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# Ribbon SBC Edge 7.0.0 IOT Cisco Unified Communication Manager 11.0 AT&T IP Flex Reach SIP Trunk Application Notes

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## Table of Contents

- [Document Overview](#)
- [Introduction](#)
  - [Audience](#)
  - [Requirements](#)
  - [Reference Configuration](#)
  - [Support](#)
  - [Third-Party Product Features](#)
  - [Not Supported Features](#)
  - [Verify License](#)
- [Cisco UCM 11 Configuration](#)
  - [1. SIP Profile](#)
  - [2. SIP Trunk Security Profile](#)
  - [3. Trunk](#)
  - [4. Route Group](#)
  - [5. Route List](#)
  - [6. Route Pattern](#)
  - [6. Meet-Me Number](#)
- [Ribbon SBC Edge Series Configuration](#)
  - [1. Easy Config Wizard](#)
  - [2. SIP Profile](#)
  - [3. Q.850 Cause Code to SIP Override Table](#)
  - [4. Tone Tables](#)
  - [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](#)
- [Conclusion](#)

# Document Overview

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This document provides a configuration guide for Ribbon Session Border Controller Edge Series (SBC) when connecting to Cisco Unified Communication Manager 11.0 (CUCM 11).

- For additional information on the Cisco Platform, visit <http://www.cisco.com>.
- For additional information on the Ribbon SBC, visit <http://ribboncommunications.com>.

## Introduction

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The interoperability compliance testing focuses on verifying inbound and outbound call flows between the Ribbon SBC Edge and Cisco Unified Communication Manager 11.0 (CUCM 11).

## 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 Reach SIP trunk group with the Cisco Unified Communication Manager 11. This configuration requires navigating 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

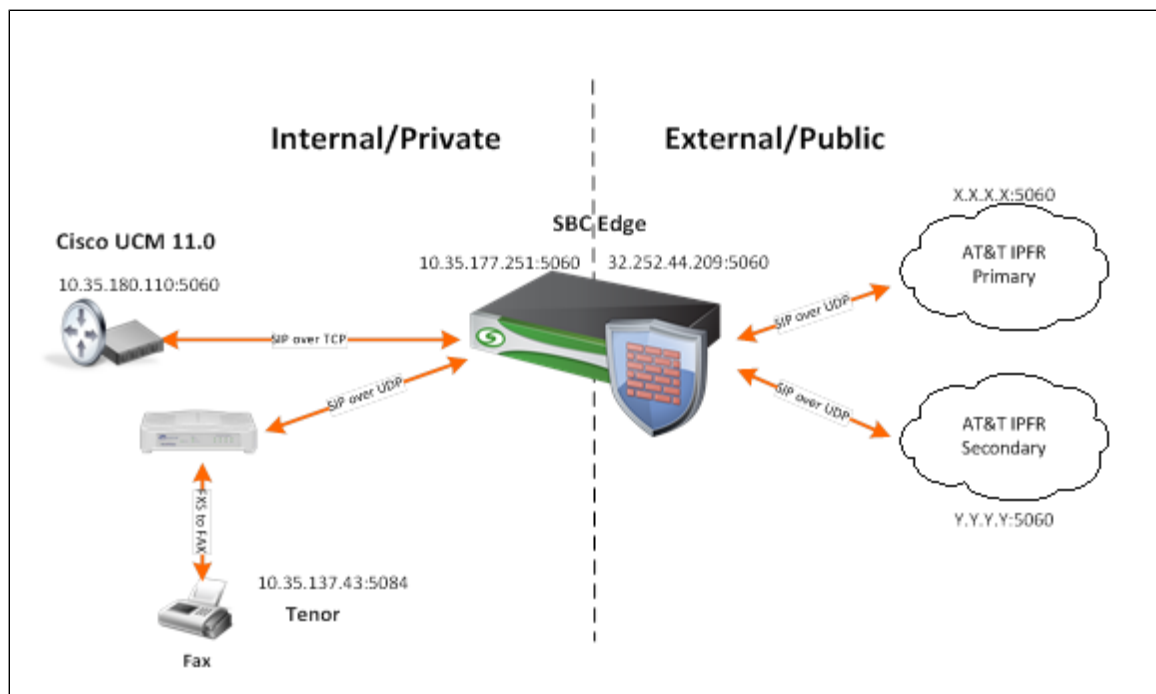
The following equipment and software were used for the sample (see [Topology](#)):

	Equipment	Software Version
<b>Ribbon Networks</b>	Ribbon SBC Edge (2000)	7.0.0b476
<b>Third-party Equipment</b>	Cisco UCM 11.0	11.0.1.21900-11
	Cisco IP Phone 7942	9.4.2

## 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
- CUCM does not support SIP REFER method for network transfer
- Voice mail is not supported on the single server deployment.
- PBX-Based Auto Attendant is not supported on the single server deployment.

## Verify License

No special licensing required.

## Cisco UCM 11 Configuration

The following new configurations are included in this section:

1. [SIP Profile](#)
2. [SIP Trunk Security Profile](#)
3. [Trunk](#)
4. [Route Group](#)
5. [Route List](#)
6. [Route Pattern](#)
7. [Meet-Me Number](#)

## 1. SIP Profile

Select **Device > Device Settings > SIP Profile**

**Figure 2:** SIP Profile

SIP Profile Information	
Name*	Newport SIP Profile
Description	Newport SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	CLEARMODE
User-Agent and Server header information*	Send Unified CM Version Information as User-Agen
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, anc
Confidential Access Level Headers*	Disabled
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> Assured Services SIP conformance	
SDP Information	
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	
Parameters used in Phone	
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default

DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- ☒ Conference Join Enabled
- ☐ RFC 2543 Hold
- ☒ Semi Attended Transfer
- ☐ Enable VAD
- ☐ Stutter Message Waiting
- ☐ MLPP User Authorization

Normalization Script

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value		
1			+	-

<b>Incoming Requests FROM URI Settings</b>	
Caller ID DN	<input type="text"/>
Caller Name	<input type="text"/>
<b>Trunk Specific Configuration</b>	
Reroute Incoming Request to new Trunk based on*	Never
Resource Priority Namespace List	< None >
SIP Rel1XX Options*	Disabled
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Mandatory (insert MTP if needed)
<input type="checkbox"/> Enable ANAT <input type="checkbox"/> Deliver Conference Bridge Identifier <input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information <input type="checkbox"/> Reject Anonymous Incoming Calls <input type="checkbox"/> Reject Anonymous Outgoing Calls <input type="checkbox"/> Send ILS Learned Destination Route String	
<b>SIP OPTIONS Ping</b>	
<input checked="" type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"	
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6
<b>SDP Information</b>	
<input checked="" type="checkbox"/> Send send-receive SDP in mid-call INVITE <input type="checkbox"/> Allow Presentation Sharing using BFCP <input type="checkbox"/> Allow iX Application Media <input checked="" type="checkbox"/> Allow multiple codecs in answer SDP	

## 2. SIP Trunk Security Profile

Select **System> Security > SIP Trunk Security Profile**

**Figure 3:** SIP Trunk Security Profile

**SIP Trunk Security Profile Information**

Name*	Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	TCP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application level authorization	
<input type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input checked="" type="checkbox"/> Accept unsolicited notification	
<input checked="" type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

### 3. Trunk

Select **Device > Trunk**

**Figure 4:** Trunk

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type

None(Default)

Device Name\*

Newport\_SIP\_Trunk

Description

SIP TG to SBC 2K Newport

Device Pool\*

711\_DP

Common Device Configuration

< None >

Call Classification\*

Use System Default

Media Resource Group List

UCM\_MRGL

Location\*

Lab

AAR Group

< None >

Tunneled Protocol\*

None

QSIG Variant\*

No Changes

ASN.1 ROSE OID Encoding\*

No Changes

Packet Capture Mode\*

None

Packet Capture Duration

0

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port

☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure\*

When using both sRTP and TLS

Route Class Signaling Enabled\*

Default

Use Trusted Relay Point\*

Default

☒ PSTN Access

☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile

< None >

MLPP and Confidential Access Level Information

MLPP Domain

< None >

Confidential Access Mode

< None >

Confidential Access Level

< None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type\*

Default

SIP Privacy\*

Default

Inbound Calls

Significant Digits\*

All

Connected Line ID Presentation\*

Default

Connected Name Presentation\*

Default

Calling Search Space

NewportISBC

AAR Calling Search Space

< None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number		0	< None >	<input checked="" type="checkbox"/>

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number		0	< None >	<input checked="" type="checkbox"/>

Connected Party Settings

Connected Party Transformation CSS - < None >

☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

< None >

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

< None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*

Originator



Calling Line ID Presentation*	Default				
Calling Name Presentation*	Default				
Calling and Connected Party Info Format*	Deliver DN only in connected party				
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound					
Redirecting Party Transformation CSS	FullCSS				
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS					
<b>Caller Information</b>					
Caller ID DN	<input type="text"/>				
Caller Name	<input type="text"/>				
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers					
<b>SIP Information</b>					
<b>Destination</b>					
<input type="checkbox"/> Destination Address is an SRV					
	<b>Destination Address</b>	<b>Destination Address IPv6</b>	<b>Destination Port</b>	<b>Status</b>	<b>Status Reason</b>
1 *	10.35.177.232		5060	up	
MTP Preferred Originating Codec*	711ulaw				
BLF Presence Group*	Standard Presence group				
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile				
Rerouting Calling Search Space	FullCSS				
Out-Of-Dialog Refer Calling Search Space	< None >				
SUBSCRIBE Calling Search Space	< None >				
SIP Profile*	Newport SIP Profile <a href="#">View Details</a>				
DTMF Signaling Method*	No Preference				
<b>Normalization Script</b>					
Normalization Script	< None >				
<input type="checkbox"/> Enable Trace					
	<b>Parameter Name</b>	<b>Parameter Value</b>			
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>		
<b>Recording Information</b>					
<input checked="" type="radio"/> None <input type="radio"/> This trunk connects to a recording-enabled gateway <input type="radio"/> This trunk connects to other clusters with recording-enabled gateways					

## 4. Route Group

Select **Call Routing > Route/Hunt > Route Group**

**Figure 5:** Route Group

**Route Group Information**

Route Group Name\*

Distribution Algorithm\* Circular

**Route Group Member Information**

**Find Devices to Add to Route Group**

Device Name contains  Find

Available Devices\*\*

SONUS8\_SIP\_Trunk
Short-SIP-Trunk
Tefnut\_SIP\_Trunk
Thames-SIP-Trunk
York\_SIP\_Trunk

Port(s) All

Add to Route Group

**Current Route Group Members**

Selected Devices (ordered by priority)\*

Newport\_SIP\_Trunk (All Ports)

Reverse Order of Selected Devices

Removed Devices\*\*\*

**Route Group Members**


[Newport\\_SIP\\_Trunk](#)

Save
Delete
Add New

## 5. Route List

Select **Call Routing > Route/Hunt > Route List**

**Figure 6:** Route List

Route List Information

Registration:

Registered with Cisco Unified Communications Manager 10.35.180.110

IPv4 Address:

10.35.180.110

☒ Device is trusted

Name\*

Newport-SIP-RL

Description

Route list to SBC Newport

Cisco Unified Communications Manager Group\*

UCM\_UCMG

☒ Enable this Route List (change effective on Save; no reset required)

☐ Run On All Active Unified CM Nodes

Route List Member Information

Selected Groups\*\*

Newport\_SIP\_RG


Removed Groups\*\*\*

▼

▲

Add Route Group

Route List Details


[Newport SIP RG](#)

Save

Delete

Copy

Reset

Apply Config

Add New

## 6. Route Pattern

Select **Call Routing > Route/Hunt > Route Patterns**



### Note

Use this procedure to create any Route Pattern configuration.

**Figure 7:** Route Pattern

Route Pattern*	214432688X		
Route Partition	NewportSBC		
Description	Newport SBC		
Numbering Plan	-- Not Selected --		
Route Filter	< None >		
MLPP Precedence*	Default		
<input type="checkbox"/> Apply Call Blocking Percentage			
Resource Priority Namespace Network Domain	< None >		
Route Class*	Default		
Gateway/Route List*	Newport-SIP-RL		
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error		
Call Classification*	OffNet		
External Call Control Profile	< None >		
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority			
<input type="checkbox"/> Require Forced Authorization Code			
Authorization Level*	0		
<input type="checkbox"/> Require Client Matter Code			

<b>Calling Party Transformations</b>			
<input type="checkbox"/> Use Calling Party's External Phone Number Mask			
Calling Party Transform Mask			
Prefix Digits (Outgoing Calls)			
Calling Line ID Presentation*	Default		
Calling Name Presentation*	Default		
Calling Party Number Type*	Cisco CallManager		
Calling Party Numbering Plan*	Cisco CallManager		

<b>Connected Party Transformations</b>			
Connected Line ID Presentation*	Default		
Connected Name Presentation*	Default		

<b>Called Party Transformations</b>			
Discard Digits	< None >		
Called Party Transform Mask			
Prefix Digits (Outgoing Calls)			
Called Party Number Type*	Cisco CallManager		
Called Party Numbering Plan*	Cisco CallManager		

<b>ISDN Network-Specific Facilities Information Element</b>			
Network Service Protocol -- Not Selected --			
Carrier Identification Code			
Network Service	Service Parameter Name	Service Parameter Value	
-- Not Selected --	< Not Exist >		

## 6. Meet-Me Number

Select **Call Routing > Meet-Me Number**

**Figure 8:** Meet-Me Number

Meet-Me Configuration	
Directory Number or Pattern*	2142592290
Description	ATT IPFR Meet-Me
Partition	NewportSBC
Minimum Security Level*	Non Secure

Save

Delete

Copy

Add New

## Ribbon SBC Edge Series Configuration

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

1. [Easy Config 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 Config Wizard

The SBC interface 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 9:** Easy Config Wizard

Step 1

Step 2

Step 3

This step takes input about the topology

Scenario Parameters

ApplicationSIP Trunk <-> IP PBX \*  
Scenario DescriptionAT&T IPFR \*  
Telephone CountryUnited States  
Emergency ServicesNone  
SIP Properties  
SIP Sessions10 \* [1..960]

SIP Trunk

NameATT SIP Trunk

IP PBX

TypeCisco CUCM

Cancel

Previous

Next

Finish

Step 1

Step 2

Step 3

This step takes input about the Provider and User side configuration

SIP Trunk: ATT SIP Trunk

Border Element ServerX.X.X.X \* FQDN or IP  
ProtocolUDP  
Port Number5060 [1024..65535]  
Use Secondary Border Element ServerEnabled  
Secondary Border Element ServerY.Y.Y.Y \* FQDN or IP  
ProtocolUDP  
Port Number5060 [1024..65535]  
AT&T Services  
AT&T Simultaneous Ring SupportedYes  
AT&T IP Toll FreeDisabled

IP PBX: Cisco CUCM

Host  
ProtocolTCP

Cancel

Previous

Next

Finish

Step 1

Step 2

Step 3

This step takes input about the Provider and User side configuration

SIP Trunk: ATT SIP Trunk

IP PBX: Cisco CUCM

Host10.35.180.110 \* FQDN or IP  
ProtocolTCP  
Port Number5060 [1024..65535]  
Use Secondary ServerDisabled

Cancel

Previous

Next

Finish

Step 1

Step 2

Step 3

This step is a summary of what will be configured

Application

SIP Trunk <-> IP PBX

Scenario Description

AT&T IPFR

Telephone Country

United States

Emergency Services

None

SIP Properties

SIP Sessions

10

SIP Trunk: ATT SIP Trunk

Border Element Server

X.X.X.X

Protocol

UDP

Port Number

5060

Use Secondary Border Element Server

Enabled

Secondary Border Element Server

Y.Y.Y.Y

Protocol

UDP

Port Number

5060

AT&T Services

AT&T Simultaneous Ring Supported

Yes

AT&T IP Toll Free

Disabled

IP PBX: Cisco CUCM

Host

10.35.180.110

Protocol

TCP

Port Number

5060

Use Secondary Server

Disabled

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 10:** AT&T-IPFR: CUCM SIP Profile

Description <b>AT&amp;T IPFR: Cisco Profile</b>			
<b>Session Timer</b>		<b>MIME Payloads</b>	
Session Timer	Enable	ELIN Identifier	LOC
Minimum Acceptable Timer	600	PIDF-LO Passthrough	Enable
Offered Session Timer	1800	Unknown Subtype Passthrough	Disable
Terminate On Refresh Failure	False		
<b>Header Customization</b>		<b>Options Tags</b>	
FQDN in From Header	Disable	100rel	Supported
FQDN in Contact Header	Disable	Path	Not Present
Send Assert Header	Trusted Only	Timer	Supported
Sonus Diagnostics Header	Enable	Update	Supported
Trusted Interface	Enable		
UA Header	Sonus SBC		
Calling Info Source	RFC Standard		
Diversion Header Selection	Last		
Record Route Header	RFC 3261 Standard		
<b>Timers</b>		<b>SDP Customization</b>	
Transport Timeout Timer	5000	Send Number of Audio Channels	True
Maximum Retransmissions	RFC Standard	Connection Info in Media Section	True
<hr/> <b>RFC Timers</b> <hr/>		Origin Field Username	SBC
Timer T1	500	Session Name	VoipCall
Timer T2	4000	Digit Transmission Preference	RFC 2833/Voice
Timer T4	5000	SDP Handling Preference	Legacy Audio/Fax
Timer D	32000		
Timer B	32000 ms		
Timer F	32000 ms		
Timer H	32000 ms (64*TimerT1)		
Timer J	4000		

**Figure 11:** 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 12:** 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 Edge administrator 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 13:** 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 14:** AT&T-IPFR (Cisco)

Voice Codec Configuration	
Description	AT&T IPFR (Cisco): G.711 A-Law
Codec	G.711 A-Law
Payload Size	20

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

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

Figure 15: 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 system wide 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 16: Media System Configuration

Port Range

Start Port

16384

\* [1024..32767]

Number of Port Pairs

1500

\* [1..4800]

Music on Hold

Music on Hold Source

File

▼

Current Music File

Echo Canceller Type Option

Standard

▼

Echo Cancel NLP Option

Mild

▼

Send STUN Packets

Enabled

▼

## 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 17: AT&T-IPFR: Cisco List

Description

AT&T IPFR: Cisco List

Media Profiles List

AT&T IPFR (Cisco): G.711 A-Law

AT&T IPFR (Cisco): G.711 Mu-Law

AT&T IPFR (Cisco): G.729

\*

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

Enabled

Fax Passthrough

Enabled

CNG Tone Detection

Enabled

Fax Tone Detection

Enabled

DTMF Signal to Noise

0

Figure 18: AT&T-IPFR: ATT Trunk List

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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 19: 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 20:** AT&T IPFR: Cisco CUCM

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	1	Keep Alive Frequency	30
Host	10.35.180.110	Recover Frequency	5
Port	5060	Local Username	Anonymous
Protocol	TCP	Peer Username	Anonymous

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 21: AT&T-IPFR: Border Element

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

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

Figure 22: 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 23:** AT&T IPFR: Cisco CUCM

Description

AT&T IPFR: Cisco CUCM

Admin State

Enabled

Service Status

Up

SIP Channels and Routing

Action Set Table

None

Call Routing Table

AT&T IPFR: From Cisco CUCM

No. of Channels

10

SIP Profile

AT&T IPFR: Cisco Profile

SIP Mode

Basic Call

Agent Type

Back-to-Back User Agent

Interop Mode

Standard

SIP Server Table

AT&T IPFR: Cisco CUCM

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: Cisco 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
10.35.180.110	255.255.255.255

Message Manipulation

Disabled

Figure 24: AT&T-IPFR: ATT Border Element

Description

AT&T-IPFR: ATT Border Element

Admin State

Enabled

Service Status

Up

SIP Channels and Routing

Action Set Table

AT&T-IPFR: ATT Action Set

Call Routing Table

AT&T-IPFR: From ATT

No. of Channels

10

SIP Profile

AT&T-IPFR: ATT Profile

SIP Mode

Basic Call

Agent Type

Back-to-Back User Agent

Interop Mode

Standard

SIP Server Table

AT&T-IPFR: Border Element

Load Balancing

First

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

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

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

AT&T-IPFR: ATT

Pass-thru Peer SIP Response Code

Disable

SIP IP Details

Signaling/Media Source IP

Auto

Signaling DSCP

10

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	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 Manipulation

Enabled

Inbound Message Manipulation

Message Table List

\*

Outbound Message Manipulation

SMM TO ATT

Message Table List

\*

Figure 25: 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 26:** 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 27: AT&T-IPFR: From Cisco CUCM

DescriptionPassthrough

Admin StateEnabled

Match TypeMandatory (Must Match)

Input Field

TypeCalled Address/Number

Value(.\*)

Output Field

TypeCalled Address/Number

Value\1

Figure 28: From Fax-TenorGW

Description	From Fax-TenorGw Passthrough	
Admin State	Enabled	
Match Type	Optional (Match One)	

Input Field		Output Field	
Type	Called Address/Number	Type	Called Address/Number
Value	(.*)	Value	\1

**Figure 29:** From ATT to FAX

Description	Call From ATT to FAX	
Admin State	Enabled	
Match Type	Optional (Match One)	

Input Field		Output Field	
Type	Called Address/Number	Type	Called Address/Number
Value	(2142592292)	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 30:** AT&T-IPFR: From Cisco CUCM

Route Details	
Description	To Outside (Passthrough)
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	AT&T IPFR: From Cisco CUCM: 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 31:** 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	<div> <div>(SIP) Fax-TenorGW</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

**Figure 32: 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 Series to successfully interoperate with AT&T IP Flex Reach SIP Trunk. All feature and serviceability test cases were completed.