Avaya Solution & Interoperability Test Lab

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.

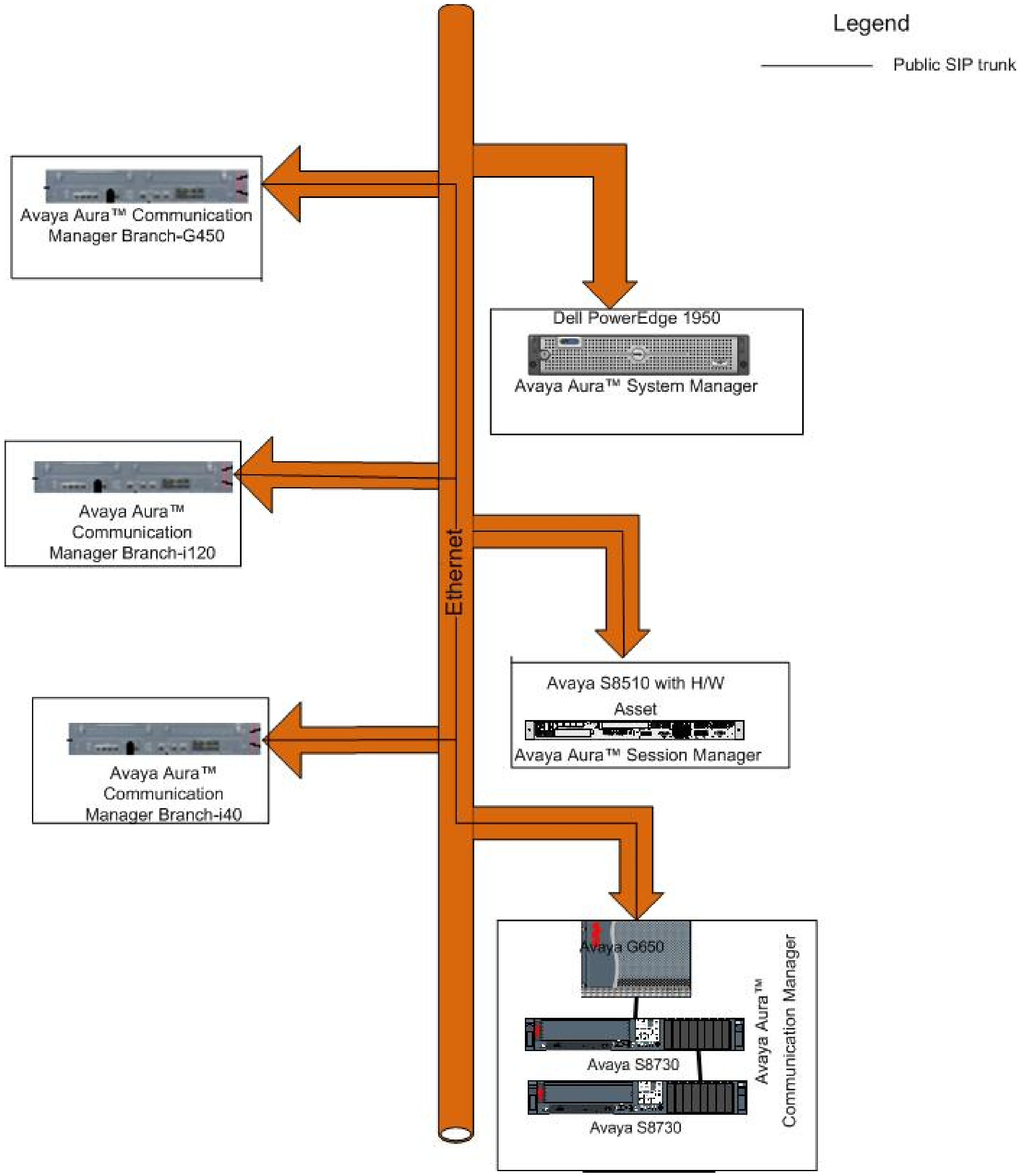


Figure 1: Configuring Avaya Aura™ Session Manager with Avaya Aura™ Communication Manager Branch

2. Equipment and Software Validated

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

Product / Hardware Platform	**Version**
Avaya Aura™ Session Manager • SM-100 Hardware • Avaya S8510 Server	Session Manager 5.2.0.1 Build: 520017
Avaya Aura™ System Manager • Dell Power Edge 1950	System Manager 5.2.0.1 Build: 520017
Avaya Aura™ Communication Manager Branch - G450	Release 2.0 SP2.0.0.17
Avaya Aura™ Communication Manager Branch - i120	Release 2.0 SP2.0.0.17
Avaya Aura™ Communication Manager Branch - i40	Release 2.0 SP2.0.0.17
Avaya Aura™ Communication Manager • G650 • Avaya S8730	Release 5.2.1

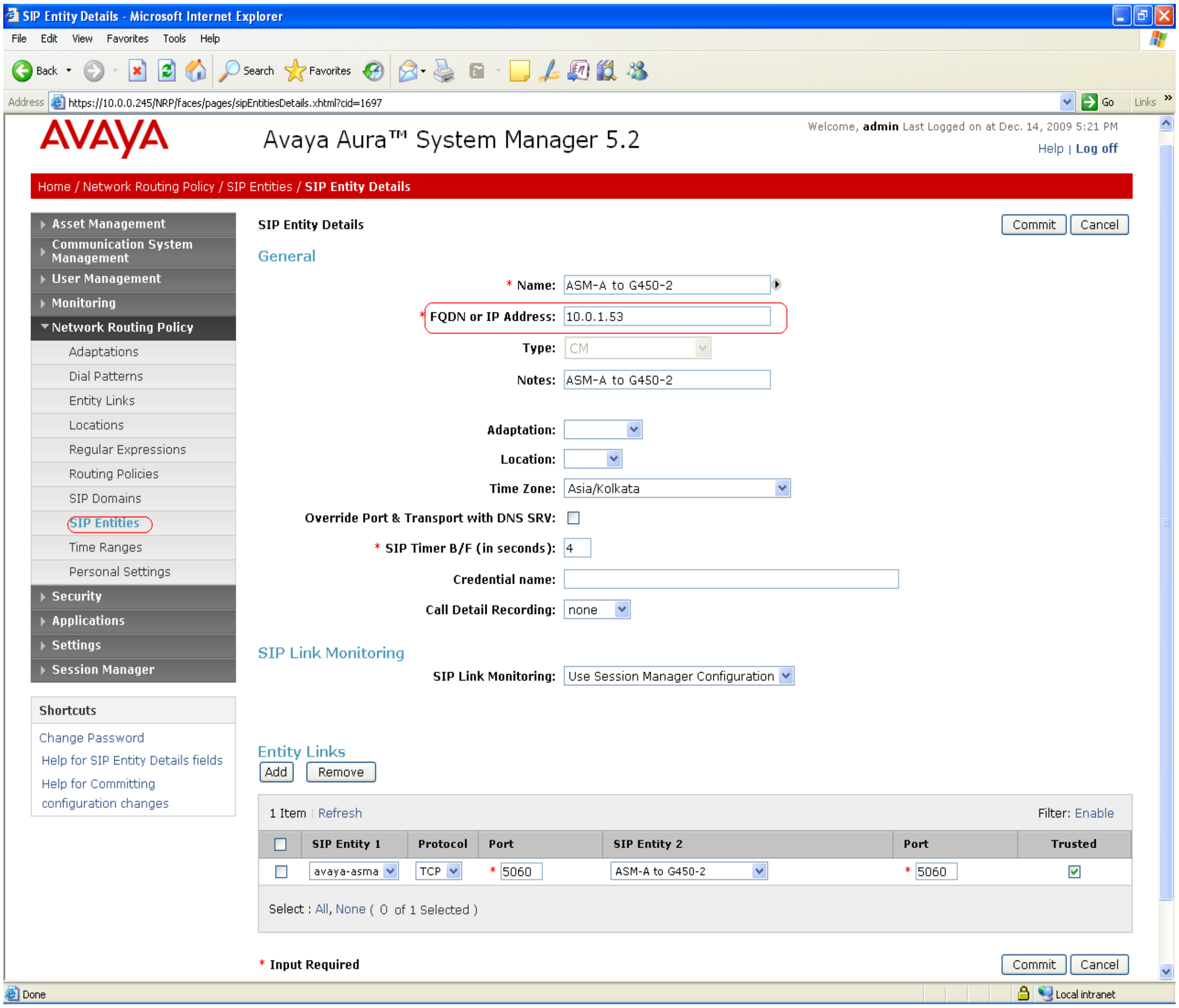
3. Configure Avaya Aura™ Session Manager

3.1 Avaya Aura™ 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.

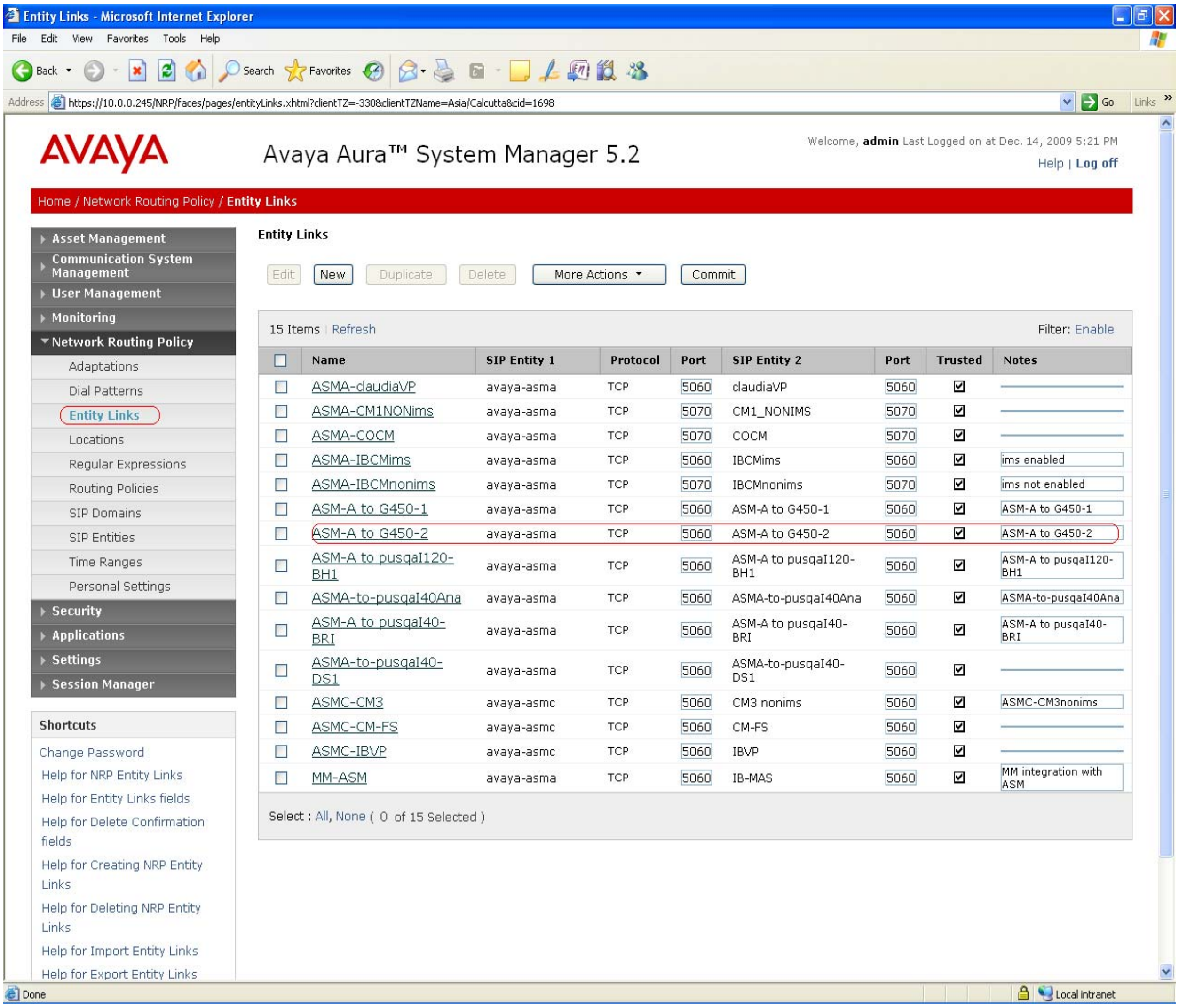
3.1.1 Add SIP entity for Communication Manager Branch

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



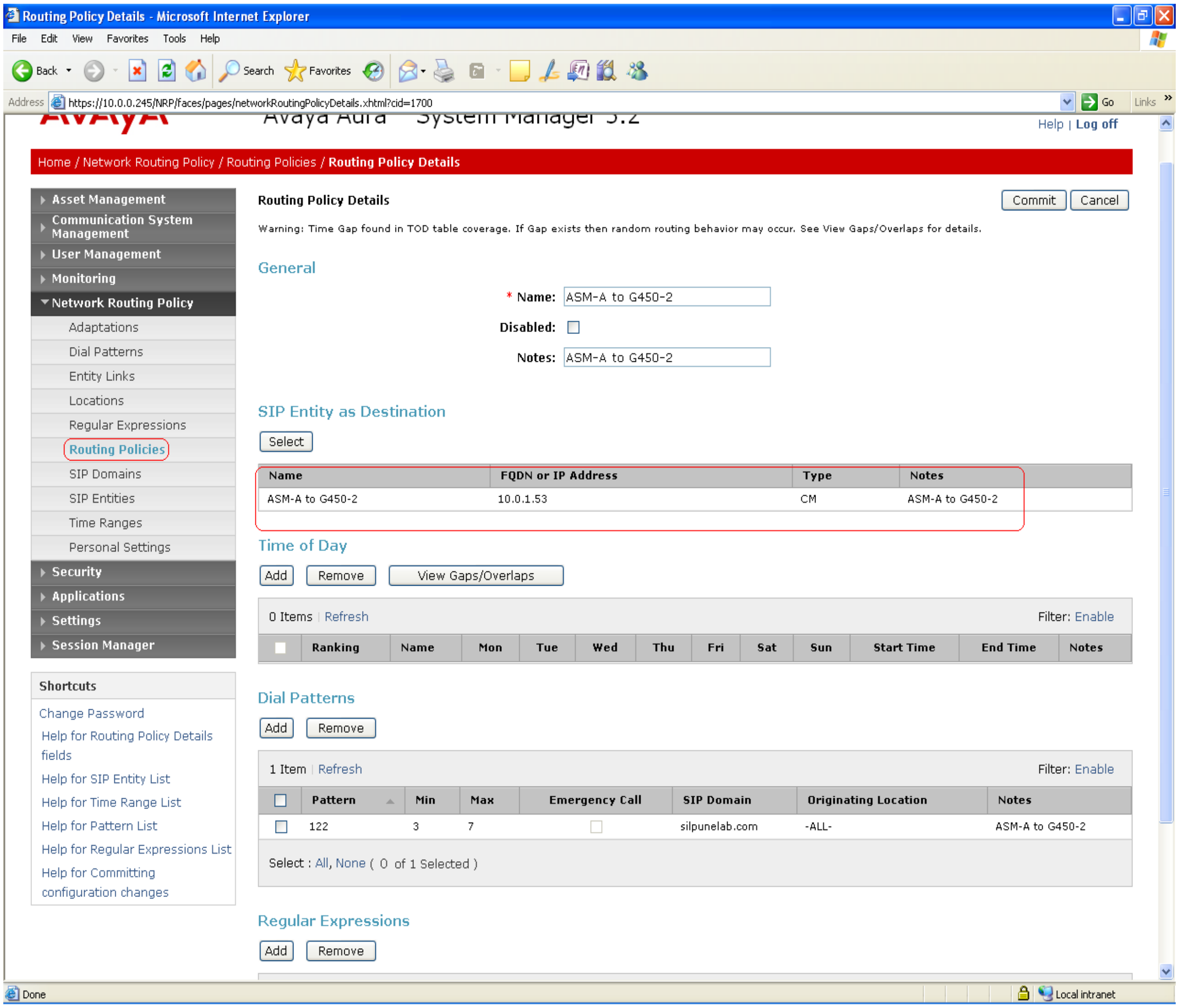
3.1.2 Add entity link for Communication Manager Branch

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.



3.1.3 Define Routing Policy

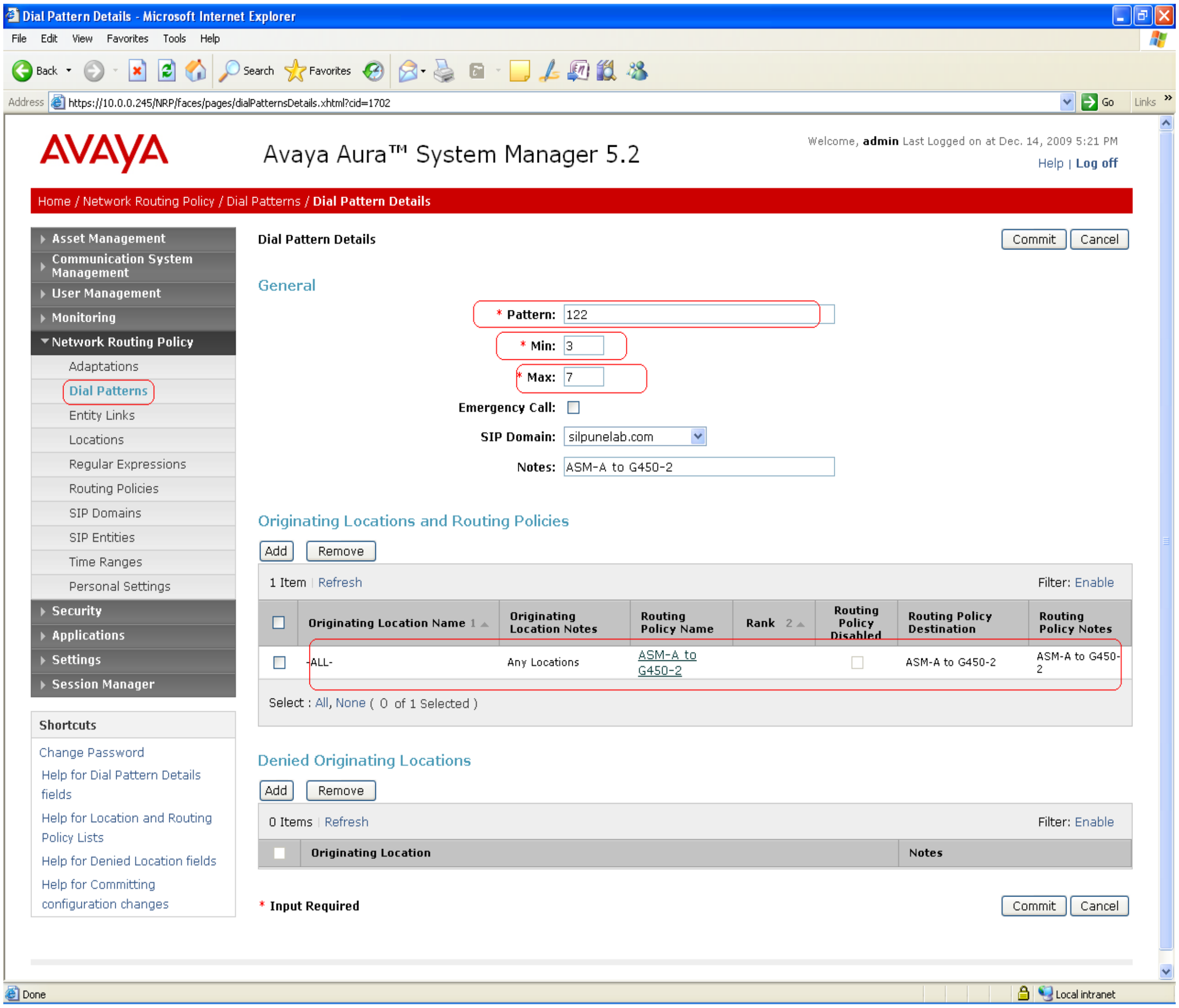
Session Manager can simply route the call based on the starting digit and the number of digits defined in the dial pattern. When a call starting with 122- xxxx reaches Session Manager, it will route the incoming call to Communication Manager Branch. Session Manager can also route calls based on different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



3.1.4 Define Dial Pattern

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

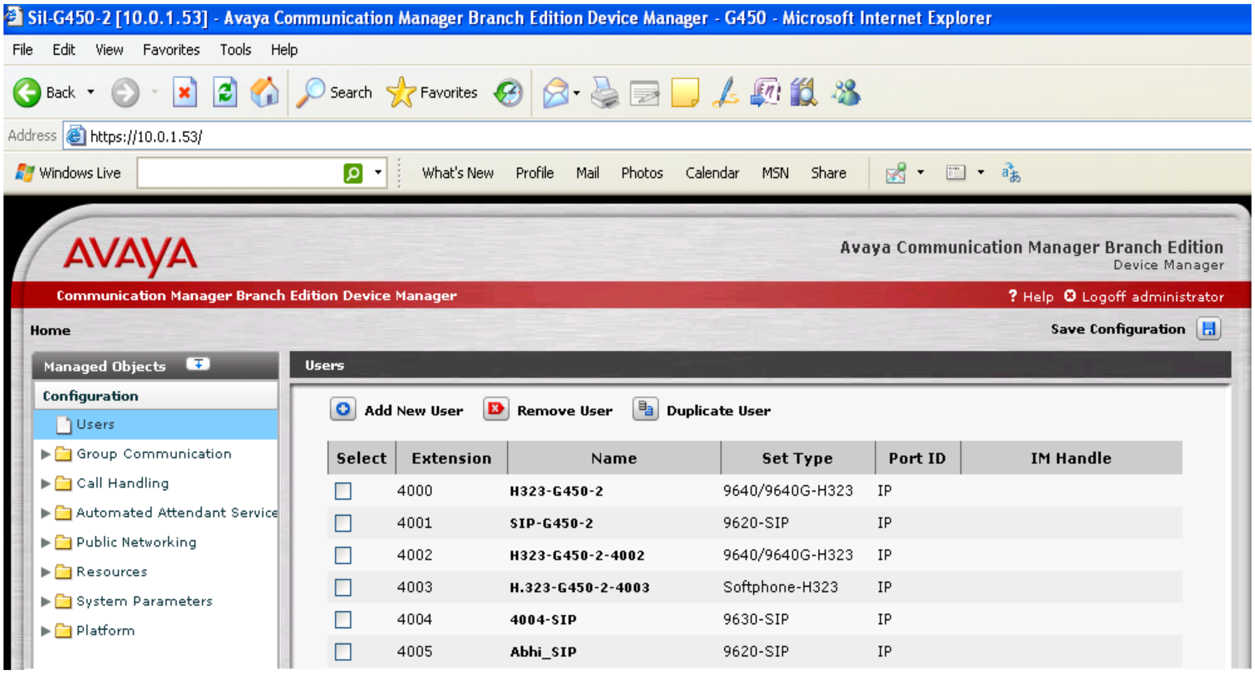
The routing capability can be limited by setting a destination domain, so the call will only route to a SIP entity with the given domain. The location value below can limit the originating SIP entity, as defined in section 3.1.1 above.


Dial Pattern Details - Microsoft Internet Explorer
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
Asset Management
Communication System Management
User Management
Monitoring
Network Routing Policy
Adaptations
Dial Patterns
Entity Links
Locations
Regular Expressions
Routing Policies
SIP Domains
SIP Entities
Time Ranges
Personal Settings
Security
Applications
Settings
Session Manager
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
Dial Pattern Details
Commit Cancel
General
* Pattern: 122
* Min: 3
* Max: 7
Emergency Call:
SIP Domain: silpunelab.com
Notes: ASM-A to G450-2
Originating Locations and Routing Policies
Add Remove
1 Item | Refresh
Filter: Enable
Originating Location Name
Originating Location Notes
Routing Policy Name
Rank
Routing Policy Disabled
Routing Policy Destination
Routing Policy Notes
-ALL-
Any Locations
ASM-A to G450-2
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
Originating Location
Notes
* Input Required
Commit Cancel
Done
Local intranet

4. Configure Avaya Aura™ Communication Manager Branch

4.1 Configure dial plan with users

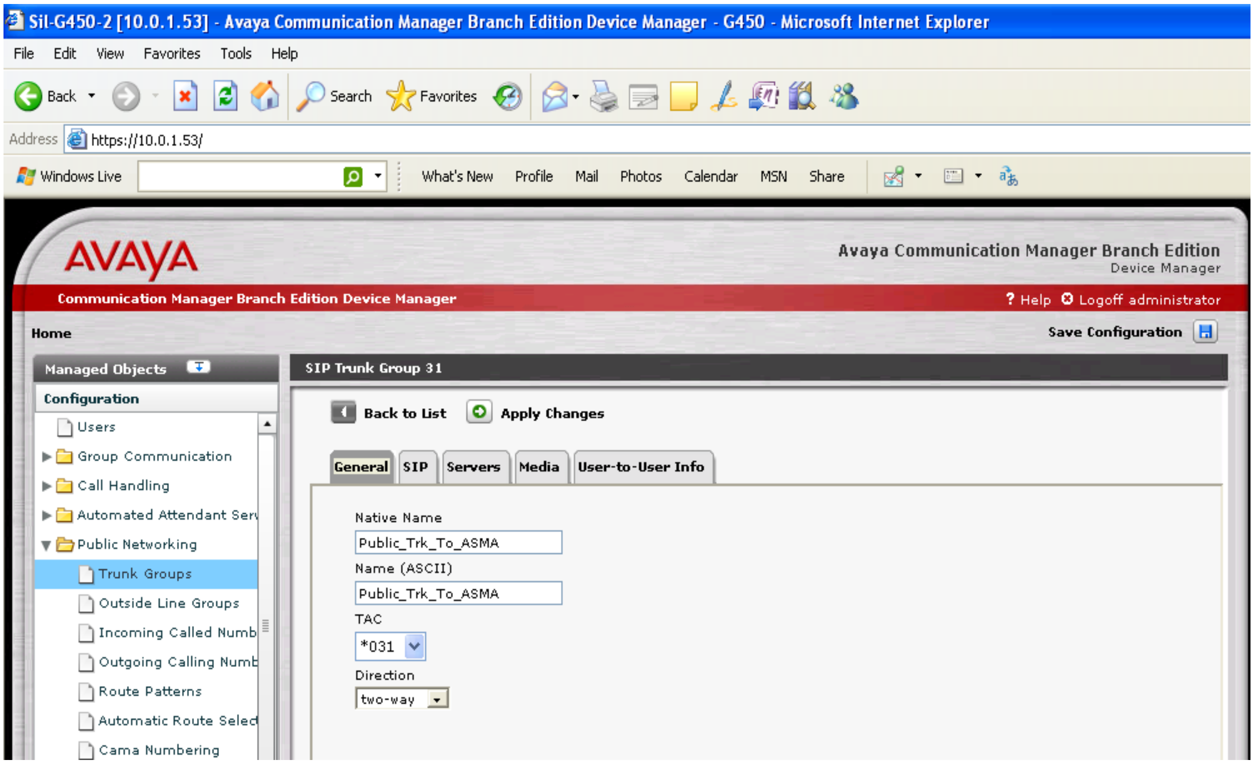
Dial plan can only be set while initializing the Communication Manager Branch Edition and cannot be changed later on. The Dial Plan allows adding new users or stations. Consider 4 digit dial plan for reference.



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.

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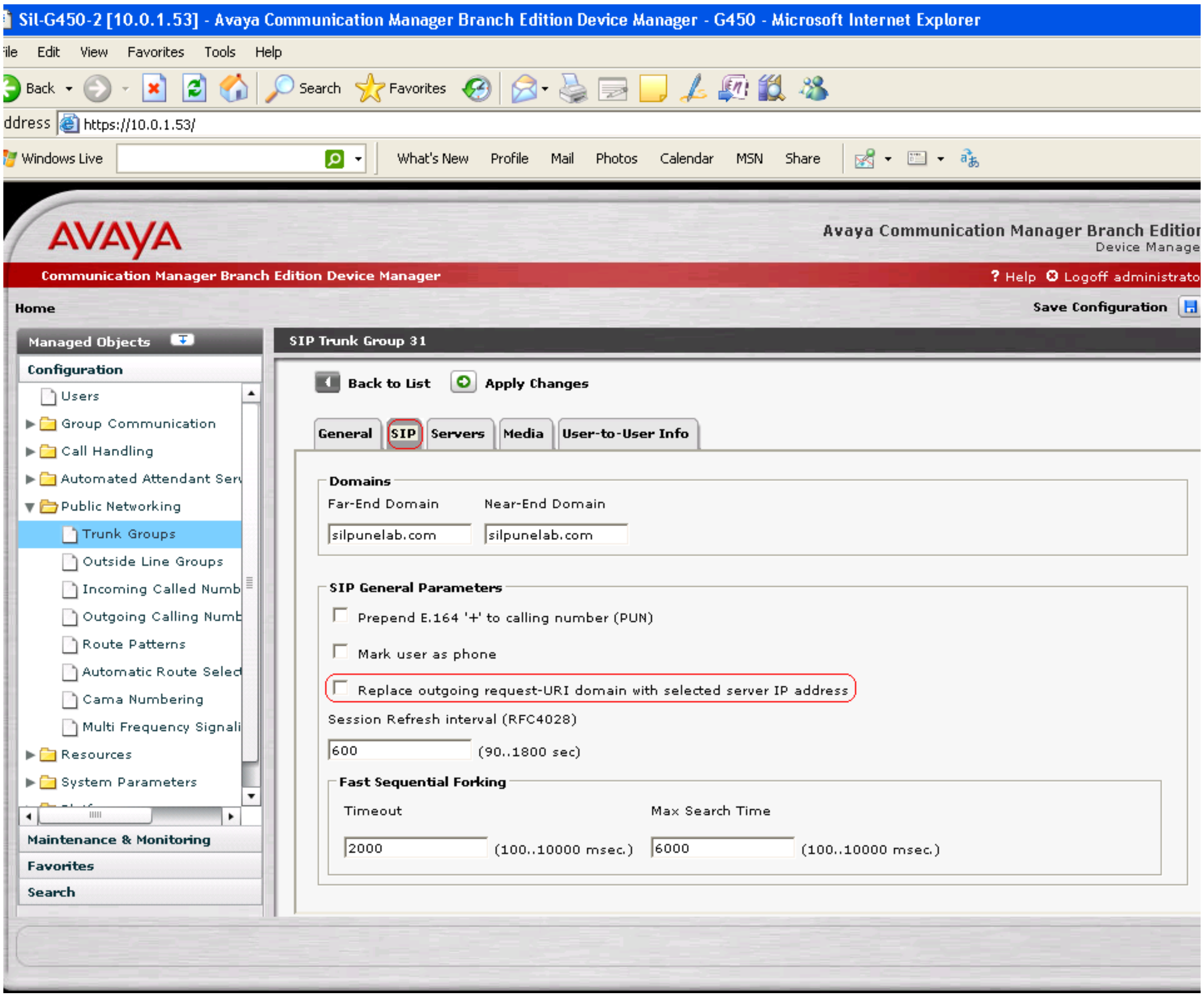
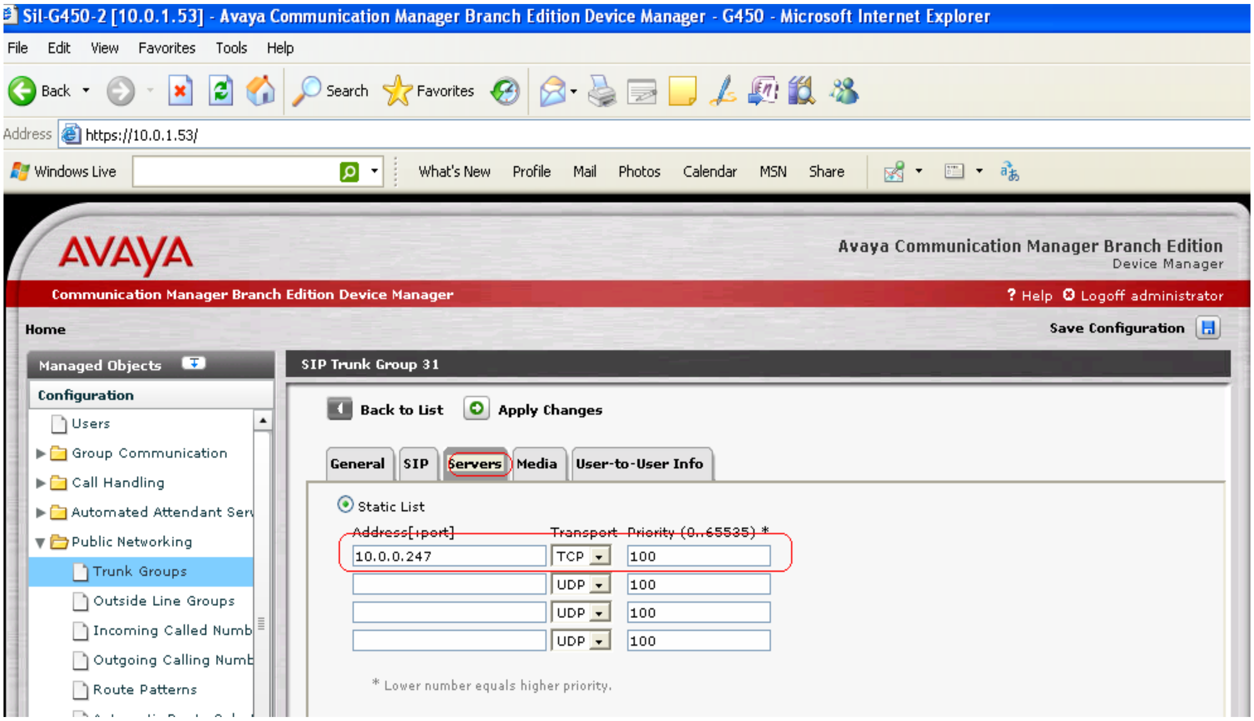


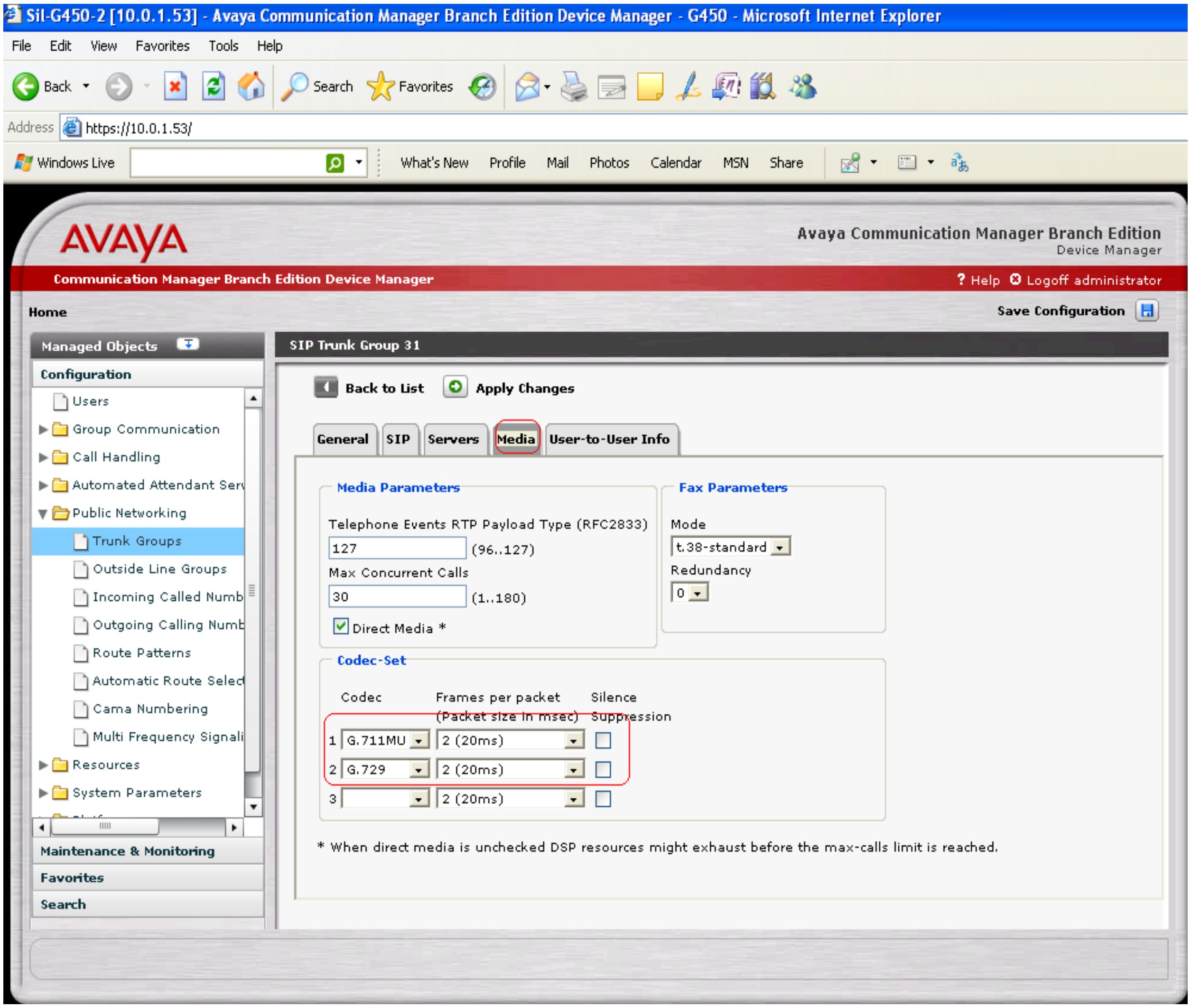
4.2.3 Servers settings

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



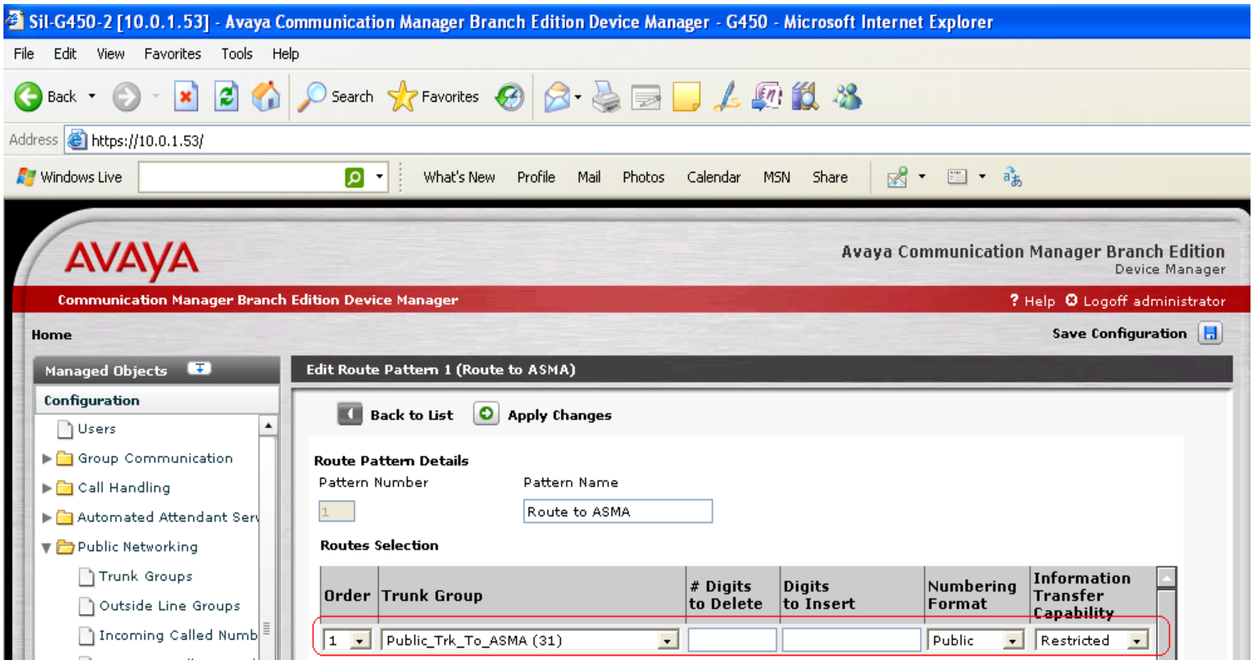
4.2.4 Media Settings

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

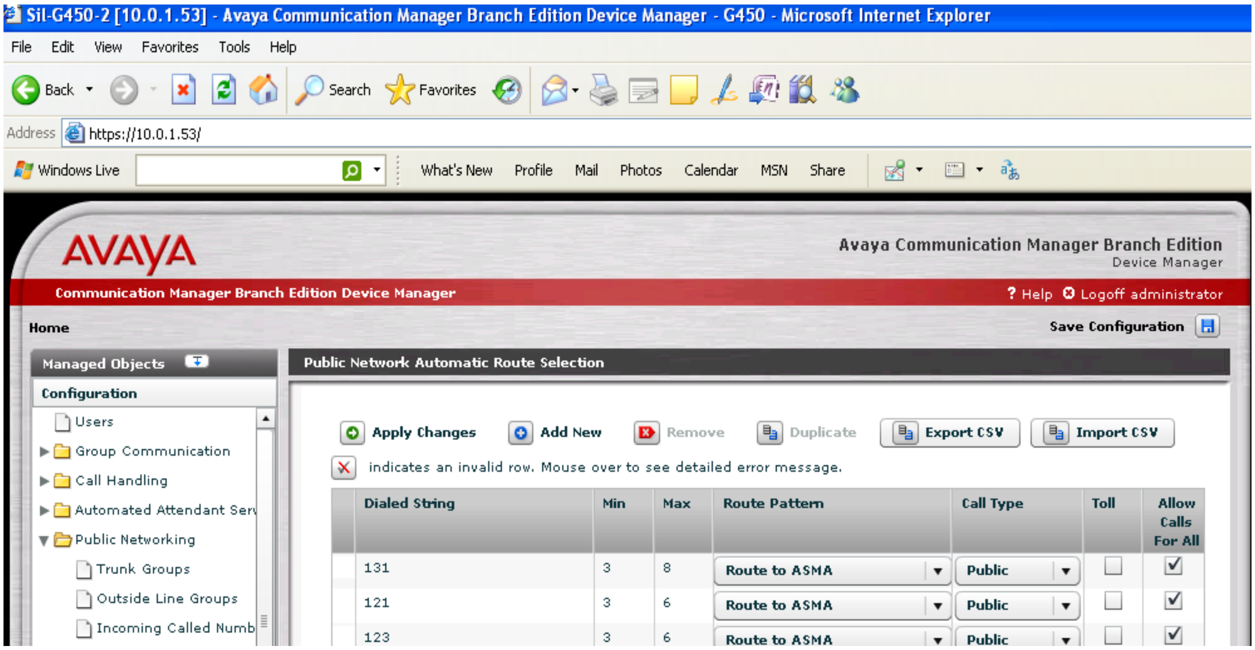


4.4 Configure Automatic Route Selection

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.



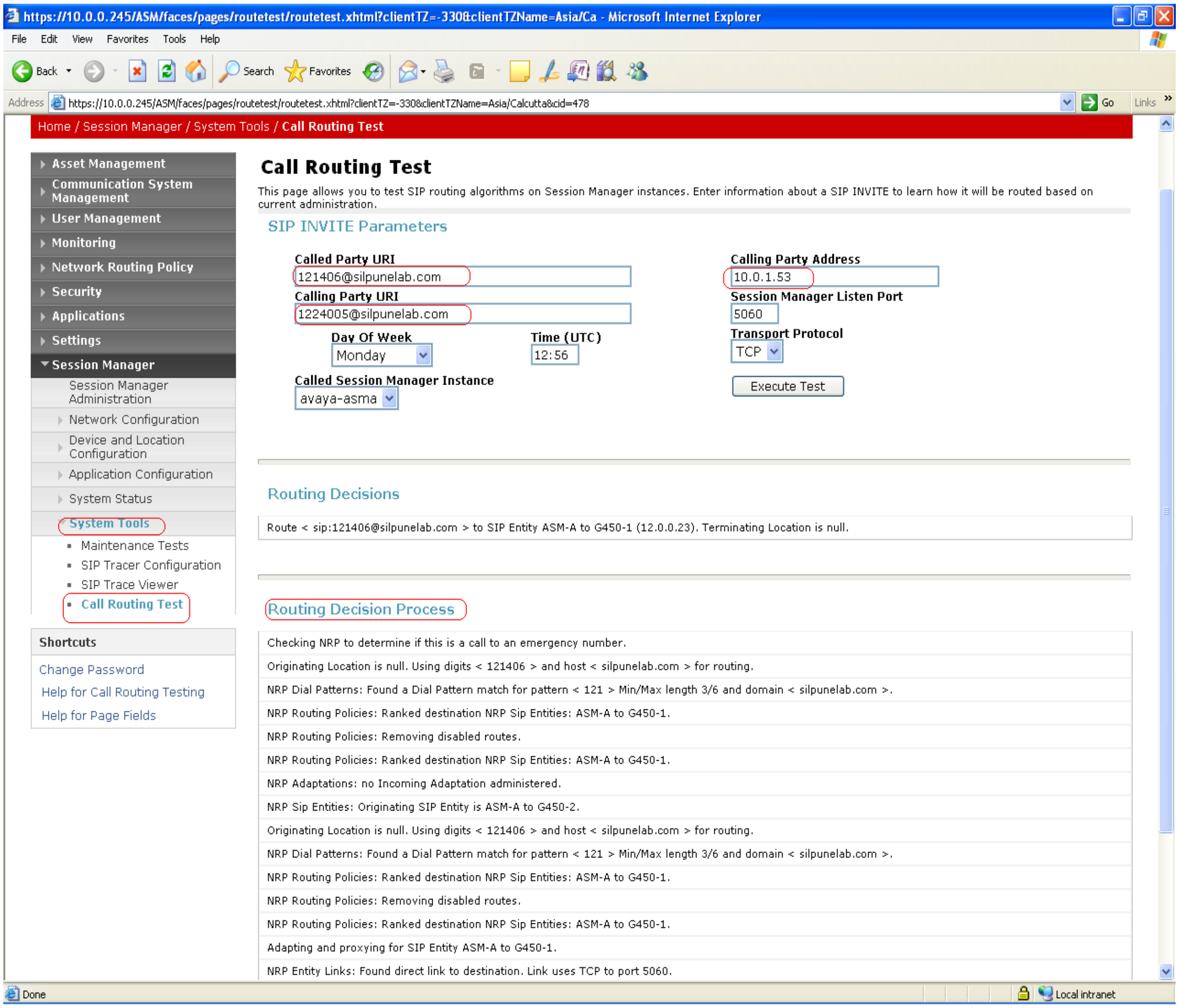
5. Verification Scenarios

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.



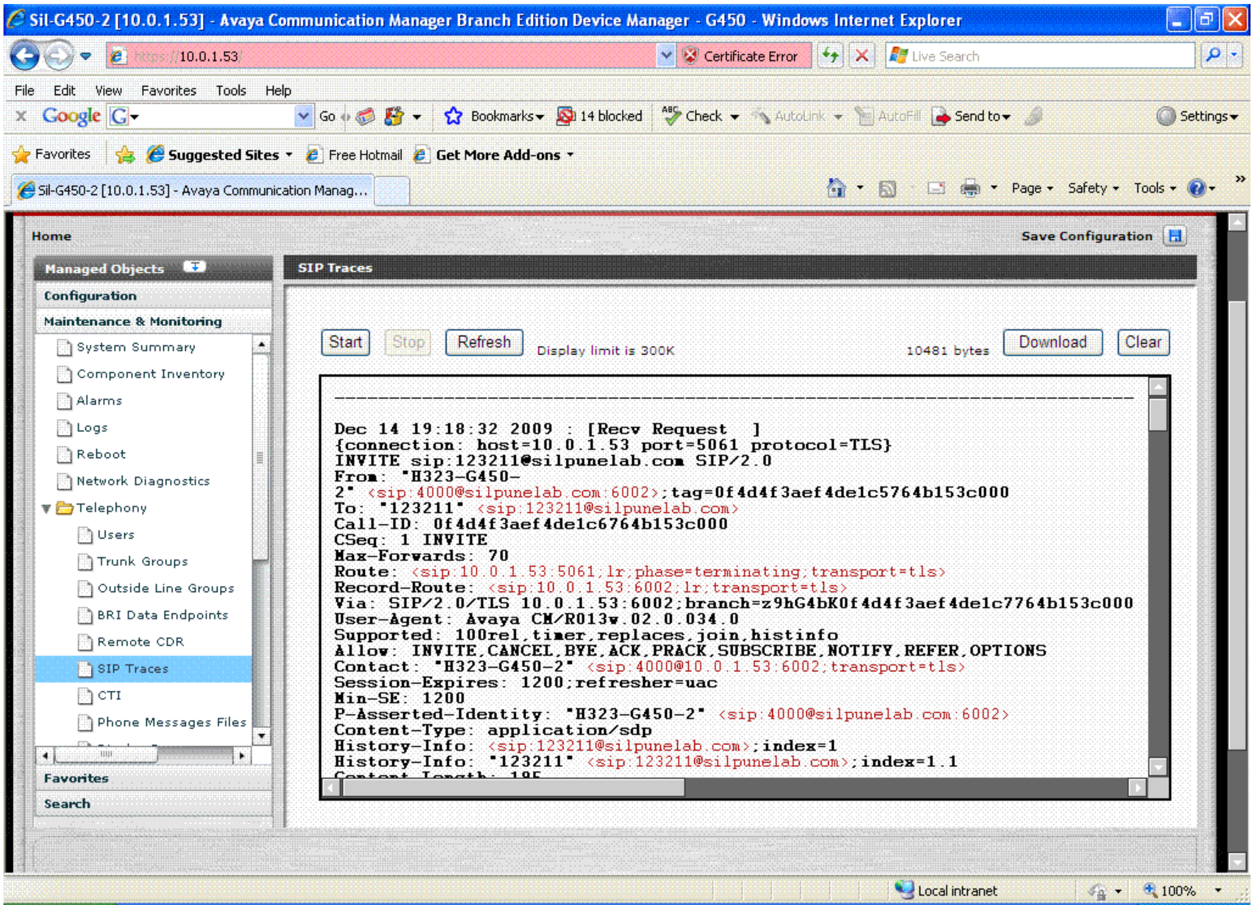
5.2 Verification from Communication Manager Branch and Session Manager

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

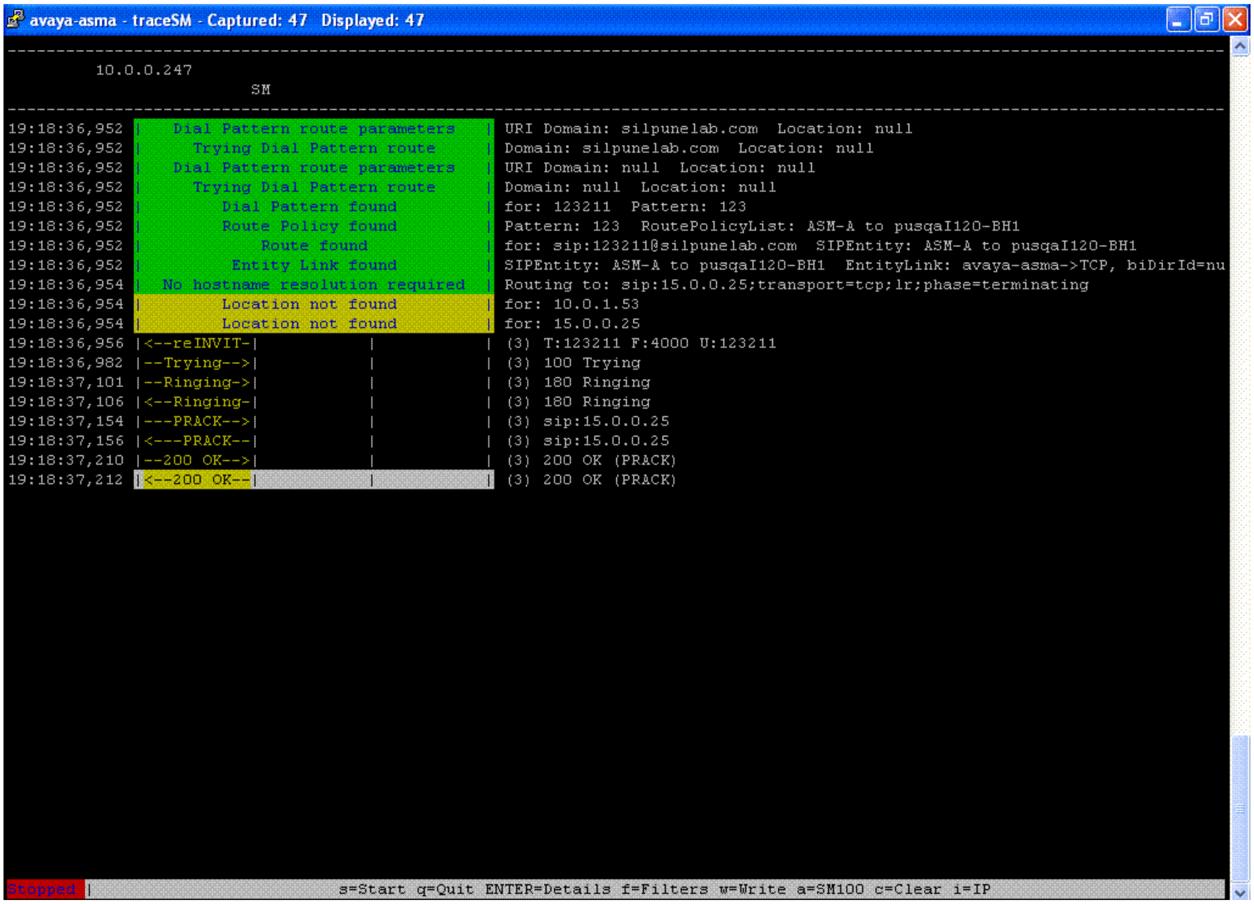
- Register extension 4000 with Communication Manager Branch-1 and extension 211 with Communication Manager Branch-2
- Log on to Communication Manager Branch-1. Go to Maintenance & Monitoring -> Telephony -> SIP Traces.
- Log on to Session Manager as **root**, run command **traceASM** to capture traces 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:



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.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com