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# Ribbon SBC Edge 7.0.0 IOT Avaya SM7 and CM7 AT&T IP Flex Reach SIP Trunk Application Notes

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# Document Overview

This document provides a configuration guide for Ribbon Session Border Controller Edge Series (SBC) when connecting to Avaya Session Manager 7 (SM7) and Avaya Communication Manager (CM7).

- For additional information on Avaya Platform, visit<http://www.avaya.com>.
- For additional information on Ribbon SBC, visit <http://ribboncommunications.com/>.

## Introduction

The interoperability compliance testing focuses on verifying inbound and outbound call flows between Ribbon SBC Edge and Avaya Session Manager 7 (SM7) and Avaya Communication Manager (CM7) .

## Audience

This technical document is intended for telecommunication engineers with the purpose of configuring the Ribbon SBC Edge series aspects of the AT&T Flex ReachSIP trunkgroup with the Avaya SM7 and CM7. This configuration requires the navigation of a third-party server and the Ribbon SBC Web browser user interface, Embedded Management Application (EMA). Understanding the basic concepts for IP/Routing, SIP, RTP, and TLS are also required for completing the configuration and any necessary troubleshooting.

## Requirements

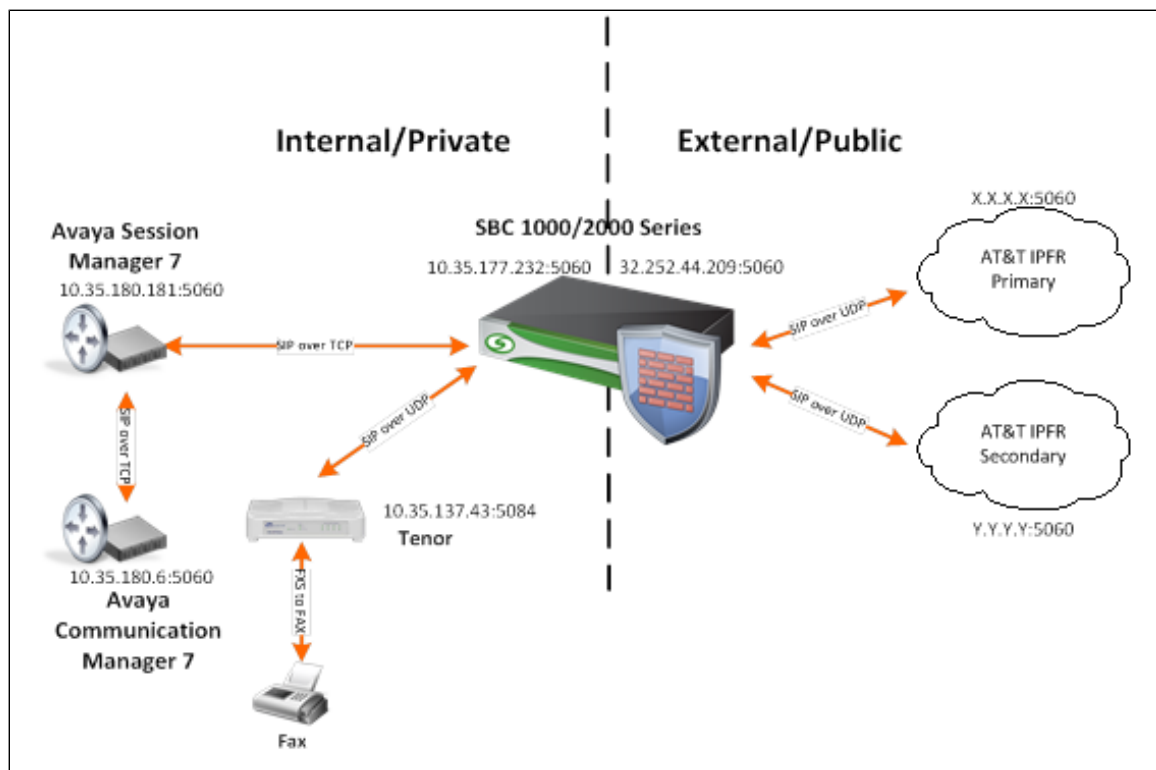
The following equipment and software were used for the sample configuration (see [Topology](#)):

	Equipment	Software Version
Ribbon Networks	Ribbon SBC Edge (2000)	7.0.0b476
Third-party Equipment	Avaya Communication Manager7.0	R017x.00.0.441.0
	Avaya Session Manager 7.0	7.0.0.0.700007
	Avaya 9608 IP Deskphone	6.3037

## Reference Configuration

The following reference configuration illustrates the connectivity between a third-party and the Ribbon SBC Edge..

Figure 1: Topology



## Support

For any questions regarding this document or the content herein, please contact your maintenance and support provider.

## Third-Party Product Features

The testing was executed with the AT&T test plan, and the following features were tested:

- Basic originated and terminated calls
- Calling Number presentation
- Hold and Resume
- Voice Mail
- Conference Call
- Call Transfer
- Call Forwarding
- Auto Attendant
- Meet-Meet Conference
- AT&T IP Teleconferencing
- N11 Calls
- FAX
- DTMF
- Network Based Enhanced Features

## Not Supported Features

- cRTP
- SBC does not send SIP with SDP without p-time
- SBC does not support network based transfer with SIP Refer method

## Verify License

No special licensing required.

## Avaya SM 7 Configuration

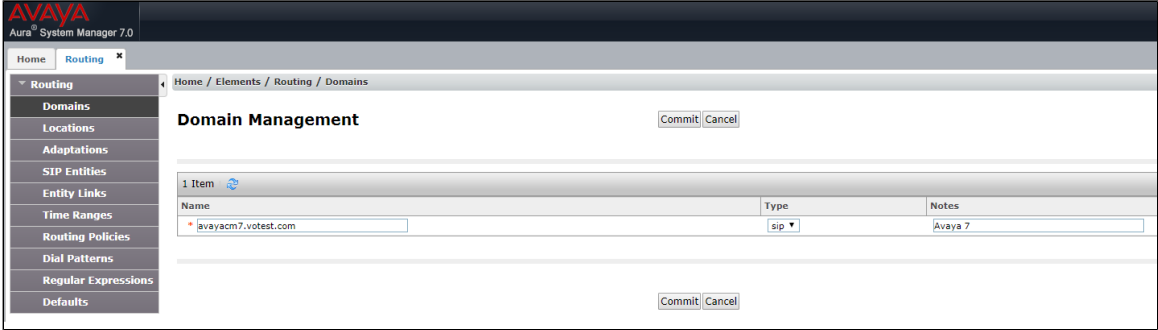
This section includes thefollowing new configurations:

- 1. Domain
- 2. Location
- 3. SIP Entity
- 4. Entity Link
- 5. Routing Policy
- 6. Dial Pattern

## 1. Domain

Select**Home> Routing > Domains**

Figure 2: Domain



## 2. Location

Select **Home> Routing > Locations**

Figure 3: Location for CM

**AVAYA**  
Aura System Manager 7.0

Home / Elements / Routing / Locations

**Location Details** [Commit] [Cancel]

**General**

\* Name: Dallas CM 7.0  
Notes:

**Dial Plan Transparency in Survivable Mode**

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

**Overall Managed Bandwidth**

Managed Bandwidth Units: Kbit/sec  
Total Bandwidth:  
Multimedia Bandwidth:  
Audio Calls Can Take Multimedia Bandwidth: ☒

**Per-Call Bandwidth Parameters**

Maximum Multimedia Bandwidth (Intra-Location): 1000 Kbit/Sec  
Maximum Multimedia Bandwidth (Inter-Location): 1000 Kbit/Sec  
\* Minimum Multimedia Bandwidth: 64 Kbit/Sec  
\* Default Audio Bandwidth: 80 Kbit/sec

**Alarm Threshold**

Overall Alarm Threshold: 80 %  
Multimedia Alarm Threshold: 80 %  
\* Latency before Overall Alarm Trigger: 5 Minutes  
\* Latency before Multimedia Alarm Trigger: 5 Minutes

**Location Pattern**

Add Remove

1 Item

☐ IP Address Pattern

☐ \* 10.35.180.1-10.35.180.30

Select : All, None

Figure 4: Location for SBC

**AVAYA**  
Aura System Manager 7.0

Home / Elements / Routing / Locations

**Location Details** [Commit] [Cancel]

**General**

\* Name: Saline SBC  
Notes: Saline SBC

**Dial Plan Transparency in Survivable Mode**

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

**Overall Managed Bandwidth**

Managed Bandwidth Units: Kbit/sec  
Total Bandwidth:  
Multimedia Bandwidth:  
Audio Calls Can Take Multimedia Bandwidth: ☒

**Per-Call Bandwidth Parameters**

Maximum Multimedia Bandwidth (Intra-Location): 2000 Kbit/Sec  
Maximum Multimedia Bandwidth (Inter-Location): 2000 Kbit/Sec  
\* Minimum Multimedia Bandwidth: 64 Kbit/Sec  
\* Default Audio Bandwidth: 80 Kbit/sec

**Alarm Threshold**

Overall Alarm Threshold: 80 %  
Multimedia Alarm Threshold: 80 %  
\* Latency before Overall Alarm Trigger: 5 Minutes  
\* Latency before Multimedia Alarm Trigger: 5 Minutes

**Location Pattern**

Add Remove

1 Item

☐ IP Address Pattern

☐ \* 10.35.177.232

### 3. SIP Entity

SelectHome> Routing > SIP Entities

Figure 5: SIP Entity for CM

AVAYA  
Aura® System Manager 7.0

Home / Elements / Routing / SIP Entities

**SIP Entity Details**

General

Name: Avaya-CH7  
FQDN or IP Address: 10.35.180.6  
Type: CH  
Notes: Avaya CH7

Adaptation: [dropdown]  
Location: Dallas CH 7.0  
Time Zone: America/Chicago

SIP Timer B/F (in seconds): 4  
Credential name: [text]  
Securable: [checkbox]  
Call Detail Recording: [none]

Loop Detection  
Loop Detection Mode: [Off]

SIP Link Monitoring  
SIP Link Monitoring: [Use Session Manager Configuration]

Supports Call Admission Control: [checkbox]  
Shared Bandwidth Manager: [checkbox]  
Primary Session Manager Bandwidth Association: [dropdown]  
Backup Session Manager Bandwidth Association: [dropdown]

Entity Links  
Override Port & Transport with DNS SRV: [checkbox]

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
Avaya-SM to Avaya-CH7	Avaya-SM	TCP	5060	Avaya-CH7	5060	Trusted	[checkbox]

SIP Responses to an OPTIONS Request

Response Code & Reason Phrase	Mark Entity Up/Down	Notes
-------------------------------	---------------------	-------

Figure 6: SIP Entity for SBC

AVAYA  
Aura® System Manager 7.0

Home / Elements / Routing / SIP Entities

**SIP Entity Details**

General

Name: Saline-SBC  
FQDN or IP Address: 10.35.177.232  
Type: SIP Trunk  
Notes: ATT IP Flex

Adaptation: Test  
Location: Saline SBC  
Time Zone: America/Chicago

SIP Timer B/F (in seconds): 4  
Credential name: [text]  
Securable: [checkbox]  
Call Detail Recording: egress

Loop Detection  
Loop Detection Mode: [Off]

SIP Link Monitoring  
SIP Link Monitoring: [Use Session Manager Configuration]

Supports Call Admission Control: [checkbox]  
Shared Bandwidth Manager: [checkbox]  
Primary Session Manager Bandwidth Association: [dropdown]  
Backup Session Manager Bandwidth Association: [dropdown]

Entity Links  
Override Port & Transport with DNS SRV: [checkbox]

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
Avaya-SM to Saline	Avaya-SM	TCP	5060	Saline-SBC	5060	Trusted	[checkbox]

### 4. Entity Link

SelectHome> Routing > Entity Links

Figure 7: Entity Link for CM

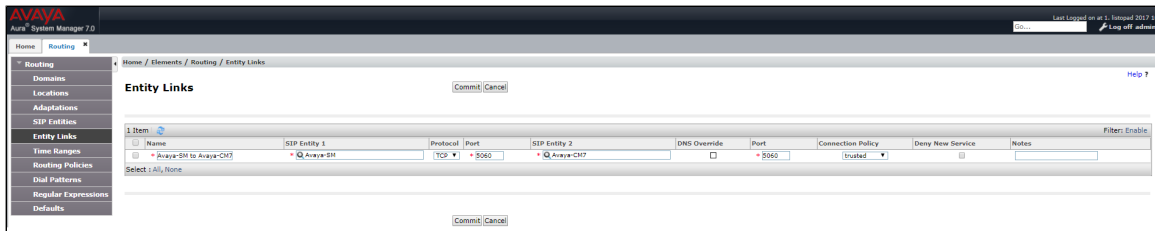
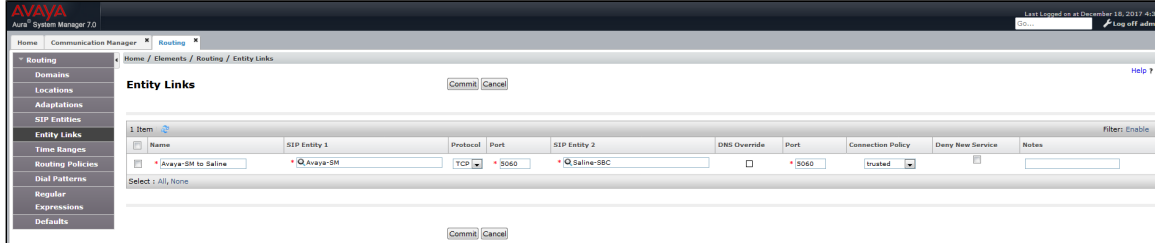


Figure 8: Entity Link for SBC



## 5. Routing Policy

SelectHome> Routing > Routing Policies

Figure 9: Route Policy for CM

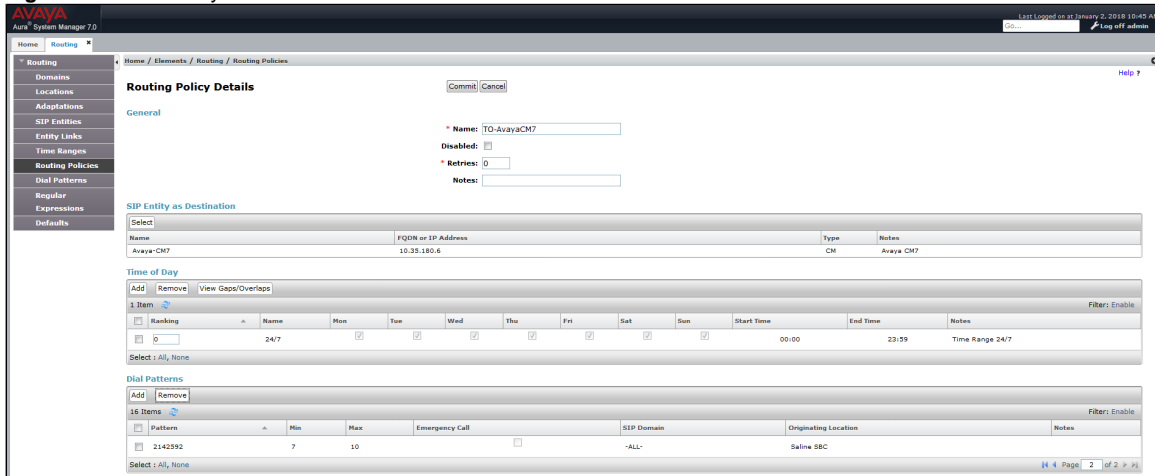
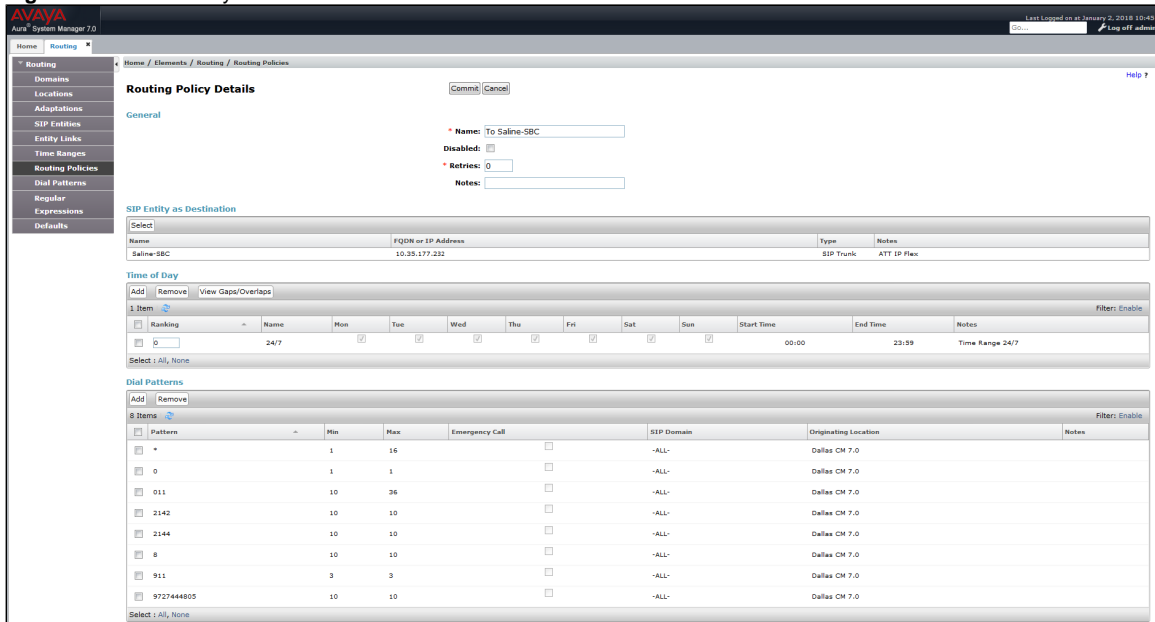


Figure 10: Route Policy for SBC



## 6. Dial Pattern

Select **Home > Routing > Dial Patterns**



### Note

Use this procedure to create any Dial Pattern configuration.

Figure 11: Dial Pattern for CM

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
Little-SBC-Loc		TO-AvayaCH7	0	<input type="checkbox"/>	Avaya-CH7	

Figure 12: Dial Pattern for SBC

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
Dallas CM 7.0		To Saline-SBC	0	<input type="checkbox"/>	Saline-SBC	

Select: All, None

Originating Location	Notes
Originating Location	

## Avaya CM 7 Configuration

This section includes the following new configurations:

1. Node Name
2. Signaling Group
3. Trunk Group
4. Route Pattern
5. ARS Digit Analysis Table
6. Station
7. Vector Directory Number
8. Vector



# 1.Node Name

- 1. Using the Site Administration, log into Avaya CM 6.0.
- 2. Type `change node-names ip` in the command line.
- 3. Press the**Down Arrow** key to a blank line and add the appropriate information.
- 4. Press **F3** to save when complete.

Figure 13: Node Name

Avaya7.0 GEDI

change node-names ip    send (return)    help (f5)    cancel (esc)    enter (f3)    schedule (f9)    next (f7)    previous (f8)

1   2

IP NODE NAMES

Name	IP Address
ASM6.3	10.35.180.41
ASM7.0	10.35.180.181
avayacm5-mas	10.35.180.11
avayacm5-ses	10.35.180.6
cd-verde	10.35.179.241
cimarron-ssm	10.35.176.32
clear-ssm	10.35.178.12
default	0.0.0.0
procr	10.35.180.6
procr6	::
rio-ssm	10.35.176.12
sbc-colby-hays	10.35.177.202
sbc-hays	10.35.177.200
sbc-little	10.35.177.247
sbc-long	10.35.177.228
sbc-newport	10.35.177.232

( 16 of 18    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

# 2. Signaling Group

- 1. Using the Site Administration, log into Avaya CM 7.0.
- 2. Type `add signaling-group next` in the command line.



**Note**

The "next" switch will auto-generate the next available group number for the Signaling Group and is the most efficient method to use when creating a new Signaling Group.

- 3. Add the appropriate information and press **F3** to save when complete. For more details, refer to [Avaya CM 7.0 guide](#).

Figure 14: Signaling Group

Avaya7.0 GEDI

change signaling-group 6   send (return)   help (f5)   cancel (esc)   enter (f3)   schedule (f9)   next (f7)   previous (f8)

1 | 2 |

### SIGNALING GROUP

Group Number: 6      Group Type: sip

IMS Enabled? ☐      Transport Method: tcp

Q-SIP? ☐

IP Video? ☐      Enforce SIPS URI for SRTP? ☒

Peer Detection Enabled? ☐      Peer Server: SM

Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? ☒

Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? ☒

Alert Incoming SIP Crisis Calls? ☐

Near-end Node Name: procr      Far-end Node Name: ASM7.0

Near-end Listen Port: 5060      Far-end Listen Port: 5060

Far-end Network Region: 1

Far-end Domain: avayacm7.votest.com

Incoming Dialog Loopbacks: eliminate      Bypass If IP Threshold Exceeded? ☐

DTMF over IP: rtp-payload      RFC 3389 Comfort Noise? ☐

Session Establishment Timer(min): 3      Direct IP-IP Audio Connections? ☒

Enable Layer 3 Test? ☒      IP Audio Hairpinning? ☐

H.323 Station Outgoing Direct Media? ☐      Initial IP-IP Direct Media? ☐

Alternate Route Timer(sec): 6

### 3. Trunk Group

1. Using the Site Administration, log into Avaya CM 7.0.
2. Type add trunk-group next in the command line.
3. Add the appropriate information and press **F3** to save when complete. For more details, refer to [Avaya CM 7.0 guide](#).

Figure 15: Trunk Group

Avaya7.0 GEDI

change trunk-group 6   send (return)   help (f5)   cancel (esc)   enter (f3)   schedule (f9)   next (f7)   previous (f8)

1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 | 13 | 14 | 15 | 16 | 17 | 18 | 19 | 20 | 21 |

### TRUNK GROUP

Group Number: 6      Group Type: sip      CDR Reports: ☒

Group Name: SIP Trunk to SM      COR: 1      TN: 1      TAC: #006

Direction: two-way      Outgoing Display? ☐

Dial Access? ☐      Night Service: ☐

Queue Length: 0

Service Type: public-ntwrk      Auth Code? ☐

Member Assignment Method: auto

Signaling Group: 6

Number of Members: 10

Avaya7.0 GEDI

display trunk-group 6
send (return)
help (f5)
cancel (esc)
enter (f3)
schedule (f9)
next (f7)
previous (f8)

123456789101112131415161718192021

### PROTOCOL VARIATIONS

Mark Users as Phone? n

Prepend '+' to Calling/Alerting/Diverting/Connected Number? n

Send Transferring Party Information? n

Network Call Redirection? y

Build Refer-To URI of REFER From Contact For NCR? n

Send Diversion Header? y

Support Request History? n

Telephone Event Payload Type: 101

Convert 180 to 183 for Early Media? y

Always Use re-INVITE for Display Updates? n

Identity for Calling Party Display: P-Asserted-Identity

Block Sending Calling Party Location in INVITE? n

Accept Redirect to Blank User Destination? n

Enable Q-SIP? n

Interworking of ISDN Clearing with In-Band Tones: keep-channel-active

Request URI Contents: may-have-extra-digits

## 4. Route Pattern

1. Using Site Administration, log into Avaya CM 7.0.
2. Type `list route-pattern` in the command line to determine the next available route pattern.
3. Identify the route pattern number you are going to use and then press **F1** to exit the current operation.
4. Type `change route-pattern` and then enter the available route pattern number. Add the appropriate information and press **F3** to save when complete.

**Figure 16:** Route Pattern

Avaya7.0 GEDI

change route-pattern 6 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8)

1 2 3

Pattern Number: 6 Pattern Name: SIP to ASM

SCCAN? n Secure SIP? n Used for SIP stations? n

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

BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	Sub	Numbering	LAR
0	1	2	M	4	W	Request	Dgts	Format		
1:	y	y	y	y	n	n	rest			none
2:	y	y	y	y	n	n	rest			none
3:	y	y	y	y	n	n	rest			none
4:	y	y	y	y	n	n	rest			none
5:	y	y	y	y	n	n	rest			none
6:	y	y	y	y	n	n	rest			none

## 5. ARS Analysis Table

1. Using the Site Administration, log into Avaya CM 7.0.
2. Type change ars analysis (dialed number) to add or change the called number handling.

Figure 17: ARS Digit Analysis Table

Avaya7.0 GEDI

change ars analysis 21443268 li send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8)

1 2

ARS DIGIT ANALYSIS TABLE

Location: all Percent Full: 3

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
21443268	10	10	6	hnpa		n

## 6. Station

1. Using the Site Administration, log into Avaya CM 7.0.
2. Type add station next to add a new station.

Figure 18: Station

Avaya7.0 GEDI
display station 12142592198
send (return)
help (f5)
cancel (esc)
enter (f3)
schedule (f9)
next (f7)
previous (f8)

1
2
3
4
5

STATION

Extension: 1214-259-2198
Type: 9608
Port: S00022
Name: IPFR3

Lock Messages? n
Security Code: \*
Coverage Path 1:
Coverage Path 2:
Hunt-to Station:

BCC: 0
TN: 1
COR: 1
COS: 1
Tests? n

STATION OPTIONS

Loss Group: 19
Speakerphone: 2-way
Display Language: english
Survivable GK Node Name:
Survivable COR: internal
Survivable Trunk Dest? y

Time of Day Lock Table:
Personalized Ringing Pattern: 1
Message Lamp Ext: 1214-259-2198
Mute Button Enabled? y
Button Modules: 0
Media Complex Ext:
IP SoftPhone? y
IP Video Softphone? n
Short/Prefixed Registration Allowed: default
Customizable Labels? y

## 7. Vector Directory Number

- Using the Site Administration, log into Avaya CM 7.0.
- Type `add vdn <extension>` to add a new vector directory number.

**Figure 19:** Vector Directory Number for Auto Attendant

Avaya7.0 GEDI

display vdn 2142592293    send (return)    help (f5)    cancel (esc)    enter (f3)    schedule (f9)    next (f7)    previous (f8)

1 2

**VECTOR DIRECTORY NUMBER**

Extension: 214-259-2293  
Name\*: AAforCM  
Destination: Vector Number      1  
Attendant Vectoring? n  
Meet-me Conferencing? n  
Allow VDN Override? n  
COR: 1  
TN\*: 1  
Measured: none

VDN of Origin Annc. Extension\*:

\* Follows VDN Override Rules

**Figure 20:** Vector Directory Number for Meet-Me Conference

Avaya7.0 GEDI

display vdn 2142592290    send (return)    help (f5)    cancel (esc)    enter (f3)    schedule (f9)    next (f7)    previous (f8)

1 2

**VECTOR DIRECTORY NUMBER**

Extension: 214-259-2290  
Name: Meet-Me  
Destination: Vector Number      2  
Meet-me Conferencing? y  
COR: 1  
TN: 1

## 8. Vector

1. Using the Site Administration, log into Avaya CM 7.0.
2. Typeadd vector next to add a new vector.

**Figure 21:** Vector for Auto Attendant

Avaya7.0 GEDI

display vector 1    send (return)    help (f5)    cancel (esc)    enter (f3)    schedule (f9)    next (f7)    previous (f8)

1 2 3 4 5 6

**CALL VECTOR**

Number: 1                      Name: AutoAttendant

Multimedia? n              Attendant Vectoring? n              Meet-me Conf? n              Lock? n

Basic? y              EAS? n              G3U4 Enhanced? y              ANI/II-Digits? y              ASAI Routing? y

Prompting? y              LAI? n              G3U4 Adv Route? y              CINFO? y              BSR? n              Holidays? y

Variables? n              3.0 Enhanced? n

01 announcement 1101

02 collect 1              digits after announcement 1100

03 goto step 6              if digits              =              1

04 goto step 7              if digits              =              2

05 wait-time 2              secs hearing silence

06 route-to number 2142592198              with cov n if unconditionally

07 route-to number 2142592199              with cov n if unconditionally

08 wait-time 5              secs hearing silence

09 stop

10

11

12

Figure 22: Vector for Meet-Me Conference

Avaya7.0 GEDI

display vector 2    send (return)    help (f5)    cancel (esc)    enter (f3)    schedule (f9)    next (f7)    previous (f8)

1 2 3 4 5 6

**CALL VECTOR**

Number: 2                      Name: Meet-Me

Multimedia? n              Attendant Vectoring? n              Meet-me Conf? y              Lock? y

Basic? y              EAS? n              G3U4 Enhanced? y              ANI/II-Digits? y              ASAI Routing? y

Prompting? y              LAI? n              G3U4 Adv Route? y              CINFO? y              BSR? n              Holidays? y

Variables? n              3.0 Enhanced? n

01 announcement 1104

02 wait-time 2              secs hearing silence

03 collect 6              digits after announcement 1103

04 wait-time 3              secs hearing silence

05 goto step 7              if digits              =              meet-me-access

06 wait-time 1              secs hearing silence

07 announcement 1102

08 route-to meetme

09

10

11

12

## Ribbon SBC Edge Series Configuration

The following steps provide an example of how to configure Ribbon SBC Edge.

1. [Easy Configuration Wizard](#)
2. [SIP Profile](#)
3. [Q.850 Cause Code to SIP Override Tables](#)
4. [Tone Table](#)
5. [Media Profile](#)

6. [Media System Configuration](#)
7. [Media List](#)
8. [Message Manipulation](#)
9. [SIP Server](#)
10. [Signaling Group](#)
11. [Transformation](#)
12. [Call Routing Table](#)

## 1. Easy Configuration Wizard

The SBCinterface includes an Easy Configuration Wizard, which enables end users to quickly deploy SBC. Based on a template, you can configure items such as endpoint (define user and provider), routing (routing configuration applied to scenario), and country (tone table parameters and emergency numbers for a particular country).

**Figure 23:** Easy Configuration Wizard

The screenshot displays the 'Easy Configuration' wizard interface. At the top, the title 'Easy Configuration' is on the left, and the date and time 'January 23, 2018 07:37:02' are on the right. Below the title bar, there are three tabs: 'Step 1' (active), 'Step 2', and 'Step 3'. To the right of the tabs, a message states 'This step takes input about the topology'. The main content area is divided into several sections. The 'Scenario Parameters' section includes dropdown menus for 'Application' (set to 'SIP Trunk <-> IP PBX'), 'Scenario Description' (set to 'AT&T IPFR'), 'Telephone Country' (set to 'United States'), and 'Emergency Services' (set to 'None'). Below this is the 'SIP Properties' section with a text input for 'SIP Sessions' set to '10' and a note '\* [1..960]'. At the bottom, there are two side-by-side sections: 'SIP Trunk' with a 'Name' dropdown set to 'ATT SIP Trunk', and 'IP PBX' with a 'Type' dropdown set to 'Avaya SM/CM'. The bottom of the window features a 'Cancel' button on the left and 'Previous', 'Next', and 'Finish' buttons on the right.



Easy Configuration

January 23, 2018 07:37:02

Step 1

Step 2

Step 3

This step takes input about the Provider and User side configuration

SIP Trunk: ATT SIP Trunk

Border Element Server

X.X.X.X

\* FQDN or IP

Protocol

UDP

Port Number

5060

[1024-65535]

Use Secondary Border Element Server

Enabled

Secondary Border Element Server

Y.Y.Y.Y

\* FQDN or IP

Protocol

UDP

Port Number

5060

[1024-65535]

AT&T Services

AT&T Simultaneous Ring Supported

Yes

AT&T IP Toll Free

Disabled

IP PBX: Avaya SM/CM

Cancel

Previous

Next

Finish

Easy Configuration

January 23, 2018 07:37:02

Step 1

Step 2

Step 3

This step takes input about the Provider and User side configuration

SIP Trunk: ATT SIP Trunk

IP PBX: Avaya SM/CM

Host

10.35.180.181

\* FQDN or IP

Protocol

TCP

Port Number

5060

[1024-65535]

Use Secondary Server

Disabled

Cancel

Previous

Next

Finish

Easy Configuration
January 23, 2018 07:37:02

Step 1
Step 2
Step 3

This step is a summary of what will be configured

SBC Setup Configuration Summary

Scenario Parameters

ApplicationSIP Trunk <-> IP PBX  
Scenario DescriptionAT&T IPFR  
Telephone CountryUnited States  
Emergency ServicesNone  
SIP Properties  
SIP Sessions10

SIP Trunk: ATT SIP Trunk

Border Element ServerX.X.X.X  
ProtocolUDP  
Port Number5060  
Use Secondary Border Element ServerEnabled  
Secondary Border Element ServerY.Y.Y.Y  
ProtocolUDP  
Port Number5060

IP PBX: Avaya SM/CM

Host10.35.180.181  
ProtocolTCP  
Port Number5060  
Use Secondary ServerDisabled

Cancel
Previous
Next
Finish

## 2.SIP Profile

SIP Profiles control how the Ribbon SBC Edge communicates with SIP devices. The SIP Profiles control characteristics such as

- Session timers
- SIP Header customization
- SIP timers
- MIME payloads
- Option tags

To configure the SIP Profiles, select **Settings > SIP > SIP Profiles**.

**Figure 24:** AT&T-IPFR: Avaya SIP Profile

Description <b>AT&amp;T-IPFR: Avaya Profile</b>		
<b>Session Timer</b>		
Session Timer	Enable	
Minimum Acceptable Timer	600	
Offered Session Timer	1800	
Terminate On Refresh Failure	False	
<b>MIME Payloads</b>		
ELIN Identifier	LOC	
PIDF-LO Passthrough	Enable	
Unknown Subtype Passthrough	Disable	
<b>Header Customization</b>		
FQDN in From Header	Disable	
FQDN in Contact Header	Disable	
Send Assert Header	Never	
Sonus Diagnostics Header	Enable	
Trusted Interface	Enable	
UA Header	Sonus SBC	
Calling Info Source	RFC Standard	
Diversion Header Selection	Last	
Record Route Header	RFC 3261 Standard	
<b>Options Tags</b>		
100rel	Supported	
Path	Not Present	
Timer	Supported	
Update	Supported	
<b>Timers</b>		
Transport Timeout Timer	5000	
Maximum Retransmissions	RFC Standard	
<b>RFC Timers</b>		
Timer T1	500	
Timer T2	4000	
Timer T4	5000	
Timer D	32000	
Timer B	32000 <i>ms</i>	
Timer F	32000 <i>ms</i>	
Timer H	32000 <i>ms (64*TimerT1)</i>	
Timer J	4000	
<b>SDP Customization</b>		
Send Number of Audio Channels	True	
Connection Info in Media Section	True	
Origin Field Username	SBC	
Session Name	VoipCall	
Digit Transmission Preference	RFC 2833/Voice	
SDP Handling Preference	Legacy Audio/Fax	

**Figure 25:** AT&T-IPFR: ATT SIP Profile

Description <b>AT&amp;T-IPFR: ATT Profile</b>	
<b>Session Timer</b> Session Timer <b>Disable</b>	<b>MIME Payloads</b> ELIN Identifier <b>LOC</b> PIDF-LO Passthrough <b>Enable</b> Unknown Subtype Passthrough <b>Disable</b>
<b>Header Customization</b> FQDN in From Header <b>Disable</b> FQDN in Contact Header <b>Disable</b> Send Assert Header <b>Trusted Only</b> Sonus Diagnostics Header <b>Disable</b> Trusted Interface <b>Enable</b> UA Header <b>Sonus SBC</b> Calling Info Source <b>RFC Standard</b> Diversion Header Selection <b>Last</b> Record Route Header <b>RFC 3261 Standard</b>	<b>Options Tags</b> 100rel <b>Supported</b> Path <b>Not Present</b> Update <b>Supported</b>
<b>Timers</b> Transport Timeout Timer <b>5000</b> Maximum Retransmissions <b>RFC Standard</b> <hr/> <b>RFC Timers</b> Timer T1 <b>500</b> Timer T2 <b>4000</b> Timer T4 <b>5000</b> Timer D <b>32000</b> Timer B <b>32000 ms</b> Timer F <b>32000 ms</b> Timer H <b>32000 ms (64*TimerT1)</b> Timer J <b>4000</b>	<b>SDP Customization</b> Send Number of Audio Channels <b>False</b> Connection Info in Media Section <b>True</b> Origin Field Username <b>SBC</b> Session Name <b>VoipCall</b> Digit Transmission Preference <b>RFC 2833/Voice</b> SDP Handling Preference <b>Legacy Audio/Fax</b>

### 3. Q.850 Cause Code to SIP Override Table

By default, the SBC Edge uses RFC 3398 cause code mappings. Q.850 Cause Code to SIP Override Table allows you to define other mappings for cause codes.

To configure the Q.850 Cause Code to SIP Override Table, select **Q.850 Cause Code to SIP Override Tables**.

**Figure 26:** Q.850 Cause Code to SIP Override Table AT&T-IPFR: ATT

Q.850 Cause Code	<b>18: No User Responding</b>
SIP Response	<b>486 - Busy Here</b>

Q.850 Cause Code 47: Resource Unavailable, Unspecified  
SIP Response 403 - Forbidden

## 4. Tone Tables

Tone tables allow the SBC Edgeadministrator to customize the tones a user hears when placing a call. You can modify the tone to match your local PSTN or PBX. The default tone table configures the following categories with the United States' values:

- Ringback
- Dial
- Busy
- Congestion
- Call Waiting
- Disconnect
- Confirmation

To configure the Tone Tables, select **Settings > Tone Tables**.

**Figure 27:** Tone Table AT&T-IPFR: United States

Tone Type	Frequency 1 (Hz)	Amplitude 1 (dBm)	Frequency 2 (Hz)	Amplitude 2 (dBm)
▶ Ringback	440	-19	480	-19
▶ Dial	350	-13	440	-13
▶ Busy	480	-24	620	-24
▶ Congestion	480	-24	620	-24
▶ Call Waiting	440	-13	0	0
▶ Disconnect	480	-24	620	-24
▶ Confirmation	350	-13	440	-13

## 5. Media Profile

Media profiles specify the individual voice and fax compression codecs, and their associated settings for inclusion into a Media list. Different codecs provide varying levels of compression, which enables the reduction of bandwidth requirements at the expense of voice quality.

To access the Media Profile, select **Settings > Media > Media Profiles**.

**Figure 28:** AT&T-IPFR (Avaya)

Voice Codec Configuration	
Description	AT&T-IPFR (Avaya): G.711 Mu-Law
Codec	G.711 $\mu$ -Law
Payload Size	20

Voice Codec Configuration	
Description	AT&T-IPFR (Avaya): G.729
Codec	G.729
Payload Size	20

**Figure 29:** AT&T-IPFR (ATT)

Fax Codec Configuration	
Description	AT&T-IPFR (ATT): Fax
Codec	T.38 Fax
Maximum Rate	14400
Signaling Packet Redundancy	3
Payload Packet Redundancy	0
Error Correction Mode	Enabled
Training Confirmation Procedure	Send Over Network
Fallback to Passthrough	Enabled
Super G3 to G3 Fallback	Disabled

Voice Codec Configuration	
Description	AT&T-IPFR (ATT): G.711 Mu-Law
Codec	G.711 $\mu$ -Law
Payload Size	20

Voice Codec Configuration	
Description	AT&T-IPFR (ATT): G.729
Codec	G.729
Payload Size	20

## 6. Media System Configuration

The Media System Configuration contains systemwide settings for the Media System. To configure the Media System, set the number of RTP/RTCP port pairs and the starting port.

To access the Media Profile, select **Settings > Media > Media System Configuration**.

**Figure 30:** Media System Configuration

Port Range

Start Port16384\* [1024..32767]

Number of Port Pairs1500\* [1..4800]

Echo Canceller Type OptionStandard▼

Echo Cancel NLP OptionMild▼

Send STUN PacketsEnabled▼

Music on Hold

Music on Hold SourceFile▼

Current Music File

7. Media List

The Media List shows the selected voice and fax compression codecs and their associated settings.

To access Media lists, select **Settings > Media > Media List**.

Figure 31: AT&T-IPFR: Avaya List

Description	AT&T-IPFR: Avaya List	
Media Profiles List	<div>AT&amp;T-IPFR (Avaya): G.711 Mu-La AT&amp;T-IPFR (Avaya): G.729 AT&amp;T-IPFR (Avaya): T.38</div>	*
Crypto Profile ID	None	
Media DSCP	46	
RTCP Mode	RTCP	
Dead Call Detection	Disabled	
Silence Suppression	Enabled	

Gain Control

Receive Gain	0
Transmit Gain	0

Digit Relay

Digit (DTMF) Relay Type	RFC 2833
Digit Relay Payload Type	101

Passthrough/Tone Detection

Modem Passthrough	Disabled
Fax Passthrough	Disabled
CNG Tone Detection	Enabled
Fax Tone Detection	Enabled
DTMF Signal to Noise	0

Figure 32: AT&T-IPFR: ATT Trunk List



Description	AT&T-IPFR: ATT Trunk List	
Media Profiles List	<div> AT&amp;T-IPFR (ATT): G.729  AT&amp;T-IPFR (ATT): G.711 Mu-Law  AT&amp;T-IPFR (ATT): Fax </div>	*
Crypto Profile ID	None	
Media DSCP	10	
RTCP Mode	RTCP	
Dead Call Detection	Disabled	
Silence Suppression	Enabled	

Gain Control	
Receive Gain	0
Transmit Gain	0

Digit Relay	
Digit (DTMF) Relay Type	RFC 2833
Digit Relay Payload Type	101

Passthrough/Tone Detection	
Modem Passthrough	Enabled
Fax Passthrough	Enabled
CNG Tone Detection	Enabled
Fax Tone Detection	Enabled
DTMF Signal to Noise	0

## 8. Message Manipulation

Condition rules are rules that apply to a specific component of a message (for example, `diversion.uri.host`, `from.uri.host`, and such) with the value in the Match Type list box. The value is matched against a literal value, token, or REGEX.

To configure Message Manipulation, select **Settings > SIP > Message Manipulation > Condition Rule Table**.

The rule on the next figure changes a host part for the PAID (P-Asserted-Identity) header for all outbound calls to ATT SIP Trunk with an IP address of public interface.

**Figure 33: SMM TO ATT**

Description

PAID Change

Condition Expression

Admin State

Enabled

Result Type

Optional

Header Action

Modify

Header Name

P-Asserted-Identity \*

Header Ordinal Number

1st

▼ Header Value

Display Name

Ignore

▼ URI

URI Scheme

Ignore

URI User Info

Ignore

URI Host

Modify

'32.252.44.210'

URI Port

Ignore

URI Parameters

Total 0 SPRUriParam Rows

Name	Value	Action
-- Table is empty --		

Header Parameters

Total 0 SPRHeaderParam Rows

Name	Value	Action
-- Table is empty --		

9. SIP Server

The SIP Server tables contain information about the SIP devices connected to the Ribbon SBC Edge. The table entries provide information about the IP addresses, ports, and protocols used to communicate with each SIP server. The entries also contain links to counters that are useful for troubleshooting.

To configure the SIP Server, select **Settings > SIP > SIP Server Tables**.

Figure 34: AT&T-IPFR: Avaya CM

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	None
Priority	1		
Host	10.35.180.181		
Port	5060		
Protocol	TCP		

Remote Authorization and Contacts		Connection Reuse	
Remote Authorization Table	None	Reuse	True
Contact Registrant Table	None	Sockets	4
Session URI Validation	Liberal	Reuse Timeout	Forever

Figure 35: AT&T-IPFR: Border Element

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	1	Keep Alive Frequency	30
Host	Y.Y.Y.Y	Recover Frequency	5
Port	5060	Local Username	Anonymous
Protocol	UDP	Peer Username	Anonymous

Remote Authorization and Contacts	
Remote Authorization Table	None
Contact Registrant Table	None
Session URI Validation	Liberal

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	1	Keep Alive Frequency	30
Host	X.X.X.X	Recover Frequency	5
Port	5060	Local Username	Anonymous
Protocol	UDP	Peer Username	Anonymous

Remote Authorization and Contacts	
Remote Authorization Table	None
Contact Registrant Table	None
Session URI Validation	Liberal

Figure 36: Fax-TenorGW

Server Host	
Server Lookup	IP/FQDN
Priority	1
Host	10.35.137.43
Port	5084
Protocol	UDP

Transport	
Monitor	None

Remote Authorization and Contacts	
Remote Authorization Table	None
Contact Registrant Table	None
Session URI Validation	Liberal

## 10. Signaling Group

Signaling Groups allow telephony channels to be grouped together for routing and shared configuration. The Signaling Groups are the entity to which calls are routed and where the Call Routes are selected. In the case of SIP, Signaling Groups will specify protocol settings and link to server, media, and mapping tables.

To configure Signaling Groups, select **Settings > Signaling Groups**.

**Figure 37:** AT&T-IPFR: Avaya CM

Description

AT&T-IPFR: Avaya CM

Admin State

Enabled

Service Status

Up

SIP Channels and Routing

Action Set Table

None

Call Routing Table

AT&T-IPFR: From Avaya CM

No. of Channels

10

SIP Profile

AT&T-IPFR: Avaya Profile

SIP Mode

Basic Call

Agent Type

Back-to-Back User Agent

Interop Mode

Standard

SIP Server Table

AT&T-IPFR: Avaya CM

Load Balancing

Round Robin

Channel Hunting

Most Idle

Notify Lync CAC Profile

Disable

Challenge Request

Disable

Outbound Proxy

Outbound Proxy Port

No Channel Available Override

34: No Circuit/Channel Available

Call Setup Response Timer

180

Call Proceeding Timer

180

QoE Reporting

Disabled

Use Register as Keep Alive

Enable

Forked Call Answered Too Soon

Disable

Media Information

Audio/Fax Stream Mode

DSP

Proxy

Direct

\*

Video/Application Stream Mode

Proxy

Direct

\*

Media List ID

AT&T-IPFR: Avaya List

Play Ringback

Auto on 180

Tone Table

AT&T-IPFR: United States

Play Congestion Tone

Disable

Early 183

Disable

Allow Refresh SDP

Enable

Music on Hold

Disabled

Mapping Tables

SIP To Q.850 Override Table

Default (RFC4497)

Q.850 To SIP Override Table

Default (RFC4497)

Pass-thru Peer SIP Response Code

Enable

SIP IP Details

Signaling/Media Source IP

Auto

Signaling DSCP

40

Static NAT - Outbound

Outbound NAT Traversal

None

Static NAT - Inbound

Detection

Disabled

Listen Ports

Total 1 SIP Listen Port Row

Port	Protocol	TLS Profile ID
5060	TCP	N/A

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
10.35.180.181	255.255.255.255

Message Manipulation

Disabled

**Figure 38:** AT&T-IPFR: ATT Border Element

DescriptionAT&T-IPFR: ATT Border Element

Admin StateEnabled

Service StatusUp

SIP Channels and Routing

Action Set TableAT&T-IPFR: ATT Action Set

Call Routing TableAT&T-IPFR: From ATT

No. of Channels10

SIP ProfileAT&T-IPFR: ATT Profile

SIP ModeBasic Call

Agent TypeBack-to-Back User Agent

Interop ModeStandard

SIP Server TableAT&T-IPFR: Border Element

Load BalancingFirst

Channel HuntingMost Idle

Notify Lync CAC ProfileDisable

Challenge RequestDisable

Outbound Proxy

Outbound Proxy Port

No Channel Available Override34: No Circuit/Channel Available

Call Setup Response Timer255

Call Proceeding Timer180

QoE ReportingDisabled

Use Register as Keep AliveEnable

Forked Call Answered Too SoonDisable

Media Information

Audio/Fax Stream Mode

DSPProxyDirect

\*

Video/Application Stream Mode

ProxyDirect

\*

Media List IDAT&T-IPFR: ATT Trunk List

Play RingbackAuto on 180

Tone TableAT&T-IPFR: United States

Play Congestion ToneDisable

Early 183Disable

Allow Refresh SDPEnable

Music on HoldDisabled

Mapping Tables

SIP To Q.850 Override TableDefault (RFC4497)

Q.850 To SIP Override TableAT&T-IPFR: ATT

Pass-thru Peer SIP Response CodeDisable

SIP IP Details

Signaling/Media Source IPAuto

Signaling DSCP10

Static NAT - Outbound

Outbound NAT TraversalNone

Static NAT - Inbound

DetectionDisabled

Listen Ports

Total 1 SIP Listen Port Row

Port	Protocol	TLS Profile ID
5060	UDP	N/A

Federated IP/FQDN

Total 2 SIP Federated IP Rows

IP/FQDN	Netmask/Prefix
12.194.18.88	255.255.255.255
12.194.20.88	255.255.255.255

Message ManipulationEnabled

Inbound Message Manipulation

Message Table List

\*

Outbound Message Manipulation

Message Table List

SMM TO ATT

\*

Figure 39: Fax-TenorGW

Description

Fax-TenorGW

Admin State

Enabled

Service Status

Up

SIP Channels and Routing

Action Set Table

None

Call Routing Table

From Fax-TenorGW

No. of Channels

60

SIP Profile

AT&T-IPFR: ATT Profile

SIP Mode

Basic Call

Agent Type

Back-to-Back User Agent

Interop Mode

Standard

SIP Server Table

Fax-TenorGW

Load Balancing

Round Robin

Channel Hunting

Most Idle

Notify Lync CAC Profile

Disable

Challenge Request

Disable

Outbound Proxy

Outbound Proxy Port

5060

No Channel Available Override

34: No Circuit/Channel Available

Call Setup Response Timer

255

Call Proceeding Timer

180

QoE Reporting

Disabled

Use Register as Keep Alive

Enable

Forked Call Answered Too Soon

Disable

Media Information

Audio/Fax Stream Mode

DSP

Proxy

Direct

\*

Video/Application Stream Mode

Proxy

Direct

\*

Media List ID

AT&T-IPFR: ATT Trunk List

Play Ringback

Auto on 180

Tone Table

Default Tone Table

Play Congestion Tone

Disable

Early 183

Disable

Allow Refresh SDP

Enable

Music on Hold

Disabled

Mapping Tables

SIP To Q.850 Override Table

Default (RFC4497)

Q.850 To SIP Override Table

Default (RFC4497)

Pass-thru Peer SIP Response Code

Enable

SIP IP Details

Signaling/Media Source IP

Ethernet 1 IP (10.35.177.232)

Signaling DSCP

40

Static NAT - Outbound

Outbound NAT Traversal

None

Static NAT - Inbound

Detection

Disabled

Listen Ports

Total 2 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	UDP	N/A
5060	TCP	N/A

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
10.35.137.43	255.255.255.255

## 11. Transformation

Transformation tables facilitate the conversion of names, numbers, and other fields when routing a call. For example, transformation table converts a public PSTN number into a private extension number or a SIP address (URI). Every entry in a Call Routing table requires Transformation tables, which are sequentially selected. In addition, Transformation tables are configurable as a reusable pool that action sets can reference.

To configure the Transformation table, select **Settings > Transformation**.

**Figure 40:** AT&T-IPFR: From ATT

DescriptionAdd CC to CLD

Admin StateEnabled

Match TypeOptional (Match One)

Input Field

TypeCalled Address/Number

Value(21425921\d{2})

Output Field

TypeCalled Address/Number

Value1\1

DescriptionPassthrough

Admin StateEnabled

Match TypeMandatory (Must Match)

Input Field

TypeCalled Address/Number

Value(.\*)

Output Field

TypeCalled Address/Number

Value\1

Figure 41: AT&T-IPFR: From Avaya CM

DescriptionStrip 1 for Diversion

Admin StateEnabled

Match TypeOptional (Match One)

Input Field

TypeRedirecting Number

Value1(\d{10})

Output Field

TypeRedirecting Number

Value\1



DescriptionPassthrough

Admin StateEnabled

Match TypeMandatory (Must Match)

Input Field

TypeCalled Address/Number

Value(.\*)

Output Field

TypeCalled Address/Number

Value\\1

DescriptionRedirecting Passthrough

Admin StateEnabled

Match TypeOptional (Match One)

Input Field

TypeRedirecting Number

Value(.\*)

Output Field

TypeRedirecting Number

Value\\1

Figure 42: From Fax-TenorGW

DescriptionFrom Fax-TenorGw Passthrough

Admin StateEnabled

Match TypeOptional (Match One)

Input Field

TypeCalled Address/Number

Value(.\*)

Output Field

TypeCalled Address/Number

Value\\1

Figure 43: From ATT to FAX

Description

Call From ATT to FAX

Admin State

Enabled

Match Type

Optional (Match One)

Input Field

Type

Called Address/Number

Value

(2142592292)

Output Field

Type

Called Address/Number

Value

\1

## 12. Call Routing Table

Call Routing allows calls to be carried between Signaling Groups, which allows calls to be carried between ports and between protocols (for example, ISDN to SIP). Routes are defined by Call Routing tables, which allows for flexible configuration of calls that are carried, as well as how the calls are translated. These tables are one of the central connection points of the system linking Transformation tables, Message translations, Cause Code Reroutes, Media lists, and the three types of Signaling Groups: ISDN, SIP, and CAS.

To configure the Call Routing Table, select **Settings > Call Routing Table**.

**Figure 44:** AT&T-IPFR: From Avaya CM

Route Details	
Description	To Outside (Passthrough)
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	AT&T-IPFR: From Avaya CM: Passthrough

Destination Information	
Destination Type	Normal
Message Translation Table	None
Cause Code Reroutes	None
Cancel Others upon Forwarding	Disabled
Fork Call	Not Licensed
Destination Signaling Groups	<div> <div>(SIP) AT&amp;T-IPFR: ATT Border Element</div> <div></div> </div>
Enable Maximum Call Duration	Disabled

Media	
Audio/Fax Stream Mode	DSP
Video/Application Stream Mode	Disabled
Media Transcoding	Enabled
Media List	None

Quality of Service	
Quality Metrics Number of Calls	10
Quality Metrics Time Before Retry	10
Min. ASR Threshold	0
Enable Min MOS Threshold	Disabled
Enable Max. R/T Delay	Enabled
Max. R/T Delay	9999
Enable Max. Jitter	Enabled
Max. Jitter	3000

**Figure 45:** AT&T-IPFR: From ATT

### Route Details

Description	To Avaya CM (Passthrough)
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	AT&T-IPFR: From ATT: Passthrough

### Destination Information

Destination Type	Normal
Message Translation Table	None
Cause Code Reroutes	None
Cancel Others upon Forwarding	Disabled
Fork Call	Not Licensed

Destination Signaling Groups

(SIP) AT&T-IPFR: Avaya CM

\*

Enable Maximum Call Duration	Disabled
------------------------------	----------

### Media

Audio/Fax Stream Mode	DSP
Video/Application Stream Mode	Disabled
Media Transcoding	Enabled
Media List	None

### Quality of Service

Quality Metrics Number of Calls	10
Quality Metrics Time Before Retry	10
Min. ASR Threshold	0
Enable Min MOS Threshold	Disabled
Enable Max. R/T Delay	Enabled
Max. R/T Delay	9999
Enable Max. Jitter	Enabled
Max. Jitter	3000

Route Details

Description

To Fax-TenorGW

Admin State

Enabled

Route Priority

1

Call Priority

Normal

Number/Name Transformation Table

AT&T-IPFR: From ATT: FAX

Destination Information

Destination Type

Normal

Message Translation Table

None

Cause Code Reroutes

None

Cancel Others upon Forwarding

Disabled

Fork Call

Not Licensed

Destination Signaling Groups

(SIP) Fax-TenorGW

\*

Enable Maximum Call Duration

Disabled

Media

Audio/Fax Stream Mode

DSP

Video/Application Stream Mode

Disabled

Media Transcoding

Enabled

Media List

None

Quality of Service

Quality Metrics Number of Calls

10

Quality Metrics Time Before Retry

10

Min. ASR Threshold

0

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

Enable Max. Jitter

Enabled

Max. Jitter

3000

Figure 46: From Fax-TenorGW

Route Details	
Description	From Fax-TenorGW
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	From Fax-TenorGW

Destination Information	
Destination Type	Normal
Message Translation Table	None
Cause Code Reroutes	None
Cancel Others upon Forwarding	Disabled
Fork Call	Not Licensed
Destination Signaling Groups	<div> <div>(SIP) AT&amp;T-IPFR: ATT Border Element</div> <div></div> </div>
Enable Maximum Call Duration	Disabled

Media	
Audio/Fax Stream Mode	DSP
Video/Application Stream Mode	Disabled
Media Transcoding	Enabled
Media List	None

Quality of Service	
Quality Metrics Number of Calls	10
Quality Metrics Time Before Retry	10
Min. ASR Threshold	0
Enable Min MOS Threshold	Disabled
Enable Max. R/T Delay	Enabled
Max. R/T Delay	65535
Enable Max. Jitter	Enabled
Max. Jitter	3000

## Conclusion

These Application Notes describe the configuration steps required for Ribbon SBC Edge Seriesto successfully interoperate with AT&T IP Flex Reach SIP Trunk. All feature and serviceability test cases were completed.