Avaya S8500 Version 4.0 to Cisco IOS Voice Gateway using SIP with T1-PRI to PSTN

February 20, 2008

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Introduction

- This Application note provides basic call interoperability and documented steps and configurations necessary for SIP integration between Avaya S8500 to Cisco IOS Voice Gateway providing PSTN (T1-PRI) connectivity.

- The SIP protocol is used between Cisco IOS Voice gateway and Avaya S8500. The connection between Cisco IOS gateway and PSTN uses T1-PRI (DMS-100 switch type).

- Features tested include Basic call, Call Transfer supervised, Call Transfer blind, Call Forward (Unconditional, Busy and No Answer), Three-way Conference, DTMF tones (in-band and relay via RFC2833), Digit Translation, and Call Hold.

- The Cisco IOS Voice Gateway offers the advantage of providing connectivity between Avaya S8500 and PSTN by offering SIP to T1-PRI inter-working functionality.

- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with the Cisco IOS Voice Gateway connected to the Avaya S8500 (10/100baseT) and connected to the PSTN via T1-PRI.

- This Application Notes uses the Cisco 3845 IOS-voice-gateway, however other Cisco voice gateways are also an option to use since the voice gateway implementation does not depend on the platform. Below is a list of Cisco platforms capable of voice gateway functionality: Care must be taken when selecting a voice gateway platform depending on the capacity and capability required for the intended deployment.

Cisco 1861 Integrated Services Router  
Cisco 1AD2400 Series Integrated Access Device  
Cisco 2800 Series Integrated Services Routers  
Cisco 3700 Series Routers  
Cisco 3800 Series Integrated Services Routers  
Cisco AS5350XM Universal Gateway  
Cisco AS5400XM Universal Gateway
**Limitations**

- The Avaya H323 phones did not work when the trunk codec was set to G.723-5.3K or G.723-6.3K.
- For inband DTMF, the Avaya "DTMF over IP" setting in the signalling group had to be set to "in-band", not "in-band-g711".
- For basic calls with RFC2833 DTMF relay, it was noticed that the PBX phone (e.g., 'A') had to send DTMF to the PSTN digital phone (e.g., 'C') before it could play tones originating from the PSTN digital phone. This was due to the H.323/SIP interworking on the PBX itself, and was also noticed on calls involving only Avaya SIP and H.323 phones.
- On a call that originates from a PSTN station to a PBX station and that is then forwarded unconditionally or on busy (e.g., phone C calls phone A, and phone A forwards to phone D), the originating PSTN user does not get ringback. Cisco IOS Voice Gateway does not support SIP 181 message (call being forwarded) which causes SDP establishment failure during a call forward between PSTN phone and Avaya stations.
System Components

Hardware requirements
Cisco equipment
• Cisco 3845 (Cisco 3800 family routers)

Avaya equipment
• Avaya S8500
• Avaya SIP Enablement Server (SES)
• (2) Avaya 4624/4612 IP phones
• TN2312BP IPSI
• TN799DP C-LAN
• TN2302AP IP Media Processor

Software Requirements
• Avaya Communications Manager Release 4.0
• Avaya SES software version 4.0
• Avaya 4612/4624 (H.323 phones) load: def24rl_8_3.bin

Features

Features Supported
• Basic call with G.729, G.729B, G.711ulaw and G.711Alaw
• Call Transfer Blind and Call Transfer Supervised
• Call Conference
• Call on Hold
• Call Forward All (See Limitations Section)
• Call Forward No Reply
• Call Forward Busy (See Limitations Section)
• Digit Translation
• DTMF in-band
• DTMF - RFC2833 (See Limitations Section)
Configuration
Avaya S8500 Configuration

Figure 2. Signaling group (in band)
Figure 3. Signaling group (RFC 2833)
Figure 4. Trunk group – p1 of 3
Figure 5. Trunk group – p2 of 3

```
<table>
<thead>
<tr>
<th>Severity</th>
<th>Date/Time</th>
<th>System</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Info</td>
<td>1/1/2008 9:14:43 AM</td>
<td>SIP2</td>
<td>change b-codes set 3</td>
</tr>
<tr>
<td>Info</td>
<td>1/1/2008 9:12:58 AM</td>
<td>SIP2</td>
<td>change b-codes set 3</td>
</tr>
<tr>
<td>Info</td>
<td>1/1/2008 9:11:12 AM</td>
<td>SIP2</td>
<td>change b-codes set 3</td>
</tr>
<tr>
<td>Info</td>
<td>1/1/2008 9:01:00 AM</td>
<td>SIP2</td>
<td>change b-codes set 3</td>
</tr>
<tr>
<td>Info</td>
<td>1/1/2008 9:00:15 AM</td>
<td>SIP2</td>
<td>change b-codes set 3</td>
</tr>
</tbody>
</table>
```

**GROUP TYPE: sip**

```
Redirect On OPTIN Failure: 5000
Preferred Minimum Session Refresh Interval(sec): 900
```

**TRUNK PARAMETERS**

- Unicode Name: y
- SCCAN: n
- Digital Loss Group: 18
Figure 6. Trunk group – p3 of 3
Figure 7.  Node-names IP

<table>
<thead>
<tr>
<th>Name</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>CMH.3</td>
<td>172.20.31.254</td>
</tr>
<tr>
<td>CMH.1</td>
<td>172.20.231.254</td>
</tr>
<tr>
<td>CMH.1.2</td>
<td>172.20.236.2</td>
</tr>
<tr>
<td>CMH.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>172.20.224.250</td>
</tr>
<tr>
<td>CM-cluster1.5</td>
<td>172.20.241.253</td>
</tr>
<tr>
<td>IP1GW</td>
<td>172.20.8.26</td>
</tr>
<tr>
<td>HA1antage</td>
<td>172.20.2.252</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>claniserver1</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>payer</td>
<td>172.20.213.240</td>
</tr>
</tbody>
</table>

(16 of 16 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 8. IP Network Region
Figure 9. IP codec set (showing G.729, annex B).
Figure 10. Uniform dialing plan

![Uniform Dialing Plan](image-url)
Figure 11. AAR analysis

![AAR analysis screenshot](image)

### AAR DIGIT ANALYSIS TABLE

<table>
<thead>
<tr>
<th>Dialed String</th>
<th>Total Min</th>
<th>Total Max</th>
<th>Route Pattern</th>
<th>Call Type</th>
<th>ANI Num</th>
<th>ANI Reqd</th>
</tr>
</thead>
<tbody>
<tr>
<td>201</td>
<td>7</td>
<td>7</td>
<td>1</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>202</td>
<td>7</td>
<td>7</td>
<td>2</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>204</td>
<td>7</td>
<td>7</td>
<td>4</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>213</td>
<td>7</td>
<td>7</td>
<td>13</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>214</td>
<td>7</td>
<td>7</td>
<td>14</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>215</td>
<td>7</td>
<td>7</td>
<td>15</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>216</td>
<td>7</td>
<td>7</td>
<td>6</td>
<td>aar</td>
<td>1</td>
<td>n</td>
</tr>
<tr>
<td>217</td>
<td>7</td>
<td>7</td>
<td>6</td>
<td>aar</td>
<td>6</td>
<td>n</td>
</tr>
<tr>
<td>224</td>
<td>7</td>
<td>7</td>
<td>224</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>7</td>
<td>7</td>
<td>999</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>7</td>
<td>7</td>
<td>999</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>7</td>
<td>7</td>
<td>999</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>7</td>
<td>7</td>
<td>999</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>7</td>
<td>7</td>
<td>7</td>
<td>999</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>7</td>
<td>7</td>
<td>999</td>
<td>aar</td>
<td>n</td>
<td></td>
</tr>
</tbody>
</table>
Figure 12. Route Pattern
Avaya SIP Enablement Server (SES) Configuration

Figure 13. SES Server

<table>
<thead>
<tr>
<th>Status</th>
<th>Commands</th>
<th>Host</th>
<th>Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>up to date</td>
<td>Edit</td>
<td>Map</td>
<td>172.20.213.254</td>
</tr>
</tbody>
</table>

Force All
Migrate Home/Edge
Figure 14. SES Trusted Hosts

### List Trusted Hosts

<table>
<thead>
<tr>
<th>Commands</th>
<th>IP Address</th>
<th>Trusted by Host</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Edit</td>
<td>172.20.15.123</td>
<td>172.20.213.254</td>
<td>Chris_GW</td>
</tr>
<tr>
<td>Edit</td>
<td>172.20.150.251</td>
<td>172.20.213.254</td>
<td>Chinh_CCM</td>
</tr>
<tr>
<td>Edit</td>
<td>172.20.212.253</td>
<td>172.20.213.254</td>
<td>Avaya CM1</td>
</tr>
<tr>
<td>Edit</td>
<td>172.20.212.254</td>
<td>172.20.213.254</td>
<td>Avaya SIP1</td>
</tr>
<tr>
<td>Edit</td>
<td>172.20.214.254</td>
<td>172.20.213.254</td>
<td>CCM-Venus</td>
</tr>
<tr>
<td>Edit</td>
<td>172.20.215.254</td>
<td>172.20.213.254</td>
<td>CM-Mercury</td>
</tr>
<tr>
<td>Edit</td>
<td>172.20.236.50</td>
<td>172.20.213.254</td>
<td>CM-Polaris</td>
</tr>
</tbody>
</table>

Add Another Trusted Host
Figure 15. SES DN Map

<table>
<thead>
<tr>
<th>Command</th>
<th>Name</th>
<th>Commands</th>
<th>Contact</th>
</tr>
</thead>
<tbody>
<tr>
<td>Edit</td>
<td>Delete</td>
<td>Avaya_SIP1</td>
<td></td>
</tr>
<tr>
<td>Edit</td>
<td>Delete</td>
<td>Avaya_SIP1_b</td>
<td></td>
</tr>
<tr>
<td>Add</td>
<td>Another Map</td>
<td>Add Another Contact</td>
<td>Delete Group</td>
</tr>
<tr>
<td>Edit</td>
<td>Delete</td>
<td>CM_Mercury_42</td>
<td></td>
</tr>
<tr>
<td>Add</td>
<td>Another Map</td>
<td>Add Another Contact</td>
<td>Delete Group</td>
</tr>
<tr>
<td>Edit</td>
<td>Delete</td>
<td>CCM-Venus</td>
<td></td>
</tr>
<tr>
<td>Add</td>
<td>Another Map</td>
<td>Add Another Contact</td>
<td>Delete Group</td>
</tr>
<tr>
<td>Edit</td>
<td>Delete</td>
<td>CCM-Venus_VM</td>
<td></td>
</tr>
<tr>
<td>Add</td>
<td>Another Map</td>
<td>Add Another Contact</td>
<td>Delete Group</td>
</tr>
<tr>
<td>Edit</td>
<td>Delete</td>
<td>Chinh_CCM</td>
<td></td>
</tr>
<tr>
<td>Add</td>
<td>Another Map</td>
<td>Add Another Contact</td>
<td>Delete Group</td>
</tr>
<tr>
<td>Edit</td>
<td>Delete</td>
<td>CM_Polaris_S00X</td>
<td></td>
</tr>
<tr>
<td>Add</td>
<td>Another Map</td>
<td>Add Another Contact</td>
<td>Delete Group</td>
</tr>
<tr>
<td>Edit</td>
<td>Delete</td>
<td>Chris_GW</td>
<td></td>
</tr>
<tr>
<td>Add</td>
<td>Another Map</td>
<td>Add Another Contact</td>
<td>Delete Group</td>
</tr>
</tbody>
</table>

Add Map in New Group
Figure 16. SES 30XX Routing Entry
Cisco IOS Voice Gateway (c3845) configuration

show version

Cisco IOS Software, 3800 Software (C3845-IPVOICE-M), Version 12.4(15)T3, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2008 by Cisco Systems, Inc.
Compiled Fri 25-Jan-08 00:11 by prod_rel_team

ROM: System Bootstrap, Version 12.3(11r)T1, RELEASE SOFTWARE (fc1)

3845_West uptime is 2 weeks, 43 minutes
System returned to ROM by reload at 18:17:14 UTC Wed Feb 6 2008
System image file is "flash:c3845-ipvoice-mz.124-15.T3.bin"

Cisco 3845 (revision 1.0) with 224256K/37888K bytes of memory.
Processor board ID FHK0847F136
2 Gigabit Ethernet interfaces
24 Serial interfaces
2 Channelized E1/PRI ports
2 Channelized T1/PRI ports
2 Voice FXS interfaces
DRAM configuration is 64 bits wide with parity enabled.
479K bytes of NVRAM.
62592K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102

show run

Building configuration...

Current configuration : 2936 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname 3845_West
!
boot-start-marker
boot-end-marker
!
 card type t1 0 0
logging buffered 1000000
no logging console
enable secret 5 $1$MFhi$AqqpDsFeO4Sb/IkzkcmO/
!
!
oo aaa new-model
 network-clock-participate wic 0
 network-clock-select 1 T1 0/0/0
 voice-card 0
no dspfarm
! ip cef
! no ip domain lookup
!
! isdn switch-type primary-dms100
isdn gateway-max-interworking
!
! voice call carrier capacity active
! voice service pots
! voice service voip
!
!
voice translation-rule 1
rule 1 /30/ /52:1/ 
!
voice translation-rule 2
rule 1 /11/ /21:1/ 
!

voice translation-profile PSTN-to-SIP
 translate called 2
!

voice translation-profile SIP-to-PSTN
translate called 1
!
!
controller T1 0/0/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
controller T1 0/0/1
 framing esf
 linecode b8zs
!
ip tcp synwait-time 13
!
!
interface GigabitEthernet0/0
 description SETH-LANS$ETH-SW-LAUNCH$INFO-GE 0/0$ 
ip address 10.10.10.1 255.255.255.248
 shutdown
duplex auto
speed auto
media-type rj45
no keepalive
!

1 The above translation rule replaces the number “30xx” with “52xx”
2 This defines the translation profile for the called number. In this particular example, the called number “30xx” is converted to “52xx”.
interface GigabitEthernet0/1
  ip address 172.20.15.123 255.255.255.0
duplex auto
  speed auto
  media-type rj45
  no keepalive

interface Serial0/0/0:23
  no ip address
  encapsulation hdlc
  isdn switch-type primary-dms100
  isdn protocol-emulate network
  isdn incoming-voice voice
  isdn channel-id invert extend-bit
  no cdp enable
  ip forward-protocol nd
  ip route 0.0.0.0 0.0.0.0 172.20.15.1
  !
ip http server
  ip http authentication local
  !
  control-plane
  
  voice-port 0/0/0:23
  
  voice-port 0/1/0
  
  voice-port 0/1/1
  
  dial-peer voice 4000 pots
  translation-profile incoming PSTN-to-SIP
  destination-pattern 5...
  incoming called-number 1...
  direct-inward-dial
  port 0/0/0:23
  forward-digits all
  
  dial-peer voice 519 voip
  translation-profile incoming SIP-to-PSTN
  destination-pattern 2...
  < rtp payload-type nte 127? >
  session protocol sipv2
  session target ipv4:172.20.213.254
  incoming called-number 3...
  dtmf-relay rtp-nte
  codec g711ulaw

3 This defines the call number translation profile for incoming calls. Here, a called number “30xx” is converted to “40xx” for an incoming call associated with this dial peer (i.e., via this SIP trunk).
4 This command is not necessary when using Avaya Communications Manager 4.0, and was not used for the creation of this application note. It is only needed with prior releases of Avaya Communications Manager.
5 This is to specify a digit string that can be matched by an incoming call to associate the call with a dial peer. For this example, user dials 3xxx will associate to this dial-peer.
6 This is for Dual-Tone Multifrequency (DTMF) tones to be sent RFC2833. It is omitted for inband DTMF.
7 Also changed to G.711A, G.729, and G.729B during testing.
banner login ^C
!
line con 0
exe-timeout 600 0
password cisco
login
line aux 0
line vty 0 4
exe-timeout 600 0
privilege level 15
password cisco
login
transport input telnet
line vty 5 15
privilege level 15
login local
transport input telnet
!
scheduler allocate 20000 1000
!
end
Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>codec</td>
<td>compressor/decompressor</td>
</tr>
<tr>
<td>PBX</td>
<td>Private Branch Exchange</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>IOS</td>
<td>Internetworking Operating System</td>
</tr>
</tbody>
</table>

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