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

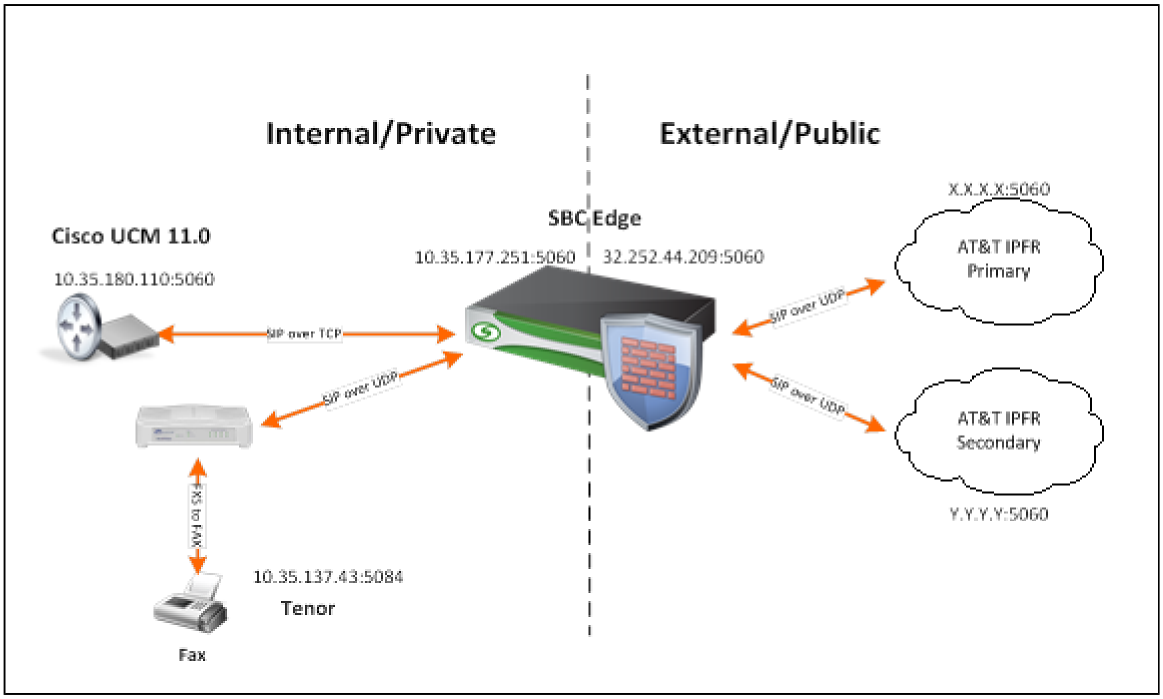
The following equipment and software were used for the sample (see Topology):

	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

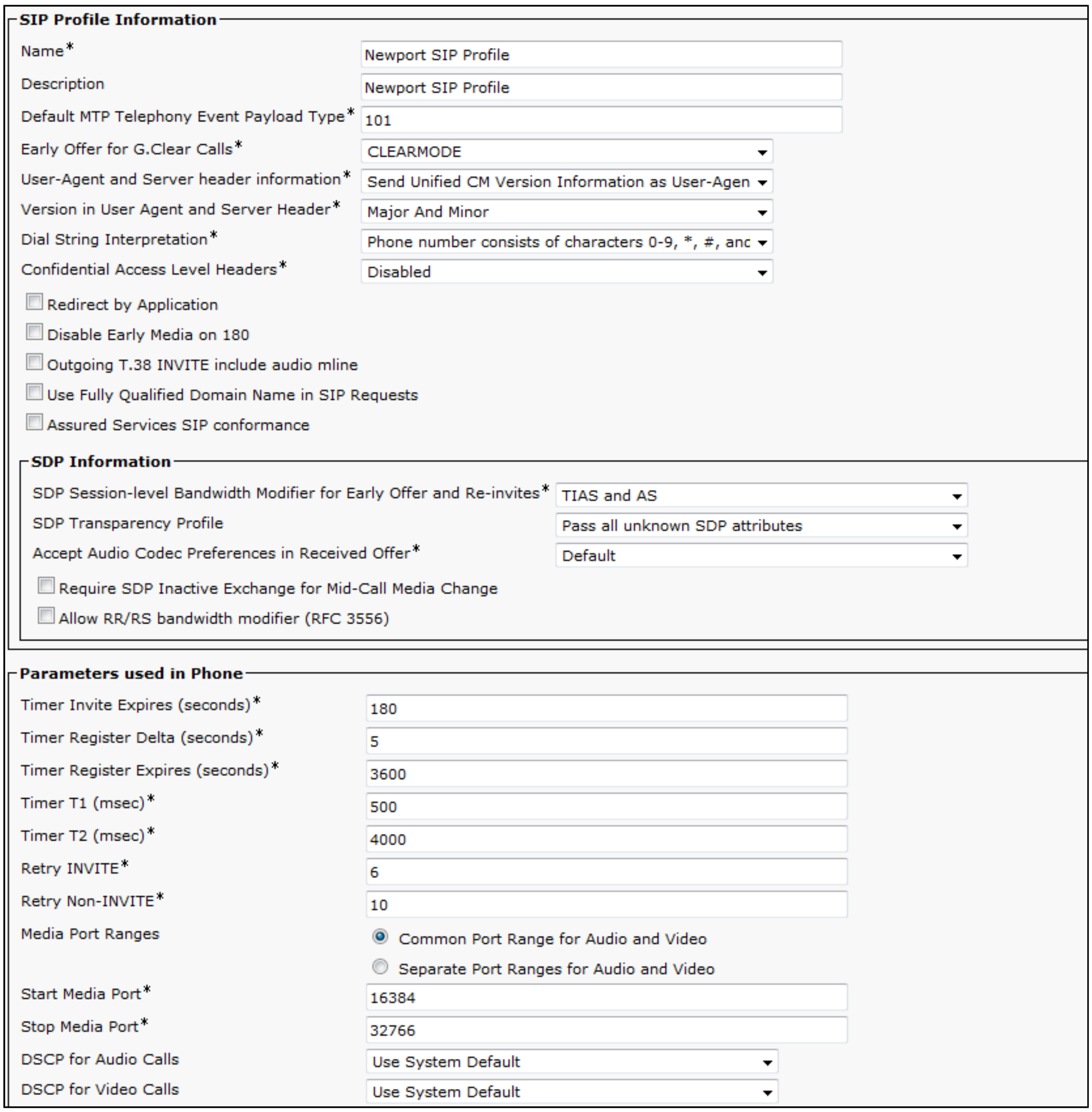
The following new configurations are included in this section:

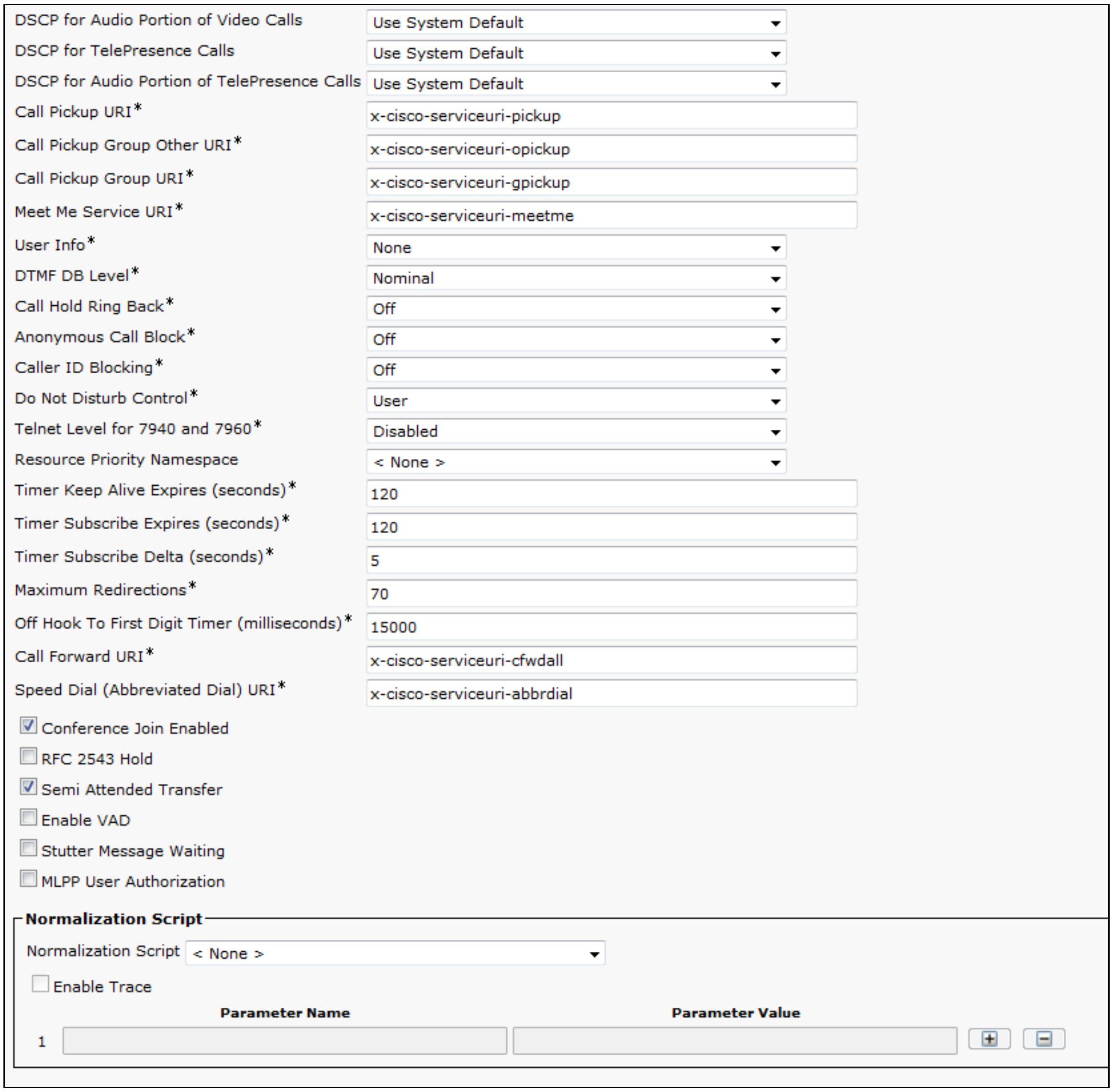
1. SIP Profile
2. SIP Trunk Security Profile
3. Trunk
4. Route Group
5. Route List
6. Route Pattern
7. Meet-Me Number

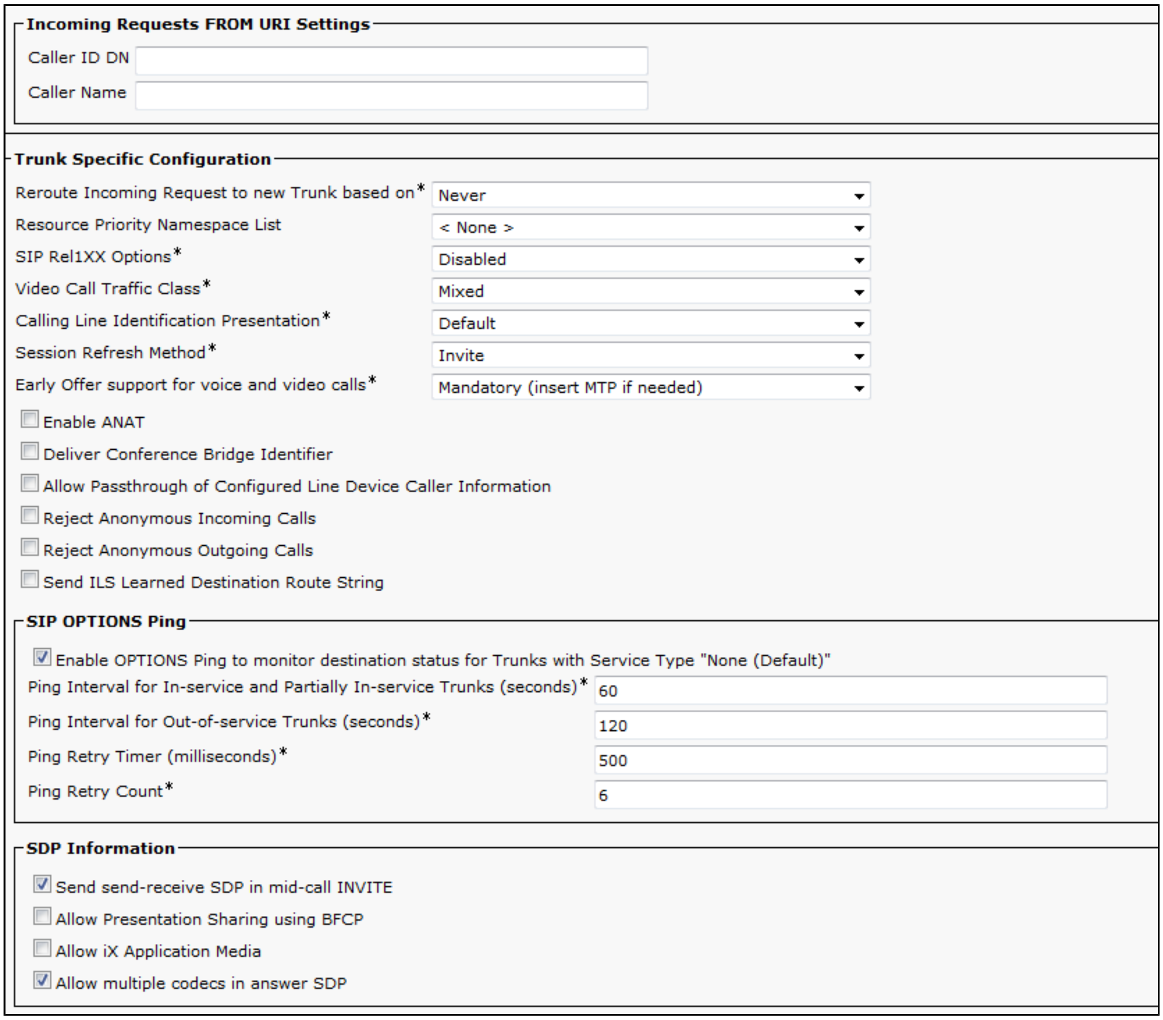
1. SIP Profile

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

Figure 2: SIP Profile



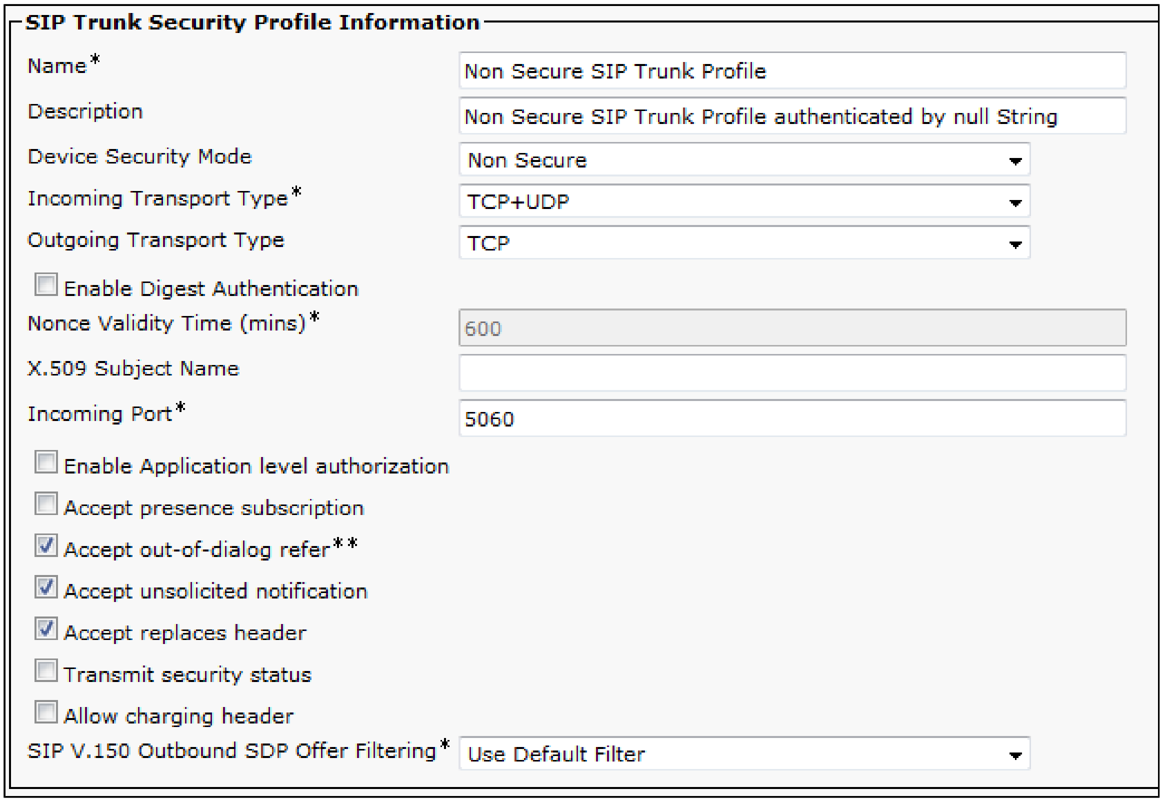

DSCP for Audio Portion of Video Calls
Use System Default
DSCP for TelePresence Calls
Use System Default
DSCP for Audio Portion of TelePresence Calls
Use System Default
Call Pickup URI*
x-cisco-serviceuri-pickup
Call Pickup Group Other URI*
x-cisco-serviceuri-opickup
Call Pickup Group URI*
x-cisco-serviceuri-gpickup
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



2. SIP Trunk Security Profile

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

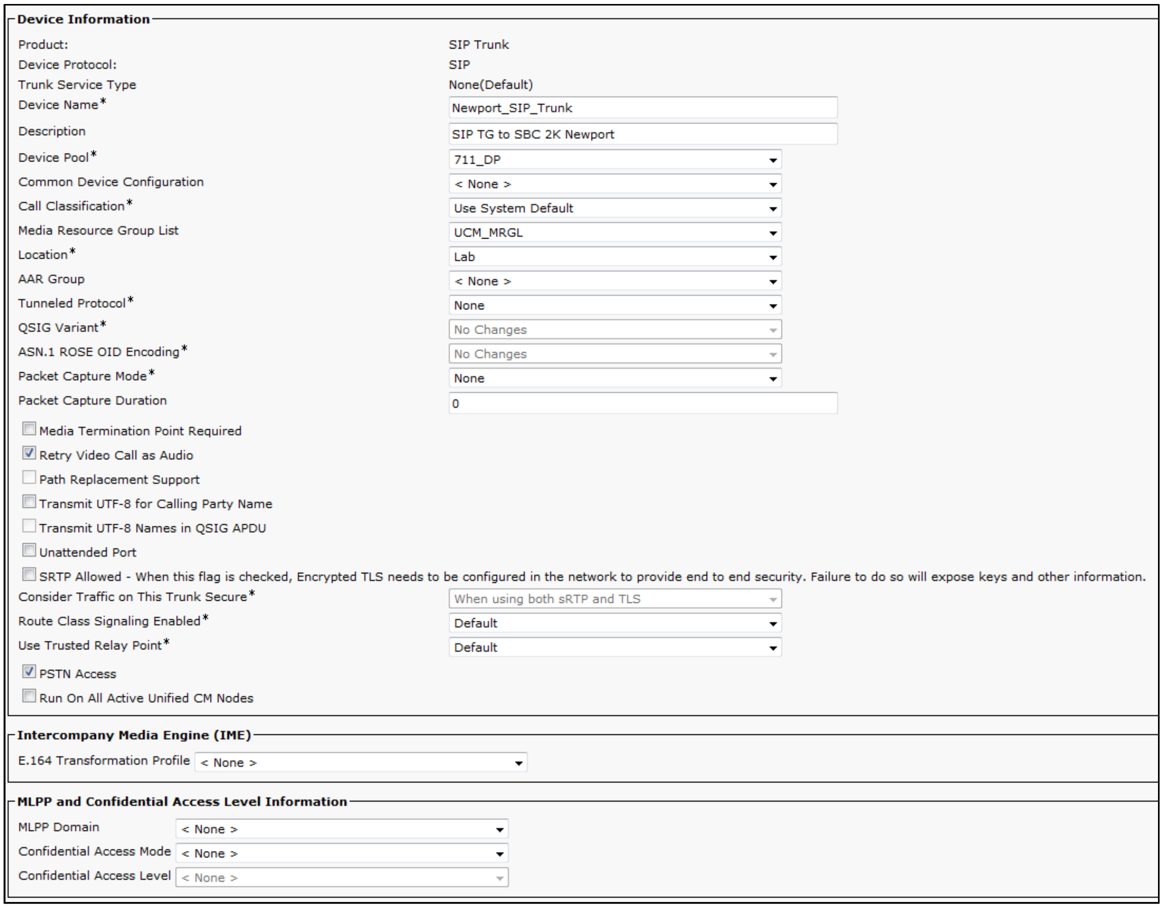
Figure 3: SIP Trunk Security Profile

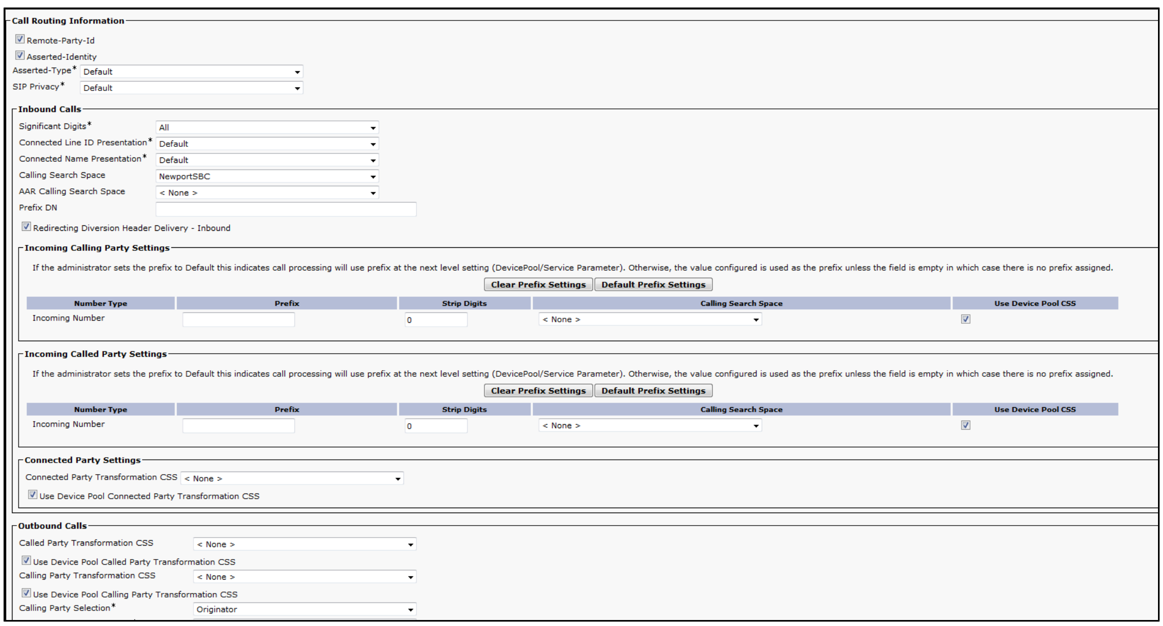


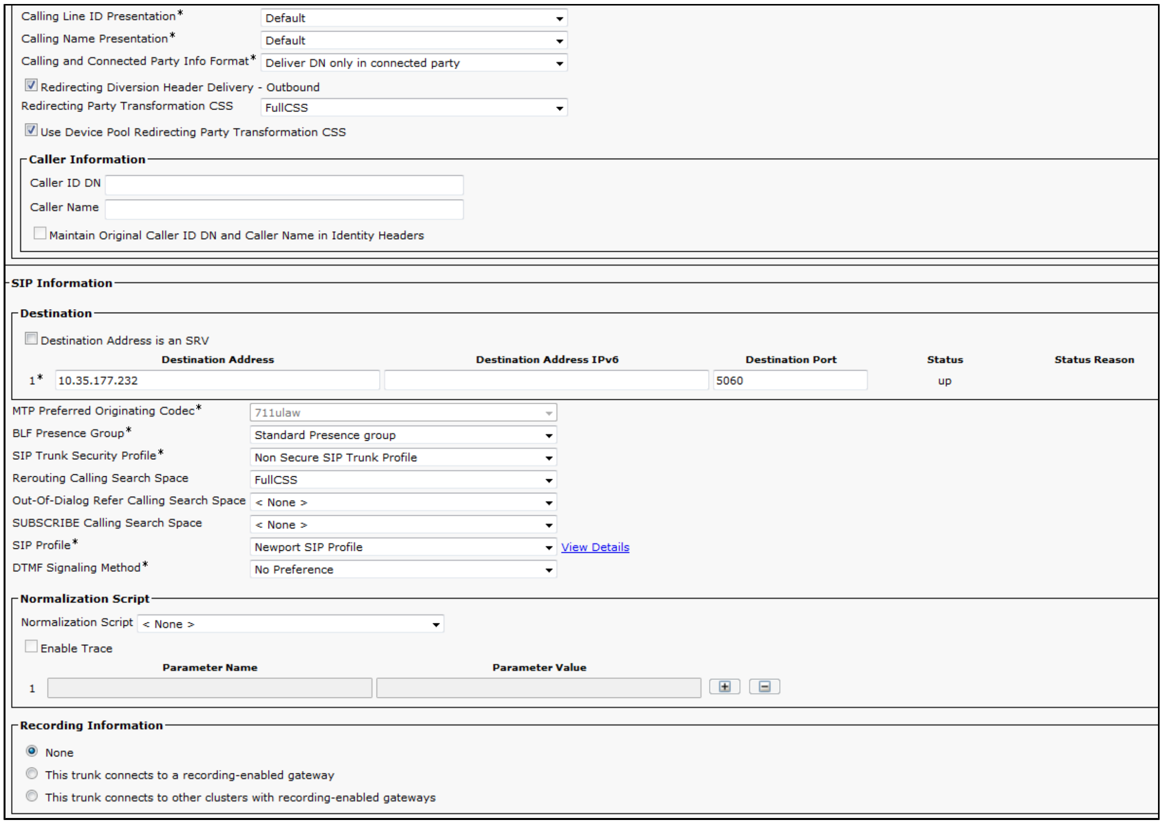
3. Trunk

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 >

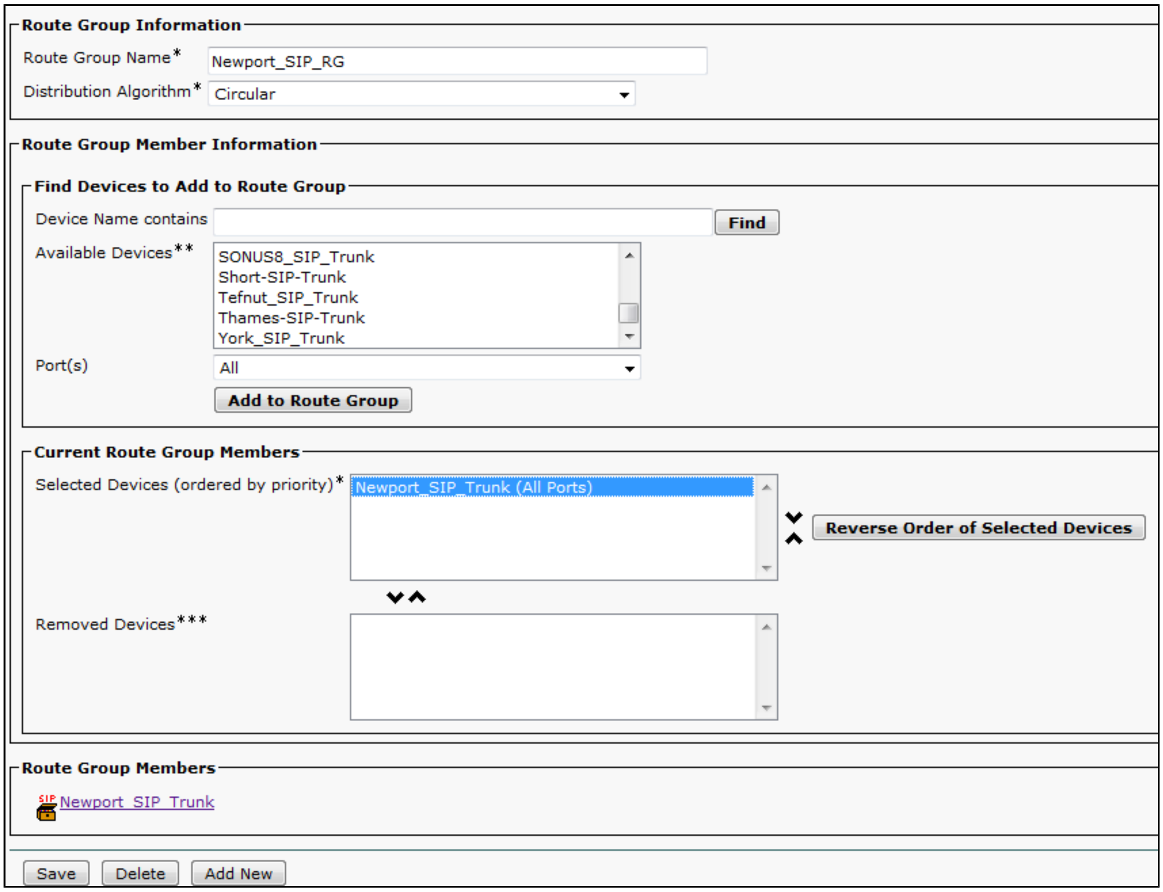

Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default
Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space NewportSBC
AAR Calling Search Space < None >
Prefix DN
Redirecting 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 >
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 >
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



4. Route Group

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

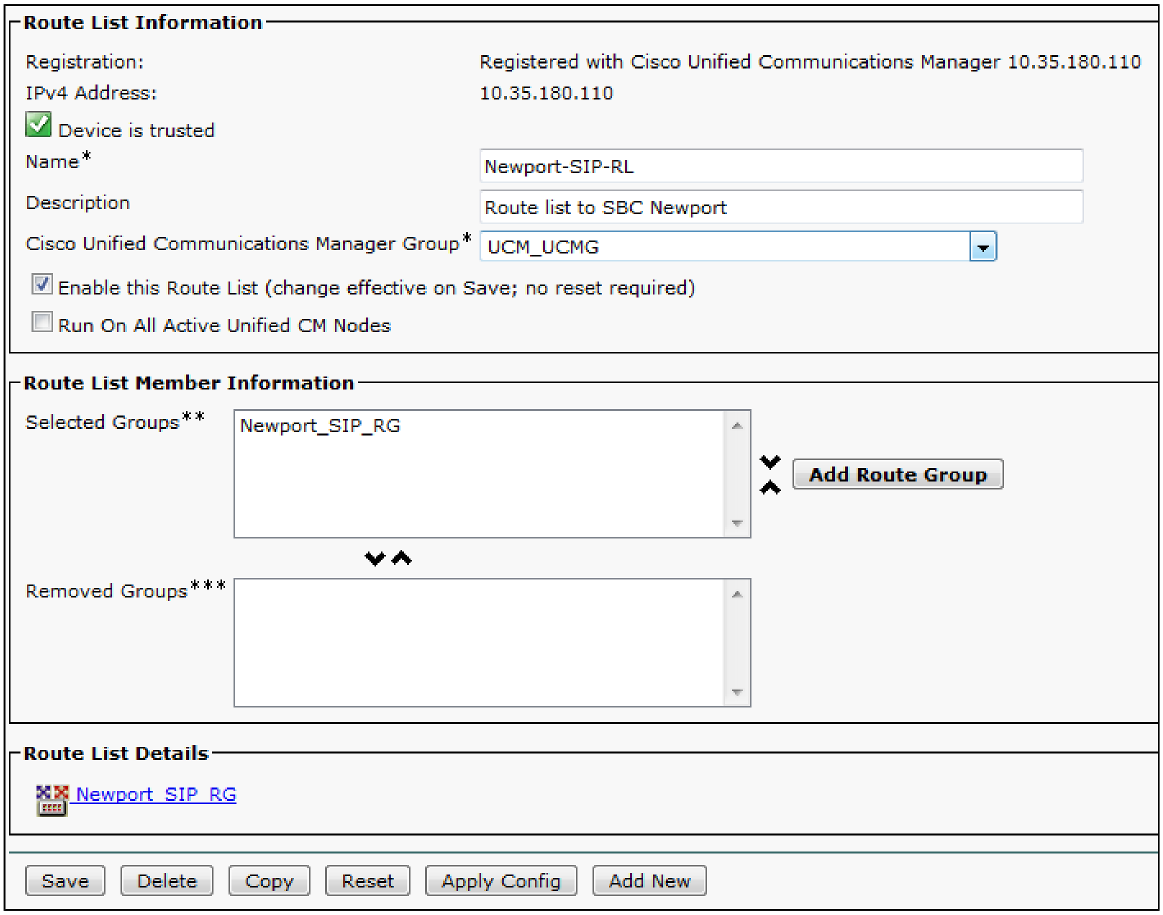
Figure 5: Route Group



5. Route List

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

Figure 6: Route List

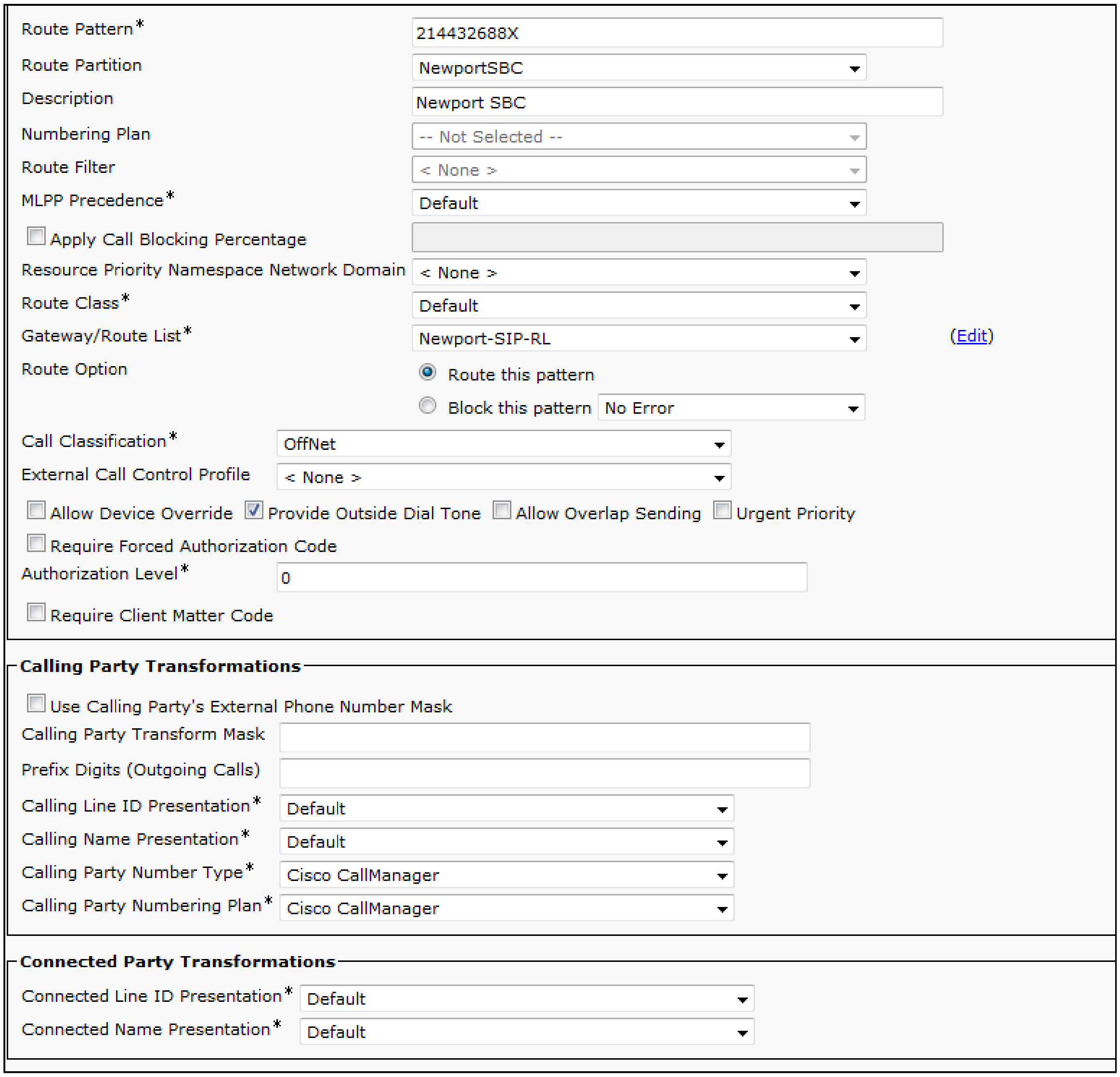


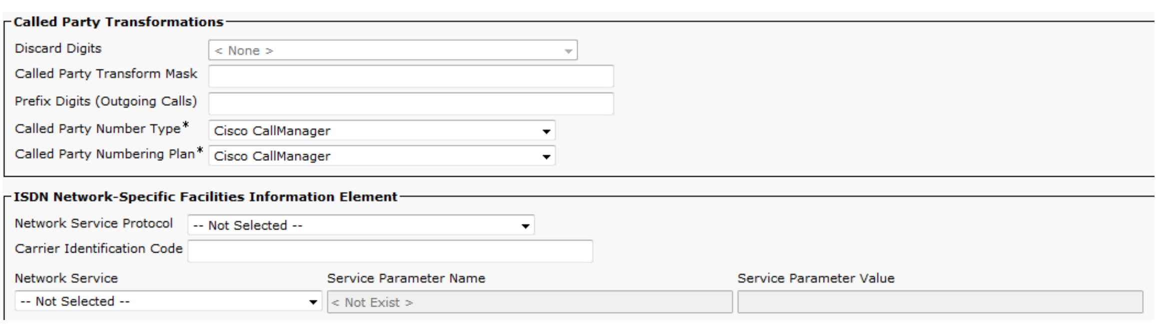
6. Route Pattern

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

Note
Use this procedure to create any Route Pattern configuration.

Figure 7: Route Pattern

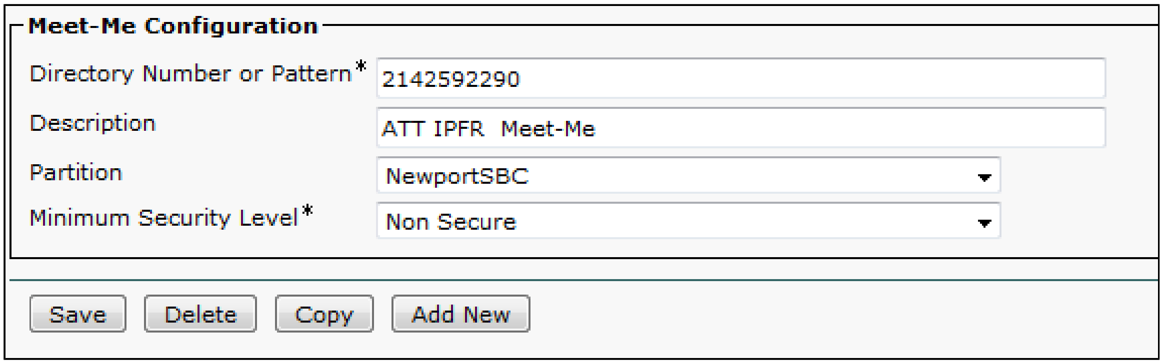




6. Meet-Me Number

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

Figure 8: Meet-Me Number



Ribbon SBC Edge Series Configuration

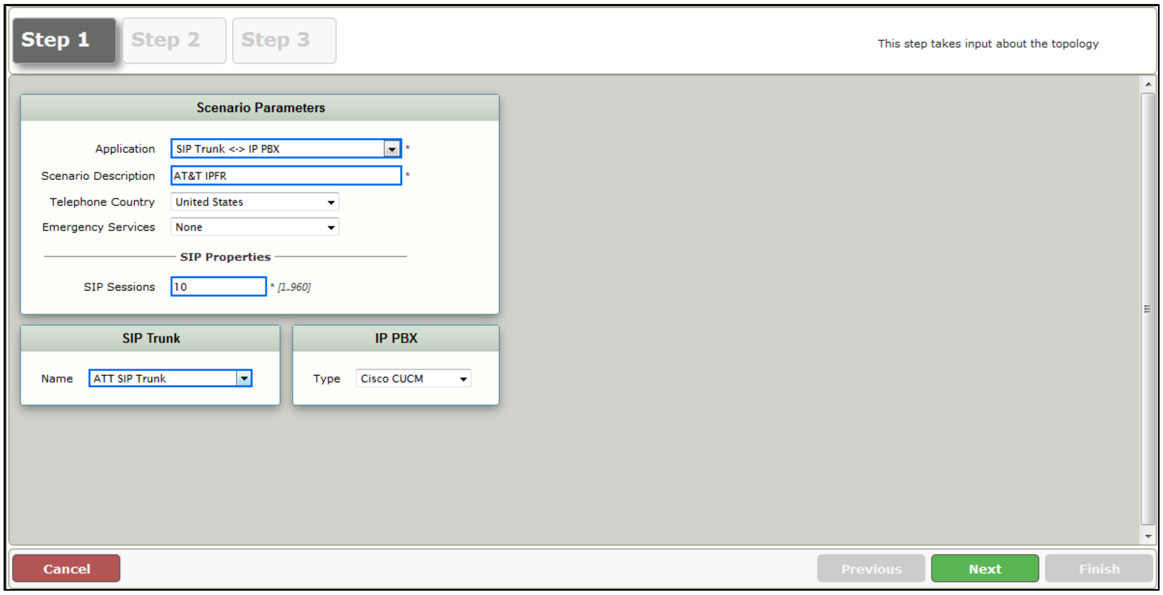
The following steps provide an example of how to configure Ribbon SBC Edge.

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

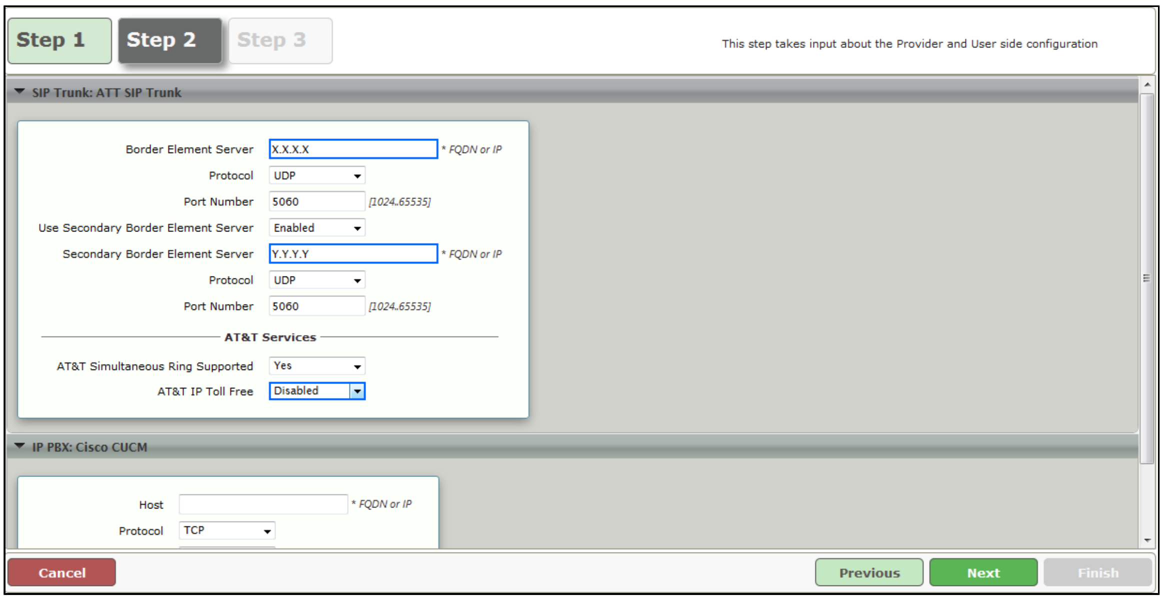
1. Easy Config Wizard

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).

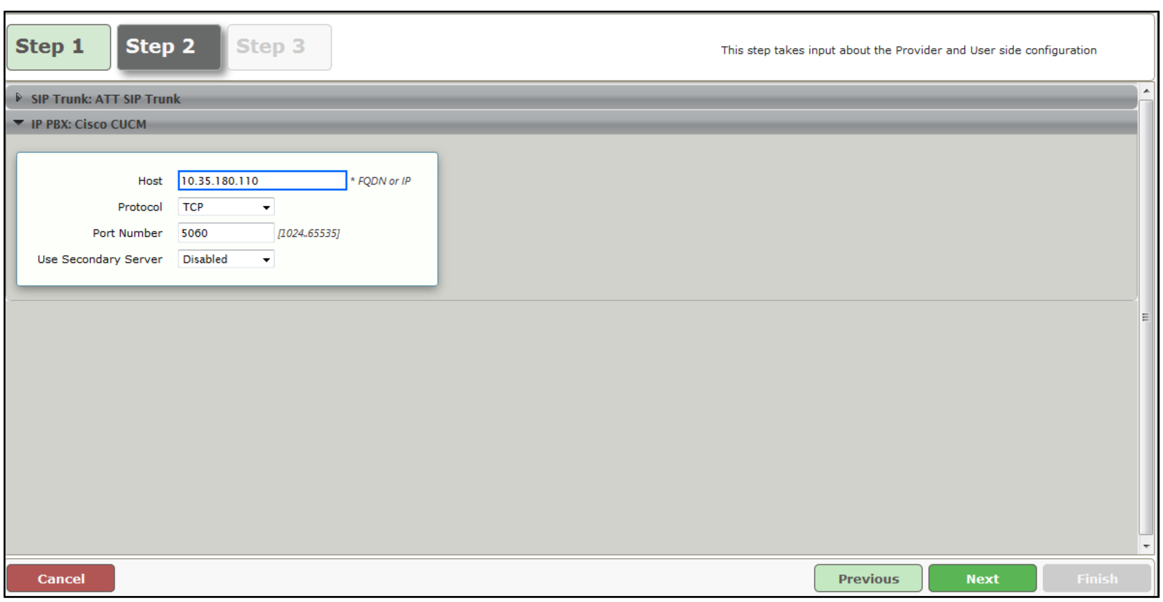
Figure 9: Easy Config Wizard



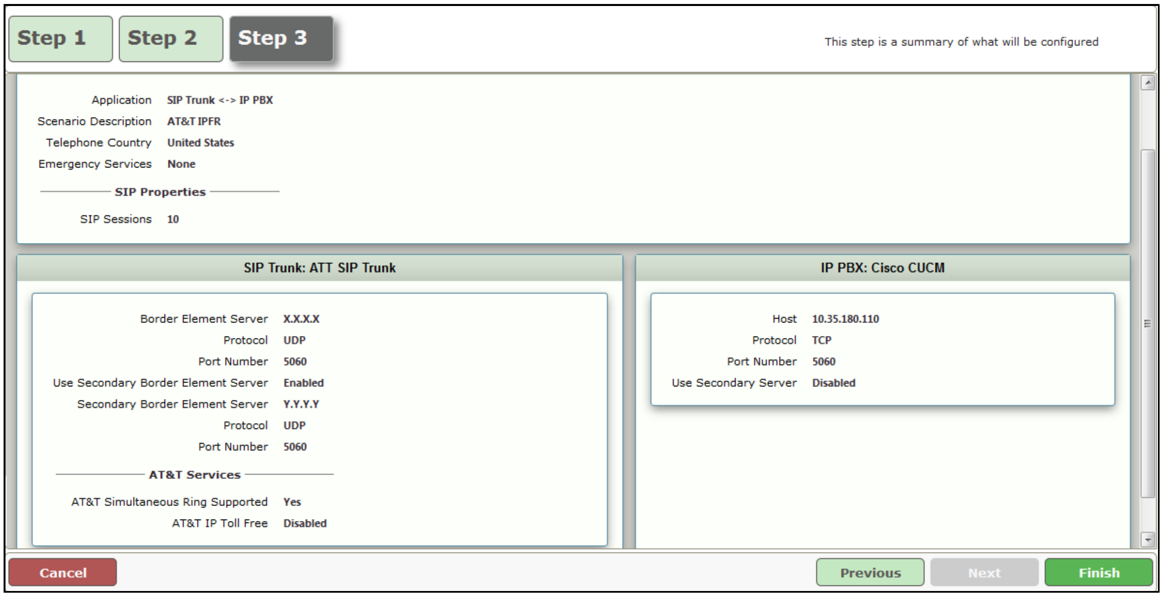
Step 1
Step 2
Step 3
This step takes input about the topology
Scenario Parameters
Application
SIP Trunk <-> IP PBX
Scenario Description
AT&T IPFR
Telephone Country
United States
Emergency Services
None
SIP Properties
SIP Sessions
10
* [1..960]
SIP Trunk
Name
ATT SIP Trunk
IP PBX
Type
Cisco CUCM
Cancel
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



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

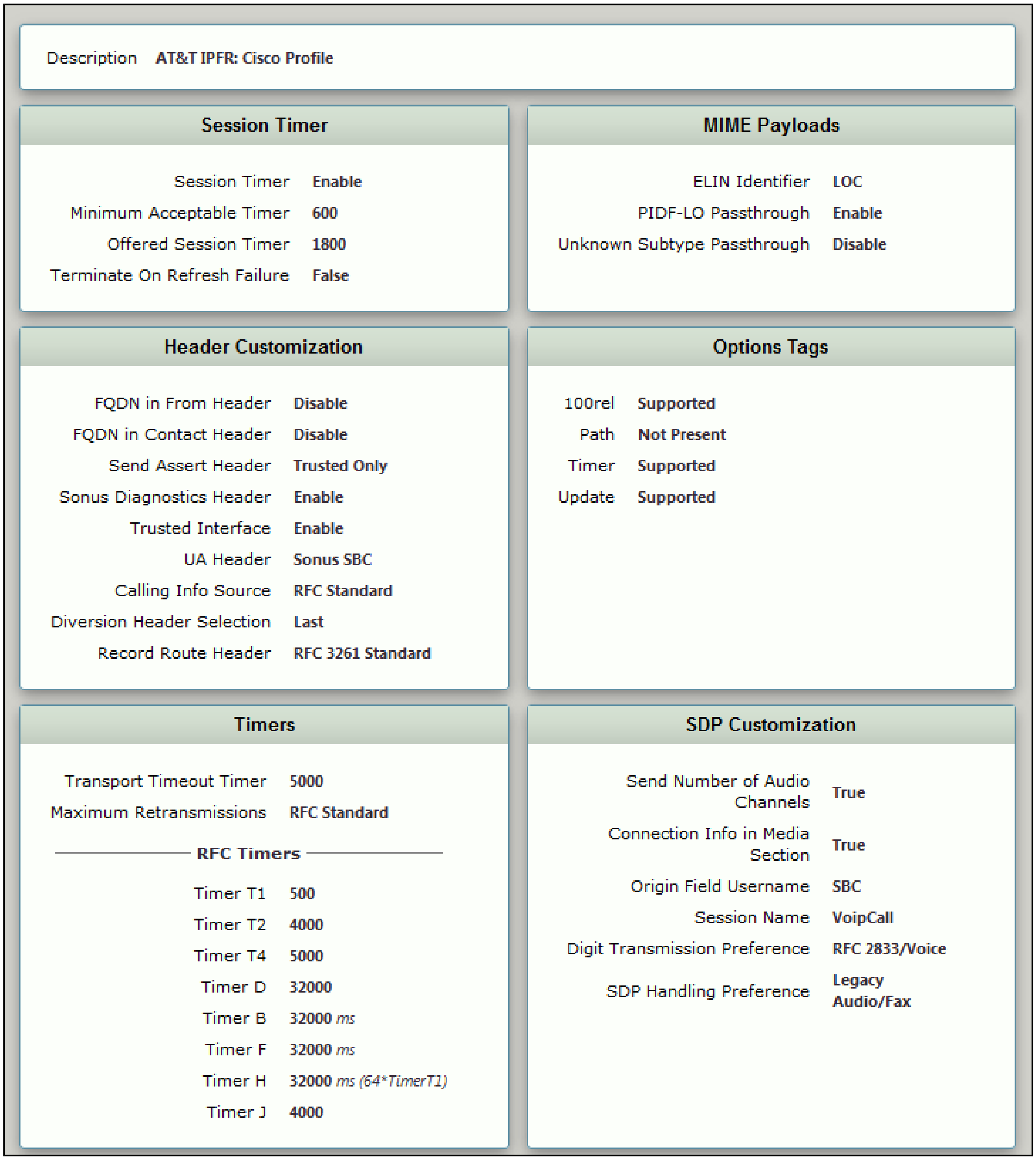
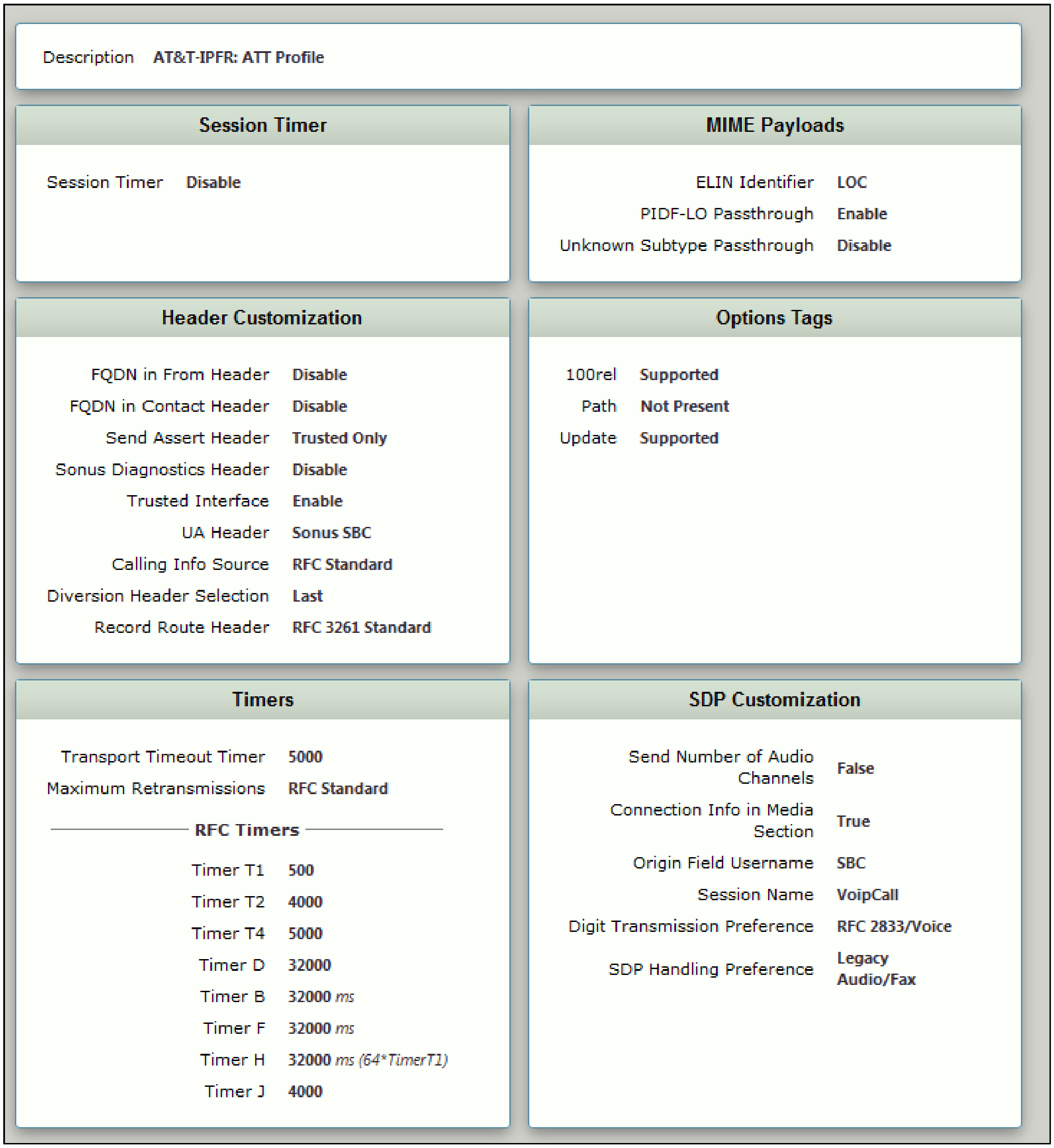


Figure 11: AT&T-IPFR: ATT SIP Profile

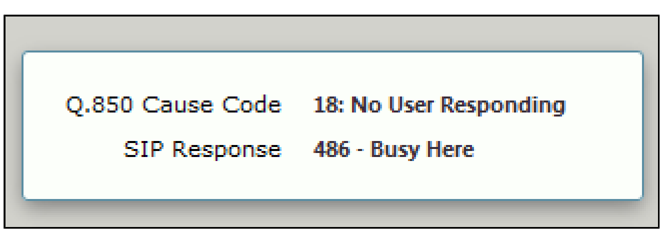


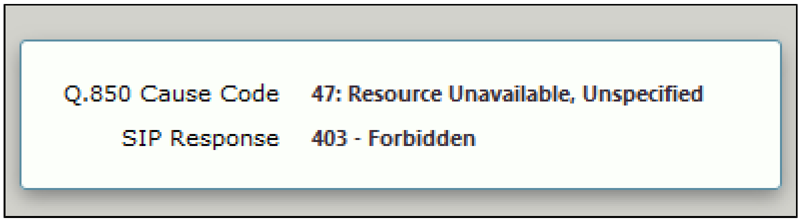
3. Q.850 Cause Code to SIP Override Table

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





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

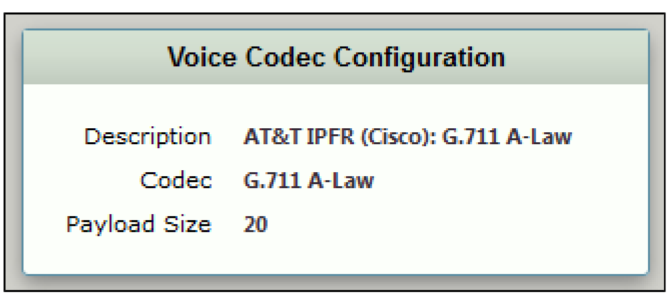
Tone Type	Frequency 1 (Hz)	Amplitude 1 (dBm)	Frequency 2 (Hz)	Amplitude 2 (dBm)
Ringback	440	-19	480	-19
Dial	350	-13	440	-13
Busy	480	-24	620	-24
Congestion	480	-24	620	-24
Call Waiting	440	-13	0	0
Disconnect	480	-24	620	-24
Confirmation	350	-13	440	-13

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

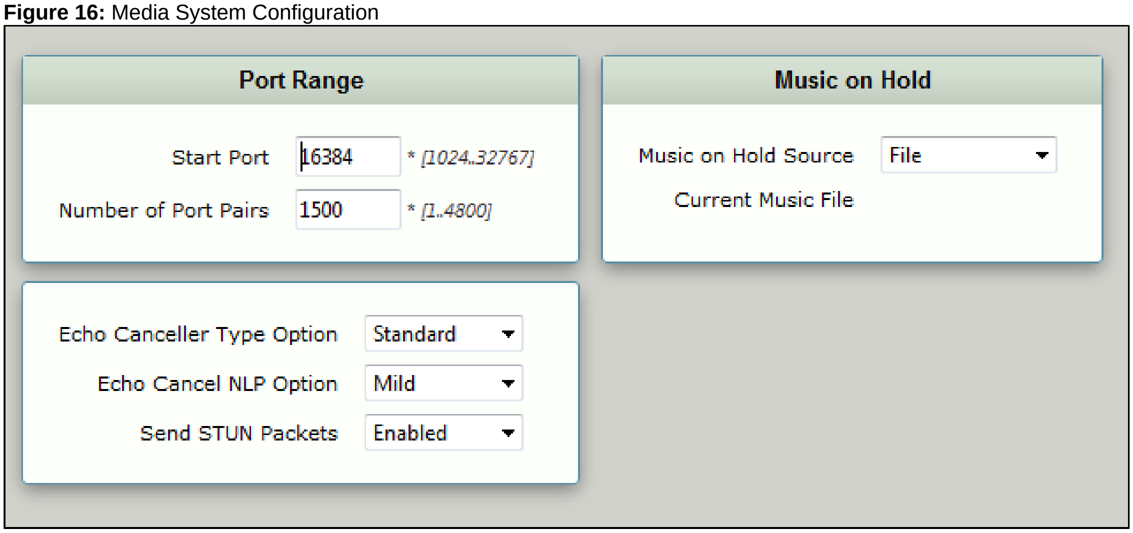
Description	AT&T-IPFR (ATT): G.729
Codec	G.729
Payload Size	20

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



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

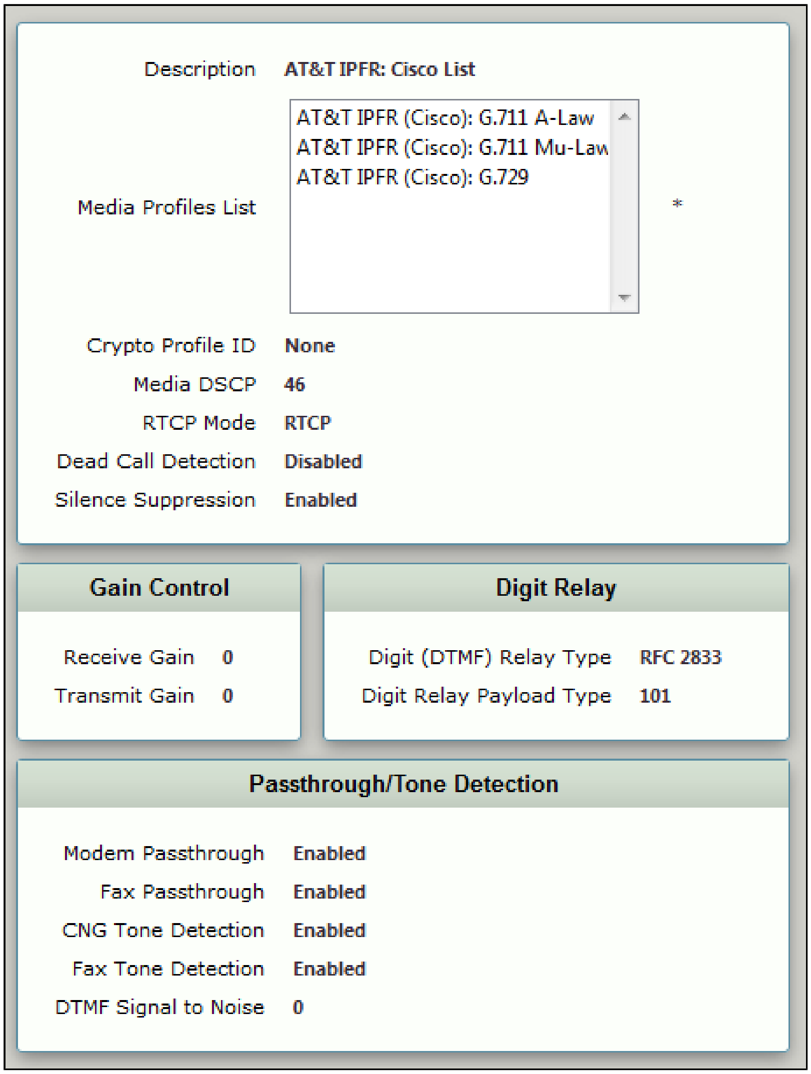
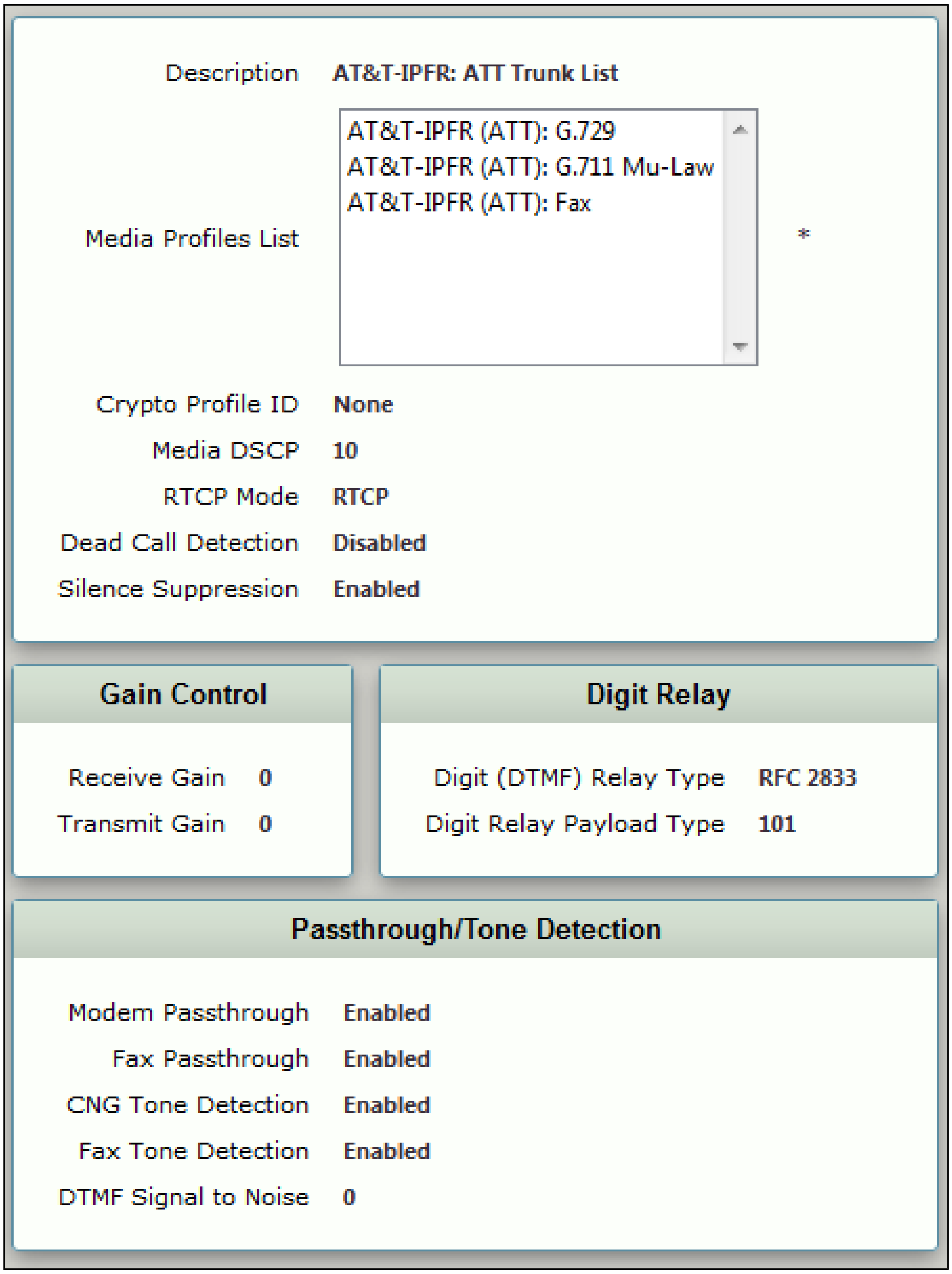


Figure 18: AT&T-IPFR: ATT Trunk List



Description	AT&T-IPFR: ATT Trunk List
Media Profiles List	AT&T-IPFR (ATT): G.729 AT&T-IPFR (ATT): G.711 Mu-Law AT&T-IPFR (ATT): Fax *
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

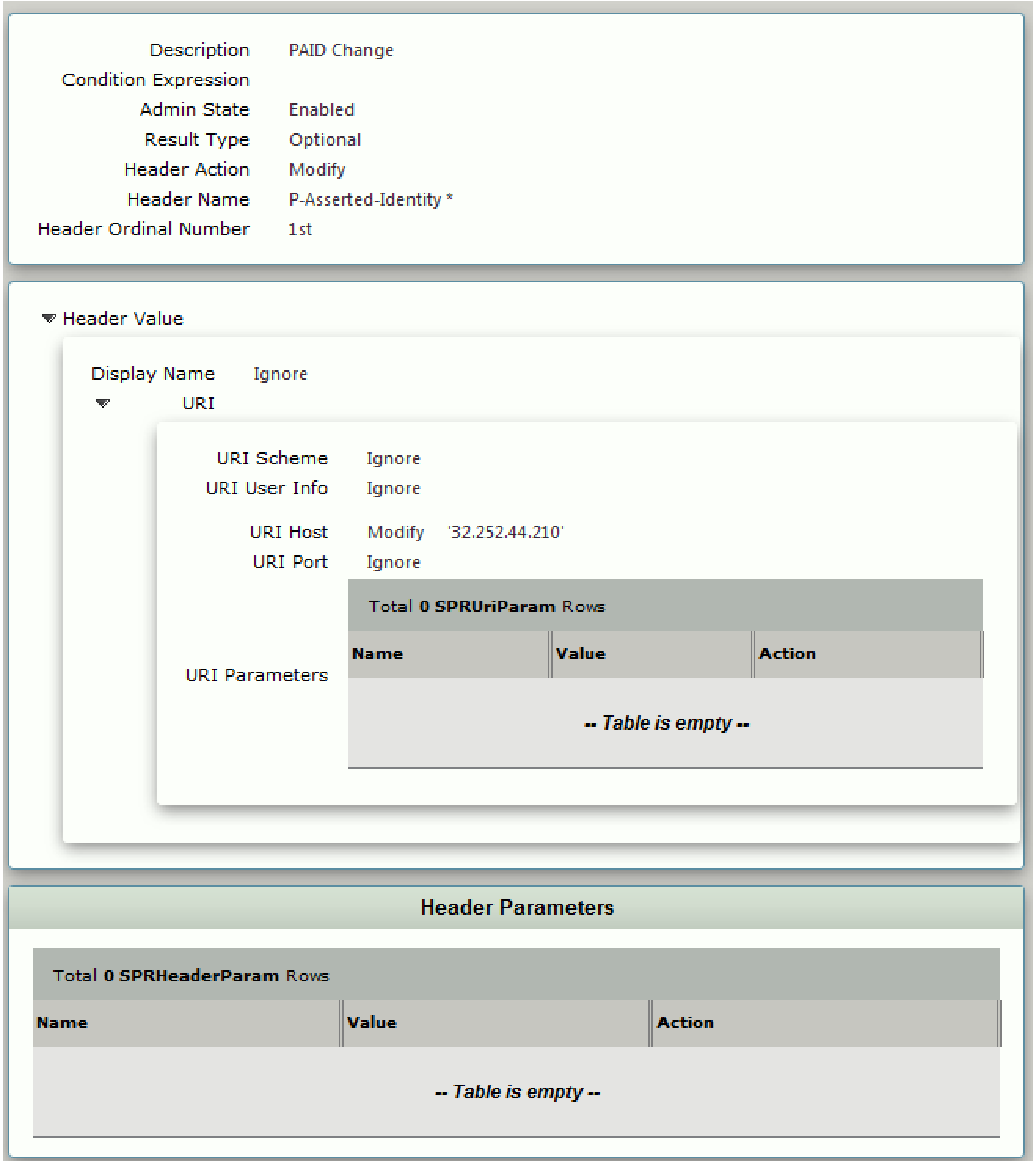
8. Message Manipulation

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



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

Server Lookup	IP/FQDN
Priority	1
Host	10.35.180.110
Port	5060
Protocol	TCP

Transport

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

Connection Reuse

Reuse	True
Sockets	4
Reuse Timeout	Forever

Figure 21: AT&T-IPFR: Border Element

Server Host

Server Lookup	IP/FQDN
Priority	1
Host	X.X.X.X
Port	5060
Protocol	UDP

Transport

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

Server Lookup	IP/FQDN
Priority	1
Host	Y.Y.Y.Y
Port	5060
Protocol	UDP

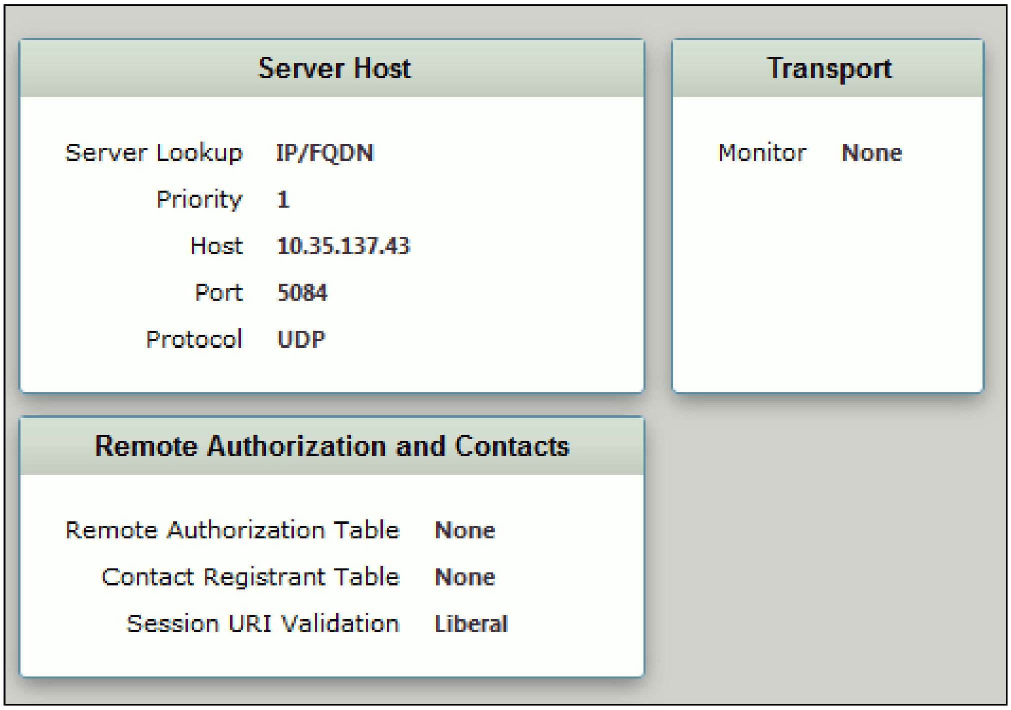
Transport

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



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

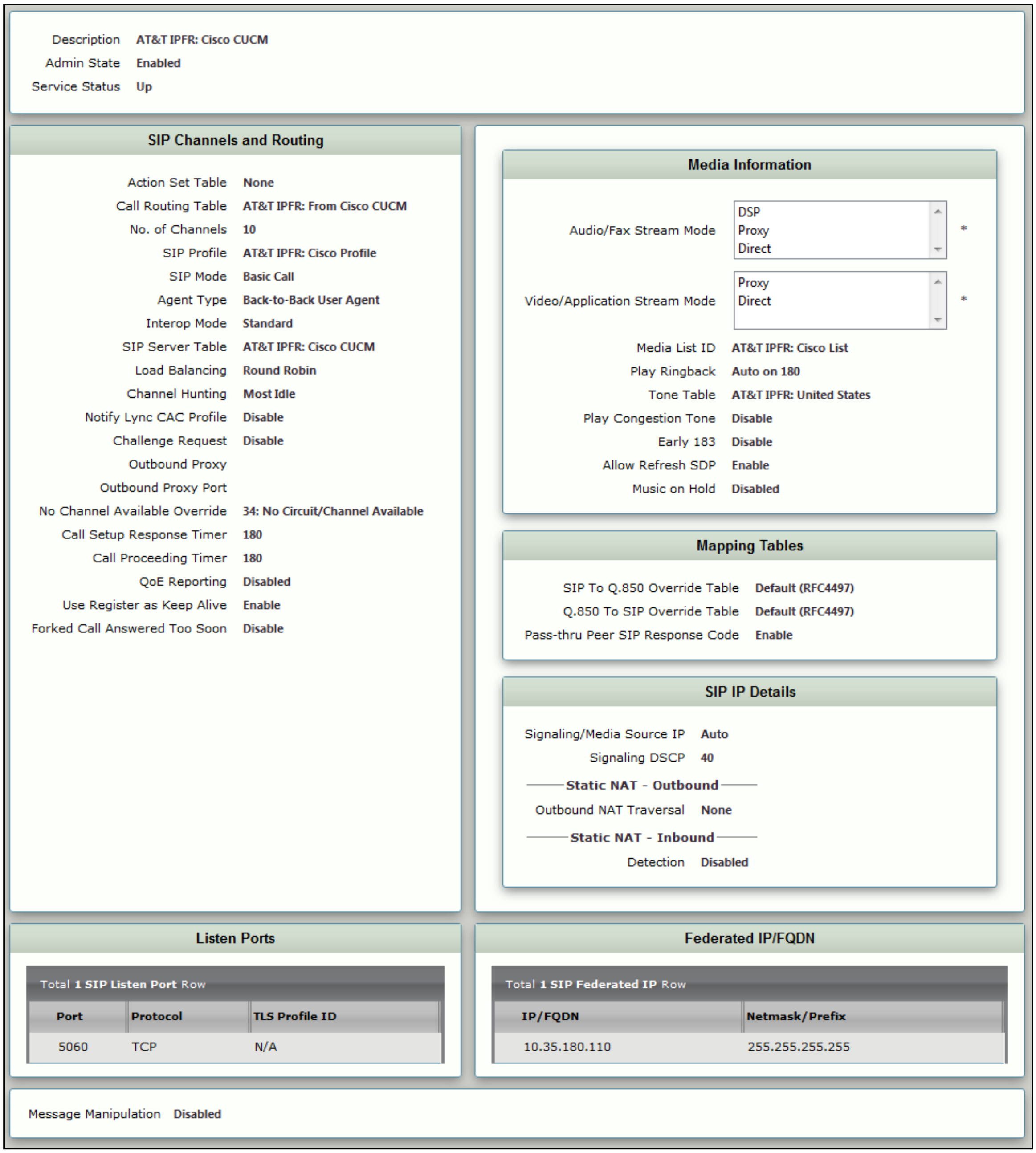


Figure 24: AT&T-IPFR: ATT Border Element

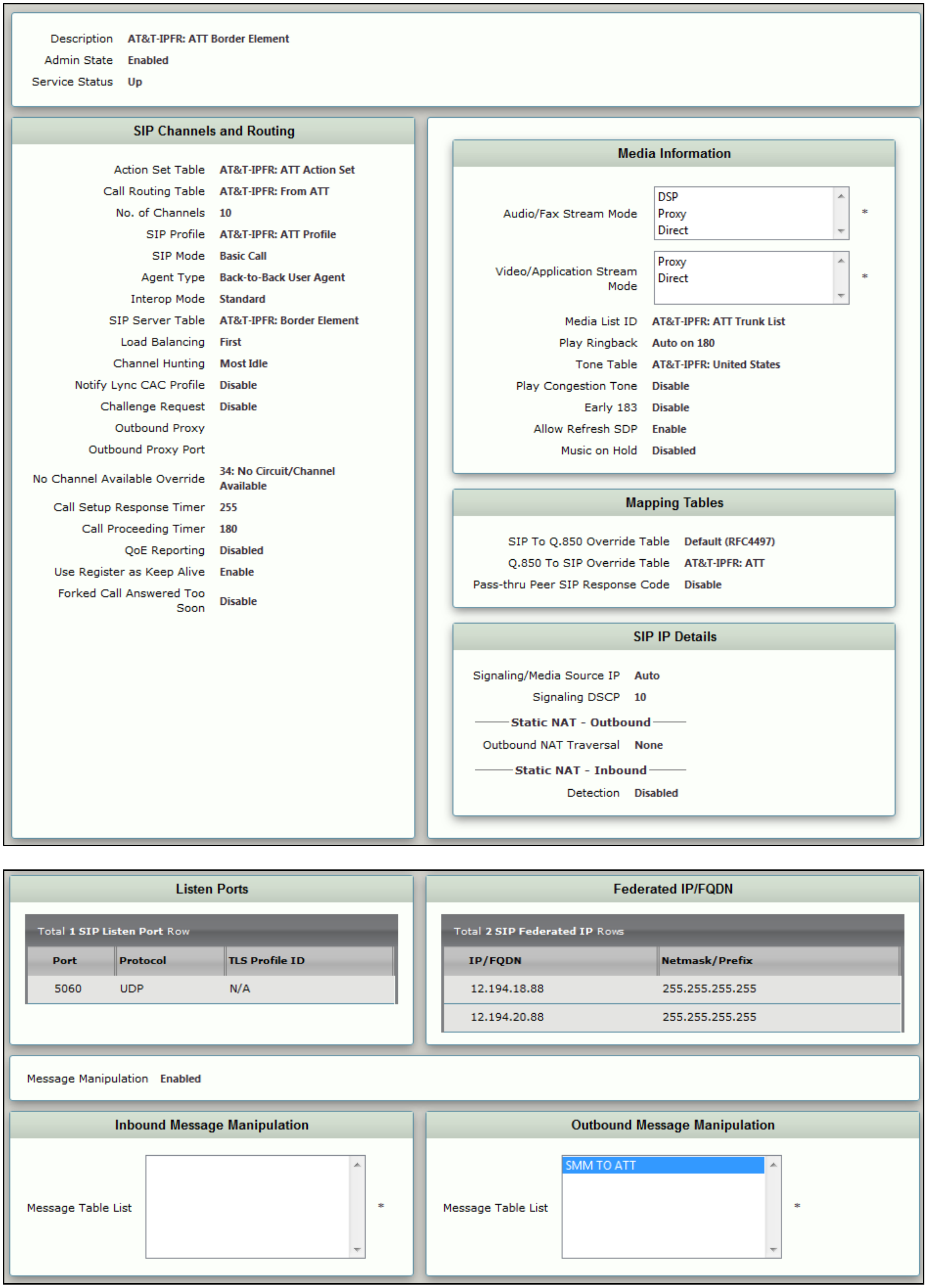
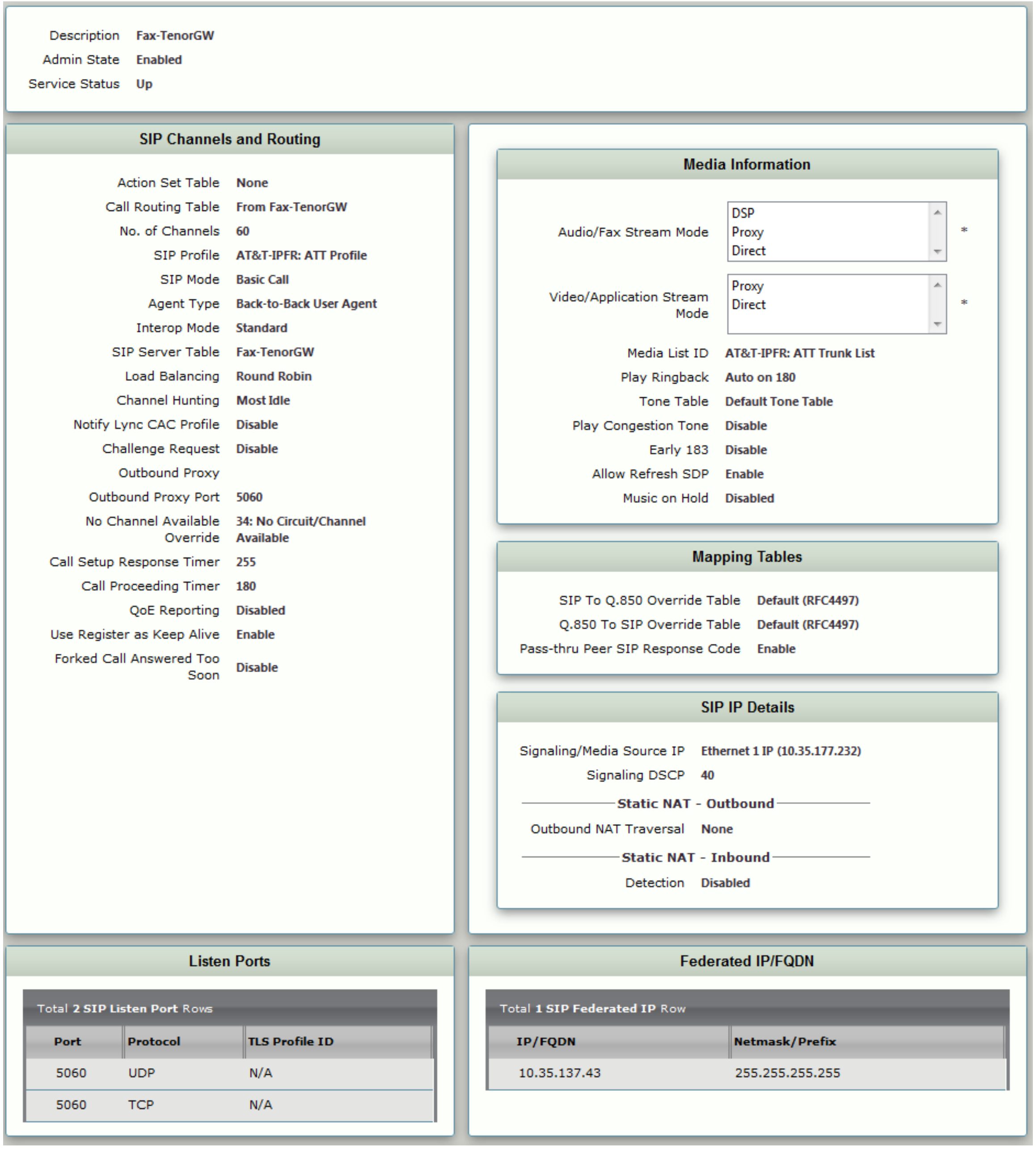


Figure 25: Fax-TenorGW

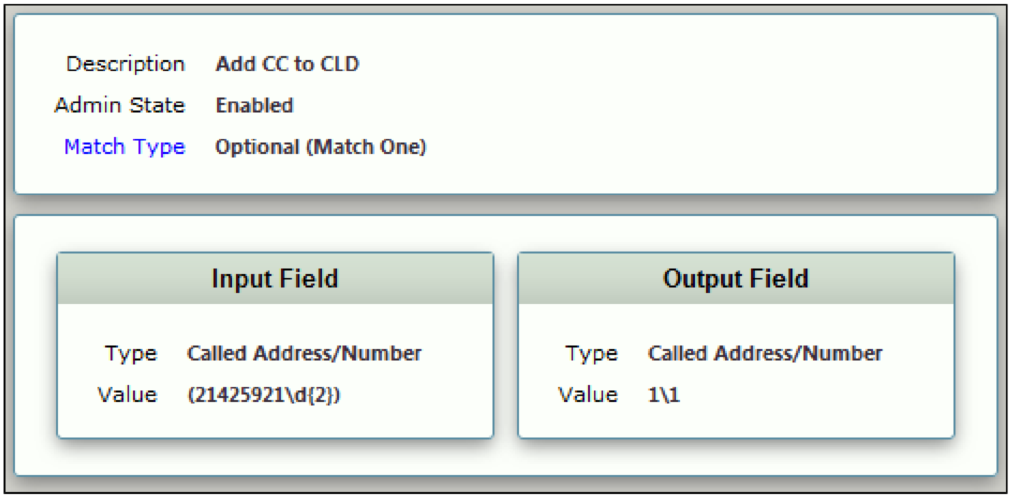


11. Transformation

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



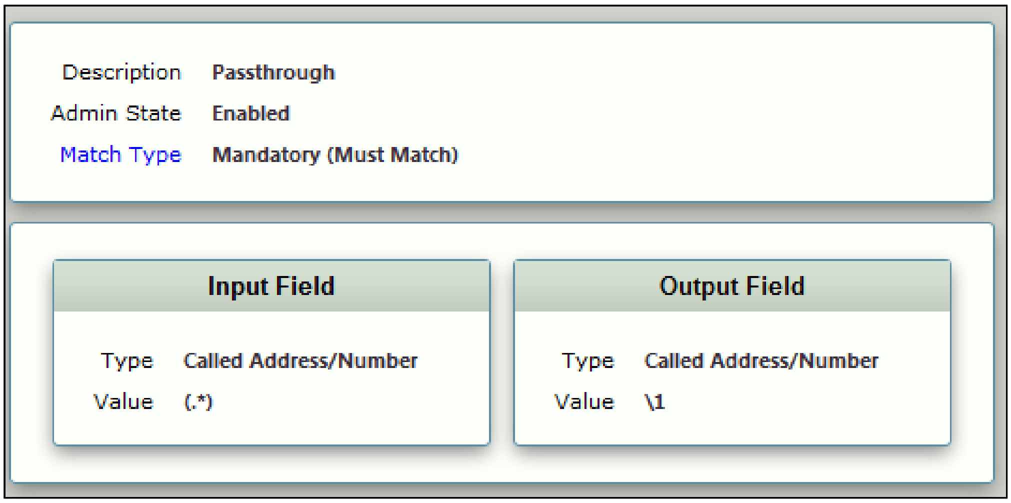


Figure 27: AT&T-IPFR: From Cisco CUCM

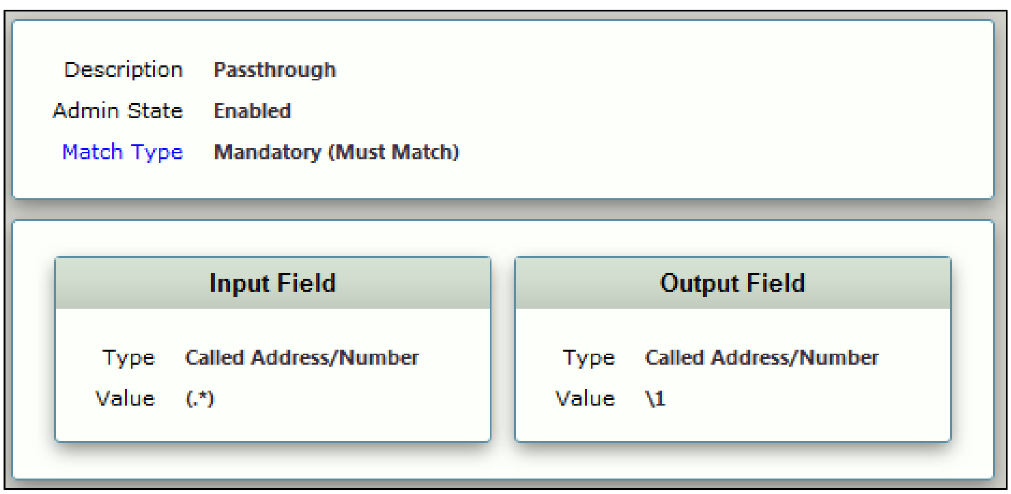


Figure 28: From Fax-TenorGW

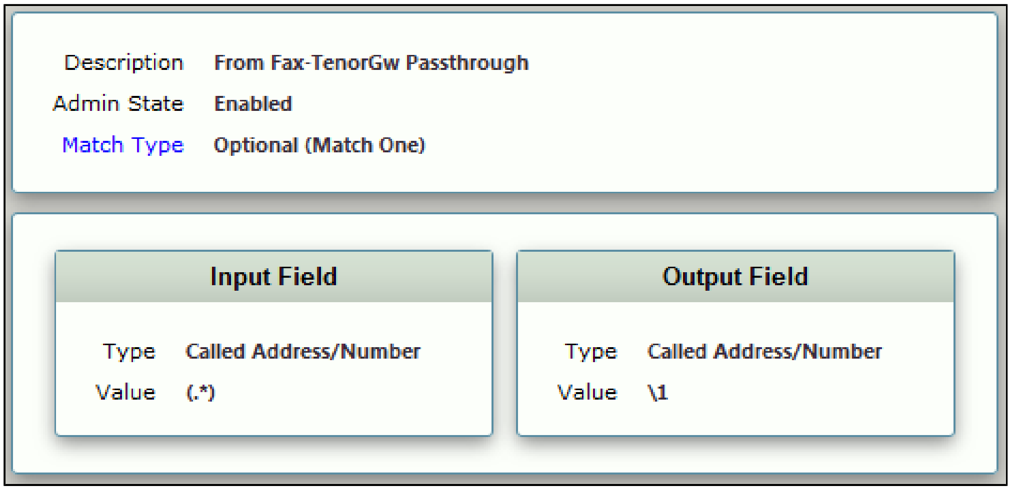
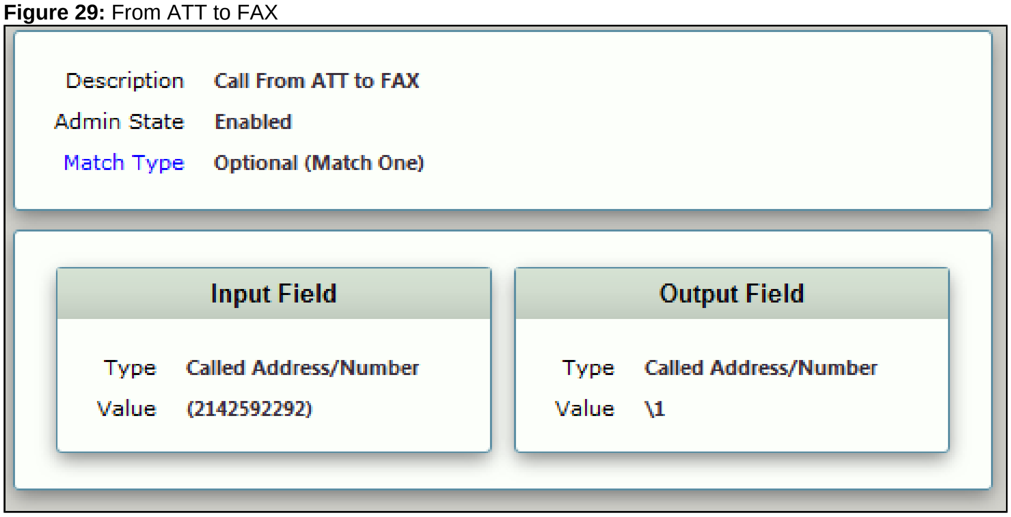


Figure 29: From ATT to FAX



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

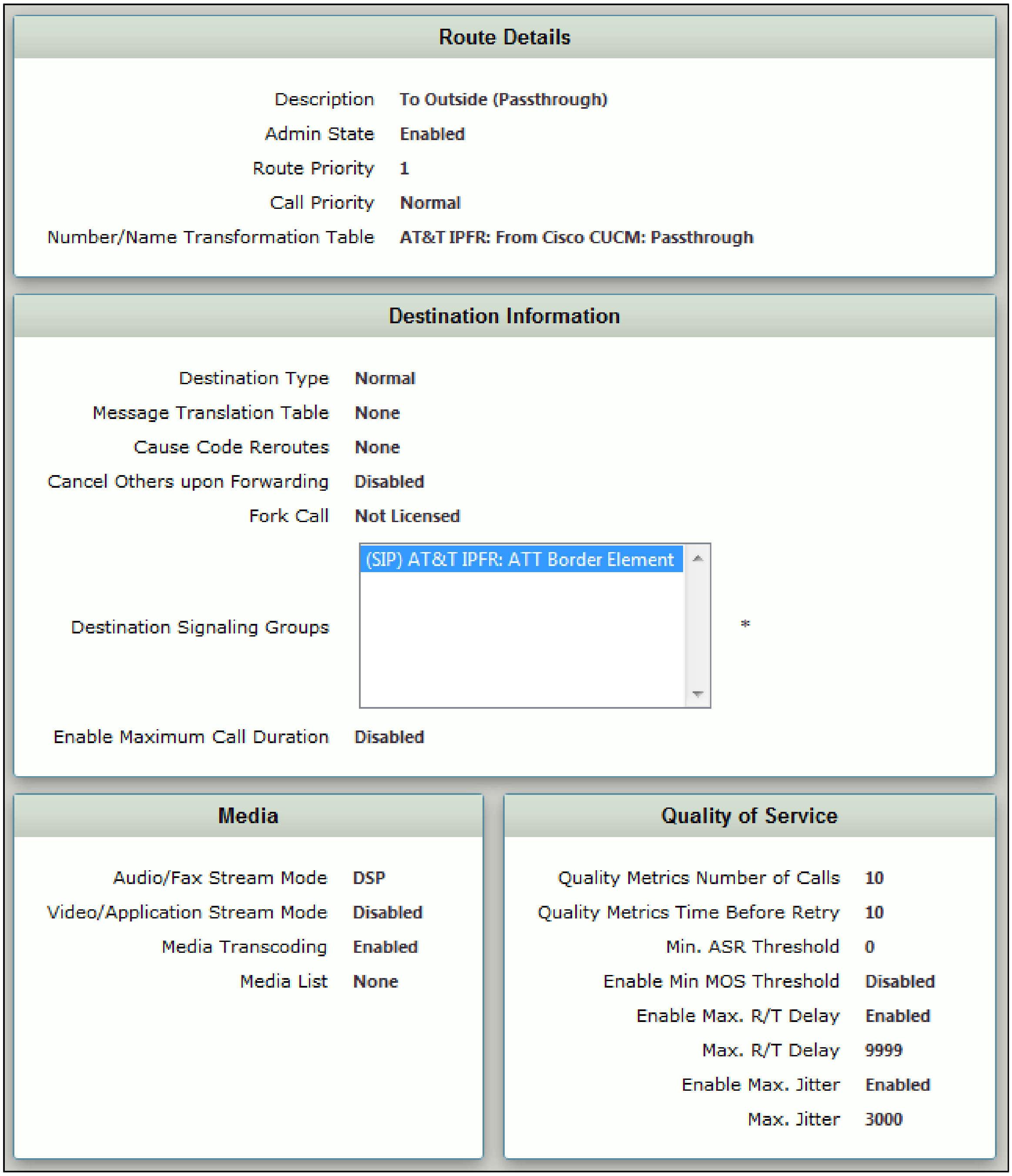


Figure 31: AT&T-IPFR: From ATT

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

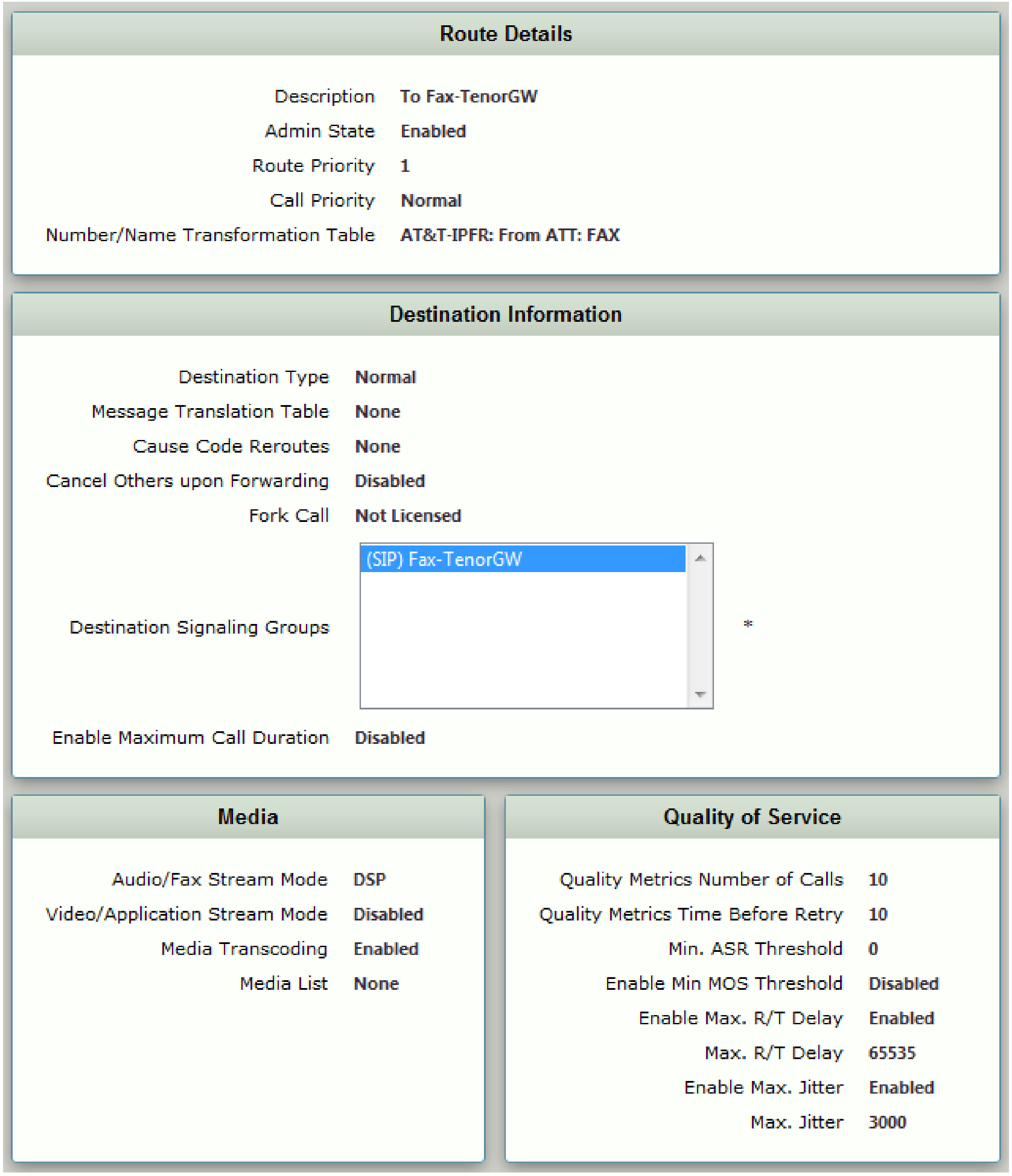
Destination Type	Normal
Message Translation Table	None
Cause Code Reroutes	None
Cancel Others upon Forwarding	Disabled
Fork Call	Not Licensed
Destination Signaling Groups	(SIP) AT&T-IPFR: Avaya CM *
Enable Maximum Call Duration	Disabled

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	(SIP) Fax-TenorGW *
Enable Maximum Call Duration	Disabled

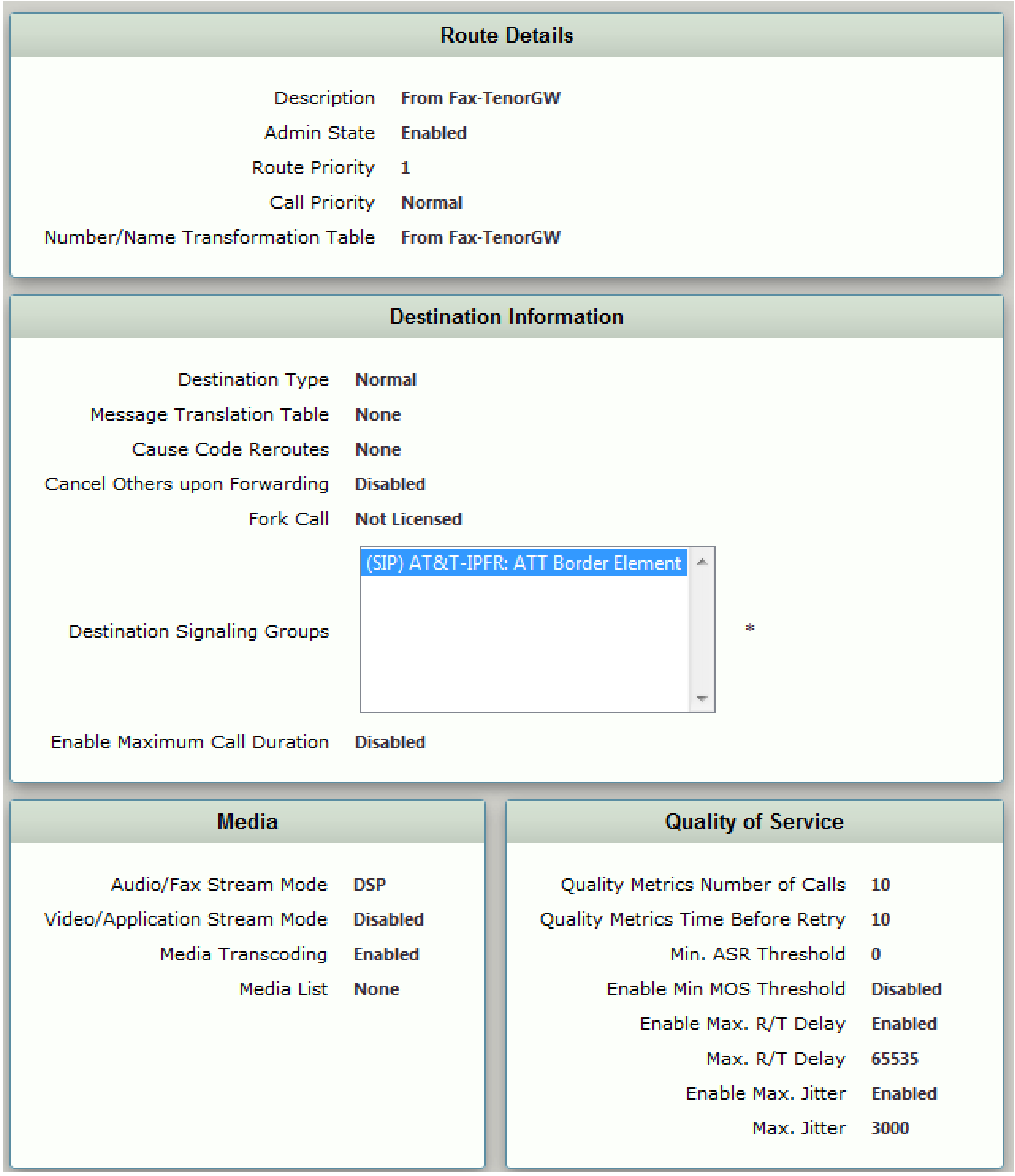
Media

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 32: From Fax-TenorGW



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.