Application Note

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

Table of Contents

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

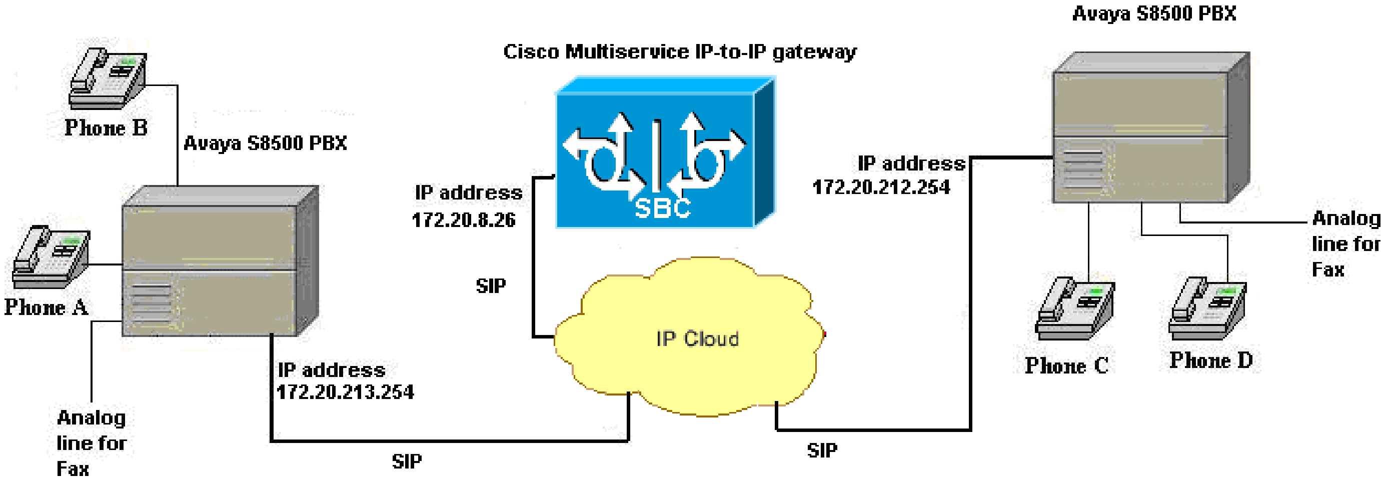

Figure 1. Network Topology or Test Setup

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.PI4.
- 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
- Cisco IOS Release: c3845-ipvoice_ivs-mz.124-9.T

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

    Port          Name
 1: T00001          OUTSIDE CA
 2: T00002          OUTSIDE CA
 3: T00003          OUTSIDE CA
 4: T00004          OUTSIDE CA
 5: T00059          OUTSIDE CA
 6: T00060          OUTSIDE CA
 7:
 8:
 9:
10:
11:
12:
13:
14:
15:
```

Node-names IP

```
          Voice System name: S8500SIP2 - IP NODE NAMES
  Name            IP Address
CCM3.3            172.20 .31 .254
CCM4.1            172.20 .231.254
CCM4.1.2          172.20 .236.2
CCM5.0-VENUS        172.20 .214.254
CM-KLINGON          172.20 .32 .254
CM-POLARIS          172.20 .236.50
IPIPGW            172.20 .8  .26
MAvantage           172.20 .7  .252
avayasip1         172.20 .212.254   → Far-end SIP Proxy
avayasip2         172.20 .213.254   → Near-end SIP Proxy
clan1           172.20 .213.253     → PBX connection to avayaSIP2 (tls)
clan1server1       172.20 .212.253
default          0  .0  .0  .0
medpro1           172.20 .213.252
procr              .   .   .

( 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              Inter-region IP-IP Direct Audio: no
      Codec Set: 1                   IP Audio Hairpinning? y
   UDP Port Min: 2048
   UDP Port Max: 3028                 RTCP Reporting Enabled? y
DIFFSERV/TOS PARAMETERS          RTCP MONITOR SERVER PARAMETERS
 Call Control PHB Value: 34       Use Default Server Parameters? y
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                              RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5

                        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
```

IP-codec

```
      Voice System name: S8500SIP2 - IP Codec Set

   Codec Set: 1

   Audio      Silence      Frames   Packet
   Codec      Suppression  Per Pkt  Size(ms)
 1: G.711MU        n         2        20
 2: G.729AB        n         2        20
 3: G.723-6.3K     n         1        30
 4:
 5:
 6:
 7:


    Media Encryption
 1: none
 2:
 3:



             IP Codec Set

               Allow Direct-IP Multimedia? n


          Mode            Redundancy
   FAX         pass-through      0
   Modem         pass-through      0
   TDD/TTY       US                3
   Clear-channel  n                0
```

→ **This field is changed to T.38 for Fax over T.38 codec**

Uniform dialing

Voice System name: S8500SIP2 - UNIFORM DIAL PLAN TABLE

Percent Full: 0

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num	Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
4154	4	0	222	aar	n				n				
4155	4	0	222	aar	n				n				
4156	4	0	222	aar	n				n				

AAR Analysis

Voice System name: S8500SIP2 - AAR DIGIT ANALYSIS TABLE

Percent Full: 1

Dialed String	Total Min	Max	Route Pattern	Call Type	Node Num	ANI Reqd
222	7	7	99	aar	n	

Route Pattern

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

SCCAN? n Secure SIP? n

Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del Dgts	Inserted Digits	DCS/ QSIG Intw	IXC
1: 1	0					3		n	user
2:								n	user
3:								n	user
4				:				n	user
5:								n	user
6:								n	user

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

Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del Dgts	Inserted Digits	DCS/ QSIG Intw	IXC
7:								n	user
8:								n	user
9:								n	user
10:								n	user
11:								n	user
12:								n	user

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.PI4a
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 dspfarm
!
!
!
!
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
!
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

Acronym	Definitions
CUBE	Cisco Unified Border Element
Cisco IOS	Cisco Internetwork Operating System
SIP	Session Initiation Protocol
RTP	Real-Time Protocol

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