>  
Via: SIP/2.0/UDP 10.1.1.124:5060;branch=z9hG4bK8383830303036666622ece.0,SIP/2.0/TLS  
10.1.1.112;psrrposn=2;received=10.1.1.112;branch=z9hG4bK80fa3796ff9fdc1a7c946da5f6500  
Content-Length: 230  
Content-Type: application/sdp  
Contact: "9640 IP H.323" <sip:7133433752@10.1.1.112:5061;transport=tls>  
Max-Forwards: 69  
User-Agent: Avaya CM/R014x.00.1.731.2  
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS  
History-Info: <sip:17328521637@englab1.mcleodusa.net>;index=1  
History-Info: "17328521637" <sip:17328521637@englab1.mcleodusa.net>;index=1.1  
Supported: 100rel,timer,replaces,join,histinfo  
Min-SE: 600  
Session-Expires: 600;refresher=uac  
P-Asserted-Identity: "9640 IP H.323" <sip:7133433752@east.devcon.com:5061>  
Route: <sip:10.2.2.10:5060;lr>

Session Description Protocol  
Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): - 1 1 IN IP4 10.1.1.112  
Session Name (s): -  
Connection Information (c): IN IP4 10.1.1.116  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 29904 RTP/AVP 18 0 8 101  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): fmp:18 annexb=no  
Media Attribute (a): rtpmap:0 PCMU/8000  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:101 telephone-event/8000

## APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the Avaya SES is a Linux regular expression used to match against the URI string found in the SIP INVITE message.

Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s).

In the pattern matching string used in the Avaya SES:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
  - A period `.` matches any character once (and only once).
  - An asterisk `*` matches zero or more of the preceding characters.
  - Square brackets enclose a list of any character to be matched. Ranges are designated by using a hyphen. Thus the expression `[12345]` or `[1-5]` both describe a pattern that will match any single digit between 1 and 5.
  - Curly brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' times. Thus `5{3}` matches '555' and `[0-9]{10}` indicates any 10 digit number.
  - The circumflex character `^` as the first character in the pattern indicates that the string must begin with the character following the circumflex.  
Putting these constructs together as used in this document, the pattern to match the SIP INVITE string for any valid 1+ 10 digit number in the North American dial plan would be:  
`^sip:1[0-9]{10}`

This reads as: "Strings that begin with exactly **sip:1** and having any 10 digits following will match.

A typical INVITE request below uses the shaded portion to illustrate the matching pattern.

INVITE sip:17325551638@proxy-McLeodUSA:5060;transport=udp SIP/2.0

## APPENDIX C: Instructions for Enabling Call Forwarding

There are two methods for enabling and disabling Call Forwarding features via the McLeodUSA SIP Trunking Service. The first method uses Feature Access Codes to enable and disable the features from the customer phone. The second method uses the McLeodUSA web portal to enable and disable the features.

### 1. Using Feature Access Codes (also known as \* codes)

#### Requirement

- The user must have the ability to invoke the Call forwarding features in the McLeodUSA call controller. Permissions to use these features are controlled by the Enterprise and/or Group administrator.
- The user must dial the Feature Access Code (FAC) from their direct extension or IP phone. The customer Avaya Communication Manager must be configured to pass the station level ANI in the SIP message to the McLeodUSA network. The McLeodUSA call controller uses the calling number to enable or disable the feature for the corresponding user.

#### Call Forwarding Feature Access Codes

The table below displays the list of valid Feature Access Codes for the Call Forwarding options available as part of the standard service.

Feature	Enable Feature	Disable Feature
Call Forwarding Always	*72	*73
Call Forwarding Busy	*90	*91
Call Forwarding No Answer	*92	*93

- **Call Forwarding Always** – This feature will automatically forward all calls to a number specified by the user.
  - To enable this feature:
    1. Dial **\*72**. Note: To eliminate post-dial delays, dial **\*72#**.
    2. The phone call will be directed to a media server, which will play the following announcement: *“Please enter the destination number for call forwarding always followed by the # key.”*
    3. The user should enter the appropriate number. The user must be sure to correctly type the number. It is recommended to always use a 10-digit number.

4. The following announcement will play: *“Your call forwarding always service has been activated successfully.”*
  - To disable this feature:
    1. Dial **\*73**. Note: To eliminate post-dial delays, dial **\*73#**.
    2. The following announcement will play: *“Your call forwarding always service has been deactivated successfully.”*
- Call Forwarding Busy – This feature will automatically forward all calls to a number specified by the user when the user phone is busy.
  - To enable this feature:
    1. Dial **\*90**. Note: To eliminate post-dial delays, dial **\*90#**.
    2. The phone call will be directed to a media server, which will play the following announcement: *“Please enter the destination number for call forwarding busy, followed by the # key.”*
    3. The user should enter the appropriate number. The user must be sure to correctly type the number. It is recommended to always use a 10-digit number.
    4. The following announcement will play: *“Your call forwarding busy service has been activated successfully.”*
  - To disable this feature:
    1. Dial **\*91**. Note: To eliminate post-dial delays, dial **\*91#**.
    2. The following announcement will play: *“Your call forwarding busy service has been deactivated successfully.”*
- Call Forwarding No Answer – This feature will automatically forward all calls to a number specified by the user when the user does not answer their phone after three rings.
  - To enable this feature:
    1. Dial **\*92**. Note: To eliminate post-dial delays, dial **\*92#**.
    2. The phone call will be directed to a media server, which will play the following announcement: *“Please enter the destination number for call forwarding no answer, followed by the # key.”*
    3. The user should enter the appropriate number. The user must be sure to correctly type the number. It is recommended to always use a 10-digit number.
    4. The following announcement will play: *“Your call forwarding no answer service has been activated successfully.”*
  - To disable this feature:
    1. Dial **\*93**. Note: To eliminate post-dial delays, dial **\*93#**.
    2. The following announcement will play: *“Your call forwarding no answer service has been deactivated successfully.”*

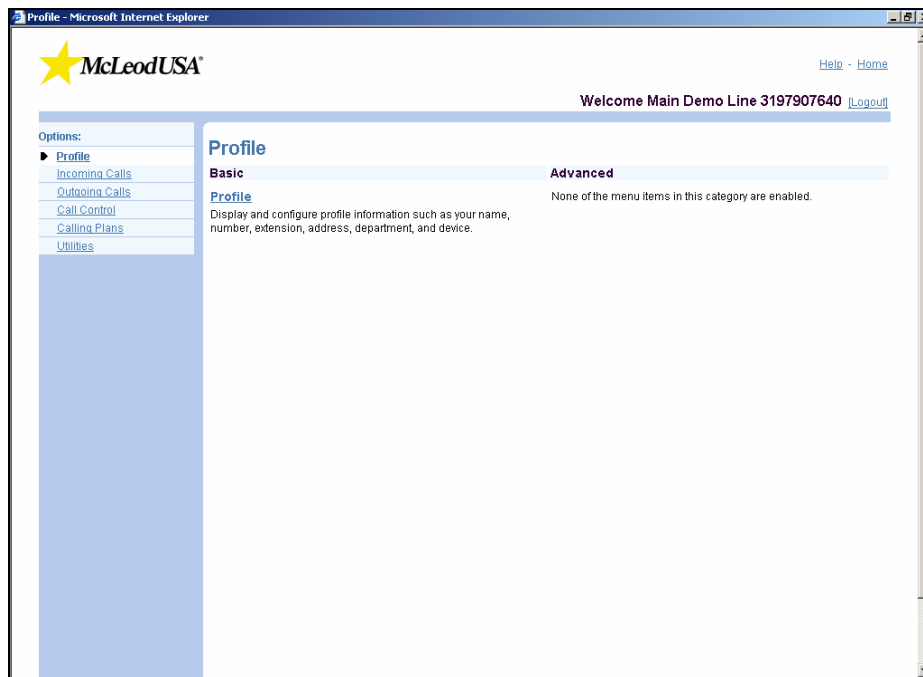


## 2. Using the Web Portal

### Requirement

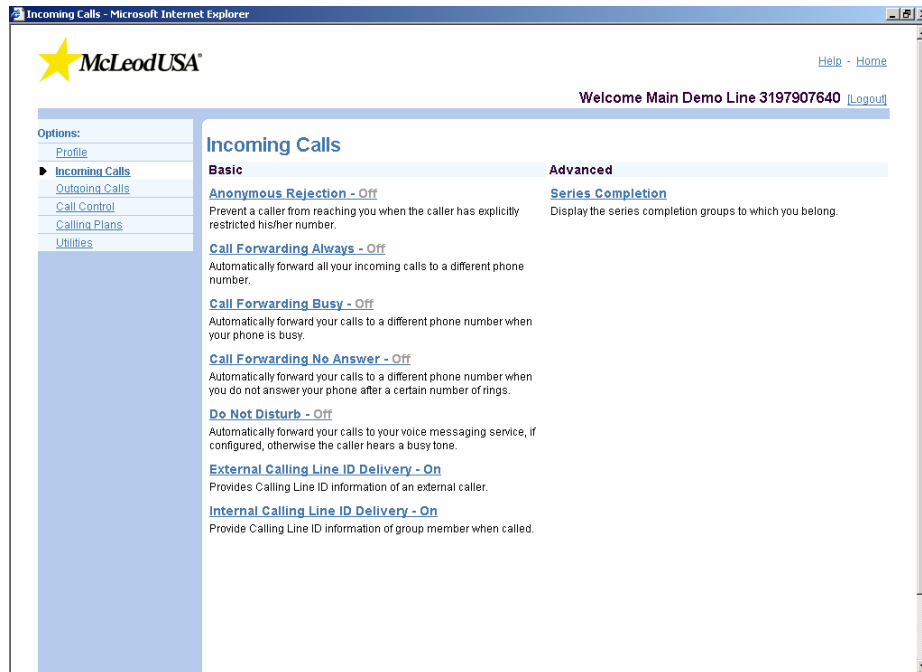
- The user must have the ability to invoke the Call forwarding features in the McLeodUSA call controller. Permissions to use these features are controlled by the Enterprise and/or Group administrator.
- The user must have the ability to log into the McLeodUSA web portal as a user. The Enterprise and/or Group administrator also controls the ability to access user information via the web portal. The web portal is accessible via the following link: <http://www.mcleodusa.com>.

After logging into the web portal as a user and entering the services section of the account, the system will bring up the initial screen showing the profile screen for the user.



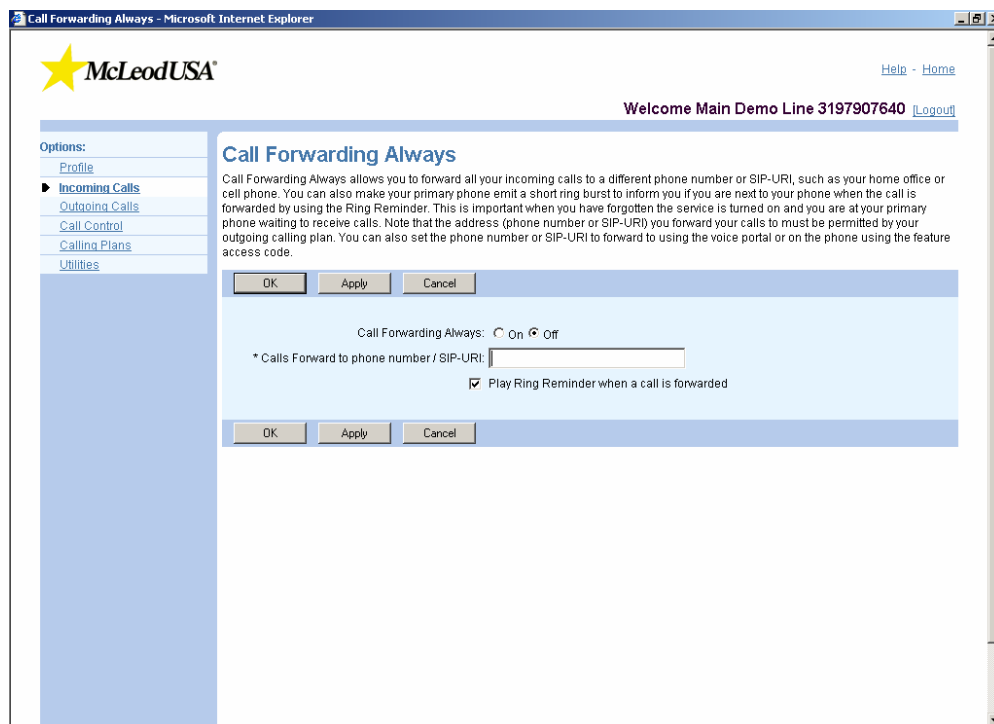
**Figure 1: User Profile Page**

To enable Call Forwarding features, click on the Incoming Calls link in the Options section on the left of the screen. This will bring up a list of all the services that can be applied to incoming calls for the user.



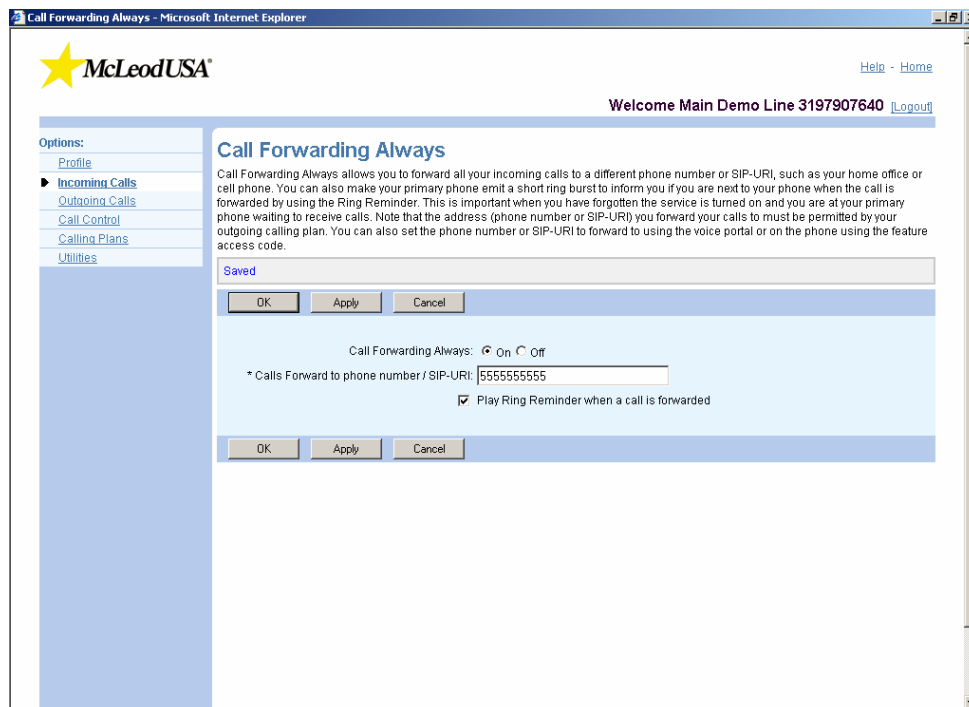
**Figure 2: Incoming Calls Services**

- **Call Forward Always** - To enable Call Forwarding Always, click on the Call Forwarding Always link. This will bring up the Call Forwarding Always page.



**Figure 3: Call Forwarding Always**

- To enable the feature:
  1. Click the On radio button.
  2. Type in the phone number in the phone number field. The user must be sure to correctly type the number without dashes or spaces. It is recommended to always use a 10-digit number. For example, when forwarding to 555-555-5555, the user should enter 5555555555.
  3. Click on the Apply button.
  4. When finished, click on OK.



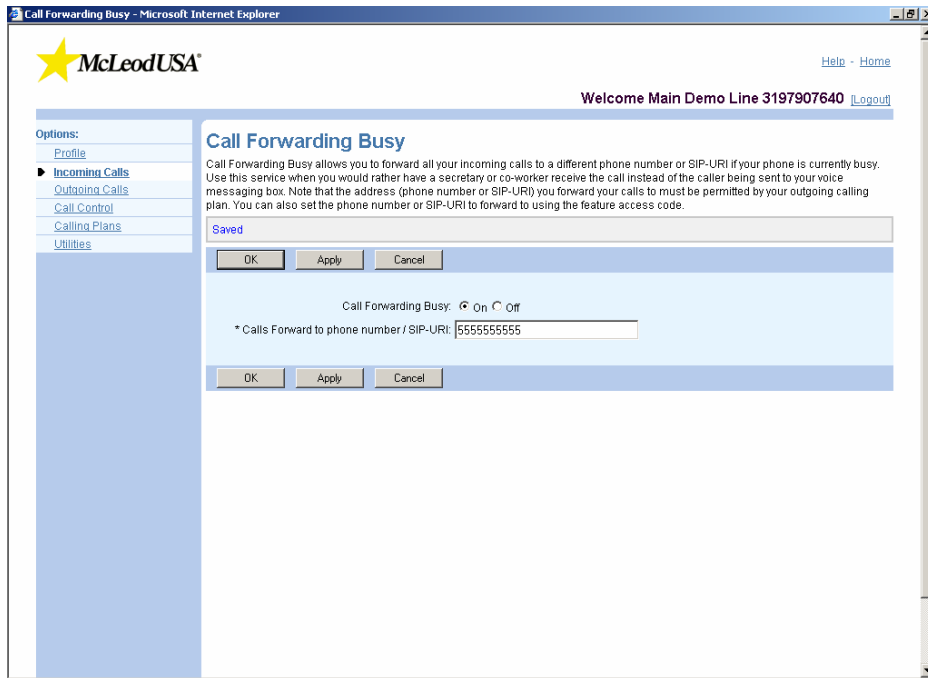
**Figure 4: Call Forwarding Always Enabled**

- To disable the feature:
  1. Click on the Off radio button.
  2. Click on the Apply button.
  3. When finished, click on OK.

Note: The user has the option of deleting the number from the phone number field. It is not necessary to do this.
- **Call Forward Busy** -To enable Call Forwarding Busy, click on the Call Forwarding Always link. This will bring up the Call Forwarding Busy page.

**Figure 5: Call Forwarding Busy**

- To enable the feature:
  1. Click the On radio button.
  2. Type in the phone number in the phone number field. The user must be sure to correctly type the number without dashes or spaces. It is recommended to always use a 10-digit number. For example, when forwarding to 555-555-5555, the user should enter 5555555555.
  3. Click on the Apply button.
  4. When finished, click on OK.



**Figure 6: Call Forwarding Busy Enabled**

- To disable the feature:
  1. Click on the Off radio button.
  2. Click on the Apply button.
  3. When finished, click on OK.

Note: The user has the option of deleting the number from the phone number field. It is not necessary to do this.
- **Call Forward No Answer** - To enable Call Forwarding No Answer, click on the Call Forwarding Always link. This will bring up the Call Forwarding No Answer page.

Call Forwarding No Answer - Microsoft Internet Explorer

McLeodUSA

Help - Home

Welcome Main Demo Line 3197907640 [Logout](#)

Options:

- Profile
- Incoming Calls
- Outgoing Calls
- Call Control
- Calling Plans
- Utilities

### Call Forwarding No Answer

Call Forwarding No Answer allows you to forward all your calls to a different phone number or SIP-URI when you do not answer your phone. Use this service when you would rather have a secretary or co-worker receive the call instead of the caller being sent to your voice messaging box if you miss a call.

OK Apply Cancel

Call Forwarding No Answer: ☐ On ☒ Off

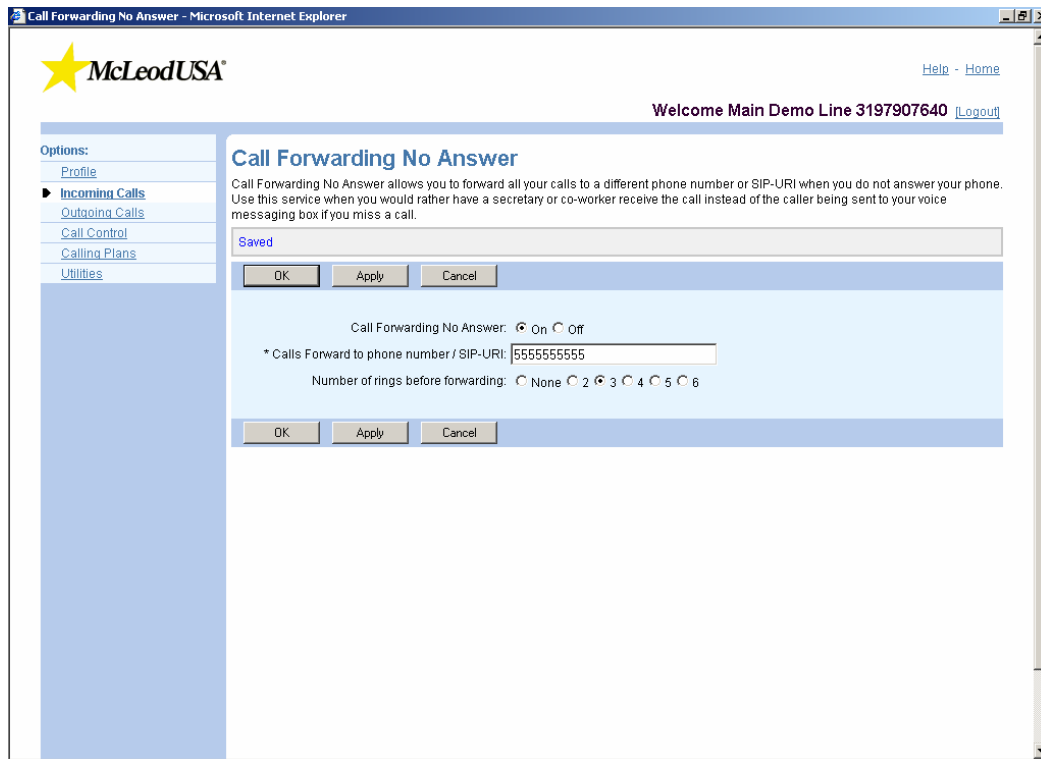
\* Calls Forward to phone number / SIP-URI:

Number of rings before forwarding: ☒ None ☐ 2 ☐ 3 ☐ 4 ☐ 5 ☐ 6

OK Apply Cancel

**Figure 7: Call Forwarding No Answer**

- To enable the feature:
  1. Click the On radio button.
  2. Type in the phone number in the phone number field. The user must be sure to correctly type the number without dashes or spaces. It is recommended to always use a 10-digit number. For example, when forwarding to 555-555-5555, the user should enter 5555555555.
  3. Specify the number of rings before forwarding. Options are None, 2, 3, 4, 5, or 6. *Note:* Optional rings can only be invoked via the web portal.
  4. Click on the Apply button.
  5. When finished, click on OK.



**Figure 8: Call Forwarding No Answer Enabled**

- To disable the feature:
  1. Click on the Off radio button.
  2. Click on the Apply button.
  3. When finished, click on OK.

Note: The user has the option of deleting the number from the phone number field. It is not necessary to do this.

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