



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

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# **Configuring Avaya Aura™ Session Manager with Avaya Aura™ Communication Manager Branch – Issue 1.0**

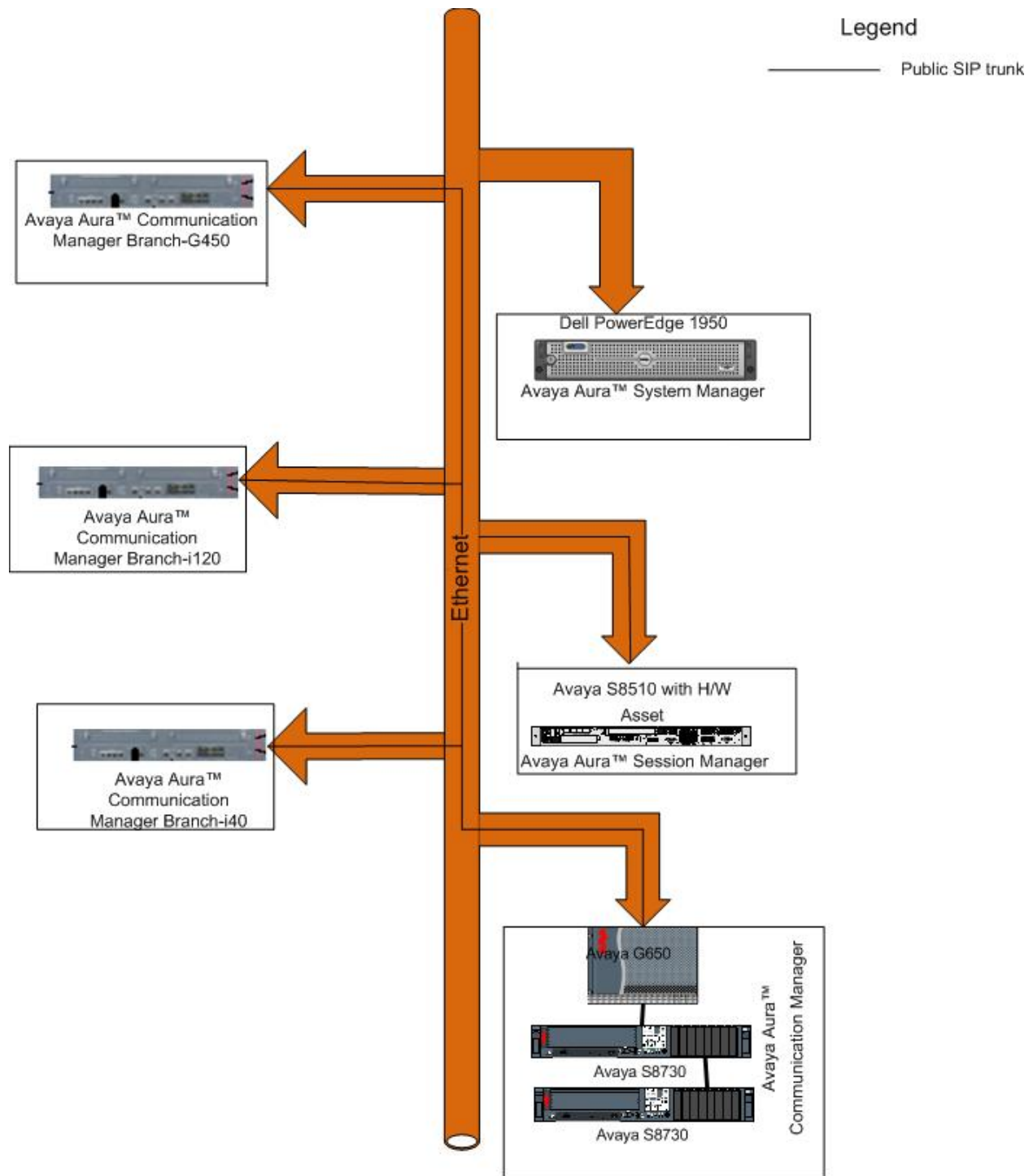
### **Abstract**

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Session Manager R5.2 to integrate with Avaya Aura™ Communication Manager Branch R 2.0 SP2 and Avaya Aura™ Communication Manager over Public SIP Trunk.

For the sample configuration, Avaya Aura™ Session Manager runs on an Avaya S8510 Server, Avaya Aura™ Communication Manager Branch on G450/ i120/i40 and Avaya Aura™ Communication Manager runs on G650 and Avaya S8730 servers. Testing was conducted via the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Session Manager **R5.2** to integrate with Avaya Aura™ Communication Manager Branch **R 2.0 SP2** over a Public SIP Trunk. The SIP trunk connects Avaya Aura™ Communication Manager Branch to Avaya Aura™ Session Manager, using its SM-100 (Security Module) network interface. The sample configuration includes two Avaya Aura™ Communication Manager Branch nodes with H.323 and SIP. All inter- Avaya Aura™ Communication Manager Branch calls are carried over public SIP trunks. Avaya Aura™ Session Manager supports inter- Avaya Aura™ Communication Manager Branch call routing based on the dialed number, ARS (Automatic Route Selection), dial pattern for the branch, and the extension number. Avaya Aura™ Session Manager is managed by Avaya Aura™ System Manager via the management network interface, and Avaya Aura™ Communication Manager Branch is administered by local device manager.



**Figure 1: Configuring Avaya Aura™ Session Manager with Avaya Aura™ Communication Manager Branch**

## 2. Equipment and Software Validated

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

Product / Hardware Platform	Version
Avaya Aura <sup>TM</sup> Session Manager <ul style="list-style-type: none"><li>• SM-100 Hardware</li><li>• Avaya S8510 Server</li></ul>	Session Manager 5.2.0.1 Build: 520017
Avaya Aura <sup>TM</sup> System Manager <ul style="list-style-type: none"><li>• Dell Power Edge 1950</li></ul>	System Manager 5.2.0.1 Build: 520017
Avaya Aura <sup>TM</sup> Communication Manager Branch - G450	Release 2.0 SP2.0.0.17
Avaya Aura <sup>TM</sup> Communication Manager Branch - i120	Release 2.0 SP2.0.0.17
Avaya Aura <sup>TM</sup> Communication Manager Branch - i40	Release 2.0 SP2.0.0.17
Avaya Aura <sup>TM</sup> Communication Manager <ul style="list-style-type: none"><li>• G650</li><li>• Avaya S8730</li></ul>	Release 5.2.1

## 3. Configure Avaya Aura<sup>TM</sup> Session Manager

### 3.1 Avaya Aura<sup>TM</sup> Session Manager Configuration

Follow the Network Routing Policy Administrator's Guide (<https://support.avaya.com/css/appmanager/public/support>) to set up Session Manager. The configuration below is required to set up a SIP trunk between Session Manager and Communication Manager Branch.

#### 3.1.1 Add SIP entity for Communication Manager Branch

Use the Network Routing Policy SIP Entities screen to administer SIP Entities. SIP Entities are all the network elements that act as the "SIP System". Enter the IP Address of the Communication Manager Branch to make it a SIP entity for Session Manager

SIP Entity Details - Microsoft Internet Explorer

Address: https://10.0.0.245/NRP/faces/pages/sipEntitiesDetails.xhtml?cid=1697

Welcome, **admin** Last Logged on at Dec. 14, 2009 5:21 PM  
Help | Log off

## AVAYA Avaya Aura™ System Manager 5.2

Home / Network Routing Policy / SIP Entities / SIP Entity Details

**SIP Entity Details** [Commit] [Cancel]

**General**

\* Name: ASM-A to G450-2

\* FQDN or IP Address: 10.0.1.53

Type: CM

Notes: ASM-A to G450-2

Adaptation: [v]

Location: [v]

Time Zone: Asia/Kolkata [v]

Override Port & Transport with DNS SRV: ☐

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

Credential name: [v]

Call Detail Recording: none [v]

**SIP Link Monitoring**

SIP Link Monitoring: Use Session Manager Configuration [v]

**Entity Links** [Add] [Remove]

1 Item Refresh Filter: Enable

<input type="checkbox"/>	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
<input type="checkbox"/>	avaya-asma [v]	TCP [v]	* 5060	ASM-A to G450-2 [v]	* 5060	<input checked="" type="checkbox"/>

Select: All, None ( 0 of 1 Selected )

\* Input Required [Commit] [Cancel]

### 3.1.2 Add entity link for Communication Manager Branch

Entity link connects two SIP entities, which enables Network Routing Policy and Session Manager to identify specific connection configuration (e.g. trusted hosts, outbound proxy, etc.) between two SIP entities.

The Trusted field means that the link between the two SIP entities is trusted.

**Entity Links**

15 Items | Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
<input type="checkbox"/>	ASMA-claudiaVP	avaya-asma	TCP	5060	claudiaVP	5060	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	ASMA-CM1NONims	avaya-asma	TCP	5070	CM1_NONIMS	5070	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	ASMA-COCM	avaya-asma	TCP	5070	COCM	5070	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	ASMA-IBCMims	avaya-asma	TCP	5060	IBCMims	5060	<input checked="" type="checkbox"/>	ims enabled
<input type="checkbox"/>	ASMA-IBCMnonims	avaya-asma	TCP	5070	IBCMnonims	5070	<input checked="" type="checkbox"/>	ims not enabled
<input type="checkbox"/>	ASMA-A to G450-1	avaya-asma	TCP	5060	ASM-A to G450-1	5060	<input checked="" type="checkbox"/>	ASM-A to G450-1
<input type="checkbox"/>	ASMA-A to G450-2	avaya-asma	TCP	5060	ASM-A to G450-2	5060	<input checked="" type="checkbox"/>	ASM-A to G450-2
<input type="checkbox"/>	ASMA-A to pusqa1120-BH1	avaya-asma	TCP	5060	ASM-A to pusqa1120-BH1	5060	<input checked="" type="checkbox"/>	ASM-A to pusqa1120-BH1
<input type="checkbox"/>	ASMA-to-pusqa140Ana	avaya-asma	TCP	5060	ASMA-to-pusqa140Ana	5060	<input checked="" type="checkbox"/>	ASMA-to-pusqa140Ana
<input type="checkbox"/>	ASMA-A to pusqa140-BRI	avaya-asma	TCP	5060	ASM-A to pusqa140-BRI	5060	<input checked="" type="checkbox"/>	ASM-A to pusqa140-BRI
<input type="checkbox"/>	ASMA-to-pusqa140-DS1	avaya-asma	TCP	5060	ASMA-to-pusqa140-DS1	5060	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	ASMC-CM3	avaya-asmc	TCP	5060	CM3 nonims	5060	<input checked="" type="checkbox"/>	ASMC-CM3nonims
<input type="checkbox"/>	ASMC-CM-FS	avaya-asmc	TCP	5060	CM-FS	5060	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	ASMC-IBVP	avaya-asmc	TCP	5060	IBVP	5060	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	MM-ASM	avaya-asma	TCP	5060	IB-MAS	5060	<input checked="" type="checkbox"/>	MM integration with ASM

Select : All, None ( 0 of 15 Selected )

### 3.1.3 Define Routing Policy

Session Manager can simply route the call based on the starting digit and the number of digits defined in the dial pattern. When a call starting with 122- xxxx reaches Session Manager, it will route the incoming call to Communication Manager Branch. Session Manager can also route calls based on different parameters (e.g. time of day, origination of the caller, SIP domain of the called party).

**Routing Policy Details - Microsoft Internet Explorer**

Address: <https://110.0.0.245/NRP/faces/pages/networkRoutingPolicyDetails.xhtml?cid=1700>

**Avaya Aura System Manager 5.2**

Home / Network Routing Policy / Routing Policies / Routing Policy Details

**Routing Policy Details** [Commit] [Cancel]

Warning: Time Gap found in TOD table coverage. If Gap exists then random routing behavior may occur. See View Gaps/Overlaps for details.

**General**

\* Name:

Disabled: ☐

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
ASM-A to G450-2	10.0.1.53	CM	ASM-A to G450-2

**Time of Day**

Add Remove View Gaps/Overlaps

0 Items Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
---------	------	-----	-----	-----	-----	-----	-----	-----	------------	----------	-------

**Dial Patterns**

Add Remove

1 Item Refresh Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
122	3	7	<input type="checkbox"/>	silpunelab.com	-ALL-	ASM-A to G450-2

Select: All, None ( 0 of 1 Selected )

**Regular Expressions**

Add Remove

Done Local intranet

### 3.1.4 Define Dial Pattern

The Network Routing Policy Dial Patterns screen is used to assign Location based Routing Policies to dial patterns.

The routing capability can be limited by setting a destination domain, so the call will only route to a SIP entity with the given domain. The location value below can limit the originating SIP entity, as defined in section 3.1.1 above.

Dial Pattern Details - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Address <https://10.0.0.245/NRP/faces/pages/dialPatternsDetails.xhtml?cid=1702>

**AVAYA** Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 14, 2009 5:21 PM  
[Help](#) | [Log off](#)

Home / Network Routing Policy / Dial Patterns / **Dial Pattern Details**

**Dial Pattern Details** [Commit](#) [Cancel](#)

**General**

\* Pattern:

\* Min:

\* Max:

Emergency Call: ☐

SIP Domain:

Notes:

**Originating Locations and Routing Policies**

[Add](#) [Remove](#)

1 Item | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	ASM-A to G450-2		<input type="checkbox"/>	ASM-A to G450-2	ASM-A to G450-2

Select : All, None ( 0 of 1 Selected )

**Denied Originating Locations**

[Add](#) [Remove](#)

0 Items | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

\* Input Required [Commit](#) [Cancel](#)

**Shortcuts**

- [Change Password](#)
- [Help for Dial Pattern Details fields](#)
- [Help for Location and Routing Policy Lists](#)
- [Help for Denied Location fields](#)
- [Help for Committing configuration changes](#)

Done Local intranet



## 4. Configure Avaya Aura™ Communication Manager Branch

### 4.1 Configure dial plan with users

Dial plan can only be set while initializing the Communication Manager Branch Edition and cannot be changed later on. The Dial Plan allows adding new users or stations. Consider 4 digit dial plan for reference.

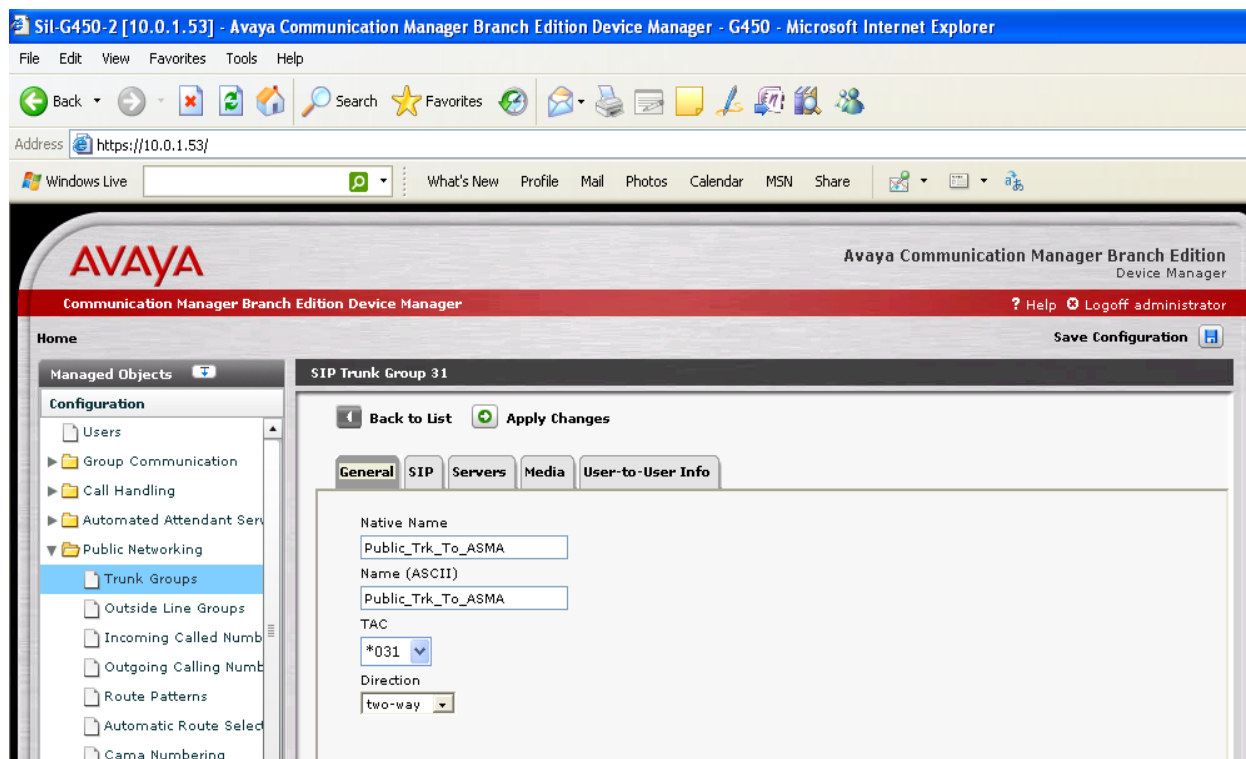
The screenshot shows the Avaya Communication Manager Branch Edition Device Manager web interface. The browser window title is "Sil-G450-2 [10.0.1.53] - Avaya Communication Manager Branch Edition Device Manager - G450 - Microsoft Internet Explorer". The address bar shows "https://10.0.1.53/". The interface has a red header bar with the Avaya logo and the text "Avaya Communication Manager Branch Edition Device Manager". Below the header, there is a navigation pane on the left with "Managed Objects" and "Configuration" sections. The "Users" link is selected under "Configuration". The main content area shows the "Users" page with buttons for "Add New User", "Remove User", and "Duplicate User". Below these buttons is a table with the following data:

Select	Extension	Name	Set Type	Port ID	IM Handle
<input type="checkbox"/>	4000	H323-G450-2	9640/9640G-H323	IP	
<input type="checkbox"/>	4001	SIP-G450-2	9620-SIP	IP	
<input type="checkbox"/>	4002	H323-G450-2-4002	9640/9640G-H323	IP	
<input type="checkbox"/>	4003	H.323-G450-2-4003	Softphone-H323	IP	
<input type="checkbox"/>	4004	4004-SIP	9630-SIP	IP	
<input type="checkbox"/>	4005	Abhi_SIP	9620-SIP	IP	

### 4.2 Configure Trunk group

#### 4.2.1 General settings

Create a SIP trunk from Communication Manager Branch to Session Manager



## 4.2.2 SIP settings

The check box for “Replace outgoing request-URI domain with selected server IP address” should be left unchecked, since Session Manager does not have URE functionalities for now. Checking this checkbox will display the selected server IP address instead of the domain name.

Add SIP domains below the SIP tab.

Far End:-

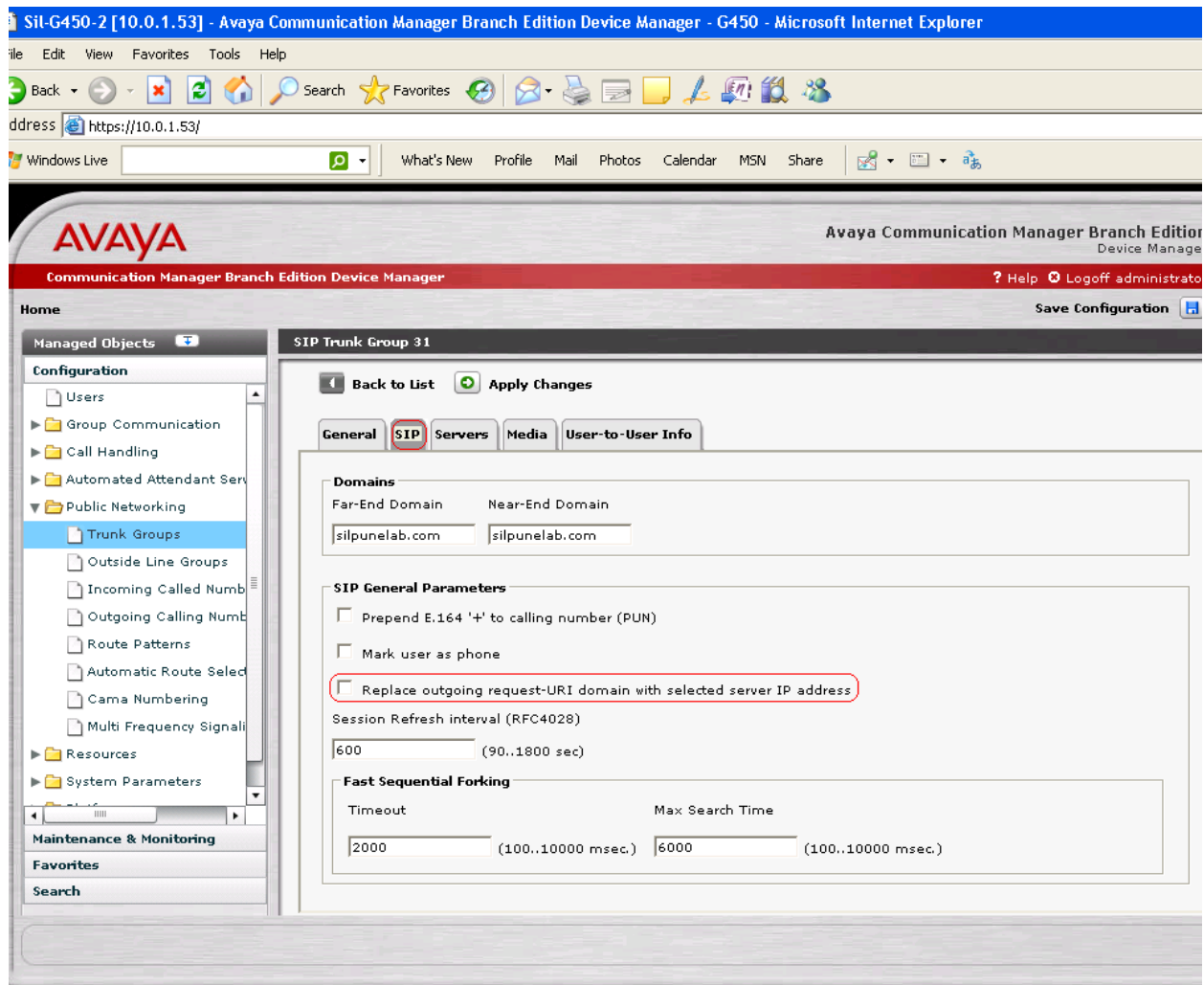
Far end SIP domain is a SIP domain configured at Session Manager. It can be set from Network Routing Policy=>SIP domains from session manager.

Near End:-

Near end SIP domain is a domain name configured at Communication Branch Manager. It can be set from

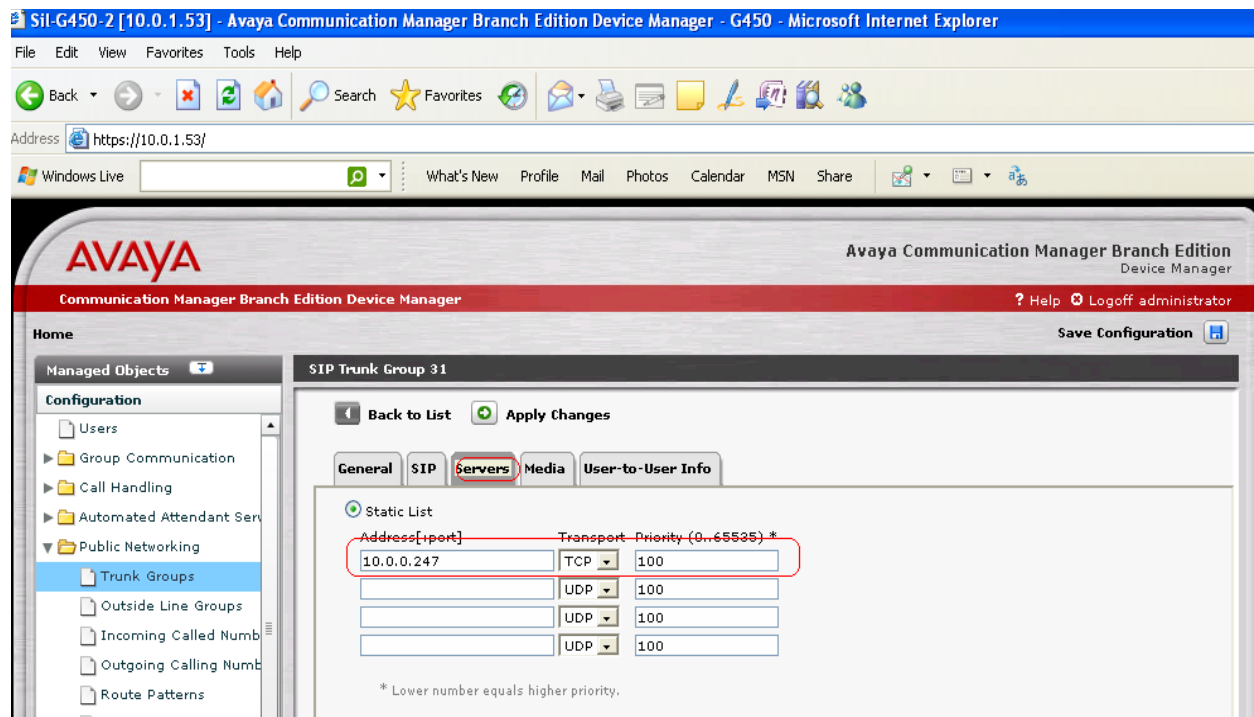
Configuration=>Platform=>Network Connection=>DNS tab.

-



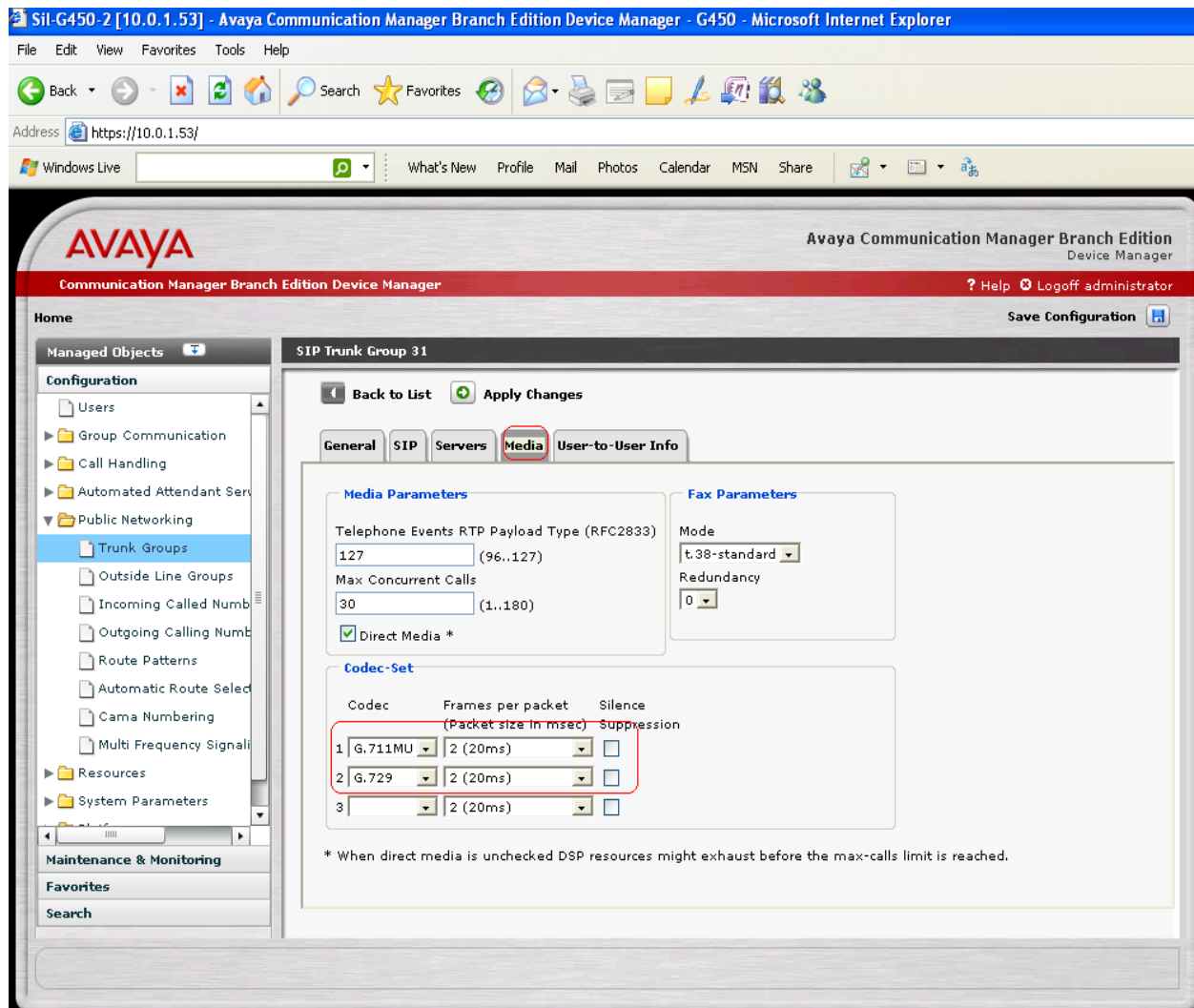
### 4.2.3 Servers settings

Enter the ip-address of Session Manager and select the Transport as 'TCP'. Let the priority be default, i.e. 100.



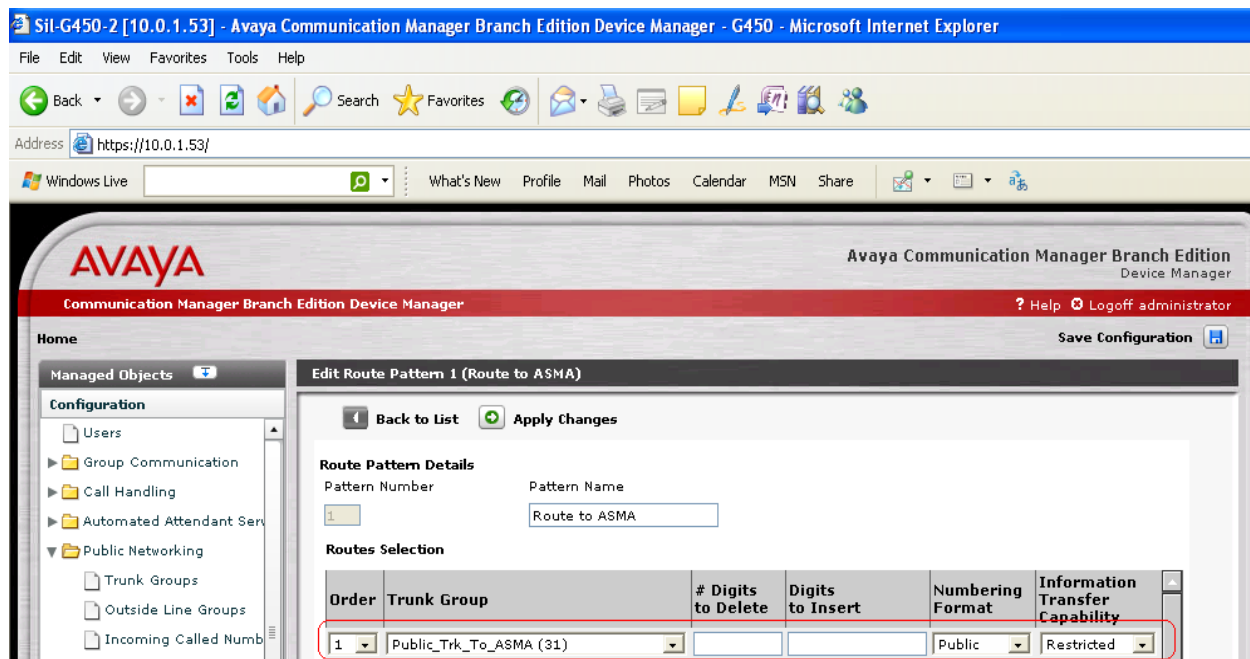
## 4.2.4 Media Settings

Select the codecs that will be used for calls on the SIP trunk to Session Manager. These are generic codecs widely supported by most of the Avaya endpoints.



## 4.3 Add Route Pattern

### 4.3.1 Add route pattern to use SIP trunk 31

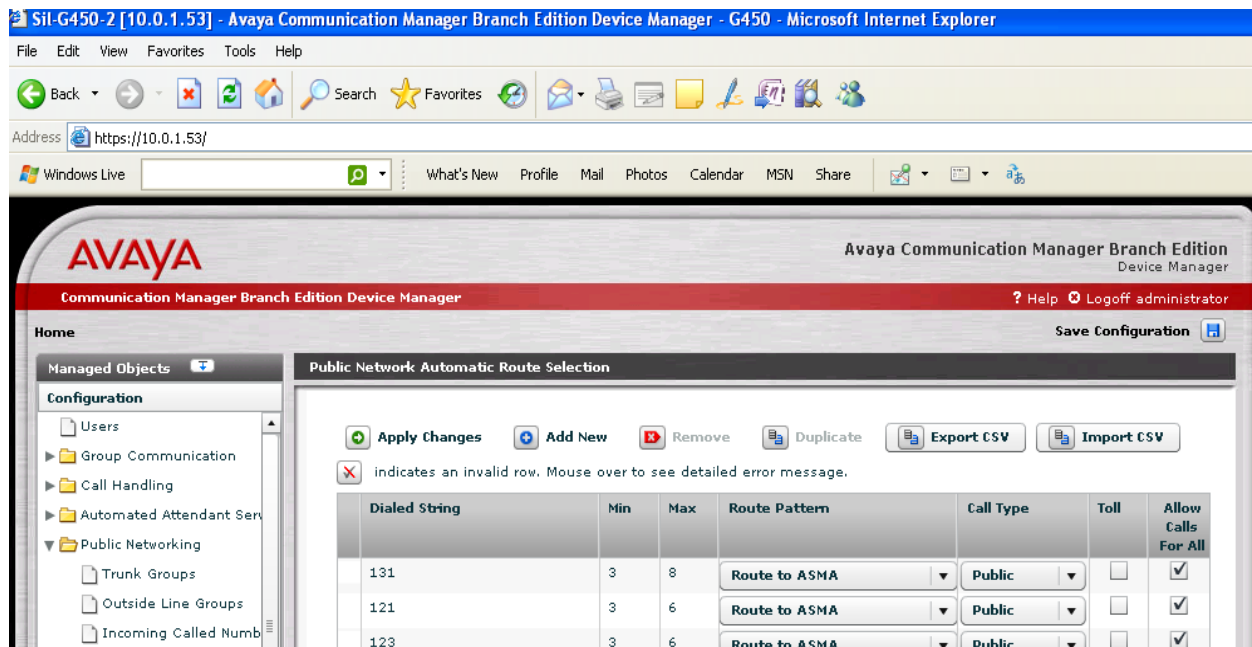


## 4.4 Configure Automatic Route Selection

Setup Automatic Route Selection to route the call from Communication Manager Branch to Session Manager over the public SIP trunk

The following guideline is used for designing the dial plan.

- For any given leading digit (0-9, \* and #), only one user will be assigned.
- All extensions within the system will be the same length. The length of extensions cannot be changed once set without erasing and re-installing the configuration, and may be preset before delivery.
- There must be at least one extension, at least one code for Feature Access Codes, at least one code for Trunk Access Codes.
- Typically, the customer will assign one single-digit code (the ARS) for the outside line FAC and one single-digit code for the inter-branch line FAC (the AAR). All the other FACs will be multiple digits, starting with "\*" and/or "#". Other schemes, however, are allowed.
- Outside line starting digit is always single digit.
- Inter-branch line starting digit is always single digit.



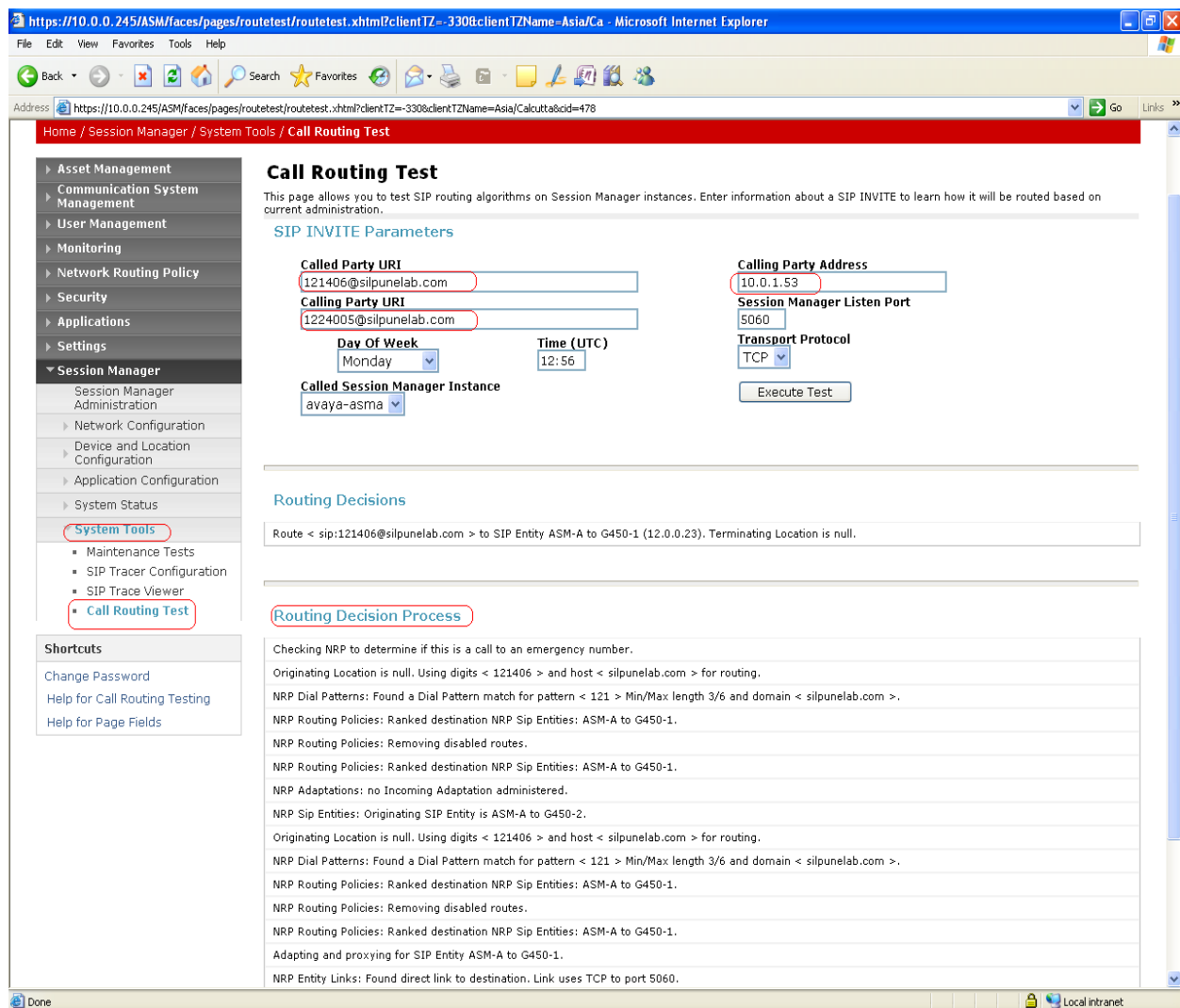
## 5. Verification Scenarios

Integrate two Communication Manager Branch devices with Session Manager over public SIP trunk as explained above. Use the following steps to verify inter-Communication Manager Branch calling.

### 5.1 Verification from Session Manager Side

#### 5.1.1 Use the Call Routing Test under Session Manager Tab -> System Tools

- Enter the Called party URI, Calling Party URI & the Calling party Address and execute the test.
- Follow the routing decision process to see that the correct route is found as configured.



## 5.2 Verification from Communication Manager Branch and Session Manager

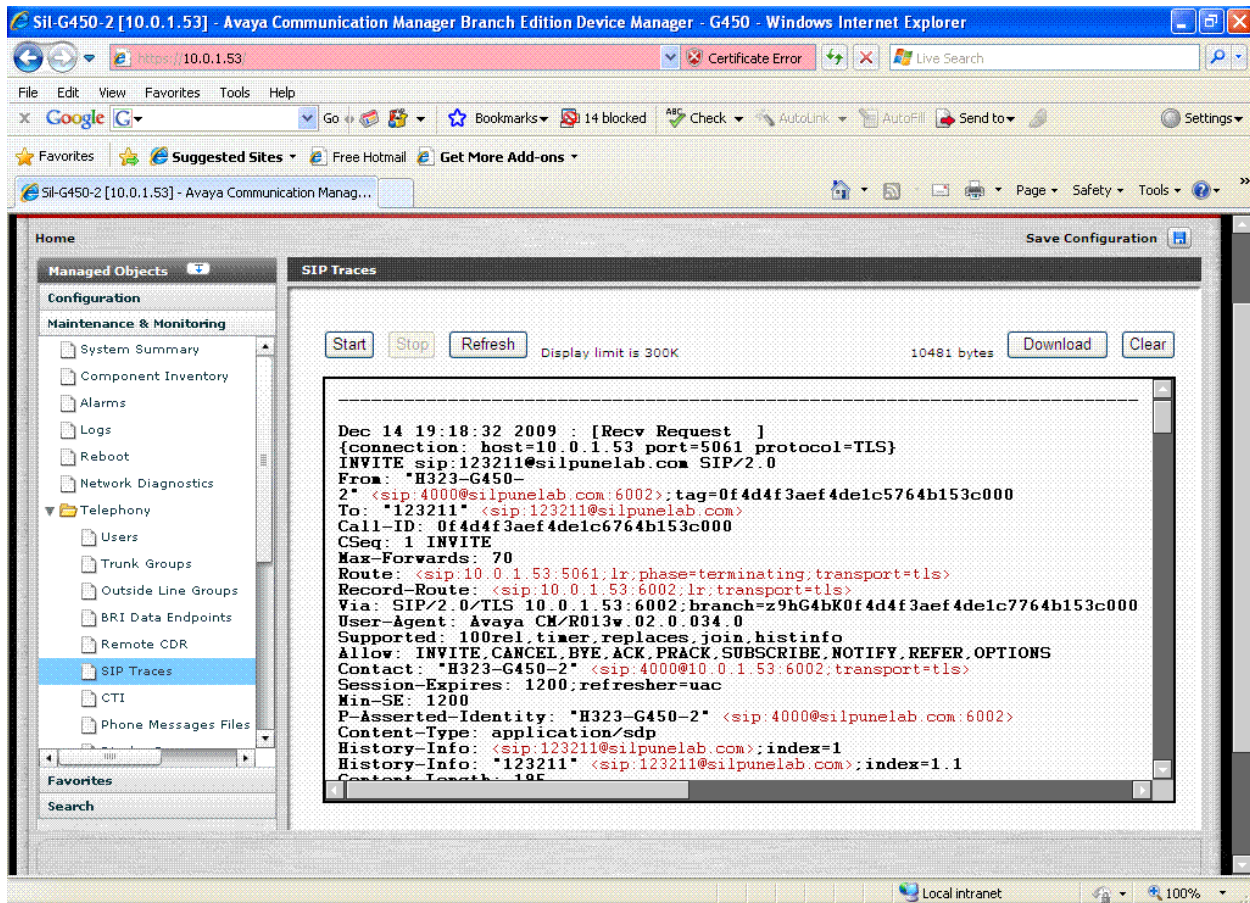
Capture Call trace/SIP trace on Communication Manager Branch for inter-Communication Manager Branch call.

- Register extension 4000 with Communication Manager Branch-1 and extension 211 with Communication Manager Branch-2
- Log on to Communication Manager Branch-1. Go to Maintenance & Monitoring -> Telephony -> SIP Traces.
- Log on to Session Manager as **root**, run command **traceASM** to capture traces on Session Manager.
- Dial <Automatic Route Selection> + < Communication Manager Branch 2 – Dial pattern, as defined in section 3.1.4 above> + 211, from ext 4000 of Communication Manager Branch-1 and place a call.(e.g.9123211)



- Capture the Communication Manager Branch and Session Manager traces for this call as mentioned in points above.  
(Kindly contact Avaya representative for the root login.)

### Communication Manager Branch Traces:



### Session Manager Traces:

```

avaya-asma - traceSM - Captured: 47 Displayed: 47
-----
10.0.0.247
SM
-----
19:18:36,952 | Dial Pattern route parameters | URI Domain: silpunelab.com Location: null
19:18:36,952 | Trying Dial Pattern route | Domain: silpunelab.com Location: null
19:18:36,952 | Dial Pattern route parameters | URI Domain: null Location: null
19:18:36,952 | Trying Dial Pattern route | Domain: null Location: null
19:18:36,952 | Dial Pattern found | for: 123211 Pattern: 123
19:18:36,952 | Route Policy found | Pattern: 123 RoutePolicyList: ASM-A to pusqa1120-BH1
19:18:36,952 | Route found | for: sip:123211@silpunelab.com SIPEntity: ASM-A to pusqa1120-BH1
19:18:36,952 | Entity Link found | SIPEntity: ASM-A to pusqa1120-BH1 EntityLink: avaya-asma->TCP, biDirId=nu
19:18:36,954 | No hostname resolution required | Routing to: sip:15.0.0.25:transport=tcp;lr;phase=terminating
19:18:36,954 | Location not found | for: 10.0.1.53
19:18:36,954 | Location not found | for: 15.0.0.25
19:18:36,956 | <--reINVIT--> | | (3) T:123211 F:4000 U:123211
19:18:36,982 | --Trying--> | | (3) 100 Trying
19:18:37,101 | --Ringing--> | | (3) 180 Ringing
19:18:37,106 | <--Ringing--> | | (3) 180 Ringing
19:18:37,154 | ---PRACK--> | | (3) sip:15.0.0.25
19:18:37,156 | <---PRACK--> | | (3) sip:15.0.0.25
19:18:37,210 | --200 OK--> | | (3) 200 OK (PRACK)
19:18:37,212 | <--200 OK--> | | (3) 200 OK (PRACK)
-----
s=Start q=Quit ENTER=Details f=Filters w=Write a=SM100 c=Clear i=IP

```

## 6. Conclusion

As illustrated in these Application Notes, Communication Manager Branch can interoperate with Session Manager using public SIP trunks. The tests were carried out using SIP Trunk with transport protocols: UDP, TCP.

## 7. Test results

The following functionalities are validated:-

Feature	CMB Platform
Automated Attendant	G450,i120
Logs and alarms	G450,i120
Backup and restore	G450
Call Hold	i40,i120
Call Drop	i40,i120
Flexible Storage Capacity	G450,i120
Whisper Page	G450
Centralized and Distributed Trunking	G450,i120
UUI/UCID testing for CMBE with ASM	G450

Call flows-Inter CMBE (G450, i120, i40), CM-SM-CMB (G450) calling, CMB (G450)-CM calling.

### Known issues:-

- 1) Call Transfer cannot be completed over the same public SIP trunk.

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