Application Note

Verizon SIP Trunking:

Avaya Aura Communication Manager 7.0 via Avaya Aura Session Manager 7.0 with Cisco Unified Border Element – CUBE 11.5.0 [v15.6.1.S1, ASR1K/XE-3.17.01.S] using SIP

May 26, 2016

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Introduction

Service Providers today, such as Verizon, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

Verizon is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Avaya Aura Communication Manager and Verizon network, Cisco Unified Border Element (Cisco UBE) 15.6(1) S1 can be used. The Cisco Unified Border Element 15.6(1) S1 provides demarcation, security, interworking and session control services for Avaya Aura Communication Manager 7.0 connected to Verizon IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Avaya Aura Communication Manager. Only configuration settings specifically required for Verizon interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure Avaya Aura Communication Manager 7.0, Avaya Aura Session Manager 7.0 and Cisco Unified Border Element (Cisco UBE) 15.6(1) S1 for connectivity to Verizon SIP trunking service. The deployment model covered in this application note is CPE (Avaya Aura Communication Manager 7.0) to PSTN (Verizon).
- Testing was performed in accordance to SIP trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and interoperability with Avaya Voicemail
- The Avaya Aura Communication Manager and Avaya Aura Session Manager configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Verizon SIP network and Avaya Aura Communication Manager. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying to interoperate Avaya Aura Communication Manager to Verizon SIP trunking network.

Network Topology

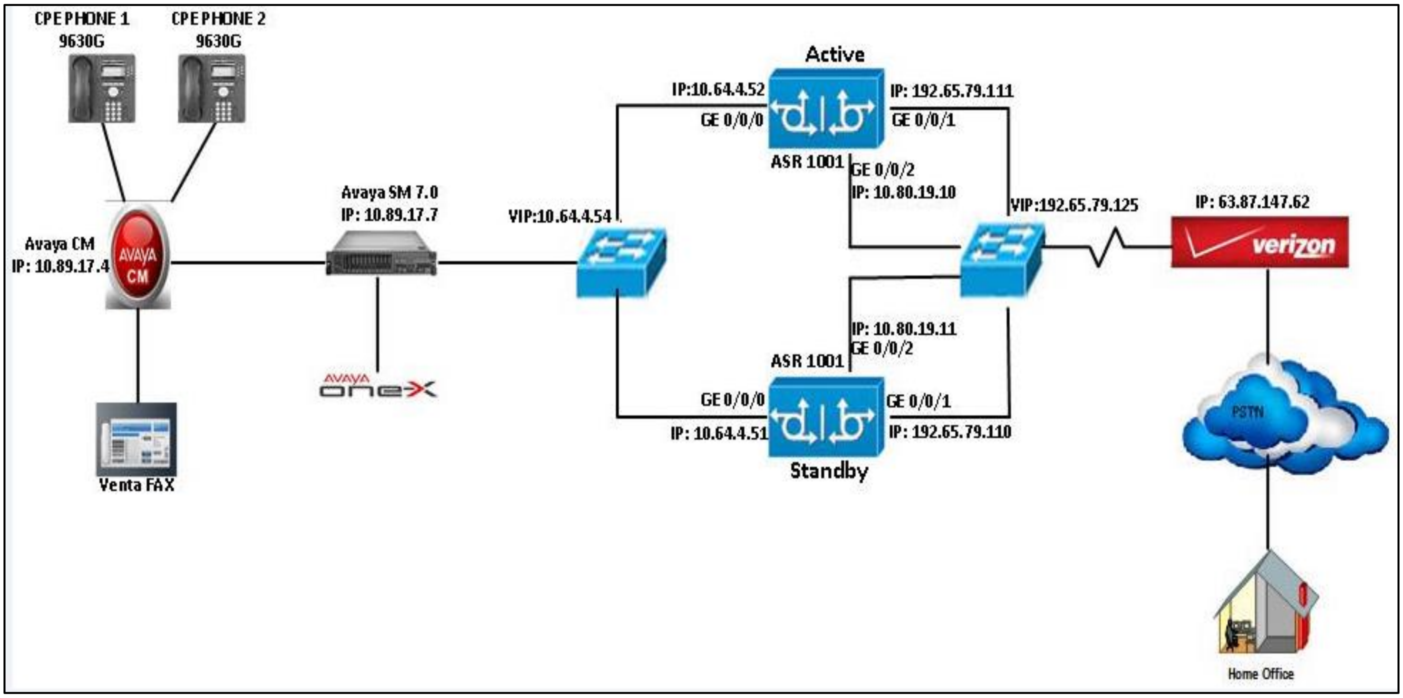

Figure 1 Network Topology

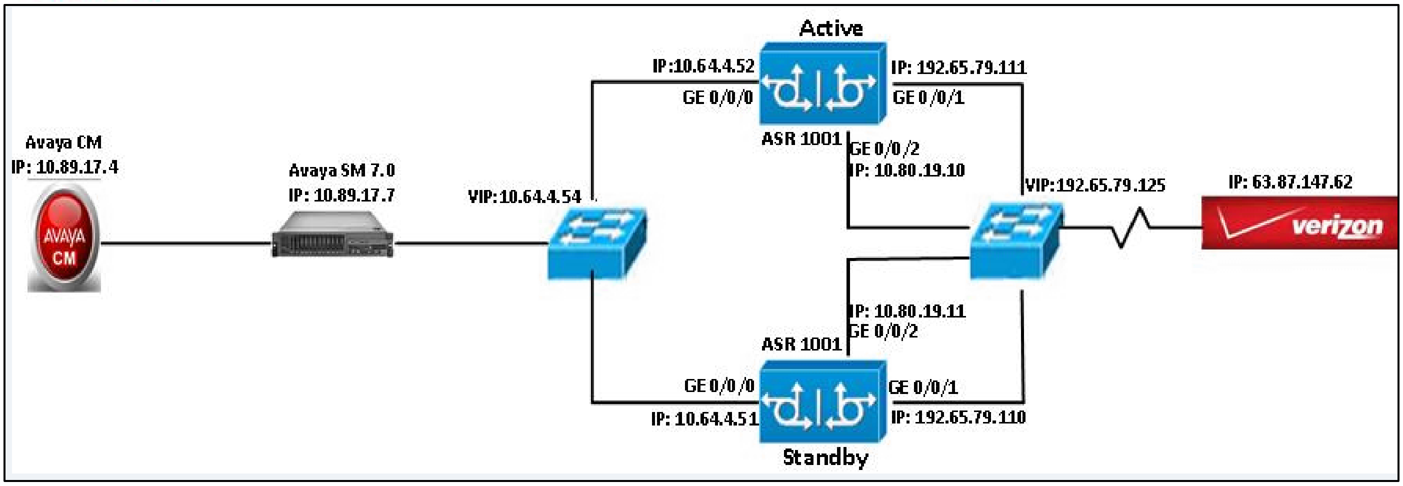

Figure 2: Cisco UBE High Availability

System Components

Hardware Requirements

- Cisco ASR1001 (1RU) processor (revision 1RU) with 1064431K/6147K bytes of memory with 4 Gigabit Ethernet interfaces.
- Processor board ID SSI17360FV1
- 2 Avaya 9630G series IP telephone (H323 and SIP)

Software Requirements

- Cisco IOS Software, ASR1000 Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.6(1)S1, RELEASE SOFTWARE (fc3)
- Cisco IOS XE Software, Version 03.17.01.S
- Avaya Aura Communication Manager release 7.0.0.2.0-SP2(22684)
- Avaya Aura Session Manager 7.0
- Avaya one-X Communicator Release 6.2.4

Features

Features Supported

- Incoming and outgoing off-net calls using G729
- Call hold
- Call transfer (unattended, attended and blind)
- Call conference
- Call forward (all, busy, no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (T38 and G711 pass through)

Features Not Supported

- In HA redundancy mode the primary cube will not take over the primary/active role after a reboot/network outage

Caveats

- Avaya does not send fax re-INVITE to network for inbound fax scenarios
- Blind transfer has been tested using One-X soft client.
- PSTN users do not hear MOH for hold and transfer scenarios due to the provider sending a=inactive attribute in the SDP. The issue at the time of testing has been identified as a service provider limitation. The issue does not impact the calls.

- Caller ID is not updated after attended or semi-attended transfers to off-net phones. This is due to a limitation on Cisco UBE and will be resolved in the next release. The issue does not impact the calls.

Configuration

Configuring the Cisco Unified Border Element

Network interface

Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration only, the actual IP address can vary. For SIP trunks two IP addresses must be configured—LAN and WAN.

interface GigabitEthernet0/0/0

```
 description Wan Interface
 ip address 192.65.79.110 255.255.255.224
 negotiation auto
 cdp enable
 redundancy rii 2
 redundancy group 1 ip 192.65.79.125 exclusive
!
interface GigabitEthernet0/0/1
 description Lan Interface
 ip address 10.64.4.51 255.255.0.0
 negotiation auto
 cdp enable
 redundancy rii 1
 redundancy group 1 ip 10.64.4.54 exclusive
```

Global Cisco UBE settings

In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

```
voice service voip
 ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
 address-hiding
```

```
mode border-element license capacity 50
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol pass-through g711ulaw
sip
 bind control source-interface GigabitEthernet0/0/0
 bind media source-interface GigabitEthernet0/0/0
 session refresh
 asserted-id pai
 privacy pstn
 early-offer forced
 no silent-discard untrusted
 midcall-signaling passthru
 g729 annexb-all!
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

Codecs

G729 is used as the preferred codec for this testing and changed the codecs according to the test plan description

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

Dial peer

Cisco UBE uses dial-peer to route the call based on the digit to route the call accordingly.

```
dial-peer voice 100 voip
 description towards Verizon-WAN side
 destination-pattern [2-9]T
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 101 voip
 description From Avaya-LAN side
 session protocol sipv2
 session transport udp
```

```
 incoming called-number [2-9]T
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 200 voip
 description from Verizon to Avaya-WAN side
 session transport udp
 incoming called-number 719.......
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 201 voip
 description from Verizon to Avaya-LAN side
 destination-pattern 719.......
 session protocol sipv2
```

```
 session target ipv4:10.89.17.7:5060
 session transport udp
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 300 voip
 description PBX to PBX via Verizon-LAN side
 session transport udp
 incoming called-number 1719377....
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 301 voip
 description PBX to PBX via Verizon-WAN side
 destination-pattern 1719377....
```

```
 translate-outgoing called 1
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 400 voip
 description International call-WAN side
 destination-pattern 011T
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
```

```
!
dial-peer voice 401 voip
 description International call-LAN side
 session protocol sipv2
 session transport udp
 incoming called-number 011T
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 500 voip
 description towards Verizon for toll free-WAN side
 destination-pattern 1800T
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
```

```
!
dial-peer voice 501 voip
 description towards Verizon for toll free-LAN side
 session protocol sipv2
 session transport udp
 incoming called-number 1800T
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!

dial-peer voice 600 voip
 description towards Verizon for Emergency and Directory service
 destination-pattern [4,9]11
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
```

Call flow

For incoming PSTN calls, the CUBE presents the full ten-digit DID number to Avaya Aura Communication Manager.

CPE callers make outbound PSTN calls by dialing a "9" prefix followed by the destination number.

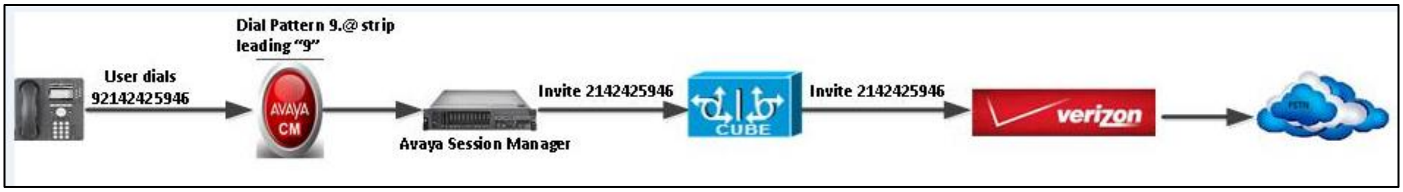

Figure 3: Outbound Voice Call

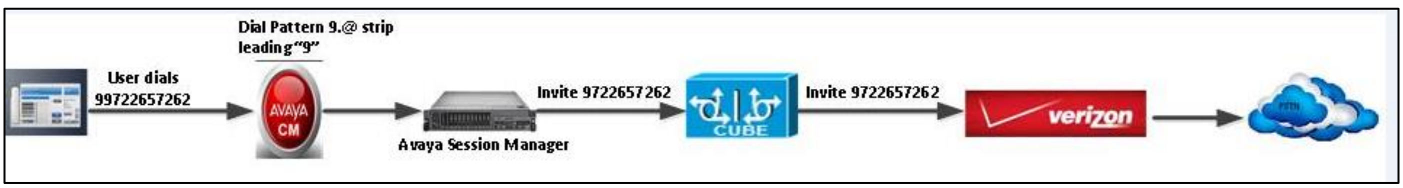

Figure 4: Outbound Fax Call

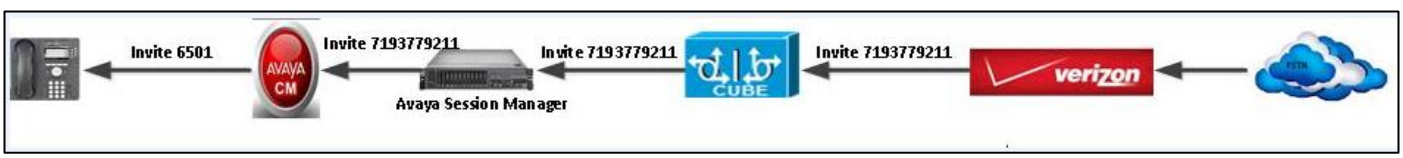

Figure 5: Inbound Voice Call

Figure 6 : Inbound Fax Call

Configuration example

The following configuration snippet contains a sample configuration of Cisco Unified Border Element with all parameters mentioned previously.

Active Cisco UBE

User Access Verification

Username: cisco

Password:

Verizon-ASR1#sh run

Building configuration...

Current configuration : 7318 bytes

!

! Last configuration change at 01:20:41 UTC Tue May 10 2016 by cisco

!

version 15.6

service timestamps debug datetime msec

service timestamps log datetime msec

no platform punt-keepalive disable-kernel-core

!

```
hostname Verizon-ASR1
!
boot-start-marker
boot system flash bootflash: asr1001-universalk9.03.17.01.S.156-1.S1-std.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging queue-limit 1000000000
logging buffered 10000000
logging rate-limit 10000
!
no aaa new-model
!
ipc zone default
 association 1
  no shutdown
!
no ip domain lookup
!
```

subscriber templating

!

multilink bundle-name authenticated

!

```
voice service voip
 ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 50
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip handle-replaces
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "<sip:\+(.*)@(.*)>" "<sip:\1@\2"
```

```
 request INVITE sip-header Diversion modify "reason=unknown" ""
!
license udi pid ASR1001 sn JAE17430GQ5
license accept end user agreement
license boot level advipservices
!
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
   track 1 shutdown
   track 2 shutdown
!
cdp run
!
translation-rule 1
 Rule 1 17193779211 7193779211
 Rule 2 17193779212 7193779212
!
interface GigabitEthernet0/0/0
```

```
 description Wan Interface
 ip address 192.65.79.111 255.255.255.224
 negotiation auto
 cdp enable
 redundancy rii 2
 redundancy group 1 ip 192.65.79.125 exclusive
!
interface GigabitEthernet0/0/1
 description Lan Interface
 ip address 10.64.4.52 255.255.0.0
 negotiation auto
 cdp enable
 redundancy rii 1
 redundancy group 1 ip 10.64.4.54 exclusive
!
interface GigabitEthernet0/0/2
 ip address 10.80.19.10 255.255.0.0
 negotiation auto
!
interface GigabitEthernet0/0/3
 no ip address
 shutdown
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
```

```
 negotiation auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.97
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 172.16.0.0 255.255.0.0 10.64.1.1
!
control-plane
!
dial-peer voice 100 voip
 description towards Verizon-WAN side
 destination-pattern [2-9]T
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
```

```
!
dial-peer voice 101 voip
 description From Avaya-LAN side
 session protocol sipv2
 session transport udp
 incoming called-number [2-9]T
 voice-class codec 1
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 voice-class sip bind media source-interface GigabitEthernet0/0/1
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 fax-relay ecm disable
 fax rate 14400
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 no vad
!
dial-peer voice 200 voip
 description from Verizon to Avaya-WAN side
 session transport udp
 incoming called-number 719.......
 voice-class codec 1
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 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
```

```
 no vad
!
dial-peer voice 301 voip
 description PBX to PBX via Verizon-WAN side
 destination-pattern 1719377....
 translate-outgoing called 1
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
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 session transport udp
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
```

```
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
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dial-peer voice 401 voip
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 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 500 voip
 description towards Verizon for toll free-WAN side
 destination-pattern 1800T
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
```

```
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 501 voip
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 session protocol sipv2
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 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 600 voip
 description towards Verizon for Emergency and Directory service
 destination-pattern [4,9]11
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 voice-class sip bind control source-interface GigabitEthernet0/0/0
```

```
 dtmf-relay rtp-nte
 no vad
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 login local
!
!
End
```

Standby Cisco UBE

User Access Verification

```
Verizon-ASR2#sh run
Building configuration...

Current configuration : 7115 bytes
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname Verizon-ASR2
!
boot-start-marker
boot system flash bootflash: asr1001-universalk9.03.17.01.S.156-1.S1-std.bin
boot-end-marker
!
aqm-register-fnf
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
```

```
 exit-address-family
!
no aaa new-model
!
ipc zone default
 association 1
  no shutdown
!
no ip domain lookup
!
ipv6 unicast-routing
!
subscriber templating
!
multilink bundle-name authenticated
!
```

```
voice service voip
 ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
 address-hiding
 mode border-element license capacity 50
 media disable-detailed-stats
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip handle-replaces
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  no silent-discard untrusted
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class sip-profiles 1
```

```
 request INVITE sip-header Diversion modify "<sip:\+(.*)@(.*)>" "<sip:\1@\2"
 request INVITE sip-header Diversion modify "reason=unknown" ""
!
license udi pid ASR1001 sn JAE174202KE
license boot level advipservices
!
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bha
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
   track 1 shutdown
   track 2 shutdown
!
cdp run
!
track 1 interface GigabitEthernet0/0/1 line-protocol
track 2 interface GigabitEthernet0/0/0 line-protocol
!
translation-rule 1
 Rule 1 17193779211 7193779211
```

Rule 2 17193779212 7193779212

!

```
interface GigabitEthernet0/0/0
 description Wan Interface
 ip address 192.65.79.110 255.255.255.224
 negotiation auto
 cdp enable
 redundancy rii 2
 redundancy group 1 ip 192.65.79.125 exclusive
!
interface GigabitEthernet0/0/1
 description Lan Interface
 ip address 10.64.4.51 255.255.0.0
 negotiation auto
 cdp enable
 redundancy rii 1
 redundancy group 1 ip 10.64.4.54 exclusive
!
interface GigabitEthernet0/0/2
 ip address 10.80.19.11 255.255.0.0
 negotiation auto
!
interface GigabitEthernet0/0/3
 no ip address
 shutdown
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
```

```
 no ip address
 negotiation auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.79.97
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 172.16.0.0 255.255.0.0 10.64.1.1
!
control-plane
!
dial-peer voice 100 voip
 description towards Verizon-WAN side
 destination-pattern [2-9]T
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
```

```
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 101 voip
 description From Avaya-LAN side
 session protocol sipv2
 session transport udp
 incoming called-number [2-9]T
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 200 voip
 description from Verizon to Avaya-WAN side
 session transport udp
 incoming called-number 719.......
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
```

```
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 201 voip
 description from Verizon to Avaya-LAN side
 destination-pattern 719.......
 session protocol sipv2
 session target ipv4:10.89.17.7:5060
 session transport udp
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 300 voip
 description PBX to PBX via Verizon-LAN side
 session transport udp
 incoming called-number 1719377....
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
```

```
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 301 voip
 description PBX to PBX via Verizon-WAN side
 destination-pattern 1719377....
 translate-outgoing called 1
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 400 voip
 description International call-WAN side
 destination-pattern 011T
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
```

```
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 401 voip
 description International call-LAN side
 session protocol sipv2
 session transport udp
 incoming called-number 011T
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 500 voip
 description towards Verizon for toll free-WAN side
 destination-pattern 1800T
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
```

```
 session transport udp
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 501 voip
 description towards Verizon for toll free-LAN side
 session protocol sipv2
 session transport udp
 incoming called-number 1800T
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!

dial-peer voice 600 voip
 description towards Verizon for Emergency and Directory service
 destination-pattern [4,9]11
 session protocol sipv2
 session target ipv4:63.87.147.62:5071
 session transport udp
 voice-class codec 1
```

```
 voice-class sip profiles 1
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
!
gateway
 timer receive-rtp 1200
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 60 0
 login local
!
!
end
```

Configuring the Avaya PBX

Avaya Aura Communication Manager Configuration

1. Configure the ip-network-region to assign to the SIP trunk.
2. Configure the ip-codec-set to assign to ip-network-region used by the SIP trunk.

3. Add the new signaling group.
4. Add the new trunk group.
5. Add the new route pattern.
6. Configure AAR/ARS Table entries.
7. Configure Uniform Dialing Plan.
8. Configure ISDN Public/Unknown Numbering Table entry.
9. Configure Incoming Call Handling Treatment for trunk group.

Software Versions

```
list configuration software-versions

                                SOFTWARE VERSIONS

    SOFTWARE VERSION
       Memory Resident: R017x.00.0.441.0
         Disk Resident: R017x.00.0.441.0

    TRANSLATION DATE
       Memory Resident: 10:00 pm  THU MAY 5, 2016
         Disk Resident: 10:00 pm  THU MAY 5, 2016
     Disk Second Copy: good
```

Figure 7 List configuration software versions

```
list configuration all

                              SYSTEM CONFIGURATION

Board                                                        Assigned Ports
Number   Board Type                Code      Vintage    u=unassigned t=tti p=psa

001V1    DCP MM                     MM712AP HW05 FW009 u  u  u  u  u  u  u  u
001V2    DS1 MM                     MM710BP HW16 FW050 u  u  u  u  u  u  u  u
                                                       u  u  u  u  u  u  u  u
                                                       u  u  u  u  u  u  u  u
                                                       u  u  u  u  u  u  u  u
001V3    ANA MM                     MM711AP HW31 FW091 01 u  u  u  u  u  u  u
001V9    MG-ANNOUNCEMENT          VMM-ANN              01 02 03 04 05 06 07 08
                                                       09 10 11 12 13 14 15 16
```

Figure 8: List configuration all

System Parameters IP Options

```
display system-parameters ip-options                            Page   1 of   3
                          IP-OPTIONS SYSTEM PARAMETERS

 IP MEDIA PACKET PERFORMANCE THRESHOLDS
    Roundtrip Propagation Delay (ms)    High: 800      Low: 400
                    Packet Loss (%)     High: 40       Low: 15
                    Ping Test Interval (sec): 20
    Number of Pings Per Measurement Interval: 10
                  Enable Voice/Network Stats? n
 RTCP MONITOR SERVER
   Server IPV4 Address:                 RTCP Report Period(secs): 5
               IPV4 Server Port: 5005
   Server IPV6 Address:
               IPV6 Server Port: 5005

AUTOMATIC TRACE ROUTE ON
           Link Failure? y
                                     H.323 IP ENDPOINT
 H.248 MEDIA GATEWAY                  Link Loss Delay Timer (min): 5
  Link Loss Delay Timer (min): 5        Primary Search Time (sec): 75
                                   Periodic Registration Timer (min): 20
                                 Short/Prefixed Registration Allowed? n
```

Figure 9: System Parameters IP Option

```
display system-parameters ip-options                                Page   2 of   3
                           IP-OPTIONS SYSTEM PARAMETERS

 Force Phones and Gateways to Active Survivable Servers? n

 Override ip-codec-set for SIP direct-media connections? n

 IP DTMF TRANSMISSION MODE
   Intra-System IP DTMF Transmission Mode: rtp-payload
                     Inter-System IP DTMF: See Signaling Group Forms

 HYPERACTIVE MEDIA GATEWAY REGISTRATIONS
   Enable Detection and Alarms? n
```

```
display system-parameters ip-options                                Page   3 of   3
                           IP-OPTIONS SYSTEM PARAMETERS

 SNMP PARAMETERS
    Download Flag? n
    Community String:

 SOURCE ADDRESSES
        1.                                  4.
        2.                                  5.
        3.                                  6.

 SERVICES DIAL PAD PARAMETERS
        Download Flag? n
        Password: *

 MUSIC/ANNOUNCEMENTS IP-CODEC PREFERENCES
                           Prefer use of G.711 by Music Sources? n
                    Prefer use of G.711 by Announcement Sources? n
         Prefer use of G.711 by IP Endpoints Listening to Music? n
 Prefer use of G.711 by IP Endpoints Listening to Announcements? n
```

Figure 10: System Parameters IP Option (cont.)

IP Nodes

```
display node-names ip
                                   IP NODE NAMES
    Name                  IP Address
ASM7                    10.89.17.7
CMM7                    10.89.17.25
default                 0.0.0.0
procr                   10.89.17.4
procr6                  ::













( 5  of 5    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 11: IP Nodes

IP Network Region

Location:1
Authoritative Domain:lab.tekvizion.com
Name: Plano
Codec Set: 1
Inter/Intra-region IP-IP Direct Audio: YES
dst rgn : codec Set is given as 1 and agl is given as ALL

```
display ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: lab.tekvizion.com
    Name: Plano                  Stub Network Region: n
MEDIA PARAMETERS                 Intra-region IP-IP Direct Audio: yes
      Codec Set: 1               Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                       IP Audio Hairpinning? n
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
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                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 12: IP Network Region

```
display ip-network-region 1                                    Page   3 of  20
                               IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
 Incoming LDN Extension:
 Conversion To Full Public Number - Delete:     Insert:
 Maximum Number of Trunks to Use for IGAR:
 Dial Plan Transparency in Survivable Mode? n

BACKUP SERVERS(IN PRIORITY ORDER)    H.323 SECURITY PROFILES
 1                                    1   challenge
 2                                    2
 3                                    3
 4                                    4
 5
 6                                    Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? y
                       Near End TCP Port Min: 61440
                       Near End TCP Port Max: 61444
```

```
display ip-network-region 1                                    Page   4 of  20

 Source Region: 1     Inter Network Region Connection Management      I       M
                                                                      G  A    t
 dst codec direct   WAN-BW-limits   Video       Intervening      Dyn  A  G    c
 rgn  set   WAN  Units     Total Norm  Prio Shr Regions          CAC  R  L    e
 1    1                                                               n all
 2
 3
 4
 5
 6
 7
 8
 9
 10
 11
 12
 13
 14
 15
```

Figure 13: IP Network Region (cont.)

IP Codec Set

Codec set 1 is configured for this test.
Audio Codec G.729 and G711MU are selected as Audio Codec

```
display ip-codec-set 1                                          Page   1 of   2

                          IP CODEC SET

    Codec Set: 1

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

Figure 14: IP Codec Set

Allow Direct-IP Multimedia set to 'y"
Set Maximum Call Rate for Direct-IP Multimedia: 4096:Kbits
Set Maximum Call Rate for Priority Direct-IP Multimedia: 4096:Kbits
Fax Protocol set as Passthrough

```
display ip-codec-set 1                                          Page   2 of   2

                          IP CODEC SET

                              Allow Direct-IP Multimedia? y
                Maximum Call Rate for Direct-IP Multimedia:   4096:Kbits
   Maximum Call Rate for Priority Direct-IP Multimedia:   4096:Kbits

                                                                    Packet
                      Mode                     Redundancy           Size(ms)
    FAX               pass-through              0
    Modem             off                       0
    TDD/TTY           US                        3
    H.323 Clear-channel n                       0
    SIP 64K Data      n                         0                   20
```

Figure 15: IP Codec Set (cont.)

Signaling Group

Set Group Type: sip IMS Enabled? N
Transport Method: tcp
Peer Detection Enabled?: y
Near-end Node Name: procr
Far-end Node Name: ASM7
Near-end Listen Port: 5060
Far-end Listen Port: 5060
Far-end Network Region: 1
DTMF over IP: rtp-payload
Direct IP-IP Audio Connections?: y

```
display signaling-group 3                                      Page   1 of   2
                                SIGNALING GROUP

 Group Number: 3                Group Type: sip
  IMS Enabled? n          Transport Method: tcp
        Q-SIP? n
     IP Video? n                                     Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                       Far-end Node Name: ASM7
 Near-end Listen Port: 5060                      Far-end Listen Port: 5060
                                              Far-end Network Region: 1

Far-end Domain:
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                      Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

Figure 16: Signaling Group

Trunk Group

Group number: 3
Group Type: sip
Group Name: to SM TCP
TAC: #003
Member Assignment Method: auto
Signaling Group: 3
Number of Members: 10

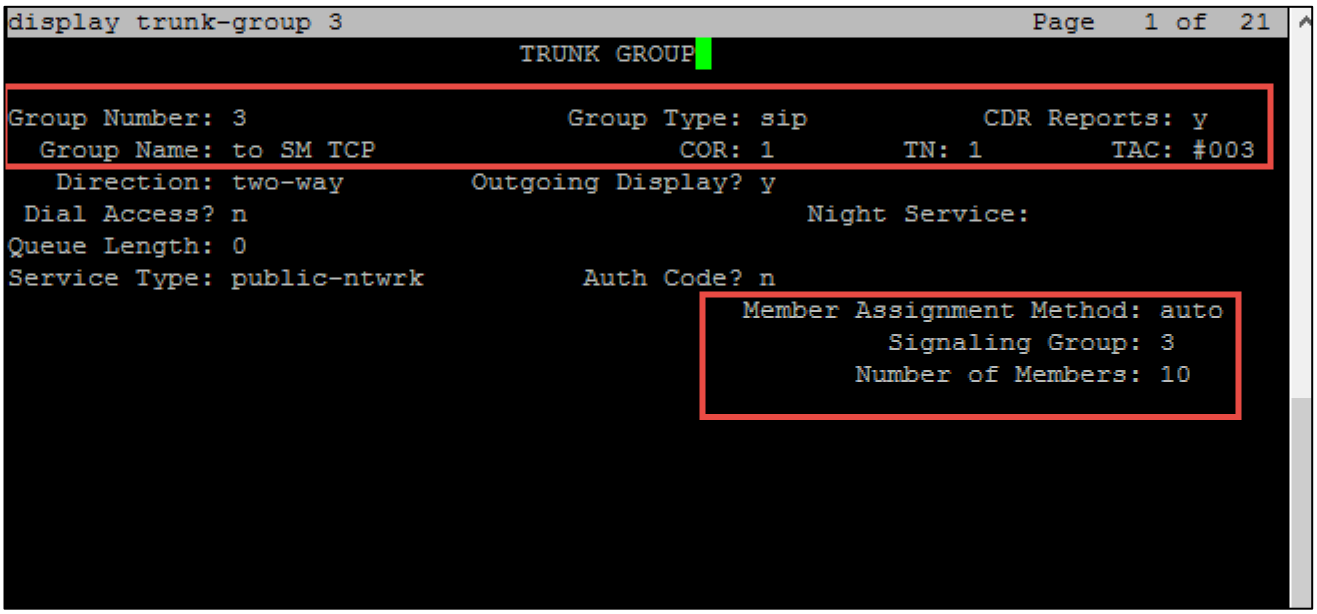


Figure 17: Trunk Group

Preferred Minimum Session Refresh Interval (sec): 900

```
display trunk-group 3                                         Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n




 Caller ID for Service Link Call to H.323 1xC: station-extension
```

Figure 18: Trunk Group (cont.)

Numbering Format: Public

```
display trunk-group 3                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n             Measured: none
                                                      Maintenance Tests? y


                     Numbering Format: public
                                                 UUI Treatment: service-provider

                                                  Replace Restricted Numbers? n
                                                 Replace Unavailable Numbers? n

                                                   Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

Figure 19: Trunk Group (cont.)

Send Diversion Header?:y
Telephone Event Payload Type: 101
Always Use re-INVITE for display Updates?:y

```
display trunk-group 3                                              Page   4 of  21
                              PROTOCOL VARIATIONS

                                      Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? y
                                  Network Call Redirection? n

                                     Send Diversion Header? y
                                   Support Request History? n
                              Telephone Event Payload Type: 101


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? y
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n

          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: called-number-only
```

Figure 20: Trunk Group (cont.)

Route Pattern

Pattern Number: 3
Pattern Name: to ASM7 tcp
Grp No: 3

```
display route-pattern 3                                              Page   1 of   3
                              Pattern Number: 3      Pattern Name: to ASM7 tcp
    SCCAN? n     Secure SIP? n      Used for SIP stations? y
    Primary SM: ASM7              Secondary SM:
    Grp FRL NPA Pfx Hop Toll No.  Inserted                                DCS/ IXC
    No          Mrk Lmt List Del  Digits                                  QSIG
                             Dgts                                         Intw
 1: 3    0                    0                                            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 Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 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 21: Route pattern

ARS Analysis

Dial String 214 is used to route calls to the Service Provider. After dialing 9 (ARS access code) plus 214xxxxxxx, all 11 digits are included in the outbound INVITE and the call is routed over Route Pattern 2.

```
display ars analysis 2                                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 2

          Dialed              Total    Route     Call    Node   ANI
          String             Min  Max  Pattern   Type    Num    Reqd
    2                        7    7     deny     hnpa           n
    214                      10   10    3        natl           n
    214242                   10   10    3        natl           n
    411                      3    3     3        svcl           n
    91                       11   11    3        natl           n
    911                      3    3     3        emer           n
    9722657262               10   10    3        natl           n
                                                                n
                                                                n
                                                                n
                                                                n
                                                                n
                                                                n
                                                                n
                                                                n
```

Figure 22: ARS Analysis

Display dialplan analysis

Dial string 65 is used to route calls to Avaya PBX extensions and dial String 9 is used for feature access code.

```
display dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all             Percent Full: 2

    Dialed    Total  Call       Dialed    Total  Call       Dialed    Total  Call
    String    Length Type       String    Length Type       String    Length Type
   0             1   attd
   5             4   ext
   65            4   ext
   7             4   ext
   8             1   fac
   9             1   fac
   *             3   fac
   #             4   dac
```

Figure 23: Display dial plan analysis

ISDN Public/Unknown Numbering Plan

The table above is used to define numbering plans.4digit extensions in the 65XX range are used by the Avaya PBX Extension and 6508 is use for Avaya one-X sip client.

```
display public-unknown-numbering 3                                Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext Ext              Trk        CPN        CPN
Len Code             Grp(s)     Prefix     Len
                                                  Total Administered: 8
 4  5                2                     4         Maximum Entries: 240
 4  53               1          571293     10
 4  748              1          972265     10     Note: If an entry applies to
 4  748              2                     4      a SIP connection to Avaya
 4  748              3          972265     10     Aura(R) Session Manager,
 4  6501             3          7193779211 10     the resulting number must
 4  6503             3          7193779213 10     be a complete E.164 number.
 4  6508             3          7193779212 10
                                                  Communication Manager
                                                  automatically inserts
                                                  a '+' digit in this case.
```

Figure 24: ISDN Public/Unknown Numbering Plan

Incoming-call-handling-treatment

The table above is used to apply changes to incoming called numbers. In this case, DID numbers provided by the Service Provider do not match extension numbers and have to be translated.

The below table is configured to change number 71937792xx to extension 65xx. This is for example and the incoming number should be translated according to the called numbers.

```
display inc-call-handling-trmt trunk-group 2                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 tie             10 5712935325       10   5003
 tie             10 5712935327       10   5001
 tie             10 5712935328       10   5000
 tie             10 5712935329       10   4006
```

Figure 25: Incoming-call-handling-treatment

Station Configuration (IP Phone)

Station:6501

Type : 9630

Port: S00023

```
display station 6501                                          Page   1 of   5
                                     STATION

Extension: 6501                          Lock Messages? n                BCC: M
     Type: 9630                          Security Code: *                 TN: 1
     Port: S00023                      Coverage Path 1:                  COR: 1
     Name:                             Coverage Path 2:                  COS: 1
                                       Hunt-to Station:                Tests? y
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 6501
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? y

                                                IP Video Softphone? y
                                 Short/Prefixed Registration Allowed: default

                                               Customizable Labels? y
```

Figure 26: Station Configuration (6501)

```
display station 6501                                                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                       Restrict Last Appearance? y
  Active Station Ringing: single
                                                         EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                     EC500 State: enabled
         Multimedia Mode: enhanced                   Audible Message Waiting? n
     MWI Served User Type:                        Display Client Redirection? n
              AUDIX Name:                        Select Last Used Appearance? n
                                                   Coverage After Forwarding? y
                                                     Multimedia Early Answer? y
 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
  Emergency Location Ext: 6501         Always Use? n IP Audio Hairpinning? y
```

Figure 27: Station Configuration 6501 (cont.)

```
display station 6501                                            Page   3 of   5
                                     STATION

             Conf/Trans on Primary Appearance? y
   Bridged Appearance Origination Restriction? y    Offline Call Logging? y


               Call Appearance Display Format: disp-param-default
                            IP Phone Group ID:
Enhanced Callr-Info Display for 1-Line Phones? n

                              ENHANCED CALL FORWARDING
                                       Forwarded Destination         Active
 Unconditional For Internal Calls To:                                   n
                   External Calls To:                                   n
          Busy For Internal Calls To:                                   n
                   External Calls To:                                   n
      No Reply For Internal Calls To:                                   n
                   External Calls To:                                   n

            SAC/CF Override: n
```

Figure 28: Station Configuration 6501 (cont.)

```
display station 6501                                          Page   4 of   5
                                   STATION
 SITE DATA
       Room:                                               Headset? n
       Jack:                                               Speaker? n
      Cable:                                              Mounting: d
      Floor:                                           Cord Length: 0
   Building:                                             Set Color: blue

ABBREVIATED DIALING
     List1:                    List2:                    List3:



BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: call-appr                          6:
 3:                                    7:
 4:                                    8:

    voice-mail
```

Figure 29: Station Configuration 6501 (cont.)

Station: 6503

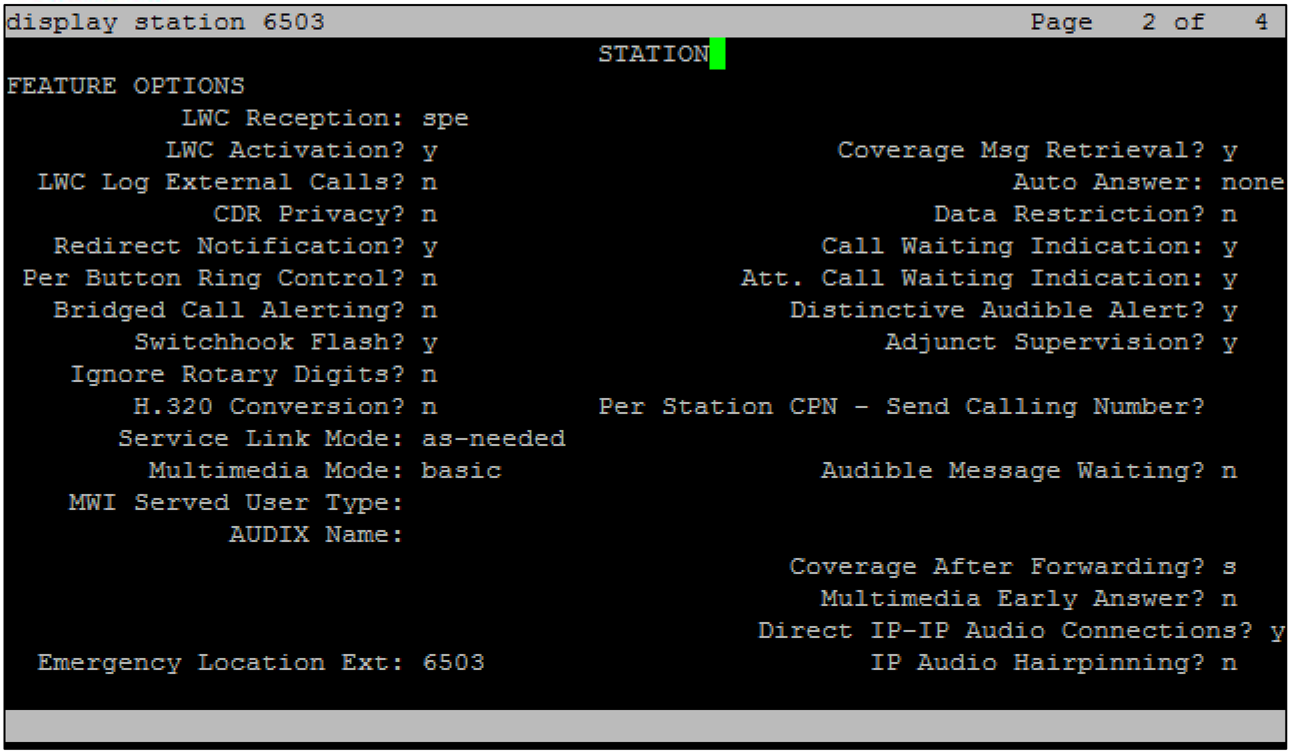
Type : 2500

Port: 001V301

Name :fax

```
display station 6503                                            Page   1 of   4
                                     STATION

Extension: 6503                         Lock Messages? n               BCC: 0
     Type: 2500                         Security Code:                  TN: 1
     Port: 001V301                    Coverage Path 1:                 COR: 1
     Name: fax                        Coverage Path 2:                 COS: 1
                                      Hunt-to Station:               Tests? y
STATION OPTIONS
      XOIP Endpoint type: auto            Time of Day Lock Table:
              Loss Group: 1            Message Waiting Indicator: none
    Off Premises Station? n



          Survivable COR: internal
   Survivable Trunk Dest? y
                                              Remote Office Phone? n



Passive Signalling Station? n
```

Figure 30: Station Configuration (6503)



```
display station 6503                                            Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                          Coverage Msg Retrieval? y
  LWC Log External Calls? n                                     Auto Answer: none
             CDR Privacy? n                                Data Restriction? n
   Redirect Notification? y                      Call Waiting Indication: y
 Per Button Ring Control? n                 Att. Call Waiting Indication: y
   Bridged Call Alerting? n                     Distinctive Audible Alert? y
        Switchhook Flash? y                           Adjunct Supervision? y
    Ignore Rotary Digits? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic                   Audible Message Waiting? n
     MWI Served User Type:
              AUDIX Name:
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
                                           Direct IP-IP Audio Connections? y
  Emergency Location Ext: 6503                          IP Audio Hairpinning? n
```

Figure 31: Station Configuration 6503 (cont.)

```
display station 6503                                            Page   3 of   4
                                     STATION


   Bridged Appearance Origination Restriction? n






                                  ENHANCED CALL FORWARDING
                                         Forwarded Destination          Active
 Unconditional For Internal Calls To:                                      n
                   External Calls To:                                      n
          Busy For Internal Calls To:                                      n
                   External Calls To:                                      n
      No Reply For Internal Calls To:                                      n
                   External Calls To:                                      n

            SAC/CF Override: n
```

Figure 32: Station Configuration 6503 (cont.)

```
display station 6503                                            Page   4 of   4
                                     STATION
 SITE DATA
       Room:                                                Headset? n
       Jack:                                                Speaker? n
      Cable:                                               Mounting: d
      Floor:                                            Cord Length: 0
   Building:                                              Set Color:



ABBREVIATED DIALING
     List1:                    List2:                    List3:

HOT LINE DESTINATION
         Abbreviated Dialing List Number (From above 1, 2 or 3):
                                                          Dial Code:

     Line Appearance: call-appr
```

Figure 33: Station Configuration 6503 (cont.)

Navigation: Home > Elements > Routing

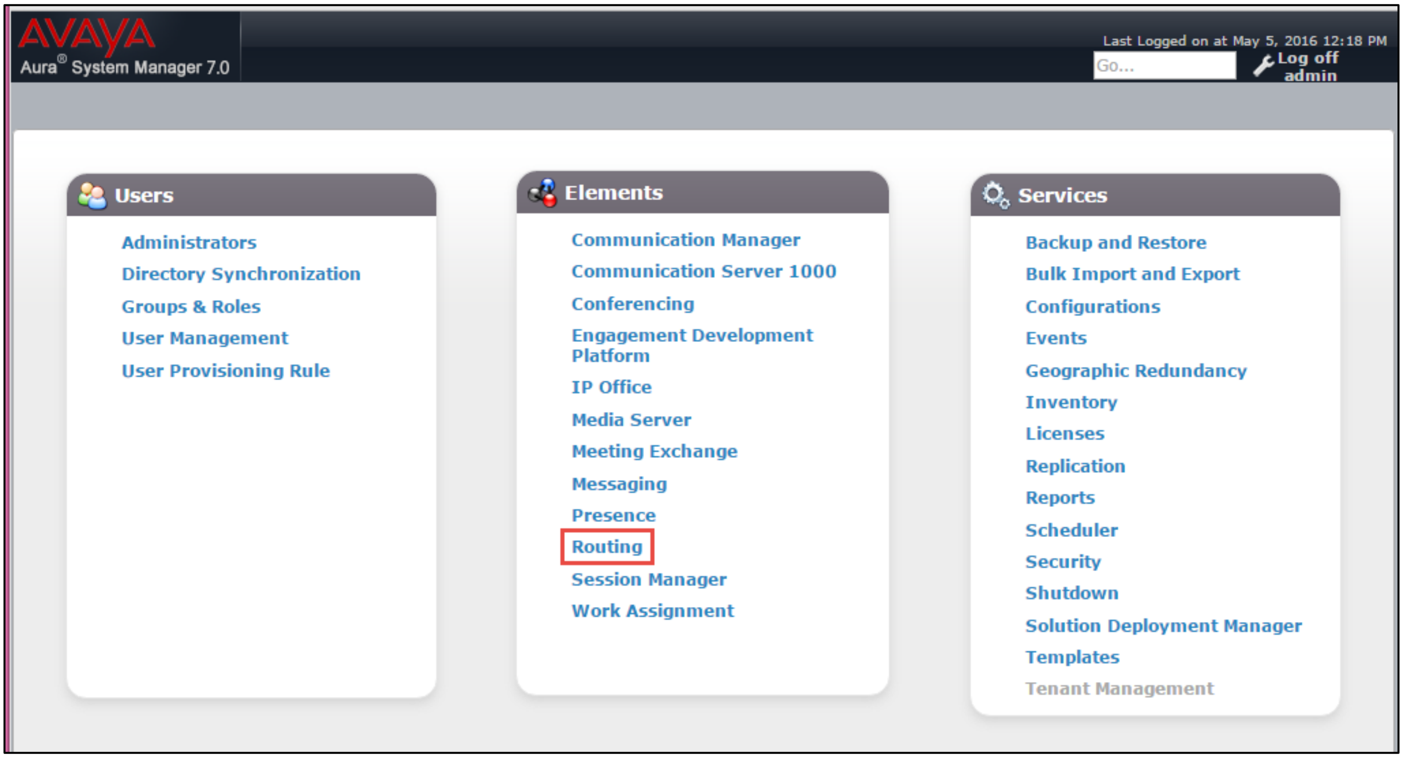


Figure 34: Avaya Aura – Session Manager Configuration

Domains

Name: lab.tekvizion.com

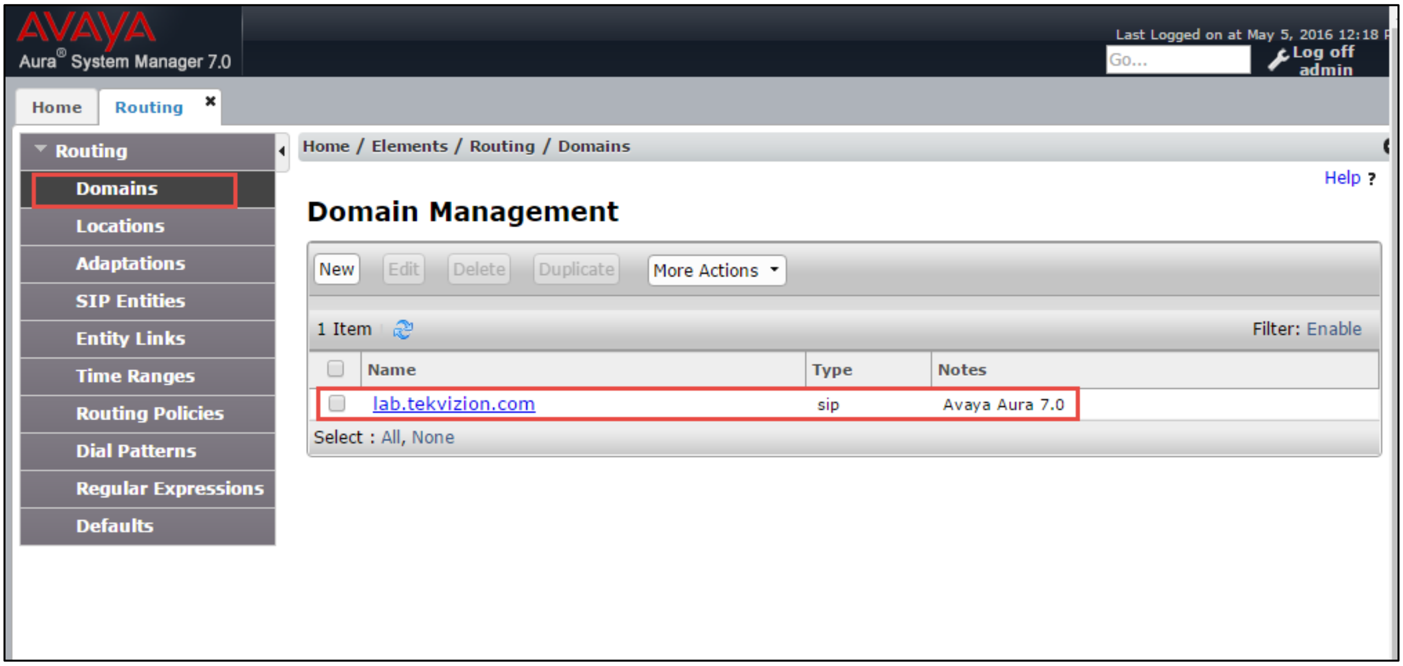


Figure 35: Avaya Aura – Domains

Locations

Name: Plano

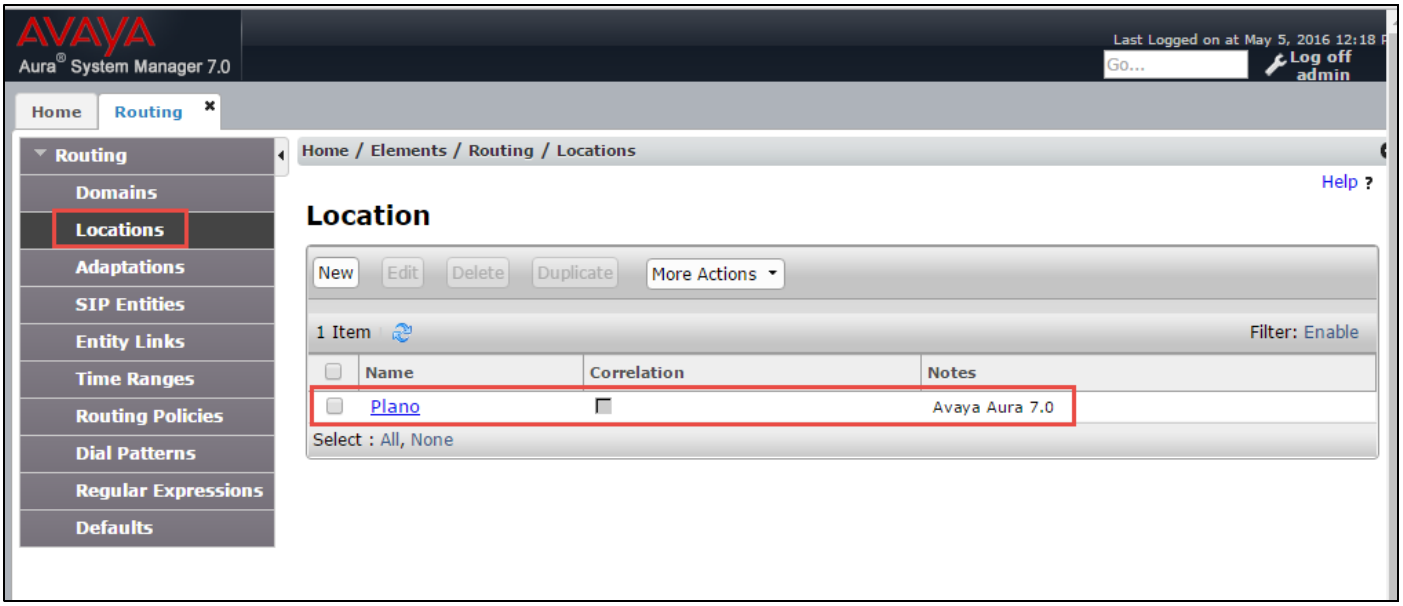
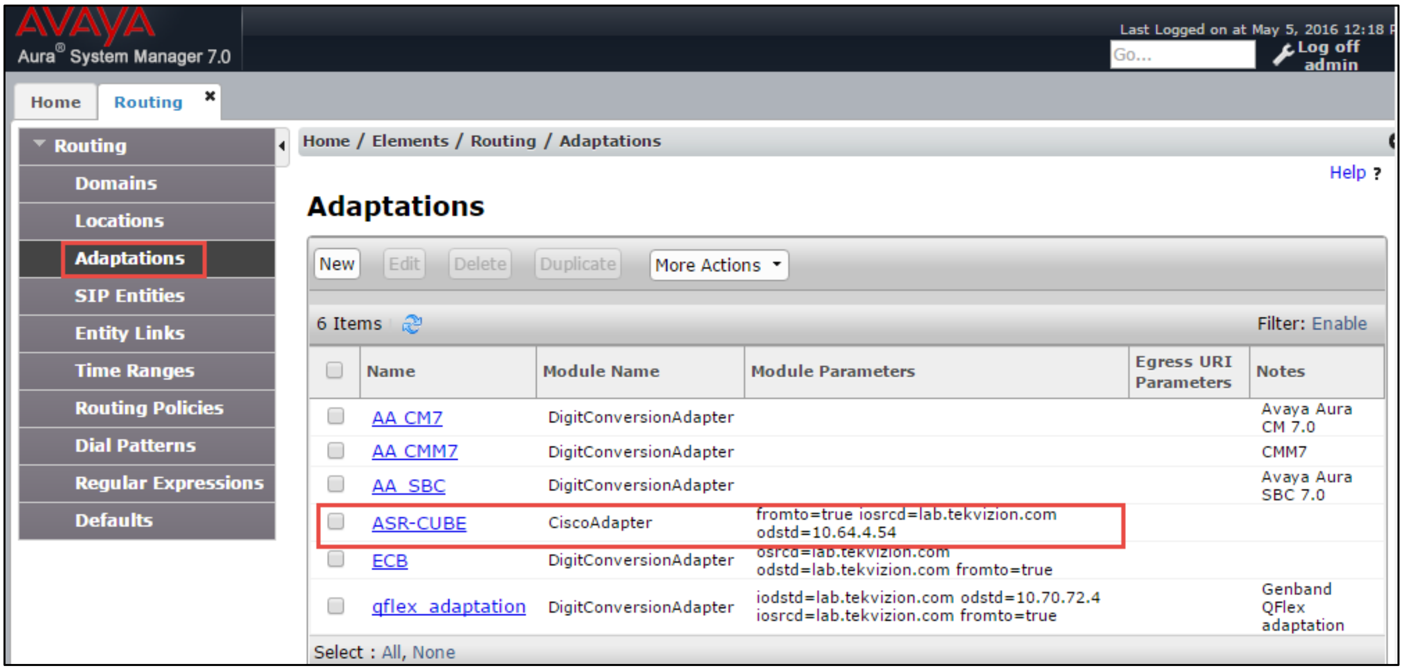


Figure 36: Avaya Aura – Locations

Module name: CiscoAdapter

Module Parameter: fromto=true odstd=10.64.4.54 iosrcd=lab.tekvizion.com



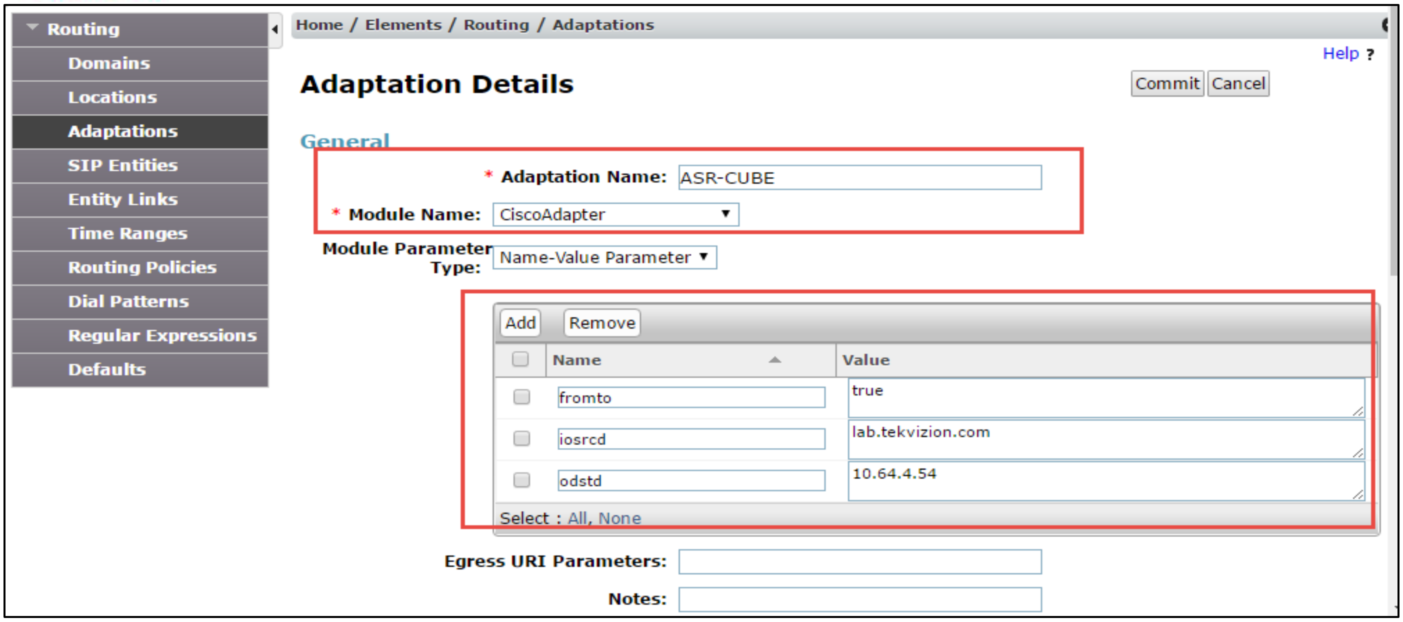


Figure 37: Avaya Aura – Adaptations

Cisco UBE IP: 10.64.4.54

Avaya CM IP: 10.89.17.4

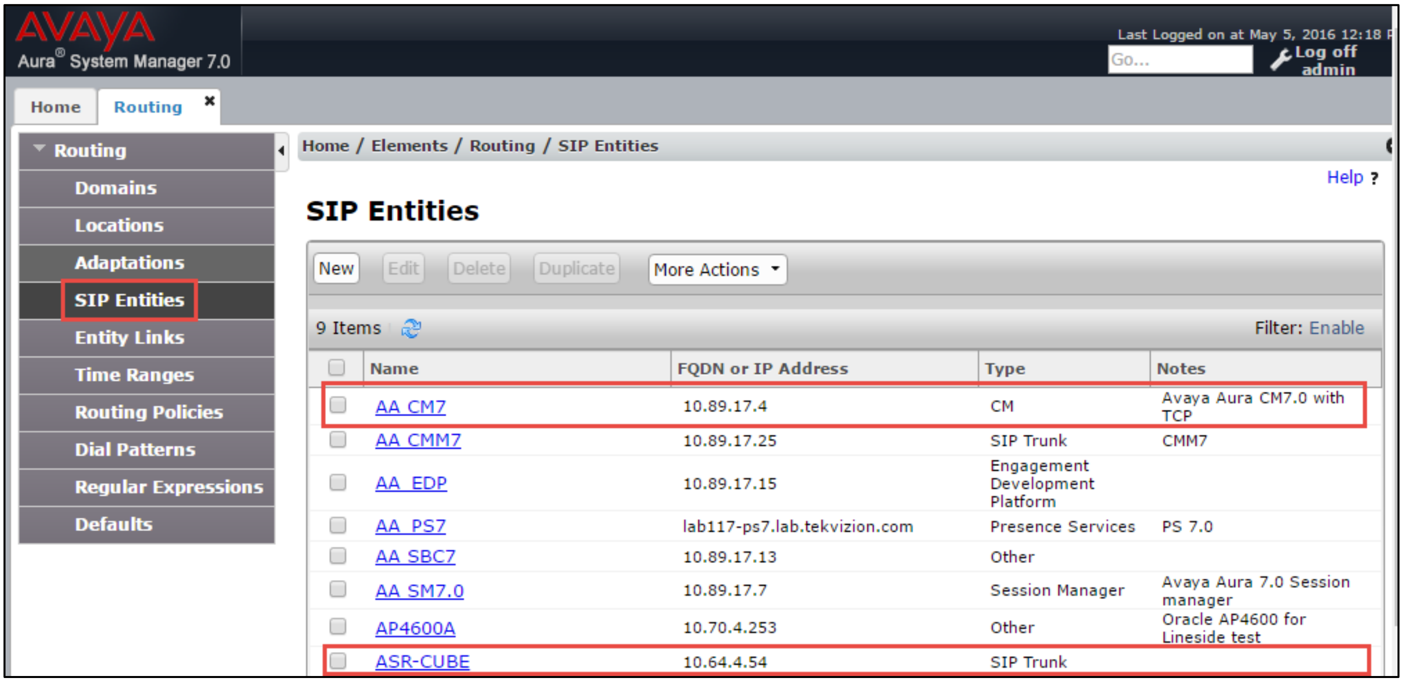


Figure 38: Avaya Aura – SIP Entities

SIP Entity for Cisco UBE

Name : ASR-CUBE

FQDN or IP Address: 10.64.4.54

Type :SIP Trunk

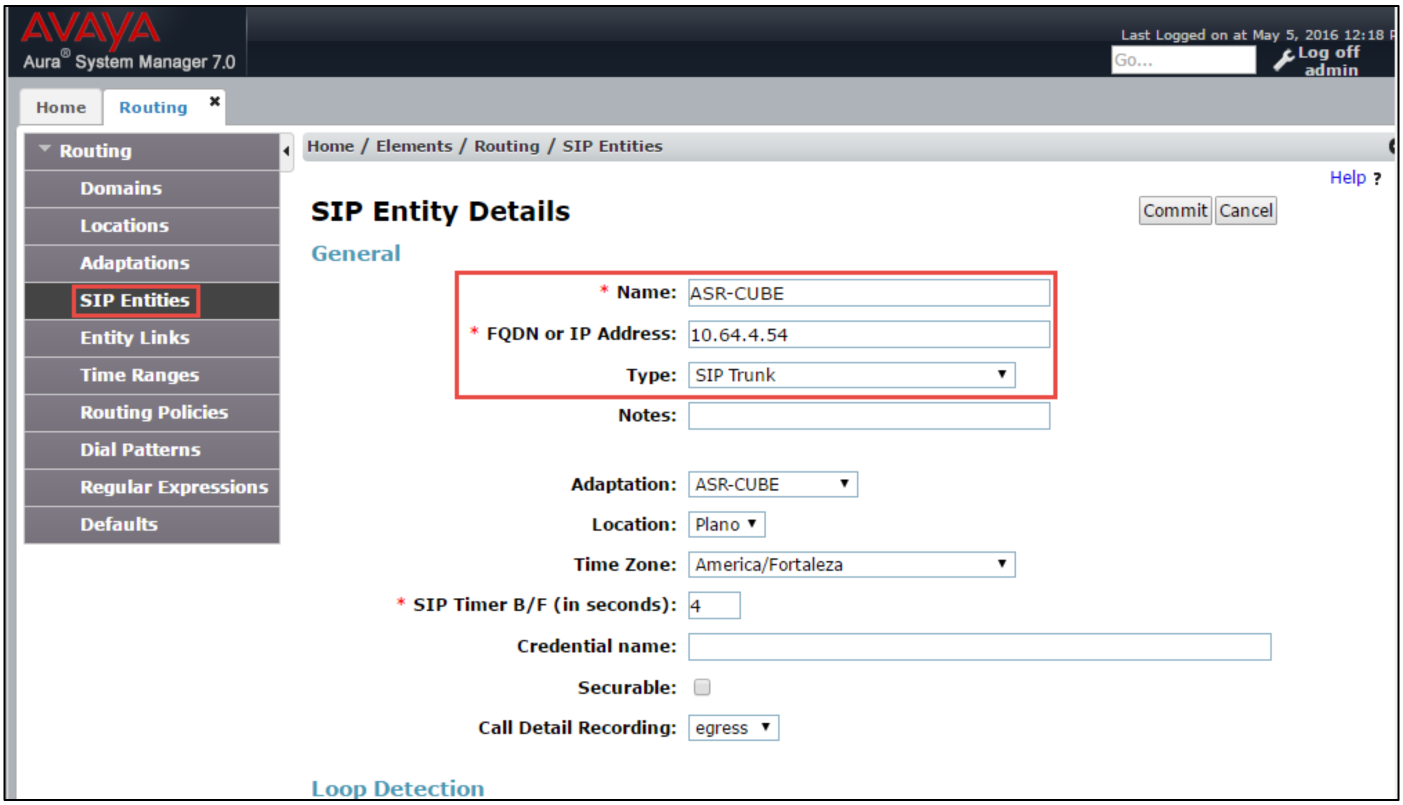


Figure 39: Avaya Aura – SIP Entities (cont.)

SIP Entity for Avaya CM

Name: AA CM7

FQDN or IP Address: 10.89.17.4

Type: CM

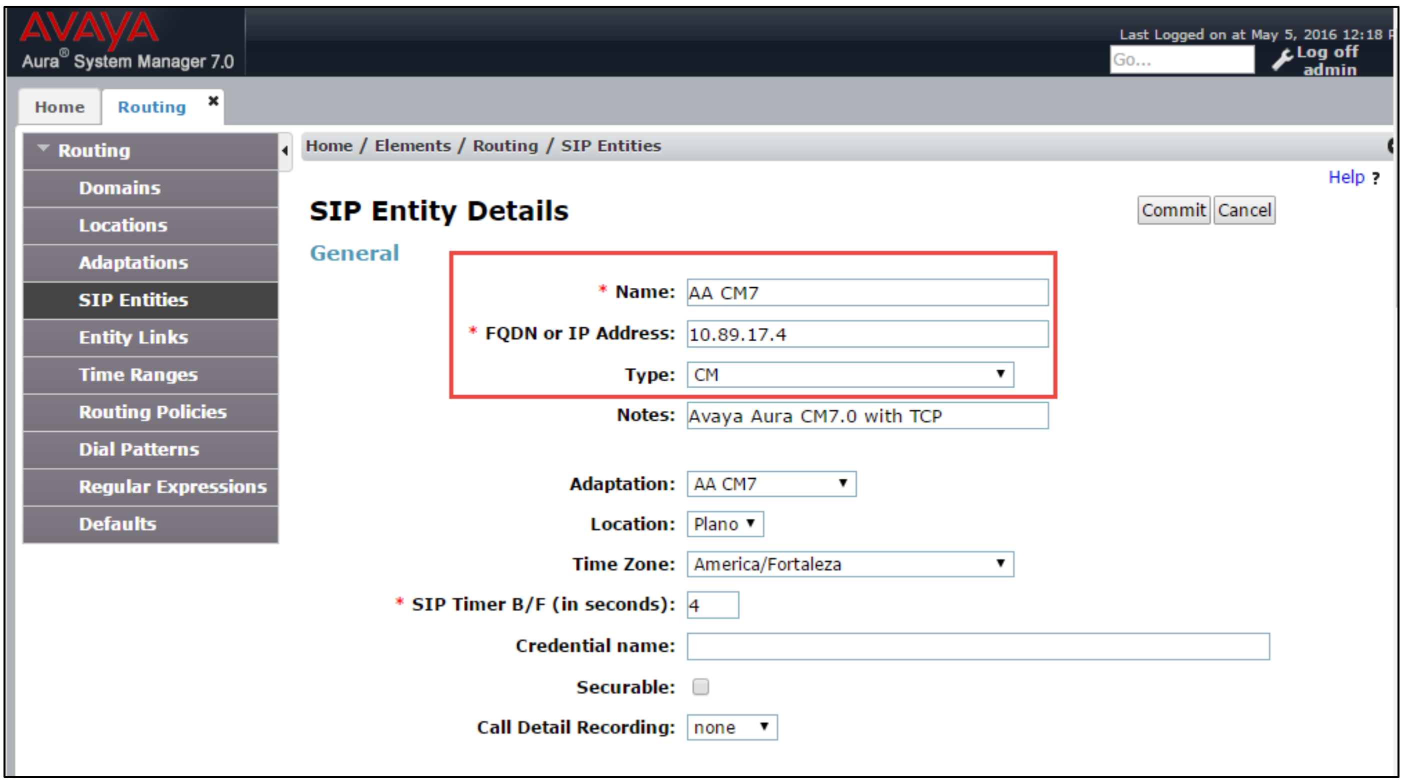


Figure 40: Avaya Aura – SIP Entities (cont.)

Entity Link

Name: Verizon_AA SM7.0_ASR-CUBE_5060_UDP , SIP Entity 1 (AA SM7.0) and SIP Entity 2(ASR-CUBE)

Name: Verizon_ASR-CUBE_to_AACM7_via_AASM7_5060_TCP, SIP Entity 1(AA SM7) and SIP Entity 2(AA CM7)

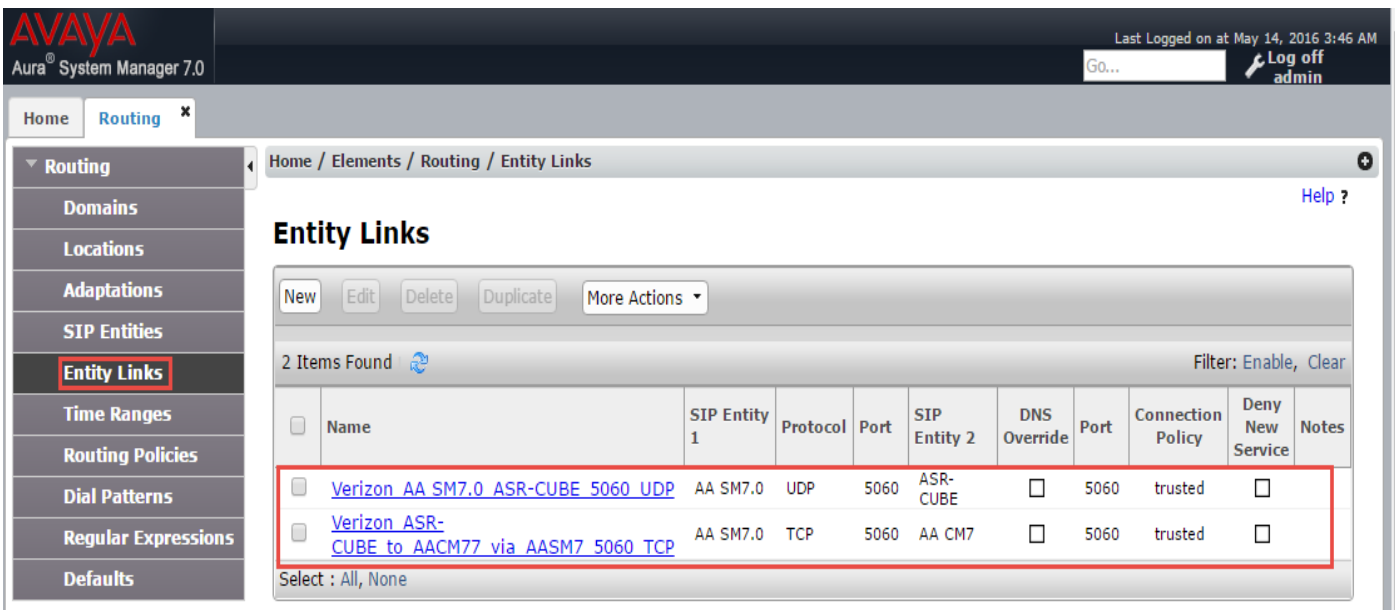


Figure 41: Avaya Aura – Entity Links

Entity Link between Avaya Session Manager and Avaya CM

SIP Entity 1: AA SM7.0

Protocol: TCP

Port: 5060

SIP Entity 2: AA CM7

Port: 5060

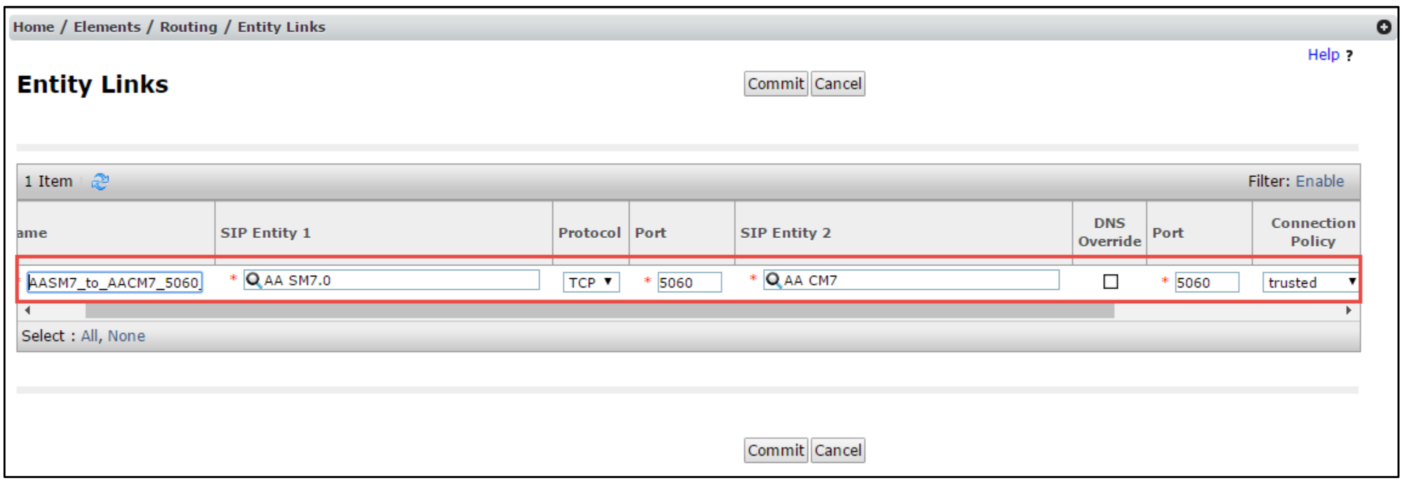


Figure 42: Avaya Aura – Entity Links cont.

Entity Link between Avaya Session Manager and Cisco UBE

SIP Entity 1: AA SM7.0

Protocol: UDP

Port: 5060

SIP Entity 2: ASR-CUBE

Port: 5060

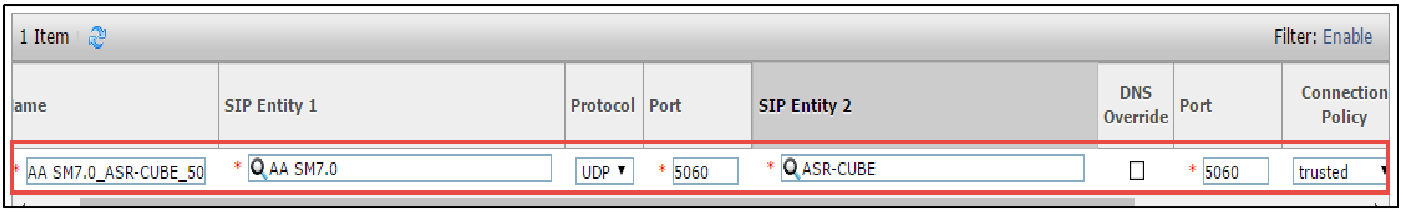


Figure 43: Avaya Aura – Entity Links cont.

Routing Policies

Routing Policy for call to go to Cisco UBE

Name: to CUBE

FQDN or IP address : 10.64.4.54

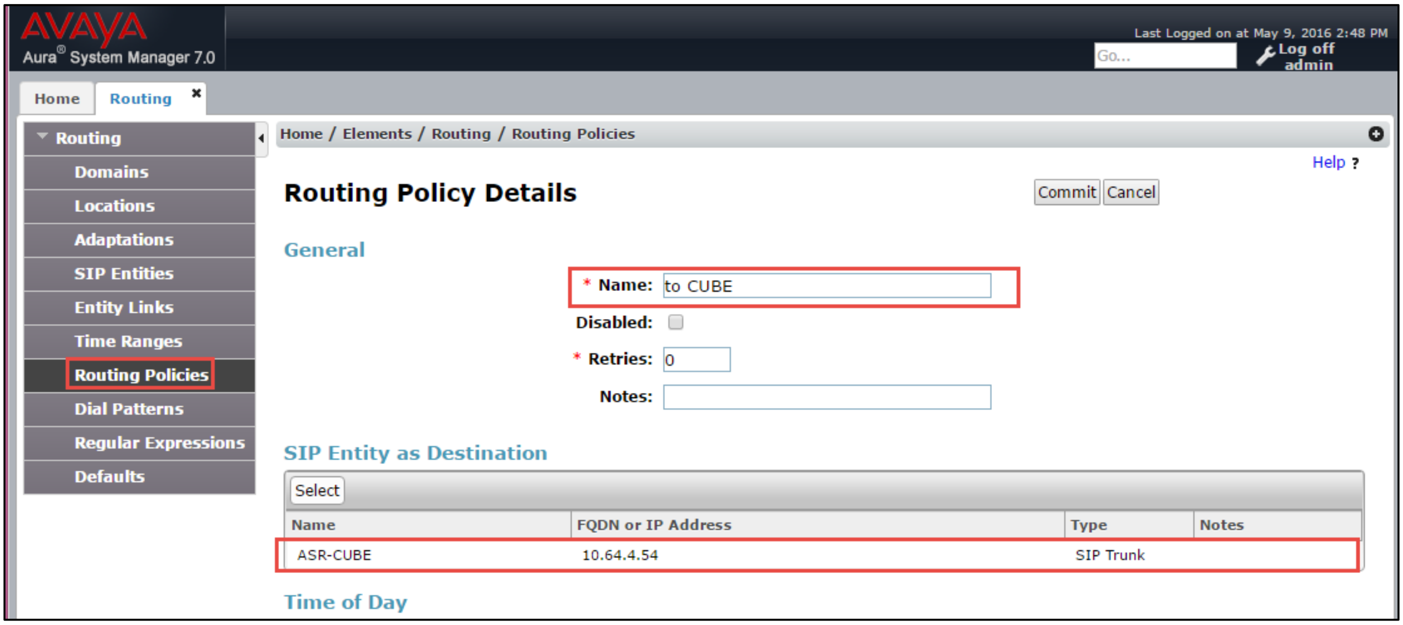


Figure 44: Avaya Aura – Routing Policies

Routing Policies Routing Policy for call to go to Avaya CM

Name: to_AACM7_TCP

FQDN or IP address : 10.89.17.4

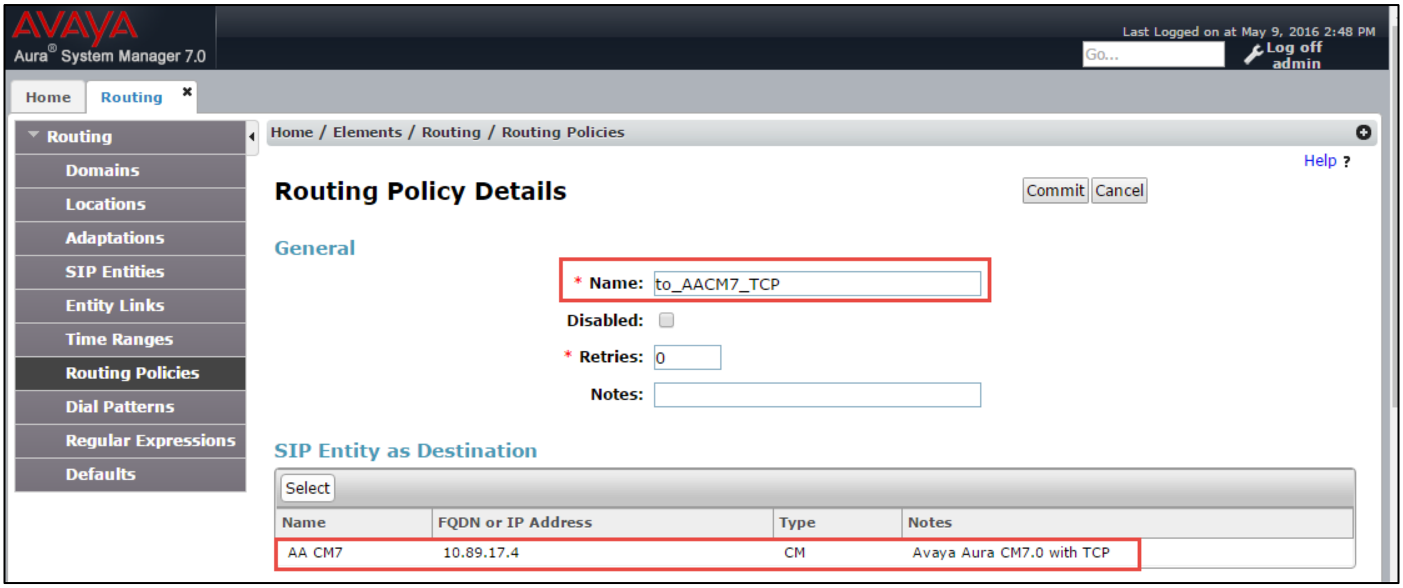


Figure 45: Avaya Aura – Routing Policies cont.

Dial Pattern

Dial Pattern to reach Cisco UBE

Dial Pattern: 214xxxxxxx

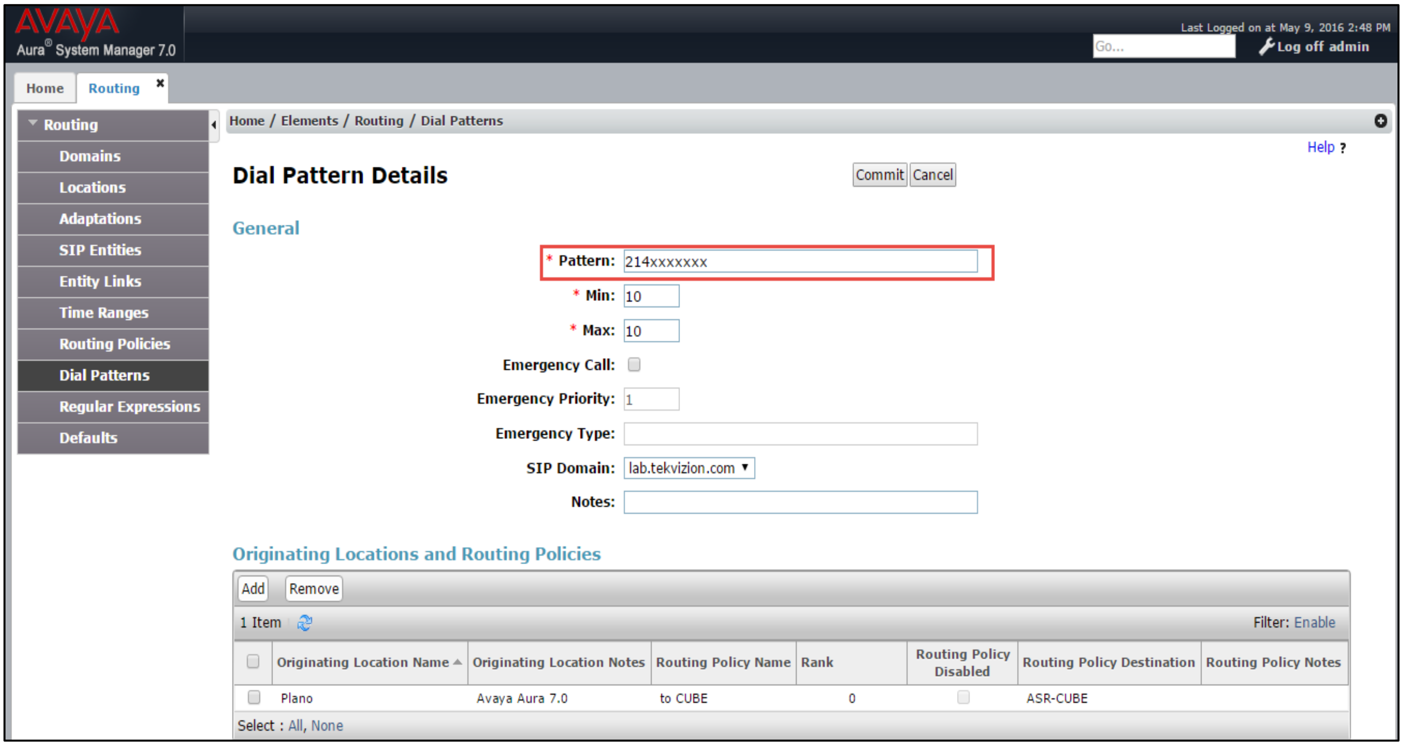


Figure 46: Avaya Aura – Dial Patterns

Dial Pattern to reach Avaya CM

Dial Pattern: 719xxxxxxx

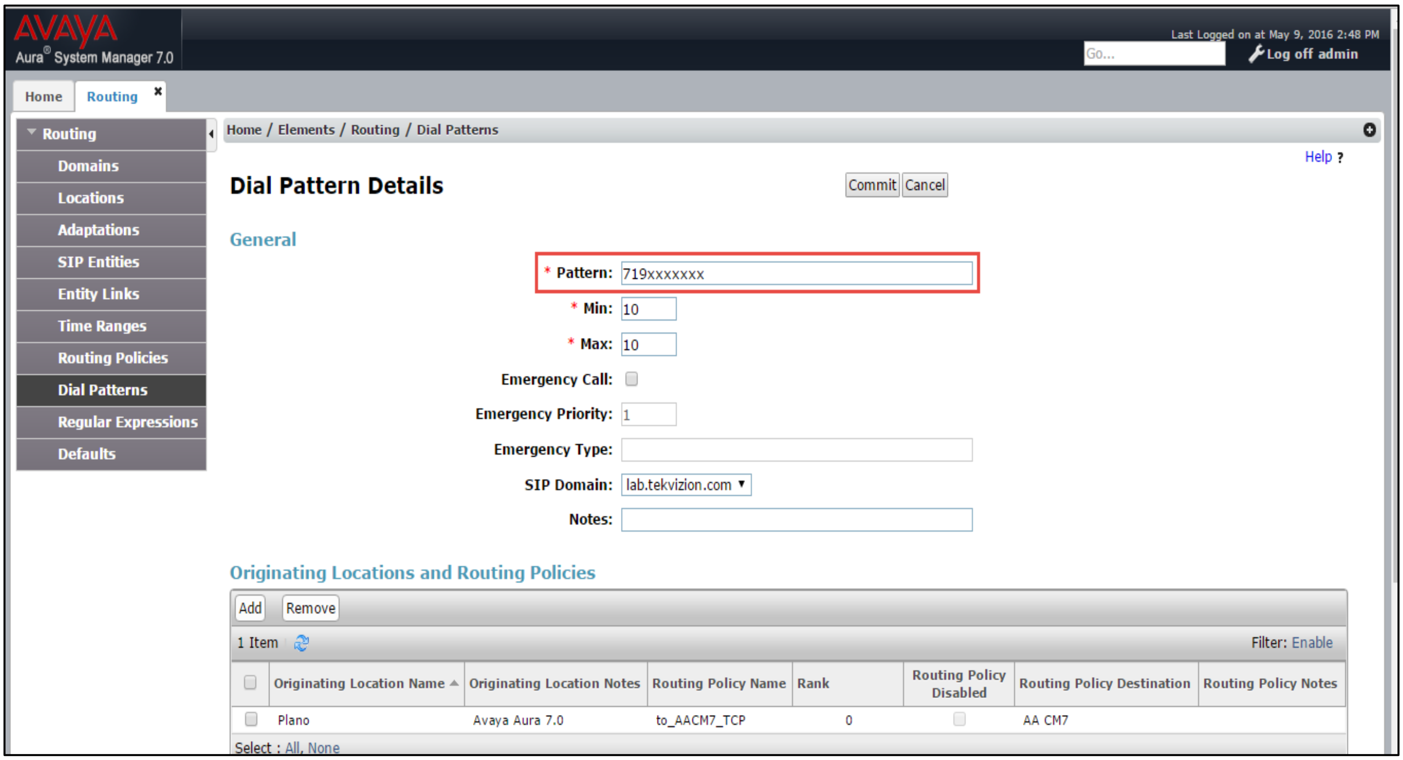


Figure 47: Avaya Aura – Dial Patterns

Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
CUBE	Cisco Unified Border Element
POP	Point of Presence
PSTN	Public Switched Telephone Network
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

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