
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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Document Overview

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

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

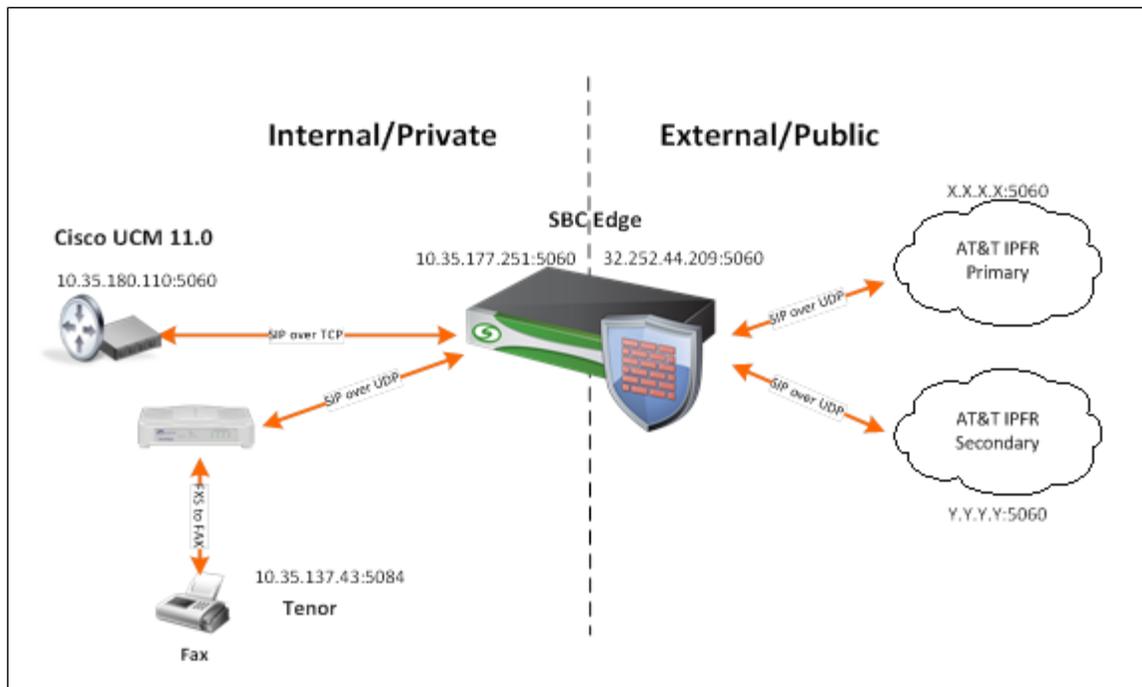
The following equipment and software were used for the sample (see [Topology](#)):

	Equipment	Software Version
Ribbon Networks	Ribbon SBC Edge (2000)	7.0.0b476
Third-party Equipment	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*	<input type="text" value="Newport SIP Profile"/>
Description	<input type="text" value="Newport SIP Profile"/>
Default MTP Telephony Event Payload Type*	<input type="text" value="101"/>
Early Offer for G.Clear Calls*	<input type="text" value="CLEARMODE"/>
User-Agent and Server header information*	<input type="text" value="Send Unified CM Version Information as User-Agen"/>
Version in User Agent and Server Header*	<input type="text" value="Major And Minor"/>
Dial String Interpretation*	<input type="text" value="Phone number consists of characters 0-9, *, #, anc"/>
Confidential Access Level Headers*	<input type="text" value="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*	<input type="text" value="TIAS and AS"/>
SDP Transparency Profile	<input type="text" value="Pass all unknown SDP attributes"/>
Accept Audio Codec Preferences in Received Offer*	<input type="text" value="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)*	<input type="text" value="180"/>
Timer Register Delta (seconds)*	<input type="text" value="5"/>
Timer Register Expires (seconds)*	<input type="text" value="3600"/>
Timer T1 (msec)*	<input type="text" value="500"/>
Timer T2 (msec)*	<input type="text" value="4000"/>
Retry INVITE*	<input type="text" value="6"/>
Retry Non-INVITE*	<input type="text" value="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*	<input type="text" value="16384"/>
Stop Media Port*	<input type="text" value="32766"/>
DSCP for Audio Calls	<input type="text" value="Use System Default"/>
DSCP for Video Calls	<input type="text" value="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	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>

Incoming Requests FROM URI Settings	
Caller ID DN	<input type="text"/>
Caller Name	<input type="text"/>
Trunk Specific Configuration	
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	
SIP OPTIONS Ping	
<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
SDP Information	
<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: NewportSBC
 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*
 Calling Name Presentation*
 Calling and Connected Party Info Format*
 Redirecting Diversion Header Delivery - Outbound
 Redirecting Party Transformation CSS
 Use Device Pool Redirecting Party Transformation CSS

Caller Information
 Caller ID DN
 Caller Name
 Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information
Destination
 Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason
1*	10.35.177.232		5060	up	

MTP Preferred Originating Codec*
 BLF Presence Group*
 SIP Trunk Security Profile*
 Rerouting Calling Search Space
 Out-Of-Dialog Refer Calling Search Space
 SUBSCRIBE Calling Search Space
 SIP Profile* [View Details](#)
 DTMF Signaling Method*

Normalization Script
 Normalization Script
 Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

Recording Information
 None
 This trunk connects to a recording-enabled gateway
 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*

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains

Available Devices**

- SONUS8_SIP_Trunk
- Short-SIP-Trunk
- Tefnut_SIP_Trunk
- Thames-SIP-Trunk
- York_SIP_Trunk

Port(s)

Current Route Group Members

Selected Devices (ordered by priority)*

- Newport_SIP_Trunk (All Ports)

Removed Devices***

Route Group Members

[Newport_SIP_Trunk](#)

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*

Description

Cisco Unified Communications Manager Group*

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

Run On All Active Unified CM Nodes

Route List Member Information

Selected Groups**

▼

▼ ▲

Removed Groups***

Route List Details

 [Newport SIP RG](#)

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

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class* Default

Gateway/Route List* Newport-SIP-RL [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern No Error

Call Classification* OffNet

External Call Control Profile < None >

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level* 0

Require Client Matter Code

Calling Party Transformations

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

Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type* Cisco CallManager

Called Party Numbering Plan* Cisco CallManager

ISDN Network-Specific Facilities Information Element

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*	<input type="text" value="2142592290"/>
Description	<input type="text" value="ATT IPFR Meet-Me"/>
Partition	<input type="text" value="NewportSBC"/>
Minimum Security Level*	<input type="text" value="Non Secure"/>

Ribbon SBC Edge Series Configuration

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

Application: SIP Trunk <-> IP PBX

Scenario Description: AT&T IPFR

Telephone Country: United States

Emergency Services: None

SIP Properties

SIP Sessions: 10 * [1..960]

SIP Trunk

Name: ATT SIP Trunk

IP PBX

Type: Cisco 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 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: Cisco CUCM

Host: * FQDN or IP

Protocol: TCP

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

Host: 10.35.180.110 * FQDN or IP

Protocol: TCP

Port Number: 5060 [1024..65535]

Use Secondary Server: Disabled

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	IP PBX: Cisco CUCM
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	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 AT&T IPFR: Cisco Profile																																							
<table border="1"> <thead> <tr> <th colspan="2">Session Timer</th> </tr> </thead> <tbody> <tr> <td>Session Timer</td> <td>Enable</td> </tr> <tr> <td>Minimum Acceptable Timer</td> <td>600</td> </tr> <tr> <td>Offered Session Timer</td> <td>1800</td> </tr> <tr> <td>Terminate On Refresh Failure</td> <td>False</td> </tr> </tbody> </table>	Session Timer		Session Timer	Enable	Minimum Acceptable Timer	600	Offered Session Timer	1800	Terminate On Refresh Failure	False	<table border="1"> <thead> <tr> <th colspan="2">MIME Payloads</th> </tr> </thead> <tbody> <tr> <td>ELIN Identifier</td> <td>LOC</td> </tr> <tr> <td>PIDF-LO Passthrough</td> <td>Enable</td> </tr> <tr> <td>Unknown Subtype Passthrough</td> <td>Disable</td> </tr> </tbody> </table>	MIME Payloads		ELIN Identifier	LOC	PIDF-LO Passthrough	Enable	Unknown Subtype Passthrough	Disable																				
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Timers																																							
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Digit Transmission Preference	RFC 2833/Voice																																						
SDP Handling Preference	Legacy Audio/Fax																																						

Figure 11: AT&T-IPFR: ATT SIP Profile

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

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	18: No User Responding
SIP Response	486 - Busy Here



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

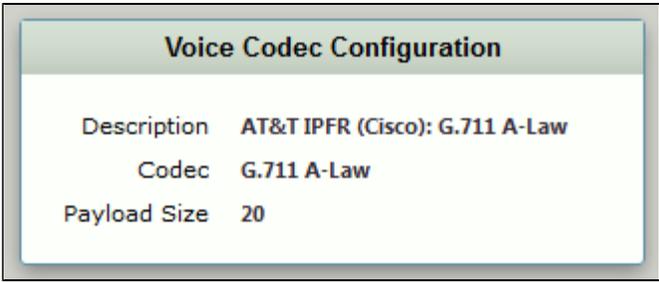
Tone Type	Frequency 1 (Hz)	Amplitude 1 (dBm)	Frequency 2 (Hz)	Amplitude 2 (dBm)
Ringback	440	-19	480	-19
Dial	350	-13	440	-13
Busy	480	-24	620	-24
Congestion	480	-24	620	-24
Call Waiting	440	-13	0	0
Disconnect	480	-24	620	-24
Confirmation	350	-13	440	-13

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 Mu-Law
Codec	G.711 μ -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 μ -Law
Payload Size	20

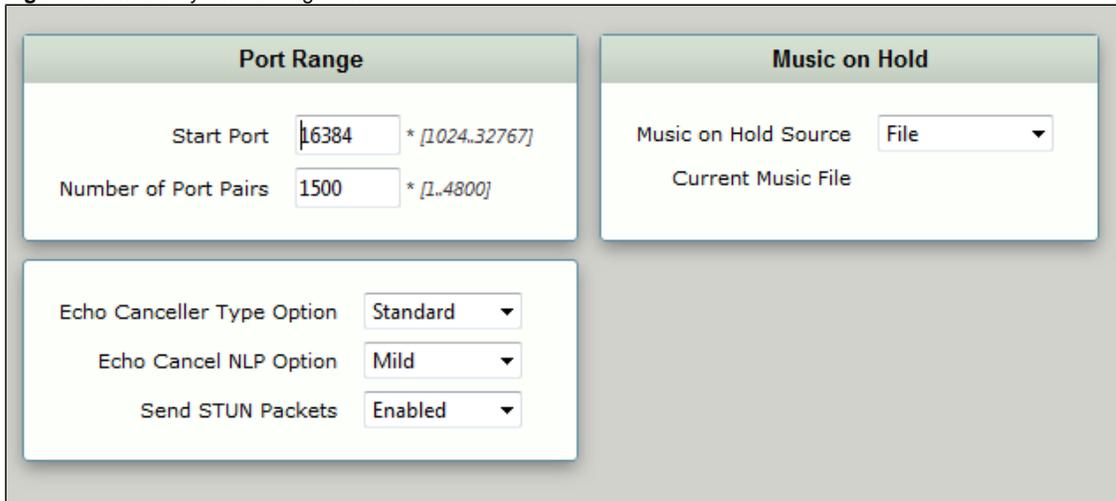
Voice Codec Configuration	
Description	AT&T-IPFR (ATT): G.729
Codec	G.729
Payload Size	20

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



The screenshot displays the Media System Configuration interface with the following settings:

- 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: (empty)
- 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	<div style="border: 1px solid gray; padding: 2px;"> AT&T IPFR (Cisco): G.711 A-Law AT&T IPFR (Cisco): G.711 Mu-Law AT&T IPFR (Cisco): G.729 </div>	*
Crypto Profile ID	None	
Media DSCP	46	
RTCP Mode	RTCP	
Dead Call Detection	Disabled	
Silence Suppression	Enabled	

Gain Control		Digit Relay	
Receive Gain	0	Digit (DTMF) Relay Type	RFC 2833
Transmit Gain	0	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

Description	AT&T-IPFR: ATT Trunk List	
Media Profiles List	<div style="border: 1px solid gray; padding: 5px;"> AT&T-IPFR (ATT): G.729 AT&T-IPFR (ATT): G.711 Mu-Law AT&T-IPFR (ATT): Fax </div>	*
Crypto Profile ID	None	
Media DSCP	10	
RTCP Mode	RTCP	
Dead Call Detection	Disabled	
Silence Suppression	Enabled	

Gain Control		Digit Relay	
Receive Gain	0	Digit (DTMF) Relay Type	RFC 2833
Transmit Gain	0	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	<table border="1"> <thead> <tr> <th colspan="3">Total 0 SPRUriParam Rows</th> </tr> <tr> <th>Name</th> <th>Value</th> <th>Action</th> </tr> </thead> <tbody> <tr> <td colspan="3" style="text-align: center;">-- Table is empty --</td> </tr> </tbody> </table>	Total 0 SPRUriParam Rows			Name	Value	Action	-- Table is empty --		
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 Priority 1 Host 10.35.180.110 Port 5060 Protocol TCP	Monitor SIP Options Keep Alive Frequency 30 Recover Frequency 5 Local Username Anonymous Peer Username Anonymous
Remote Authorization and Contacts	Connection Reuse
Remote Authorization Table None Contact Registrant Table None Session URI Validation Liberal	Reuse True Sockets 4 Reuse Timeout Forever

Figure 21: AT&T-IPFR: Border Element

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

Video/Application Stream Mode **Proxy** *

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**

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

Message Table List *

SMM TO ATT

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 34: No Circuit/Channel Available
Override 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

Listen Ports

Total 2 SIP Listen Port Rows

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

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

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

Description	Add CC to CLD
Admin State	Enabled
Match Type	Optional (Match One)

Input Field		Output Field	
Type	Called Address/Number	Type	Called Address/Number
Value	(21425921\d{2})	Value	1\1

Description	Passthrough
Admin State	Enabled
Match Type	Mandatory (Must Match)

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

Figure 27: AT&T-IPFR: From Cisco CUCM

Description	Passthrough
Admin State	Enabled
Match Type	Mandatory (Must Match)

Input Field		Output Field	
Type	Called Address/Number	Type	Called Address/Number
Value	(.*)	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 style="border: 1px solid black; padding: 2px;"> (SIP) AT&T IPFR: ATT Border Element </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 style="border: 1px solid gray; padding: 2px;"> <div style="background-color: #e0f0ff; padding: 2px;">(SIP) Fax-TenorGW</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 style="border: 1px solid black; padding: 2px;"> (SIP) AT&T-IPFR: ATT Border Element </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.