Verizon SIP Trunking:

Avaya Aura Communication Manager 6.3 Via Avaya Aura Session Manager 6.3 with Cisco Unified Border Element 15.5(1) T1 using SIP to Verizon Service Provider.

April 24, 2015
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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) 15.5(1)T1 can be used. The Cisco Unified Border Element 15.5(1)T1 provides demarcation, security, interworking and session control services for Avaya Aura Communication Manager 6.3 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 6.3, Avaya Aura Session Manager 6.3 and Cisco Unified Border Element (Cisco UBE) 15.5(1)T1 for connectivity to Verizon SIP trunking service. The deployment model covered in this application note is CPE (Avaya Aura Communication Manager 6.3.0) 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 UBE (Active) on Cisco ISR 2951/K9 (revision 1.1) with 1835264K/261888K bytes of memory with 3 Gigabit Ethernet interfaces
- Cisco UBE (Standby) on Cisco ISR 2921/K9 (revision 1.0) with 483328K/40960K bytes of memory with 3 Gigabit Ethernet interfaces.
- 2 Avaya 4610sw series IP telephone (H323)

Software Requirements
- IOS Version 15.5(1)T1 for Cisco ISR 2951 Cisco Unified Border Element.
- Avaya Aura Communication Manager release 6.3 Service Pack 10 (System Platform 6.3.10.0)
- Avaya Aura Session Manager 6.3
- Avaya one-X Communicator Release 6.2.4

Features
This section lists supported and unsupported features. No deviation from the configuration presented in this document will be supported by Cisco. Please see the Caveats section below for more information.

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 G711 pass through

Features Not Supported
- T.38 fax is not supported on the Verizon SIP trunk

Caveats
- Avaya does not send Fax re-INVITE for any inbound Fax scenarios however Fax re-INVITE is sent from network instead of Avaya PBX
- Inbound Fax over G711 pass through fails when CPE is configured to SG3 but passed in case of G3.
- Attended and blind Call transfer scenarios work only if SIP REFER is disabled in Avaya Aura Communication Manager
- Blind transfer scenario is supported only on Avaya one-X soft client.
After an attended or unattended call transfer from a PBX phone to an off-net phone, the caller ID on the off-net phone is not updated as expected.

**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.

```plaintext
interface GigabitEthernet0/0
description LAN to Avaya
ip address 10.64.3.228 255.255.0.0
standby 6 ip 10.64.3.43
standby 6 priority 50
standby 6 preempt
standby 6 track 2 decrement 10
duplex auto
speed auto
!
interface GigabitEthernet0/1
description WAN to Verizon
ip address 192.65.79.110 255.255.255.224
standby 1 ip 192.65.79.126
standby 1 priority 50
standby 1 preempt
standby 1 name SB
standby 1 track 1 decrement 10
duplex auto
speed auto
```
Global Cisco UBE settings

voice service voip

no ip address trusted authenticate
address-hiding
mode border-element
allow-connections sip to sip
redundancy
fax protocol pass-through g711ulaw

sip
session refresh
asserted-id pai
privacy pstn
early-offer forced
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

Verizon offers only G.729 codec for voice call. It allows codecs other than G.729 but will only accept G.729.

For customers using **G.729** codec:
voice class codec 1
codec preference 1 g729r8

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

For Incoming calls from Verizon:
dial-peer voice 2001 voip
description incoming call from Verizon facing PBX
session protocol sipv2
incoming called-number 57129353..
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 1999 voip
description incoming call from Verizon facing Verizon
destination-pattern 5712.....
session protocol sipv2
session target ipv4:10.70.4.7:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad

For outgoing calls to Verizon:
dial-peer voice 2000 voip
description outgoing call to Verizon facing Verizon network
translation-profile outgoing remv_plus
huntstop
destination-pattern [2-9]T
session protocol sipv2
session target ipv4:63.87.147.48:5071
voice-class codec 1
voice-class sip rel1xx supported "100rel"
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 12345 voip
description outgoing call to Verizon facing PBX
translation-profile outgoing remv_plus
session protocol sipv2
incoming called-number [2-9]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
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**
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:
Avaya_CUBE_VZ#sh run
Building configuration...

Current configuration : 6226 bytes

! Last configuration change at 10:42:49 UTC Tue Apr 21 2015 by cisco

! version 15.5
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers

! hostname Avaya_CUBE_VZ

! boot-start-marker
boot system flash c2951-universalk9-mz.SPA.154-2.T1.bin
boot-end-marker

! no logging rate-limit
no logging console
enable secret 4 sKPgCY/XPea3wk8xoeSWo7UGFaNVwzXDEyXWhuDjeLk

! ipc zone default
association 1
no shutdown
protocol sctp
  local-port 5000
  local-ip 10.64.3.228
  remote-port 5000
  remote-ip 10.64.3.229

! no aaa new-model

bsd-client server url https://cloudsso.cisco.com/as/token.oauth2

! no ip domain lookup
ip name-server 10.64.1.3
ip cef
no ipv6 cef

multilink bundle-name authenticated
password encryption aes
ccts logging verbose
!
!
voice-card 0
dspfarm
dsp services dspfarm
!
!
voice service voip
address-hiding
mode border-element
allow-connections sip to sip
redundancy
no supplementary-service sip handle-replaces
fax protocol pass-through g711ulaw
sip
session refresh
asserted-id pai
privacy pstn
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
g729 annexb-all
!
voice class codec 2
codec preference 1 g711ulaw
codec preference 2 g729r8 bytes 30
codec preference 3 g726r32
codec preference 4 g729br8
!
voice class codec 3
 codec preference 1 g729br8
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g726r32
 codec preference 4 g729br8
!
voice class sip-profiles 10
  request REFER sip-header Referred-By remove
!
voice class sip-profiles 109
!
voice class sip-profiles 110
!
!
!
!
!
!
!
!
!
!
!
!
voice translation-rule 1
 rule 1 /\+\[(571.......)\]/ /\1/ 
!
voice translation-rule 2
rule 1 /\+/\([571........]/\1/  

!  

!  

voice translation-profile remv_plus  
translate calling 2  
translate called 1  

!  

!  

!  

license udi pid CISCO2951/K9 sn FTX1509AJJV  
hw-module pvdm 0/0  

!  

!  

!  

username CISCO privilege 15 password 0 tekV1z10n  

!  

redundancy inter-device  
scheme standby SB  

!  

!  

redundancy  

!  

!  

track 1 interface GigabitEthernet0/1 line-protocol  

!  

track 2 interface GigabitEthernet0/0 line-protocol  

!  

!  

translation-rule 10  
Rule 1 2142425980 5712935329
interface Embedded-Service-Engine0/0
  no ip address
  shutdown

interface GigabitEthernet0/0
  description LAN to Avaya
  ip address 10.64.3.228 255.255.0.0
  standby 6 ip 10.64.3.43
  standby 6 priority 50
  standby 6 preempt
  standby 6 track 2 decrement 10
  duplex auto
  speed auto

interface GigabitEthernet0/1
  description WAN to Verizon
  ip address 192.65.79.110 255.255.255.224
  standby 1 ip 192.65.79.126
  standby 1 priority 50
  standby 1 preempt
  standby 1 name SB
  standby 1 track 1 decrement 10
  duplex auto
  speed auto

interface GigabitEthernet0/2
no ip address
shutdown
duplex auto
speed 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 10.0.0.0 255.0.0.0 10.64.1.1
ip route 172.16.0.0 255.255.0.0 10.64.1.1
!
!

lsl resp-timeout 1

control-plane
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
mgcp behavior rsip-range tgc-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
sccp local GigabitEthernet0/0
sccp ccm 10.80.18.3 identifier 1 version 7.0
sccp
!
sccp ccm group 1
  bind interface GigabitEthernet0/0
  associate ccm 1 priority 1
  associate profile 1 register conference
!
!
!
dspfarm profile 1 conference
  codec g729br8
  codec g729r8
  codec g729abr8
  codec g729ar8
  codec g711alaw
  codec g711ulaw
  codec g722-64
  associate application SCCP
  shutdown
!
dial-peer voice 2001 voip
  description incoming call from Verizon facing PBX
  session protocol sipv2
  incoming called-number 57129353..
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 1999 voip
description incoming call from Verizon facing Verozon network
destination-pattern 5712......
session protocol sipv2
session target ipv4:10.70.4.7:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 2000 voip
description outgoing call to Verizon facing Verizon network
translation-profile outgoing remv_plus
huntstop
destination-pattern [2-9]T
session protocol sipv2
session target ipv4:63.87.147.48:5071
voice-class codec 1
voice-class sip rel1xx supported "100rel"
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 12345 voip
description outgoing call to Verizon facing PBX
translation-profile outgoing remv_plus
session protocol sipv2
incoming called-number [2-9]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad

gateway
  media-inactivity-criteria all
  timer receive-rtcp 5
  timer receive-rtp 86400
!
sip-ua
  no remote-party-id
!
!
!
gatekeeper
  shutdown
!
!
!
line con 0
line aux 0
line 2
  no activation-character
  no exec
  transport preferred none
  transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
  stopbits 1
line vty 0 4
  exec-timeout 960 0
  logging synchronous
  login local
  transport input all
! scheduler allocate 20000 1000
!
end

Standby Cisco UBE
User Access Verification
Username: cisco
Password:
Avaya_Cube_VZ2#sh run
Building configuration...

Current configuration : 5365 bytes
!
version 15.5
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname Avaya_Cube_VZ2
!
boot-start-marker
boot system flash c2900-universalk9-mz.SSA-eng-sp-153-3.M1.bin
boot-end-marker
!
aqm-register-fnf
!
logging queue-limit 1000000000
logging buffered 30000000
logging rate-limit 10000
no logging console
no logging monitor
enable secret 4 Pe0NhiiWw5iXZpE.k5VhTSCoGPcuVeRyrer9kEPz20Z6

ipc zone default
association 1
no shutdown
protocol sctp
local-port 5000
  local-ip 10.64.3.229
remote-port 5000
  remote-ip 10.64.3.228

no aaa new-model
bsd-client server url https://cloudsso.cisco.com/as/token.oauth2
no ip domain lookup
ip cef
no ipv6 cef

multilink bundle-name authenticated

cts logging verbose

voice-card 0
dspfarm
dsp services dspfarm

voice service voip
no ip address trusted authenticate
address-hiding
mode border-element
allow-connections sip to sip
redundancy
no supplementary-service sip handle-replaces
fax protocol pass-through g711ulaw
sip
session refresh
asserted-id pai
privacy pstn
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
g729 annexb-all
sip-profiles 1
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g726r32
codec preference 4 g729br8
!
voice class codec 2
codec preference 1 g711ulaw
codec preference 2 g729r8 bytes 30
codec preference 3 g726r32
codec preference 4 g729br8
!
voice class codec 3
codec preference 1 g729br8
!
!
!
!
!
!
!
!
!
!
!
voice translation-rule 1
rule 1 \+/\(571\ldots\)/ \+/1/

voice translation-rule 2

rule 1 \+/\(571\ldots\)/ \+/1/

voice translation-profile remv_plus

translate calling 2

translate called 1

license udi pid CISCO2921/K9 sn FTX1746AJCB

hw-module pvdm 0/0

username cisco privilege 15 password 0 cisco

redundancy inter-device

scheme standby SB

redundancy

track 1 interface GigabitEthernet0/1 line-protocol
track 2 interface GigabitEthernet0/0 line-protocol

interface Embedded-Service-Engine0/0
  no ip address
  shutdown

interface GigabitEthernet0/0
  description WAN to Verizon
  ip address 192.65.79.111 255.255.255.224
  standby 1 ip 192.65.79.126
  standby 1 priority 50
  standby 1 preempt delay minimum 10
  standby 1 name SB
  standby 1 track 1 decrement 10
duplex auto
  speed auto

interface GigabitEthernet0/1
  description LAN Interface
ip address 10.64.3.229 255.255.0.0
standby 6 ip 10.64.3.43
standby 6 priority 50
standby 6 preempt delay minimum 10
standby 6 track 2 decrement 10
duplex auto
speed auto
!
interface GigabitEthernet0/2
no ip address
shutdown
duplex auto
speed auto
!
interface Ethernet0/2
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 172.16.0.0 255.255.0.0 10.64.1.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
dial-peer voice 2001 voip
description incoming call from Verizon facing PBX
session protocol sipv2
incoming called-number 57129353..
voice-class codec 1
dtmf-relay rtp-npe
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 1999 voip
description incoming call from Verizon facing Verozon network
destination-pattern 5712......
session protocol sipv2
session target ipv4:10.70.4.7:5060
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-npe
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 2000 voip
description outgoing call to Verizon facing Verizon network
translation-profile outgoing remv_plus
huntstop
destination-pattern [2-9]T
session protocol sipv2
session target ipv4:63.87.147.48:5071
voice-class codec 1
voice-class sip rel1xx supported "100rel"
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-npe
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 12345 voip
description outgoing call to Verizon facing PBX
session protocol sipv2
incoming called-number [2-9]T
voice-class codec 1
dtmf-relay rtp-nce
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 86400
!
sip-ua
no remote-party-id
timers expires 1800000
!
!
!
gatekeeper
shutdown
!
!
!
line con 0
logging synchronous
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

![Figure 7 List configuration software versions](image)

![Figure 8: List configuration all](image)
System Parameters IP Options

Figure 9: System Parameters IP Option
Figure 10: System Parameters IP Option (cont.)

IP Nodes
**IP Network Region**

Location: 1  
Authoritative Domain: lab.tekvizion.com  
Name: tekvizion  
Codec Set: 1  
Inter/Intra-region IP-IP Direct Audio: YES  
H.323 SECURITY PROFILES: any-auth  
dst rgn : codec Set is given as 1 and agl is given as ALL
Figure 12: IP Network Region

Figure 13: IP Network Region (cont.)
**IP Codec Set**

<table>
<thead>
<tr>
<th>Source Region</th>
<th>Inter Network Region</th>
<th>Connection Management</th>
<th>WAN-BW-limits</th>
<th>Video</th>
<th>Intervening</th>
<th>Dyn</th>
<th>Ag</th>
<th>CAC</th>
<th>R</th>
<th>L</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>11</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Figure 14: IP Network Region (cont.)
Codec set 1 is configured for this test.
Audio Codec G.729 and G711MU are selected as Audio Codec
Media Encryption is given None

Figure 15: 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
Note: The ip-codec-set configuration above is assigned to SIP trunk(s) using codecs G.729, G.711Mulaw, and T.38 fax relay. If the Service Provider does not support T.38 fax relay, a trunk using G.711 codec is required, with FAX Mode set to “off”.

### 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: SM1
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 17: Signaling Group

Trunk Group

Group number: 2
Group Type: sip
Group Name: SIP to Cisco
TAC: #102
Member Assignment Method: auto
Service Type: tie
Signaling Group: 2
Number of Members: 10

<table>
<thead>
<tr>
<th>Change trunk-group 2</th>
<th>Change trunk-group 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group Number: 2</td>
<td>Group Type: sip</td>
</tr>
<tr>
<td>Group Name: SIP TRUNK</td>
<td>CDR Reports: y</td>
</tr>
<tr>
<td></td>
<td>COR: 1</td>
</tr>
<tr>
<td></td>
<td>TN: 1</td>
</tr>
<tr>
<td></td>
<td>TAC: #102</td>
</tr>
<tr>
<td>Direction: two-way</td>
<td>Outgoing Display? n</td>
</tr>
<tr>
<td>Dial Access? n</td>
<td>Night Service:</td>
</tr>
<tr>
<td>Queue Length: 0</td>
<td></td>
</tr>
<tr>
<td>Service Type: tie</td>
<td>Auth Code? n</td>
</tr>
<tr>
<td></td>
<td>Member Assignment Method: auto</td>
</tr>
<tr>
<td></td>
<td>Signaling Group: 2</td>
</tr>
<tr>
<td></td>
<td>Number of Members: 10</td>
</tr>
</tbody>
</table>

Preferred Minimum Session Refresh Interval (sec): 900

Figure 18: Trunk Group
Numbering Format: private
Mark Users as Phone?: y
Support Request History?: y
Telephone Event Payload Type: 101
Figure 21: Trunk Group (cont.)

**Route Pattern**

Pattern Number: 2
**Pattern Name**: SIP trunk  
**Grp No:** 2

![Figure 22: Route pattern](image_url)

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

<table>
<thead>
<tr>
<th>Dialed String</th>
<th>Total Min</th>
<th>Route Pattern</th>
<th>Call Type</th>
<th>Node Num</th>
<th>ANI Regd</th>
</tr>
</thead>
<tbody>
<tr>
<td>214</td>
<td>10</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>2142425920</td>
<td>10</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>2142425942</td>
<td>10</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>2142425944</td>
<td>10</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>2142425999</td>
<td>10</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>7</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>311</td>
<td>3</td>
<td>2</td>
<td>svcl</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>7</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>411</td>
<td>3</td>
<td>2</td>
<td>svcl</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>10</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>5003</td>
<td>4</td>
<td>4</td>
<td>locl</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>511</td>
<td>3</td>
<td>2</td>
<td>svcl</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>555</td>
<td>7</td>
<td>7</td>
<td>deny</td>
<td>hnpa</td>
<td>n</td>
</tr>
<tr>
<td>5551212</td>
<td>7</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>5712935327</td>
<td>10</td>
<td>2</td>
<td>hnpa</td>
<td>n</td>
<td></td>
</tr>
</tbody>
</table>

Figure 23: ARS Analysis

Display dialplan analysis

Dial string 5 is used to route calls to Avaya PBX extensions and dial String 9 is used for feature access code.
### Figure 24: Display dial plan analysis

<table>
<thead>
<tr>
<th>Dialed String</th>
<th>Total Length</th>
<th>Call Type</th>
<th>Dialed String</th>
<th>Total Length</th>
<th>Call Type</th>
<th>Dialed String</th>
<th>Total Length</th>
<th>Call Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>1</td>
<td>attd</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1000</td>
<td>4</td>
<td>udp</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>5</td>
<td>ext</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>20</td>
<td>4</td>
<td>udp</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2302</td>
<td>4</td>
<td>ext</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>25</td>
<td>4</td>
<td>ext</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>26</td>
<td>4</td>
<td>ext</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>4</td>
<td>ext</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>41</td>
<td>4</td>
<td>ext</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>4</td>
<td>ext</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5712935329</td>
<td>10</td>
<td>ext</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>1</td>
<td>fac</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>9</td>
<td>1</td>
<td>fac</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>*</td>
<td>4</td>
<td>fac</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>#</td>
<td>4</td>
<td>dac</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**ISDN Public/Unknown Numbering Plan**
The table above is used to define numbering plans. 4-digit extensions in the 5XXX range are used by the Avaya PBX Extn. and 4006 is used for Avaya one-X sip client.

![Figure 25: ISDN Public/Unknown Numbering Plan](image-url)

*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 5712935xxx to ext. 5xxx and 4xxx. This is for example and the incoming number should be translated according to the called numbers.

![Figure 26: Incoming-call-handling-treatment](image)

*Station Configuration (IP Phone)*
Station: 5000

Type: 2500

Port: 001v302

Name: fax

Figure 27: Station Configuration (5000)
### Figure 28: Station Configuration 5000 (cont.)

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

**Emergency Location Ext:** 5000

---

### Figure 29: Station Configuration 5000 (cont.)

**ENHANCED CALL FORWARDING**

<table>
<thead>
<tr>
<th>Forwarded Destination</th>
<th>Active</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unconditional For Internal Calls To</td>
<td>n</td>
</tr>
<tr>
<td>External Calls To</td>
<td>n</td>
</tr>
<tr>
<td>Busy For Internal Calls To</td>
<td>n</td>
</tr>
<tr>
<td>External Calls To</td>
<td>n</td>
</tr>
<tr>
<td>No Reply For Internal Calls To</td>
<td>n</td>
</tr>
<tr>
<td>External Calls To</td>
<td>n</td>
</tr>
</tbody>
</table>

**SAC/CF Override:** n
Figure 30: Station Configuration 5000 (cont.)
Station: 5001
Type: 4610
Port: sooo03

Figure 31: Station Configuration (5001)
Figure 32: Station Configuration 5001 (cont.)

Figure 33: Station Configuration 5001 (cont.)
Figure 34: Station Configuration 5001 (cont.)

Figure 35: Station Configuration 5001 (cont.)
Station: 5002
Type: 4610
Port: S00004
Name: 4610

<table>
<thead>
<tr>
<th>Extension: 5002</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type: 4610</td>
</tr>
<tr>
<td>Port: S00004</td>
</tr>
<tr>
<td>Name: 4610</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Lock Messages?</th>
<th>n</th>
<th>BCC: 0</th>
</tr>
</thead>
<tbody>
<tr>
<td>Security Code:</td>
<td>*</td>
<td>TN: 1</td>
</tr>
<tr>
<td>Coverage Path 1:</td>
<td>COR: 1</td>
<td></td>
</tr>
<tr>
<td>Coverage Path 2:</td>
<td>COS: 1</td>
<td></td>
</tr>
<tr>
<td>Hunt-to Station:</td>
<td>Tests? y</td>
<td></td>
</tr>
</tbody>
</table>

**Figure 36: Station Configuration (5002)**
Figure 37: Station Configuration 5002 (cont.)

<table>
<thead>
<tr>
<th>Feature Options</th>
<th>Setting</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>Active Station Ringing</td>
<td>single</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>enhanced</td>
</tr>
<tr>
<td>MWI Served User Type</td>
<td></td>
</tr>
<tr>
<td>AUDIX Name</td>
<td></td>
</tr>
<tr>
<td>Emergency Location Ext</td>
<td>5002</td>
</tr>
<tr>
<td>Auto Select Any Idle Appearance</td>
<td>n</td>
</tr>
<tr>
<td>Coverage Mag 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>Idle Appearance Preference</td>
<td>n</td>
</tr>
<tr>
<td>Bridged Idle Line Preference</td>
<td>n</td>
</tr>
<tr>
<td>Restrict Last Appearance</td>
<td>y</td>
</tr>
<tr>
<td>EMU Login Allowed</td>
<td>n</td>
</tr>
<tr>
<td>Per Station CNP - Send Calling Number</td>
<td></td>
</tr>
<tr>
<td>EC500 State</td>
<td>enabled</td>
</tr>
<tr>
<td>Audible Message Waiting</td>
<td>n</td>
</tr>
<tr>
<td>Display Client Redirection</td>
<td>n</td>
</tr>
<tr>
<td>Select Last Used Appearance</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 Connections</td>
<td>n</td>
</tr>
<tr>
<td>Always Use</td>
<td>n</td>
</tr>
<tr>
<td>IP Audio Hairpinning</td>
<td>n</td>
</tr>
</tbody>
</table>

Figure 38: Station Configuration 5002 (cont)

<table>
<thead>
<tr>
<th>Configuration Options</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conf/Trans on Primary Appearance</td>
<td>n</td>
</tr>
<tr>
<td>Bridged Appearance Origination Restriction</td>
<td>n</td>
</tr>
<tr>
<td>Call Appearance Display Format</td>
<td>disp-param-default</td>
</tr>
<tr>
<td>IP Phone Group ID</td>
<td></td>
</tr>
<tr>
<td>Enhanced Caller-Info Display for 1-Line Phones</td>
<td>n</td>
</tr>
<tr>
<td>ENHANCED CALL FORWARDING</td>
<td></td>
</tr>
<tr>
<td>Forwarded Destination</td>
<td></td>
</tr>
<tr>
<td>Active</td>
<td></td>
</tr>
<tr>
<td>Unconditional For Internal Calls To</td>
<td>n</td>
</tr>
<tr>
<td>External Calls To</td>
<td>n</td>
</tr>
<tr>
<td>Busy For Internal Calls To</td>
<td>n</td>
</tr>
<tr>
<td>External Calls To</td>
<td>n</td>
</tr>
<tr>
<td>No Reply For Internal Calls To</td>
<td>n</td>
</tr>
<tr>
<td>External Calls To</td>
<td>n</td>
</tr>
<tr>
<td>SAC/CF Override</td>
<td>n</td>
</tr>
</tbody>
</table>
Figure 39: Station Configuration 5002 (cont.)

display station 5002

SITE DATA
  Room:  
  Jack:  Speaker? n
  Cable: Mounting: d
  Floor: Cord Length: 0
  Building: Set Color:

ABBREVIATED DIALING
  List1:  
  List2:  
  List3:

BUTTON ASSIGNMENTS
  1: call-appr  
  2: call-appr  
  3: call-appr  
  4:  

FEATURE BUTTON ASSIGNMENTS
  9:  
  10:  
  11:  
  12:  
  13:  
  14:  
  15:  
  16:  
  17:  
  18:  
  19:  
  20:  
  21:  
  22:  
  23:  
  24:  

Figure 40: Station Configuration 5002 (cont.)
Station: 5003
Type: 9630
Port: S00064
Name: 1x

Figure 41: Station Configuration (5003)
Figure 42: Station Configuration 5003 (cont.)

display station 5003

FEATURE OPTIONS
LWC Reception: spe
LWC Activation? y
LWC Log External Calls? n
CDR Privacy? n
Redirect Notification? y
Per Button Ring Control? n
Bridged Call Alerting? n
Active Station Ringing: single

Auto Select Any Idle Appearance? n
Coverage Msg Retrieval? y
Auto Answer: none
Data Restriction? n
Idle Appearance Preference? n
Restrict Last Appearance? y
EMU Login Allowed? n
Per Station CNF - Send Calling Number?
Service Link Mode: as-needed
Multimedia Mode: enhanced
MWI Served User Type: sip-adjunct
Display Client Redirection? n
Select Last Used Appearance? n
Coverage After Forwarding? s
Multimedia Early Answer? n
Remote Softphone Emergency Calls: as-on-local
Direct IP-IP Audio Connections? y
Emergency Location Ext: 5003
Always Use? n
IP Audio Hairpinning? n

Figure 43: Station Configuration 5003 (cont.)

display station 5003

CONF/TRANS on Primary Appearance? n
Bridged Appearance Origination Restriction? n
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:
External Calls To:
Busy For Internal Calls To:
External Calls To:
No Reply For Internal Calls To: 92142425980
External Calls To:

SAC/CF Override: n

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Figure 44: Station Configuration 5003 (cont.)

Figure 45: Station Configuration 5003 (cont.)
Type: 9630
Port: S00067
Station: 5004
Name: 1x

Figure 46: Station Configuration (5004)
Figure 49: Station Configuration (cont.)

Figure 50: Station Configuration 5004 (cont.)
Station: 4006
Type: 9630SIP
Port: S00046
Name: Avaya, SIP2

Figure 51: Station Configuration 4006
<table>
<thead>
<tr>
<th>Feature Options</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>LWC Reception: spe</td>
<td>y</td>
</tr>
<tr>
<td>LWC Activation? y</td>
<td></td>
</tr>
<tr>
<td>CDR Privacy? n</td>
<td></td>
</tr>
<tr>
<td>Per Button Ring Control? n</td>
<td></td>
</tr>
<tr>
<td>Bridged Call Alerting? n</td>
<td></td>
</tr>
<tr>
<td>Active Station Ringing: single</td>
<td></td>
</tr>
<tr>
<td>H.320 Conversion? n</td>
<td></td>
</tr>
<tr>
<td>Per Station CPN - Send Calling Number?</td>
<td></td>
</tr>
<tr>
<td>EC500 State: enabled</td>
<td></td>
</tr>
<tr>
<td>MWI Served User Type:</td>
<td></td>
</tr>
<tr>
<td>AUDIX Name:</td>
<td></td>
</tr>
<tr>
<td>Coverage After Forwarding? s</td>
<td></td>
</tr>
<tr>
<td>Remote Softphone Emergency Calls: as-on-local</td>
<td></td>
</tr>
<tr>
<td>Direct IP-IP Audio Connections? y</td>
<td></td>
</tr>
<tr>
<td>Emergency Location Ext: 4006</td>
<td></td>
</tr>
<tr>
<td>Always Use? n IP Audio Hairpinning? n</td>
<td></td>
</tr>
</tbody>
</table>
Figure 52: Station Configuration 4006 (cont.)

Figure 53: Station Configuration 4006 (cont.)
Figure 54: Station Configuration 4006 (cont.)

Figure 55: Station Configuration 4006 (cont.)
Avaya Aura Session Manager Configuration

Navigation: Home > Elements > Routing

Figure 56: Avaya Aura – Session Manager Configuration
Domains

Name: lab.tekvizion.com

Figure 57: Avaya Aura – Domains
**Locations**

Name: Richardson

![Avaya Aura – Locations](image)

**Figure 58: Avaya Aura – Locations**
Adaptations
Adaptation for Cisco UBE

Module name: CiscoAdapter

Module Parameter: fromto=true ostd=10.64.3.43 iosrcd=lab.tekvizion.com

Figure 59: Avaya Aura – Adaptations
Adaptation for Avaya Aura CM

Adaptation name: Avaya_CM
Module name: DigitConversionAdapter
Module Parameter: fromto = true

Figure 60: Avaya Aura – Adaptations
SIP Entities

Cisco UBE IP: 10.64.3.43
Avaya CM IP: 10.70.4.4

Figure 61: Avaya Aura – SIP Entities
FQDN or IP Address: 10.64.3.43

Figure 62: Avaya Aura – SIP Entities (cont.)
SIP Entity for Avaya CM

Name: tekaacm

FQDN or IP Address: 10.70.4.4

Type: CM

Figure 63: Avaya Aura – SIP Entities (cont.)
**Entity Link**

Name: tekaasm_cube, SIP Entity 1 (tekaasm) and SIP Entity 2 (cisco UBE)

Name: tekaasm_tekaacm_5060_tcp, SIP Entity 1 (tekaasm) and SIP Entity 2 (tekaacm)

![Figure 64: Avaya Aura – Entity Links](image-url)
Entity Link between Avaya Session Manager and Avaya CM

SIP Entity 1: tekaasm
Protocol: TCP
Port: 5060

SIP Entity 2: tekaacm
Port: 5060

Figure 65: Avaya Aura – Entity Links cont.
Entity Link between Avaya Session Manager and Cisco UBE

SIP Entity 1: tekaasm
Protocol: UDP
Port: 5060

SIP Entity 2: Cisco cube
Port: 5060

Figure 66: Avaya Aura – Entity Links cont.
Routing Policies
Routing Policy for call to go to Cisco UBE

Name: to_cisco_ube

FQDN or IP address: 10.64.3.43

Figure 67: Avaya Aura – Routing Policies
Routing Policies Routing Policy for call to go to Avaya CM

Name: to_tekaaCM
FQDN or IP address: 10.70.4.4

Figure 68: Avaya Aura – Routing Policies cont.
Dial Pattern to reach Cisco UBE

Dial Pattern: 214242

Figure 69: Avaya Aura – Dial Patterns
## Acronyms

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