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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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

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

### 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-npe
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-npe
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-nce
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-nce
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
dt mf-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
dt mf-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-npe
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-npe
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
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.156-1.51-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
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
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description from Verizon to Avaya-WAN side
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fax rate 14400
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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-nge
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
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destination-pattern 1719377....
translate-outgoing called 1
session protocol sipv2
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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
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no vad

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dial-peer voice 401 voip
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!
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
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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-nce

no vad
!
line con 0
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
login local
!
!
End
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.156-1.51-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
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

!

ccontrol-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-nnte
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-nnte
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-npe
dtmf-relay rtp-npe
dtmf-relay rtp-npe
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-npe
dtmf-relay rtp-npe
dtmf-relay rtp-npe
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
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 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
Figure 8: List configuration all

### System Parameters IP Options

<table>
<thead>
<tr>
<th>Board Number</th>
<th>Board Type</th>
<th>Code</th>
<th>Vintage</th>
<th>Assigned Ports</th>
</tr>
</thead>
<tbody>
<tr>
<td>001V1</td>
<td>DCP MM</td>
<td>MM712AP</td>
<td>HW05</td>
<td>u u u u u u u u</td>
</tr>
<tr>
<td>001V2</td>
<td>DS1 MM</td>
<td>MM710BP</td>
<td>HW16</td>
<td>u u u u u u u u</td>
</tr>
<tr>
<td>001V3</td>
<td>ANA MM</td>
<td>MM711AP</td>
<td>HW31</td>
<td>u u u u u u u u</td>
</tr>
<tr>
<td>001V9</td>
<td>MG-ANNOUNCEMENT</td>
<td>VMM-ANN</td>
<td>u u u u u u u u</td>
<td></td>
</tr>
</tbody>
</table>
Figure 9: System Parameters IP Option

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Roundtrip Propagation Delay (ms)</td>
<td>High: 800</td>
</tr>
<tr>
<td></td>
<td>Low: 400</td>
</tr>
<tr>
<td>Packet Loss (%)</td>
<td>High: 40</td>
</tr>
<tr>
<td></td>
<td>Low: 15</td>
</tr>
<tr>
<td>Ping Test Interval (sec)</td>
<td>20</td>
</tr>
<tr>
<td>Number of Pings Per Measurement Interval</td>
<td>10</td>
</tr>
<tr>
<td>Enable Voice/Network Stats?</td>
<td>n</td>
</tr>
<tr>
<td>RTCP Monitor Server</td>
<td></td>
</tr>
<tr>
<td>Server IPv4 Address</td>
<td></td>
</tr>
<tr>
<td></td>
<td>IPv4 Server Port: 5005</td>
</tr>
<tr>
<td>RTCP Report Period (secs)</td>
<td>5</td>
</tr>
<tr>
<td>Server IPv6 Address</td>
<td></td>
</tr>
<tr>
<td></td>
<td>IPv6 Server Port: 5005</td>
</tr>
<tr>
<td>Automatic Trace Route On</td>
<td></td>
</tr>
<tr>
<td>Link Failure?</td>
<td>y</td>
</tr>
<tr>
<td>H.323 IP Endpoint</td>
<td></td>
</tr>
<tr>
<td>Link Loss Delay Timer (min)</td>
<td>5</td>
</tr>
<tr>
<td></td>
<td>Primary Search Time (sec): 75</td>
</tr>
<tr>
<td>Periodic Registration Timer (min)</td>
<td>20</td>
</tr>
<tr>
<td>Short/Prefixed Registration Allowed?</td>
<td>n</td>
</tr>
</tbody>
</table>
display system-parameters ip-options

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

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
**IP Nodes**

<table>
<thead>
<tr>
<th>Name</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASM7</td>
<td>10.89.17.7</td>
</tr>
<tr>
<td>CMM7</td>
<td>10.89.17.25</td>
</tr>
<tr>
<td>default</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>procr</td>
<td>10.89.17.4</td>
</tr>
<tr>
<td>procr6</td>
<td>::</td>
</tr>
</tbody>
</table>

(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 10: System Parameters IP Option (cont.)**

**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

Figure 12: 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)**

<table>
<thead>
<tr>
<th>Backup Server</th>
<th>H.323 Security Profiles</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1 - challenge</td>
</tr>
<tr>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>3</td>
<td>3</td>
</tr>
<tr>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>5</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>Allow SIP URI Conversion? y</td>
</tr>
</tbody>
</table>

**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:** 61441
**IP Codec Set**

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

Figure 15: IP Codec Set (cont.)
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

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](image-url)
Preferred Minimum Session Refresh Interval (sec): 900

Figure 18: Trunk Group (cont.)
Numbering Format: Public

```
display trunk-group 3
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
Figure 20: Trunk Group (cont.)
Route Pattern

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

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.

<table>
<thead>
<tr>
<th>Dailed String</th>
<th>Total Min</th>
<th>Total Max</th>
<th>Route Pattern</th>
<th>Call Type</th>
<th>Node Num</th>
<th>ANI Reqd</th>
</tr>
</thead>
<tbody>
<tr>
<td>214</td>
<td>10</td>
<td>10</td>
<td>3</td>
<td>natl</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>214242</td>
<td>10</td>
<td>10</td>
<td>3</td>
<td>natl</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>411</td>
<td>3</td>
<td>3</td>
<td>3</td>
<td>svcl</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>91</td>
<td>11</td>
<td>11</td>
<td>3</td>
<td>natl</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>911</td>
<td>3</td>
<td>3</td>
<td>3</td>
<td>emerg</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>9722657262</td>
<td>10</td>
<td>10</td>
<td>3</td>
<td>natl</td>
<td>n</td>
<td></td>
</tr>
</tbody>
</table>

Figure 22: ARS Analysis
Dial string 65 is used to route calls to Avaya PBX extensions and dial String 9 is used for feature access code.

<table>
<thead>
<tr>
<th>Dialed String</th>
<th>Total Call Length</th>
<th>Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>1</td>
<td>attd</td>
</tr>
<tr>
<td>5</td>
<td>4</td>
<td>ext</td>
</tr>
<tr>
<td>65</td>
<td>4</td>
<td>ext</td>
</tr>
<tr>
<td>7</td>
<td>4</td>
<td>ext</td>
</tr>
<tr>
<td>0</td>
<td>1</td>
<td>fac</td>
</tr>
<tr>
<td>9</td>
<td>1</td>
<td>fac</td>
</tr>
<tr>
<td>*</td>
<td>3</td>
<td>fac</td>
</tr>
<tr>
<td>#</td>
<td>4</td>
<td>dac</td>
</tr>
</tbody>
</table>

Figure 23: Display dial plan analysis
ISDN Public/Unknown Numbering Plan

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

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.

![Figure 25: Incoming-call-handling-treatment](image-url)
**Station Configuration (IP Phone)**

Station: 6501  
Type: 9630  
Port: S00023

![Station Configuration Screen](image)

---

Figure 26: Station Configuration (6501)
Figure 27: Station Configuration 6501 (cont.)
Figure 28: Station Configuration 6501 (cont.)

```
display station 6501

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

<table>
<thead>
<tr>
<th>ENHANCED CALL FORWARDING</th>
<th>Forwarded Destination</th>
<th>Active</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unconditional For Internal Calls To:</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>External Calls To:</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>Busy For Internal Calls To:</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>External Calls To:</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>No Reply For Internal Calls To:</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>External Calls To:</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>SAC/CF Override:</td>
<td>n</td>
<td></td>
</tr>
</tbody>
</table>
```
Figure 29: Station Configuration 6501 (cont.)
Station: 6503
Type: 2500
Port: 001V301
Name: fax

Figure 30: Station Configuration (6503)
<table>
<thead>
<tr>
<th>Feature Options</th>
<th>Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>LWC Reception</td>
<td>spe</td>
</tr>
<tr>
<td>LWC Activation</td>
<td>y</td>
</tr>
<tr>
<td>LWC Log External Calls</td>
<td>n</td>
</tr>
<tr>
<td>CDR Privacy</td>
<td>n</td>
</tr>
<tr>
<td>Redirect Notification</td>
<td>y</td>
</tr>
<tr>
<td>Per Button Ring Control</td>
<td>n</td>
</tr>
<tr>
<td>Bridged Call Alerting</td>
<td>n</td>
</tr>
<tr>
<td>Switchhook Flash</td>
<td>y</td>
</tr>
<tr>
<td>Ignore Rotary Digits</td>
<td>n</td>
</tr>
<tr>
<td>H.320 Conversion</td>
<td>n</td>
</tr>
<tr>
<td>Service Link Mode</td>
<td>as-needed</td>
</tr>
<tr>
<td>Multimedia Mode</td>
<td>basic</td>
</tr>
<tr>
<td>MWI Served User Type</td>
<td></td>
</tr>
<tr>
<td>AUDIX Name</td>
<td></td>
</tr>
<tr>
<td>Coverage Msg Retrieval</td>
<td>y</td>
</tr>
<tr>
<td>Auto Answer</td>
<td>none</td>
</tr>
<tr>
<td>Data Restriction</td>
<td>n</td>
</tr>
<tr>
<td>Call Waiting Indication</td>
<td>y</td>
</tr>
<tr>
<td>Att. Call Waiting Indication</td>
<td>y</td>
</tr>
<tr>
<td>Distinctive Audible Alert</td>
<td>y</td>
</tr>
<tr>
<td>Adjunct Supervision</td>
<td>y</td>
</tr>
<tr>
<td>Per Station CPN</td>
<td></td>
</tr>
<tr>
<td>Send Calling Number</td>
<td></td>
</tr>
<tr>
<td>Audible Message Waiting</td>
<td>n</td>
</tr>
<tr>
<td>Coverage After Forwarding</td>
<td>s</td>
</tr>
<tr>
<td>Multimedia Early Answer</td>
<td>n</td>
</tr>
<tr>
<td>Direct IP-IP Audio Conn</td>
<td>y</td>
</tr>
<tr>
<td>IP Audio Hairpinning</td>
<td>n</td>
</tr>
</tbody>
</table>

Figure 31: Station Configuration 6503 (cont.)
Figure 32: Station Configuration 6503 (cont.)
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
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

![Entity Link Diagram]

**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: 214xxxxxxx

Figure 46: Avaya Aura – Dial Patterns
Dial Pattern to reach Avaya CM

Dial Pattern: 719xxxxxxx

Figure 47: Avaya Aura – Dial Patterns
## Acronyms

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<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
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<td>CUBE</td>
<td>Cisco Unified Border Element</td>
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<tr>
<td>POP</td>
<td>Point of Presence</td>
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<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
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<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
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<td>SIP</td>
<td>Session Initiation Protocol</td>
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