Avaya S8500 Communications Manager 3.0 with Cisco Unified Border Element for SIP-to-SIP Calls

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Introduction

- This is an application note for connectivity of Avaya S8500 Communications Manager 3.0 with Cisco Unified Border Element via SIP (10/100baseT).
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with the Cisco Unified Border Element (CUBE) connected to the IP PBX via SIP (10/100baseT). Connectivity is achieved by using the SIP protocol.
- This Application Note uses the c3845 IOS-voice-gateway, however other Cisco voice gateways are also an option to use since CUBE implementation does not depend on the platform. Here is a list of Cisco Products capable of CUBE functionality:

  Cisco 2800 Series Integrated Services Routers
  Cisco 3800 Series Integrated Services Routers
  Cisco AS5350XM Universal Gateway
  Cisco AS5400XM Universal Gateway
Network Topology

Figure 1. Network Topology or Test Setup

Limitations

- Connected Name is not presented to the originating (calling) Phone display. CUBE does not relay the destination "contact" (URI) info from the 180 Ringing message sent by the Avaya PBX.

- Basic Call using G.726 codec fail. Avaya PBX rejects G.726 codec, even when the Avaya is set for G.726. (This limitation as of version G3V13 of the Avaya PBX)

- Call Transfer Name and Number updates do not occur

- Calling Number Restricted is not honored by the Avaya PBX (This limitation as of version G3V13 of the Avaya PBX)

- On Call forward all and Call forward busy the originating phone does not hear ringback, even though the final destination rings and the call is established if final destination answers. Avaya SIP supports STATUS message 181 "Call is being forwarded" to cut-through the ringback, CUBE IOS does not support this message as of 124-7.24.P14.

- DTMF relay using RFC2833 requires the IOS CUBE to configure the appropriate dial-peer for “dtmf-relay rtp-nte”, “rtp payload-type 127”. Avaya utilizes RTP payload type value 127 (hardcoded). (This limitation as of version G3V13 of the Avaya PBX).
System Components

Hardware Requirements
Cisco equipment
- Cisco 3845 (Cisco 3800 family routers)
- Cisco Catalyst 6500

Avaya equipment
- Avaya S8500
- TN2312BP IPSI
- TN799DP C-LAN
- TN2302AP IP Media Processor
- TN746B Analog
- TN2224B 2-wire Digital
- 2 - Digital stations 8410D
- 2 - Digital stations 6408D+

Software Requirements
- PBX Software: G3 version: V13

Features

Features Supported
- G711u and A law, G729 and G723 codecs
- Call Transfer blind and Call Transfer supervised
- Call Conference
- Call on-hold
- Call Forward No Reply
- FAX integrity
- DTMF (RFC2833) or inband (G711)

Features Not Supported
- Connected Name
- Calling Number Restriction
- Call Forward all
- Call Forward Busy
Configuration

Configuration Sequence and Tasks

Configuration Menus and Commands

Avaya Configuration

**Signaling-Group**

Voice System name: S8500SIP2 - SIGNALING GROUP

Group Number: 1  Group Type: sip  Transport Method: tls

Near-end Node Name: clan1  Far-end Node Name: avayasip2
Near-end Listen Port: 5061  Far-end Listen Port: 5061
Far-end Network Region: 1
Far-end Domain: lab2.com

Bypass If IP Threshold Exceeded? n

DTMF over IP: rtp-payload  Direct IP-IP Audio Connections? n
IP Audio Hairpinning? n
Session Establishment Timer(min): 120

**Trunk-Group**

Voice System name: S8500SIP2 - TRUNK GROUP

Group Number: 1  Group Type: sip  CDR Reports: y
Group Name: OUTSIDE CALL  COR: 1  TN: 1  TAC: 801
Direction: two-way  Outgoing Display? n
Dial Access? n  Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie  Auth Code? n
Signaling Group: 1
Number of Members: 6

TRUNK PARAMETERS

Unicode Name? y

Redirect On OPTIM Failure: 5000

SCCAN? n  Digital Loss Group: 18

TRUNK FEATURES

ACA Assignment? n  Measured: none
Maintenance Tests? y

Numbering Format: public  Replace Unavailable Numbers? n
Trunk-Group

TRUNK GROUP
Administered Members (min/max): 1/6
GROUP MEMBER ASSIGNMENTS
Total Administered Members: 6

<table>
<thead>
<tr>
<th>Port</th>
<th>Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>1:</td>
<td>T00001</td>
</tr>
<tr>
<td>2:</td>
<td>T00002</td>
</tr>
<tr>
<td>3:</td>
<td>T00003</td>
</tr>
<tr>
<td>4:</td>
<td>T00004</td>
</tr>
<tr>
<td>5:</td>
<td>T00059</td>
</tr>
<tr>
<td>6:</td>
<td>T00060</td>
</tr>
<tr>
<td>7:</td>
<td></td>
</tr>
<tr>
<td>8:</td>
<td></td>
</tr>
<tr>
<td>9:</td>
<td></td>
</tr>
<tr>
<td>10:</td>
<td></td>
</tr>
<tr>
<td>11:</td>
<td></td>
</tr>
<tr>
<td>12:</td>
<td></td>
</tr>
<tr>
<td>13:</td>
<td></td>
</tr>
<tr>
<td>14:</td>
<td></td>
</tr>
<tr>
<td>15:</td>
<td></td>
</tr>
</tbody>
</table>

Node-names IP

Voice System name: S8500SIP2 - IP NODE NAMES

<table>
<thead>
<tr>
<th>Name</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>CCM3.3</td>
<td>172.20.31.254</td>
</tr>
<tr>
<td>CCM4.1</td>
<td>172.20.231.254</td>
</tr>
<tr>
<td>CCM5.0-VENUS</td>
<td>172.20.214.254</td>
</tr>
<tr>
<td>CM-KLINGON</td>
<td>172.20.32.254</td>
</tr>
<tr>
<td>CM-POLARIS</td>
<td>172.20.236.50</td>
</tr>
<tr>
<td>IPIPGW</td>
<td>172.20.8.26</td>
</tr>
<tr>
<td>MAvantage</td>
<td>172.20.7.252</td>
</tr>
<tr>
<td>avayasip1</td>
<td>172.20.212.254</td>
</tr>
<tr>
<td>avayasip2</td>
<td>172.20.213.254</td>
</tr>
<tr>
<td>clan1</td>
<td>172.20.213.253</td>
</tr>
<tr>
<td>clan1server1</td>
<td>172.20.212.253</td>
</tr>
<tr>
<td>default</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>medpro1</td>
<td>172.20.213.252</td>
</tr>
<tr>
<td>procr</td>
<td>. . .</td>
</tr>
</tbody>
</table>

(15 of 15 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
IP Network Region

Voice System name: S8500SIP2 - IP NETWORK REGION
Region: 1
Location: 1   Authoritative Domain: lab2.com
Name: CiscoLAB2

Intra-region IP-IP Direct Audio: no

MEDIA PARAMETERS
Codec Set: 1
UDP Port Min: 2048
UDP Port Max: 3028
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 34
Audio PHB Value: 46
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 7
Audio 802.1p Priority: 6

IP Network Region

INTER-GATEWAY ALTERNATE ROUTING
Incoming LDN Extension:
Conversion To Full Public Number - Delete: Insert:
Maximum Number of Trunks to Use:

LSP NAMES IN PRIORITY ORDER
1
2
3
4
5
6
### Voice System name: S8500SIP2 - IP Codec Set

Codec Set: 1

<table>
<thead>
<tr>
<th>Codec</th>
<th>Audio</th>
<th>Silence</th>
<th>Frames</th>
<th>Packet</th>
<th>Suppression</th>
<th>Per Pkt</th>
<th>Size(ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1: G.711MU</td>
<td>n</td>
<td>2</td>
<td>20</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2: G.729AB</td>
<td>n</td>
<td>2</td>
<td>20</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3: G.723-6.3K</td>
<td>n</td>
<td>1</td>
<td>30</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>4:</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5:</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6:</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>7:</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Media Encryption
1: none  
2:  
3:  

### IP Codec Set

| Allow Direct-IP Multimedia? | n |

#### Mode | Redundancy
---|---
FAX | pass-through | 0 ➔ **This field is changed to T.38 for Fax over T.38 codec**  
Modem | pass-through | 0  
TDD/TTY | US | 3  
Clear-channel | n | 0
### Uniform dialing
Voice System name: S8500SIP2 - UNIFORM DIAL PLAN TABLE
Percent Full: 0

<table>
<thead>
<tr>
<th>Pattern Len Del Digits Net Conv Num</th>
<th>Pattern Len Del Digits Net Conv Num</th>
</tr>
</thead>
<tbody>
<tr>
<td>4154 4 0 222 aar n</td>
<td>n</td>
</tr>
<tr>
<td>4155 4 0 222 aar n</td>
<td>n</td>
</tr>
<tr>
<td>4156 4 0 222 aar n</td>
<td>n</td>
</tr>
</tbody>
</table>

### AAR Analysis
Voice System name: S8500SIP2 - AAR DIGIT ANALYSIS TABLE
Percent Full: 1

<table>
<thead>
<tr>
<th>Dialed String</th>
<th>Total Min</th>
<th>Max</th>
<th>Route Pattern</th>
<th>Call Type</th>
<th>Node ANI Reqd</th>
</tr>
</thead>
<tbody>
<tr>
<td>222</td>
<td>7</td>
<td>7</td>
<td>99</td>
<td>aar</td>
<td>n</td>
</tr>
</tbody>
</table>

### Route Pattern
Voice System name: S8500SIP2 - 99ttPattern Name: CCS Sever 2

<table>
<thead>
<tr>
<th>Grp FRL NPA Pfx HOP Toll No. Inserted Digits</th>
<th>DCS/ IXC QSIG Dgts Format Intw Subaddress</th>
</tr>
</thead>
<tbody>
<tr>
<td>1: 1 0 3</td>
<td>n user</td>
</tr>
<tr>
<td>2:</td>
<td>n user</td>
</tr>
<tr>
<td>3:</td>
<td>n user</td>
</tr>
<tr>
<td>4:</td>
<td>n user</td>
</tr>
<tr>
<td>5:</td>
<td>n user</td>
</tr>
<tr>
<td>6:</td>
<td>n user</td>
</tr>
</tbody>
</table>

BCC VALUE  TSC CA-TSC  ITC BCIE Service/Feature BAND No. Numbering LAR 0 1 2 3 4 W Request Dgts Format Subaddress

| 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 |

Pattern Number: 99

<table>
<thead>
<tr>
<th>Grp FRL NPA Pfx Hop Toll No. Inserted Digits</th>
<th>DCS/ IXC QSIG</th>
</tr>
</thead>
<tbody>
<tr>
<td>7:</td>
<td>n user</td>
</tr>
<tr>
<td>8:</td>
<td>n user</td>
</tr>
<tr>
<td>9:</td>
<td>n user</td>
</tr>
<tr>
<td>10:</td>
<td>n user</td>
</tr>
<tr>
<td>11:</td>
<td>n user</td>
</tr>
<tr>
<td>12:</td>
<td>n user</td>
</tr>
</tbody>
</table>
Cisco 3845 IOS Configuration

tony_3845#sh run
Building configuration...

Current configuration : 2831 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname tony_3845
!
boot-start-marker
boot system flash: c3845-ipvoice_ivs-mz.124-7.9.P14a
boot-end-marker
!
logging buffered 100000000 debugging
no logging console
enable password cisco
!
no aaa new-model
!
resource policy
!
ip cef
!
!
!
no ip domain lookup
!
voice-card 0
no dspsfarm
!
!
!
voice service voip
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
signaling forward unconditional
fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw h323
h225 id-passthru
h225 connect-passthru
sip
min-se 240
!
!
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729br8
!
! interface GigabitEthernet0/0
  ip address 172.20.8.26 255.255.255.0
  duplex auto
  speed auto
  media-type rj45
  negotiation auto

! interface GigabitEthernet0/1
  no ip address
  shutdown
duplex auto
  speed auto
  media-type rj45
  negotiation auto
duplex auto
  speed auto
  media-type rj45
  negotiation auto

  ip default-gateway 172.20.8.1
  ip route 0.0.0.0 0.0.0.0 172.20.8.1

  ip http server

  !
  control-plane

  !
  dial-peer voice 3000 voip
  destination-pattern 30...
  rtp payload-type nte 127 This must be set when Avaya is set to DTMF “rtp-payload”
  voice-class codec 1
  session protocol sipv2
  session target ipv4:172.20.213.254
  session transport tcp
dtmf-relay rtp-nte
  fax-relay ecm disable
  fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
  no vad

  dial-peer voice 4150 voip
  destination-pattern 41...
  rtp payload-type nte 127 This must be set when Avaya is set to DTMF “rtp-payload”
  voice-class codec 1
  session protocol sipv2
  session target ipv4:172.20.212.254
  session transport tcp
dtmf-relay rtp-nte
  fax-relay ecm disable
  fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
  no vad

  !
  !
  gatekeeper
  shutdown

  !
  telephony-service
  max-conferences 12 gain -6
  transfer-system full-consult
  !
line con 0
password cisco
stopbits 1
line aux 0
stopbits 1
line vty 0 4
timeout login response 300
password cisco
login
!
scheduler allocate 20000 1000
!
end

Tony_3845#

Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>Cisco IOS</td>
<td>Cisco Internetwork Operating System</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>RTP</td>
<td>Real-Time Protocol</td>
</tr>
</tbody>
</table>