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

Table of Contents

Document Overview

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

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

Introduction

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

Audience

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

Requirements

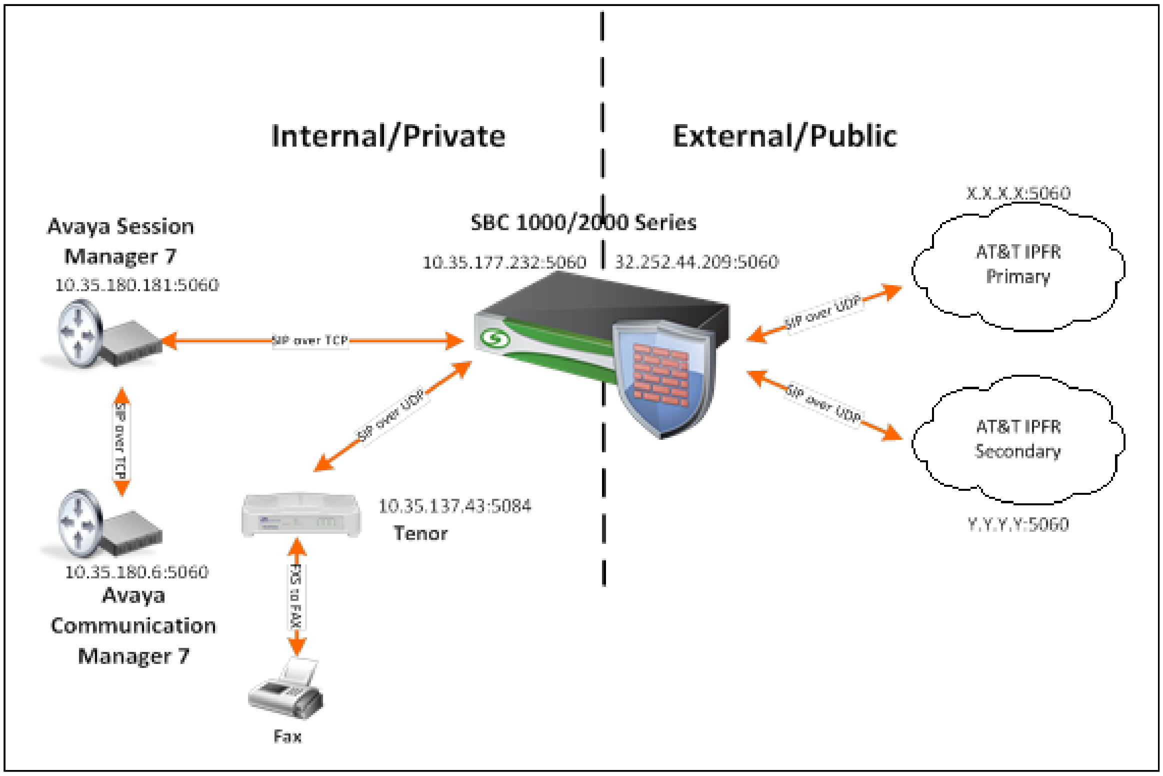
The following equipment and software were used for the sample configuration (see Topology):

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

Reference Configuration

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

Figure 1: Topology



Support

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

Third-Party Product Features

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

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

Not Supported Features

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

Verify License

No special licensing required.

Avaya SM 7 Configuration

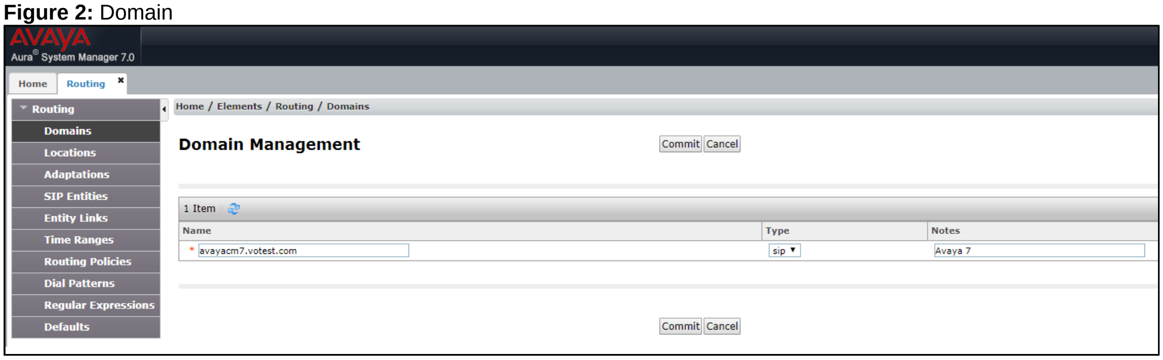
This section includes thefollowing new configurations:

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

1. Domain

Select**Home> Routing > Domains**

Figure 2: Domain



2. Location

Select **Home> Routing > Locations**

Figure 3: Location for CM

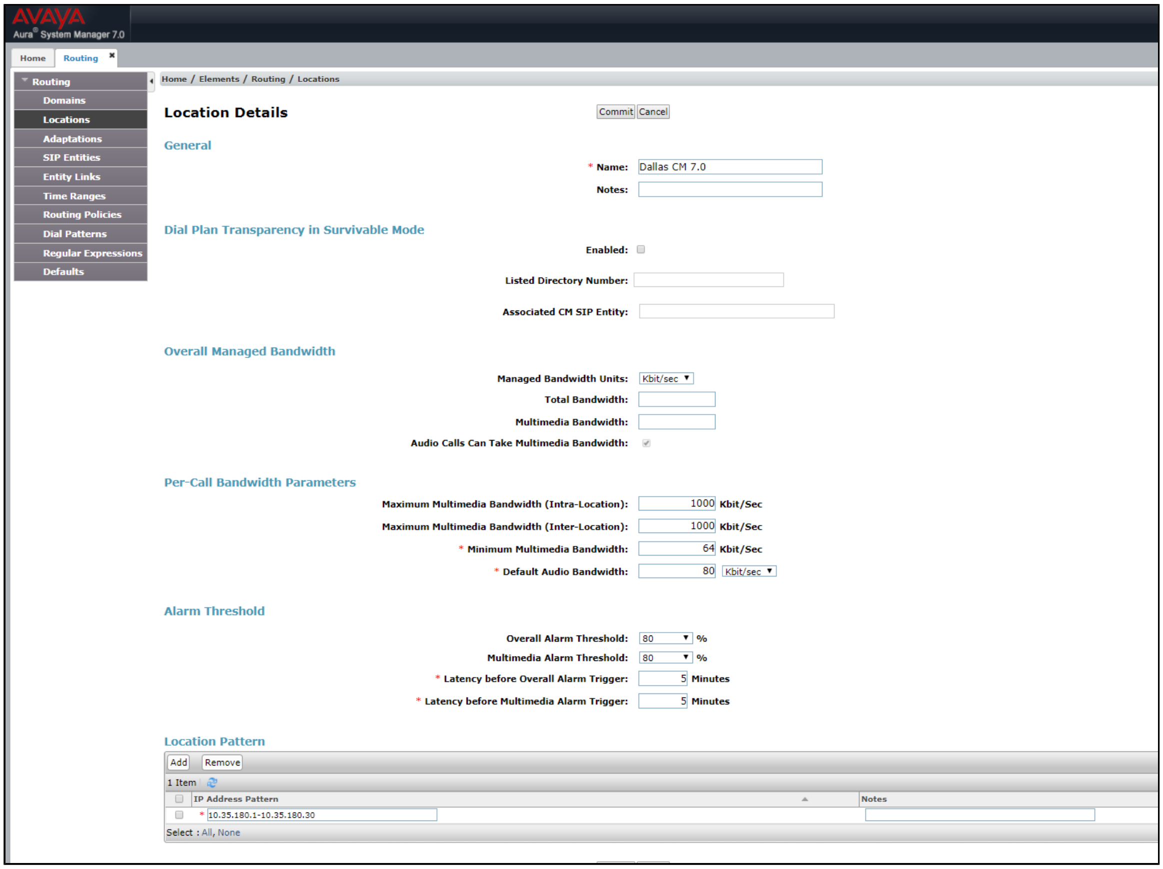
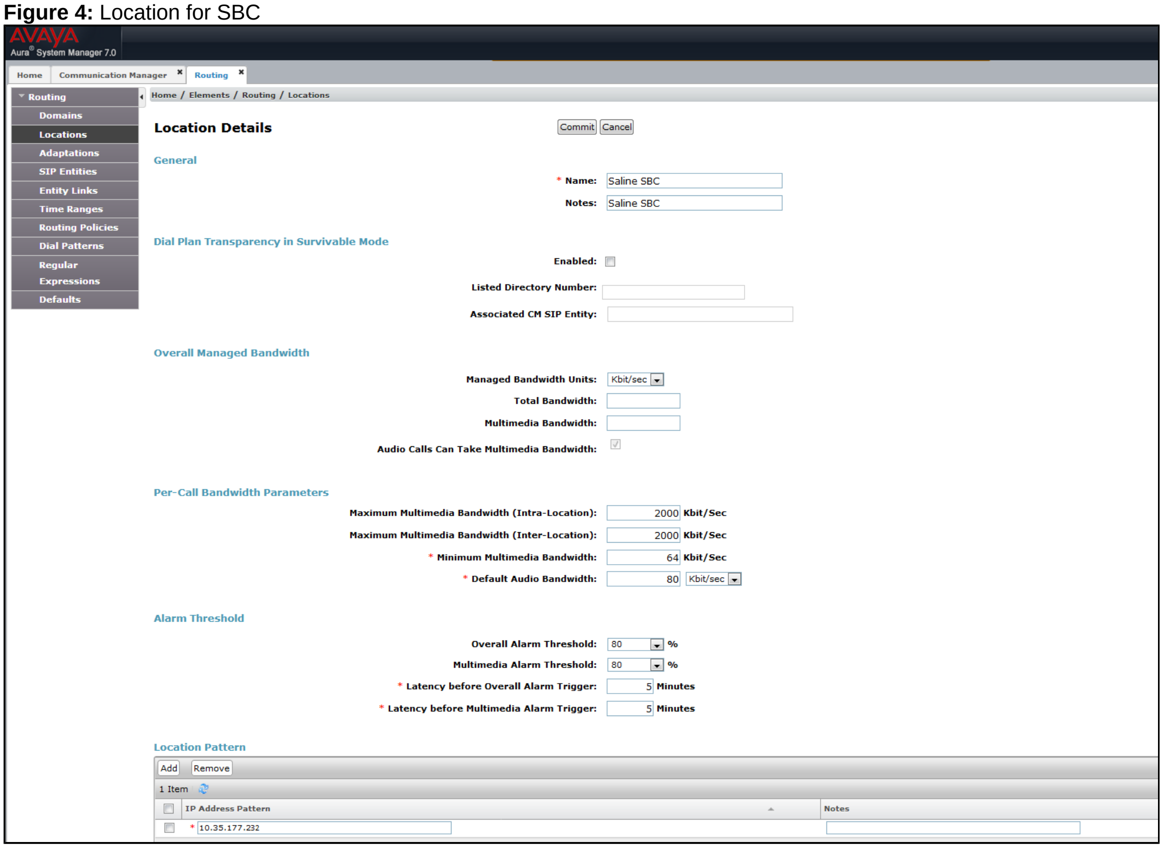


Figure 4: Location for SBC



3. SIP Entity

Select**Home> Routing > SIP Entities**

Figure 5: SIP Entity for CM

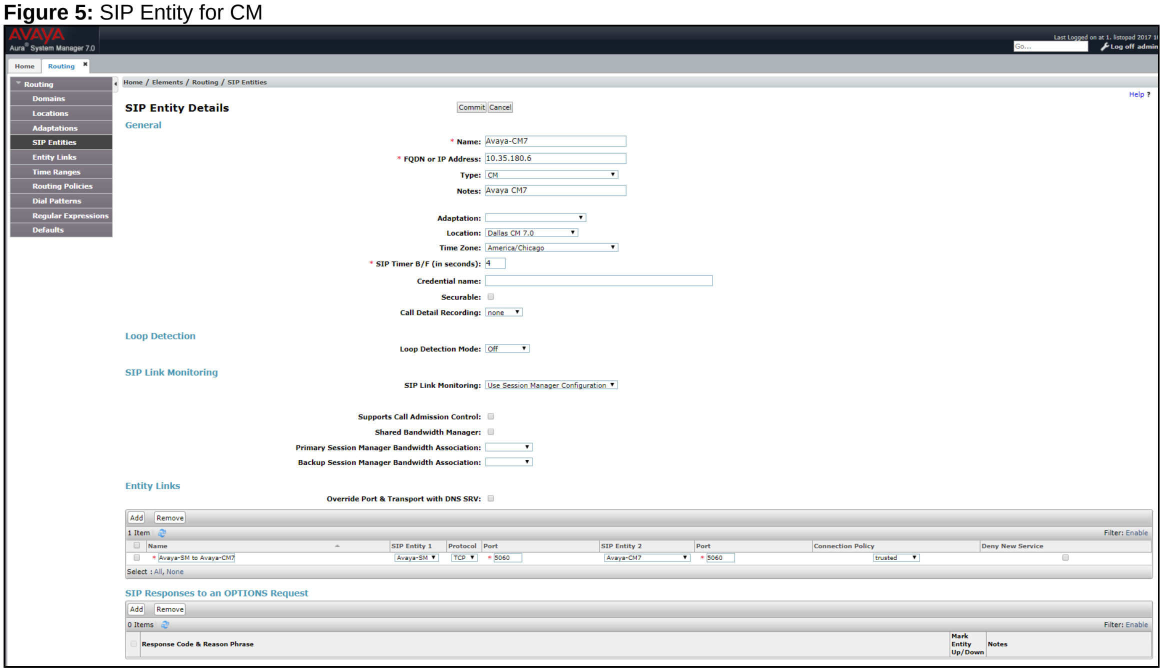
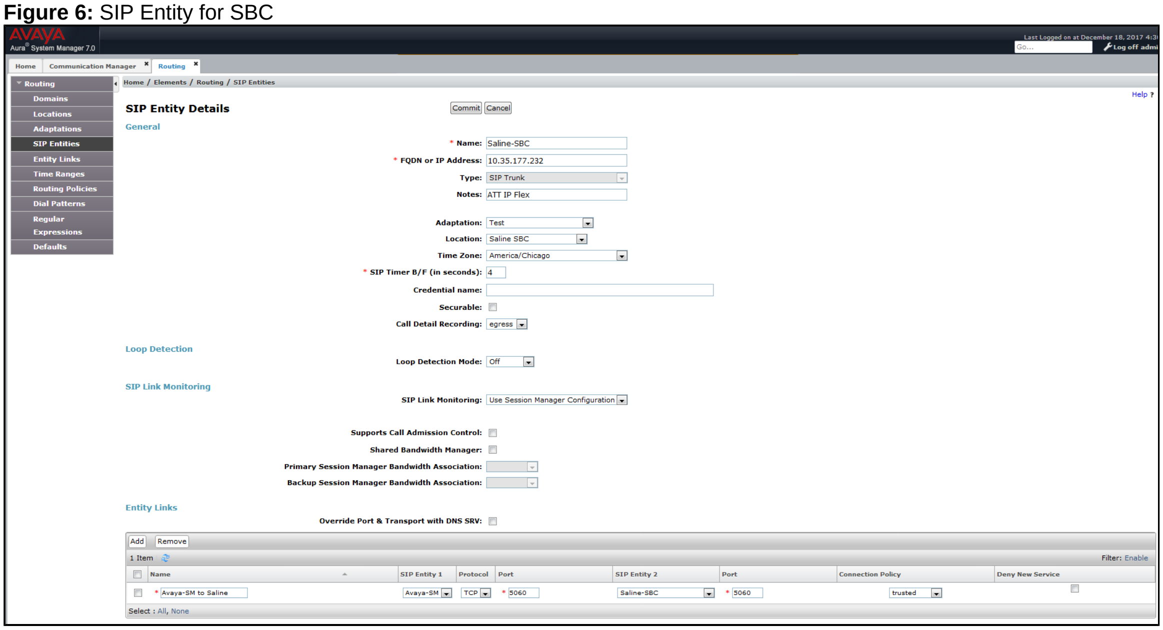


Figure 6: SIP Entity for SBC



4. Entity Link

Select**Home> Routing > Entity Links**

Figure 7: Entity Link for CM

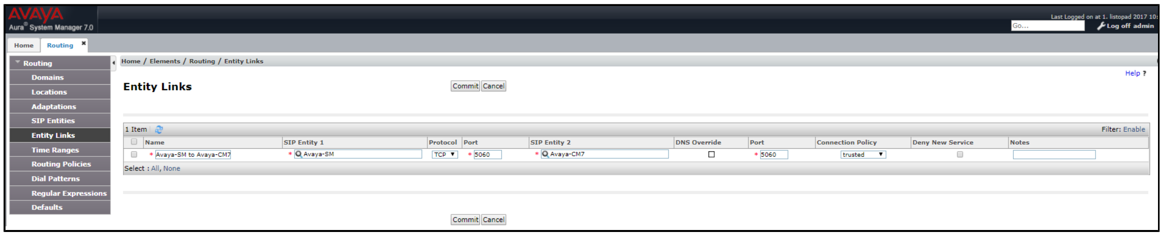
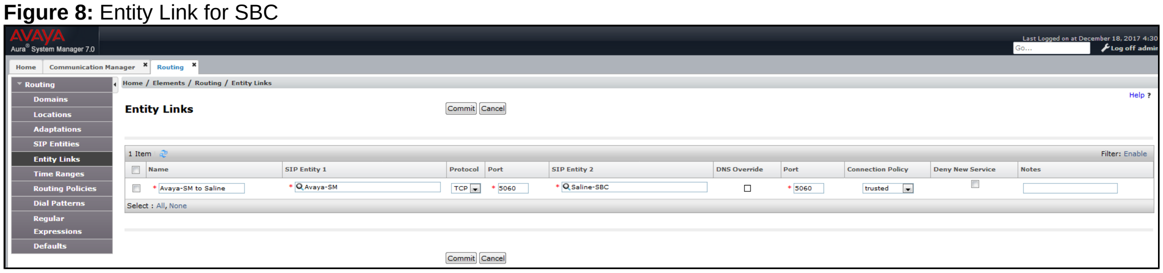


Figure 8: Entity Link for SBC



5. Routing Policy

Select**Home> Routing > Routing Policies**

Figure 9: Route Policy for CM

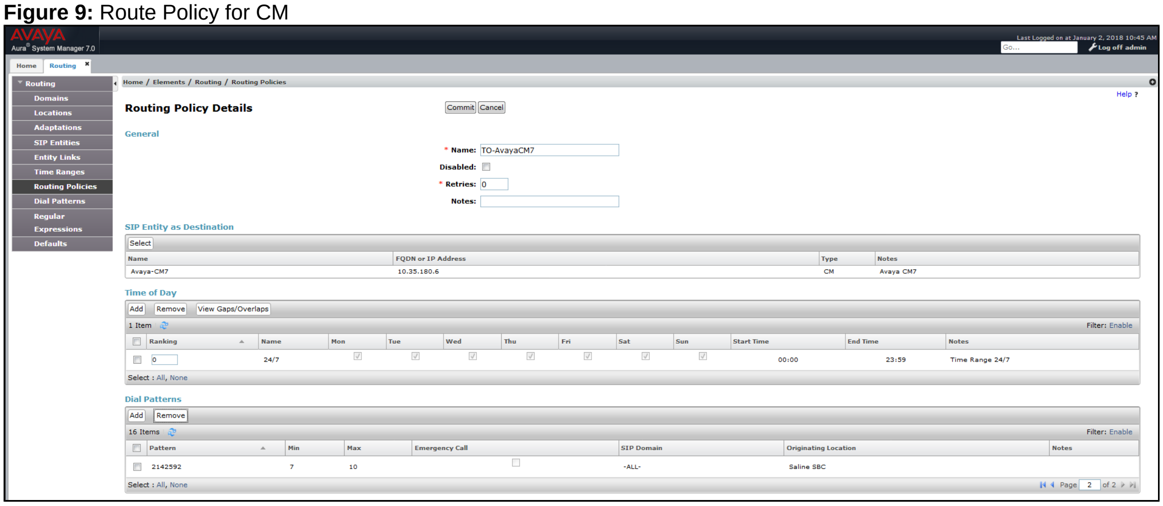
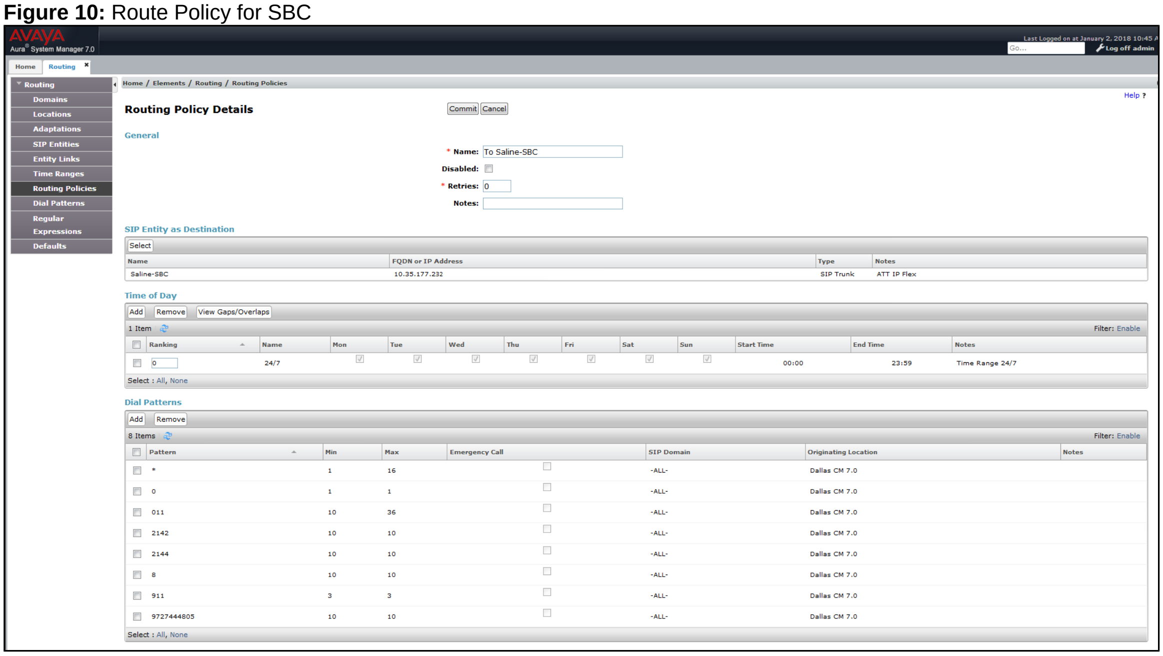


Figure 10: Route Policy for SBC



6. Dial Pattern

Select**Home> Routing > Dial Patterns**

> **Note**
> Use this procedure to create any Dial Pattern configuration.

Figure 11: Dial Pattern for CM

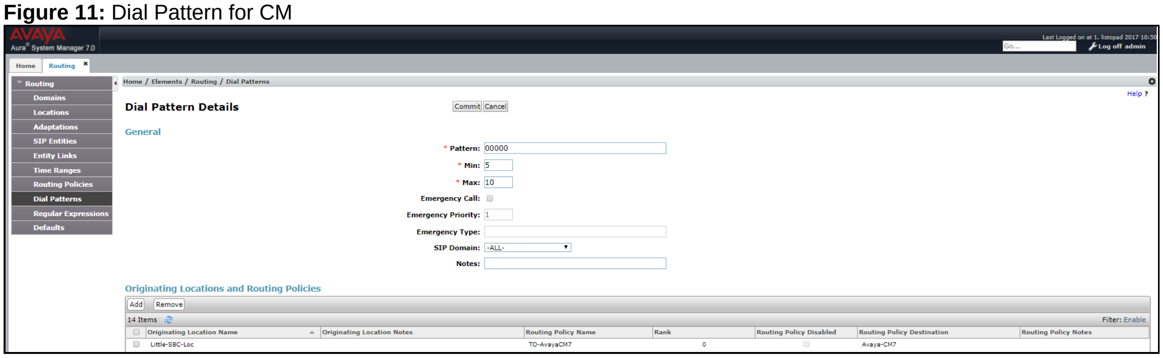
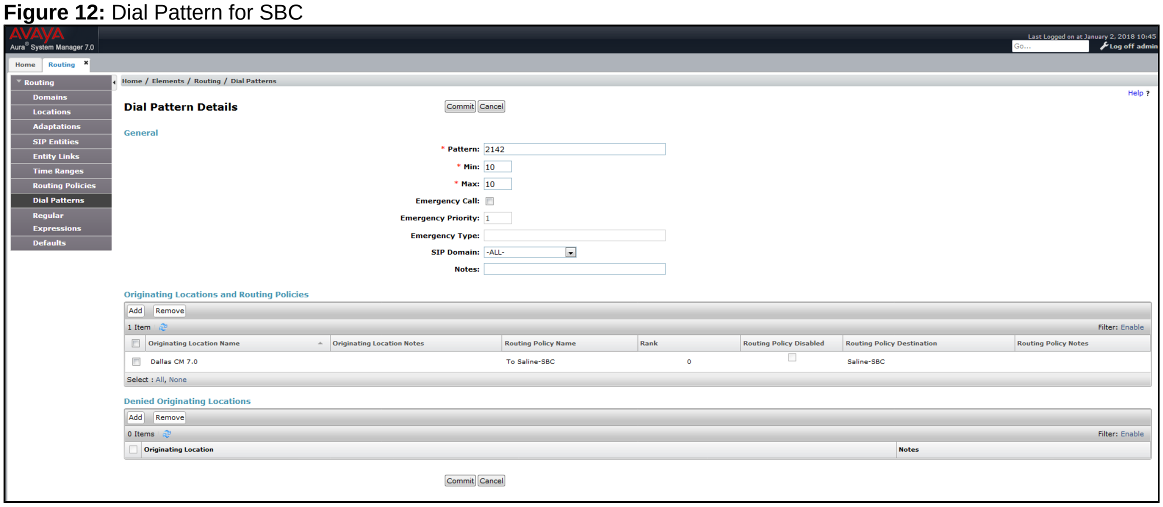


Figure 12: Dial Pattern for SBC



Avaya CM 7 Configuration

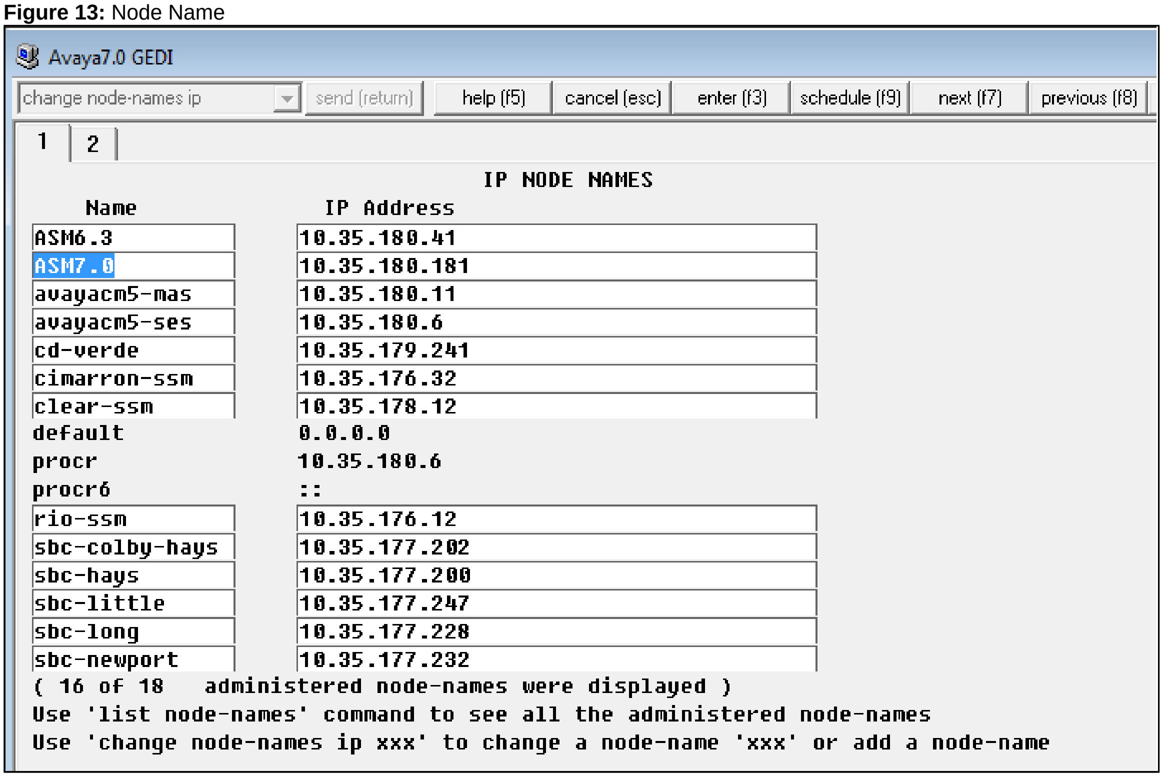
This section includes thefollowing new configurations:

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

1.Node Name

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

Figure 13: Node Name



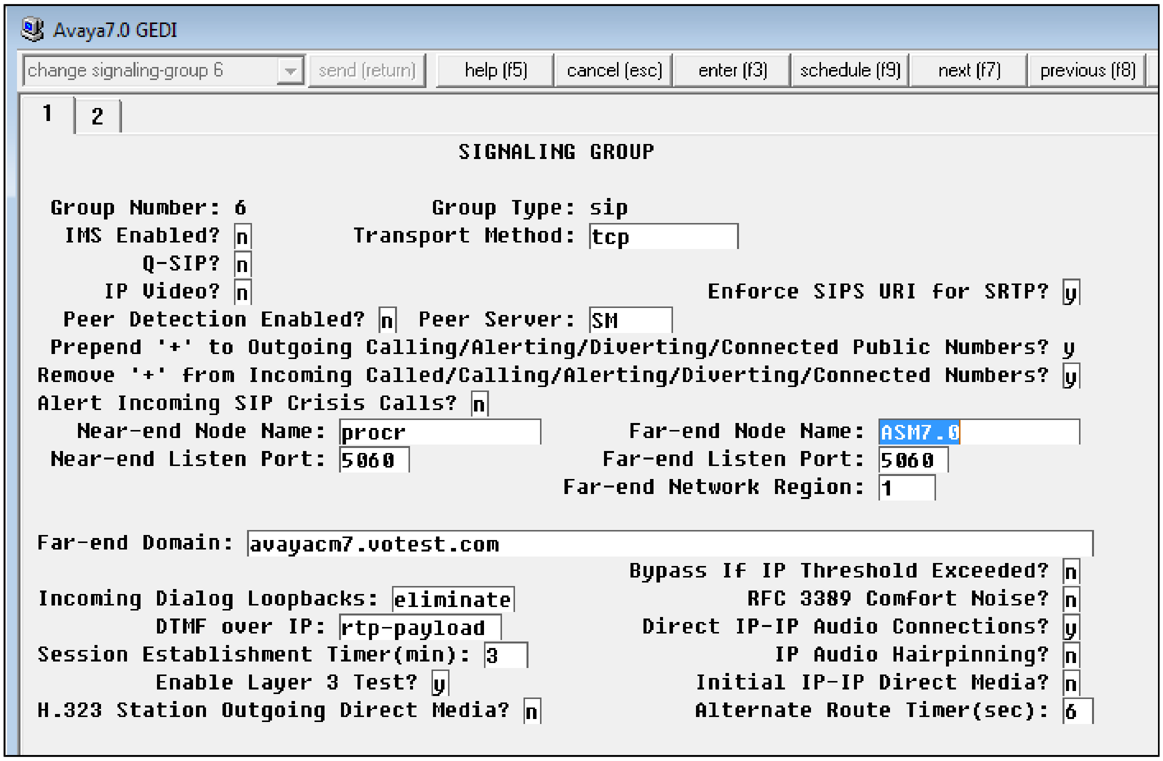
2. Signaling Group

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

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

3. Add the appropriate information and press **F3** to save when complete. For more details, refer to Avaya CM 7.0 guide.

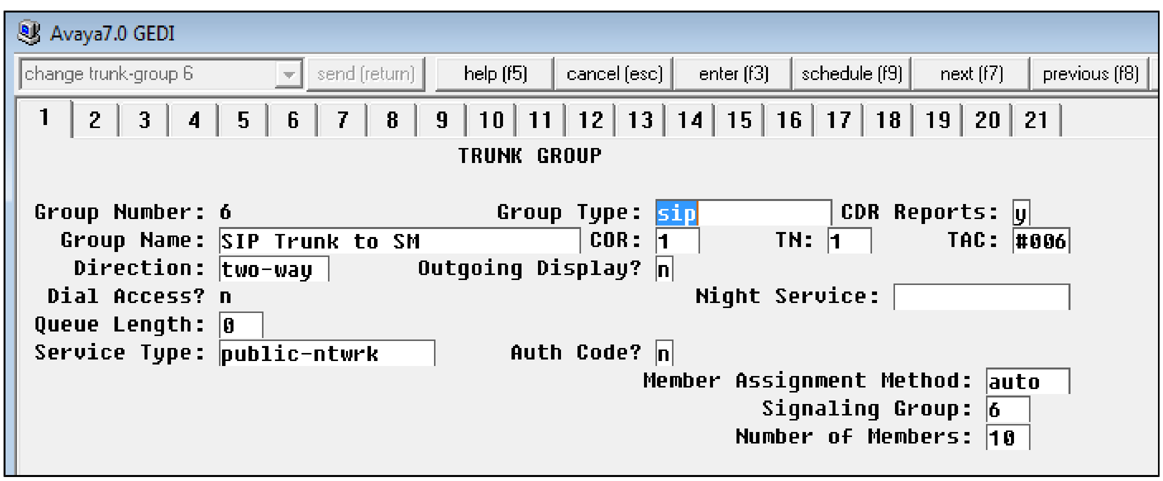
Figure 14: Signaling Group

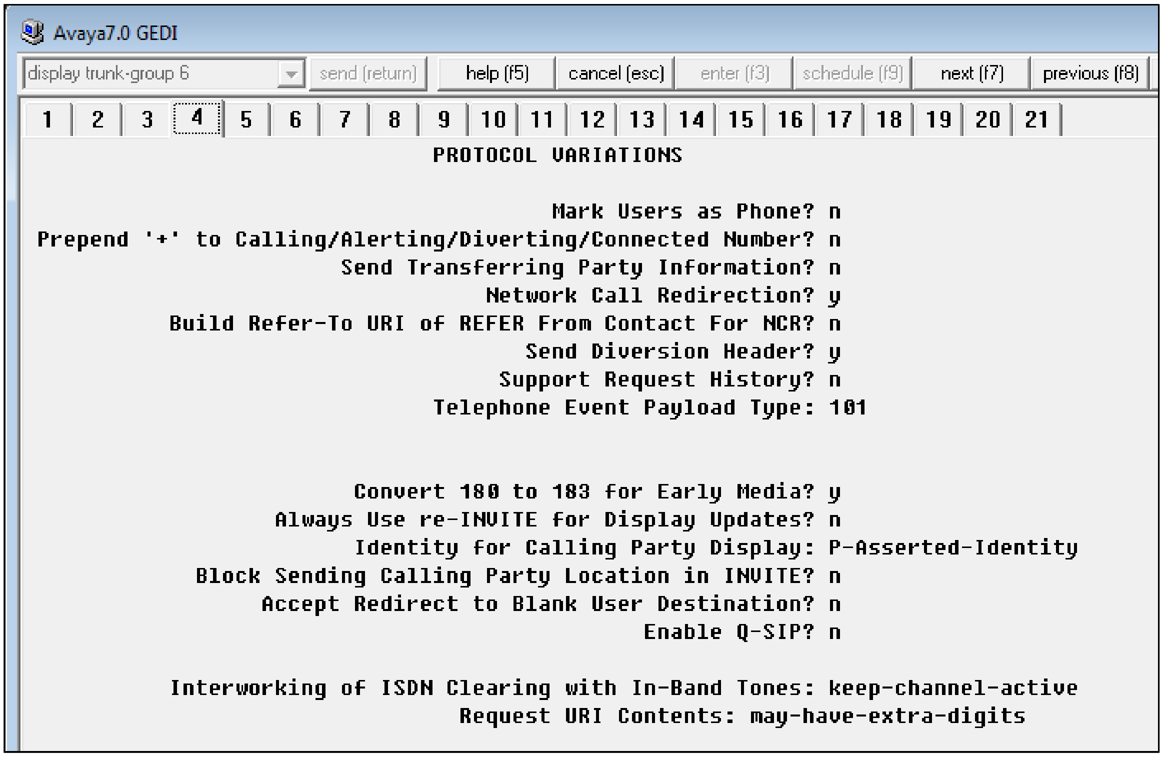


3. Trunk Group

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

Figure 15: Trunk Group

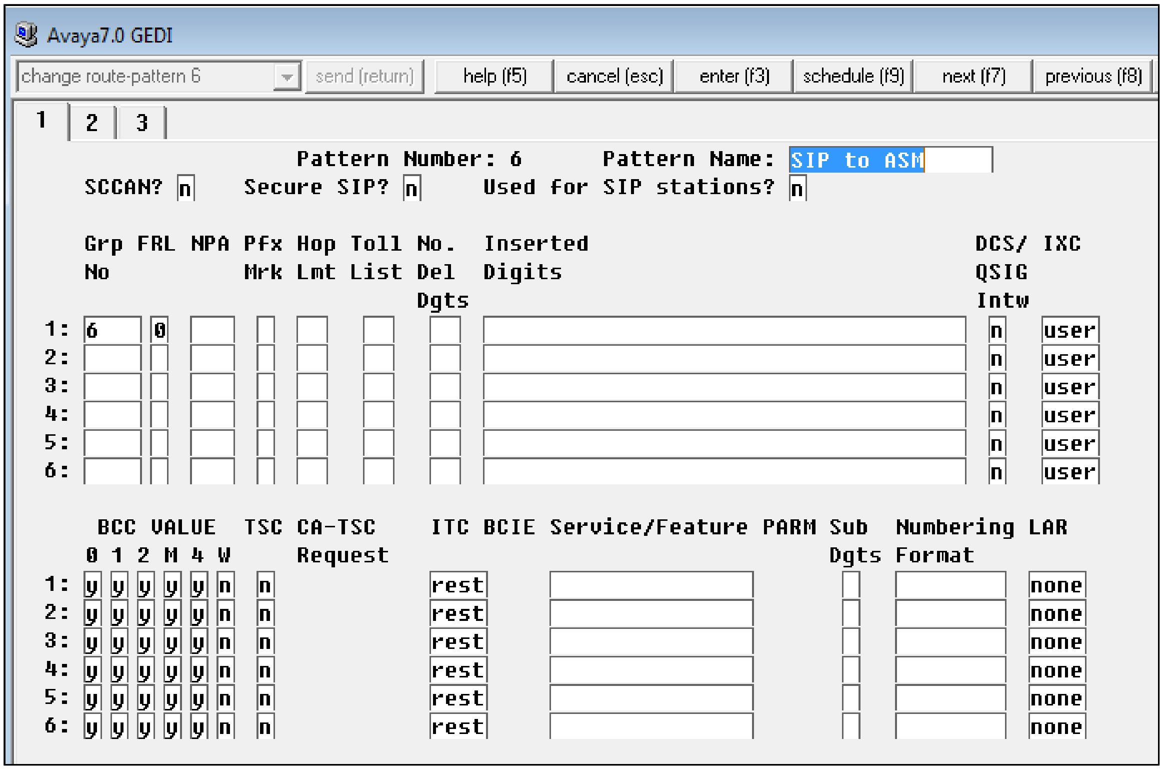




4. Route Pattern

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

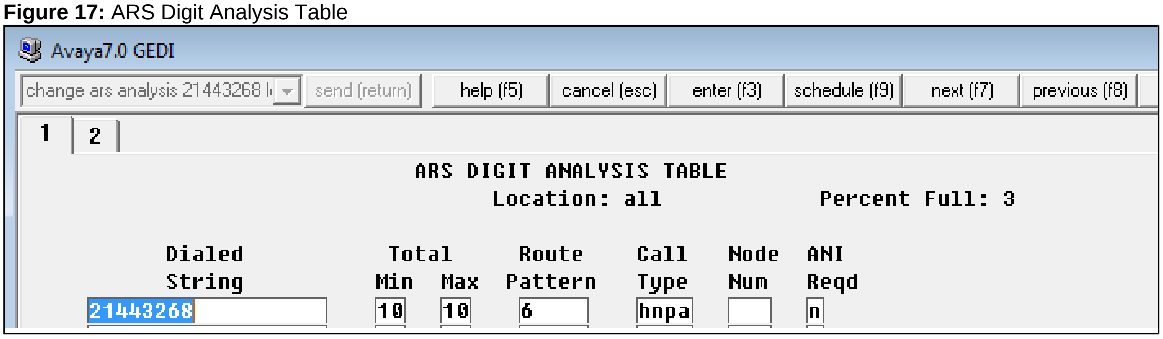
Figure 16: Route Pattern



5. ARS Analysis Table

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

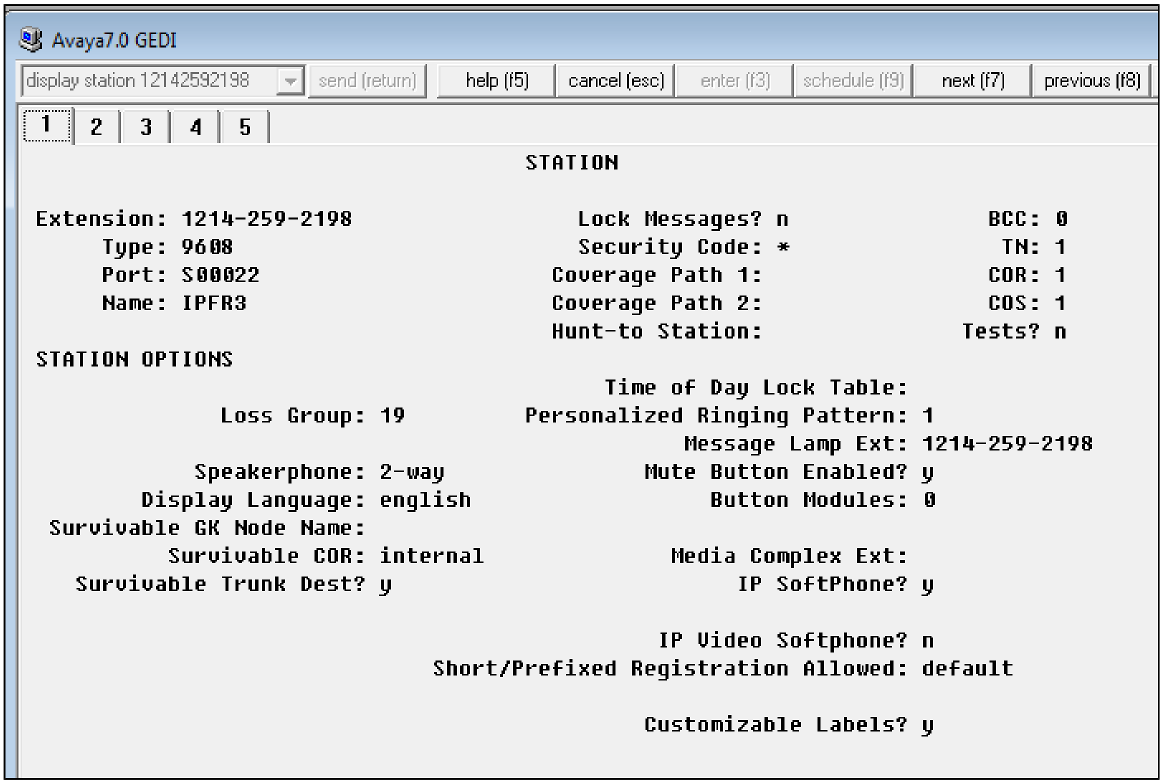
Figure 17: ARS Digit Analysis Table



6. Station

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

Figure 18: Station



7. Vector Directory Number

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

Figure 19: Vector Directory Number for Auto Attendant

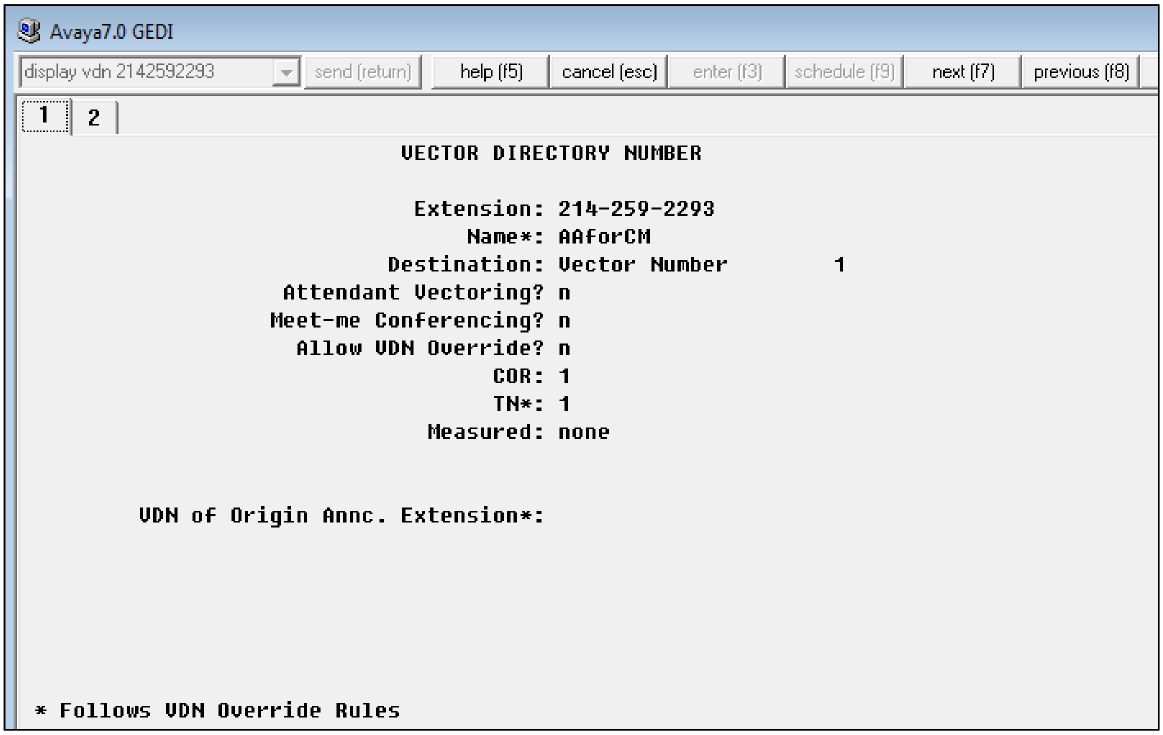
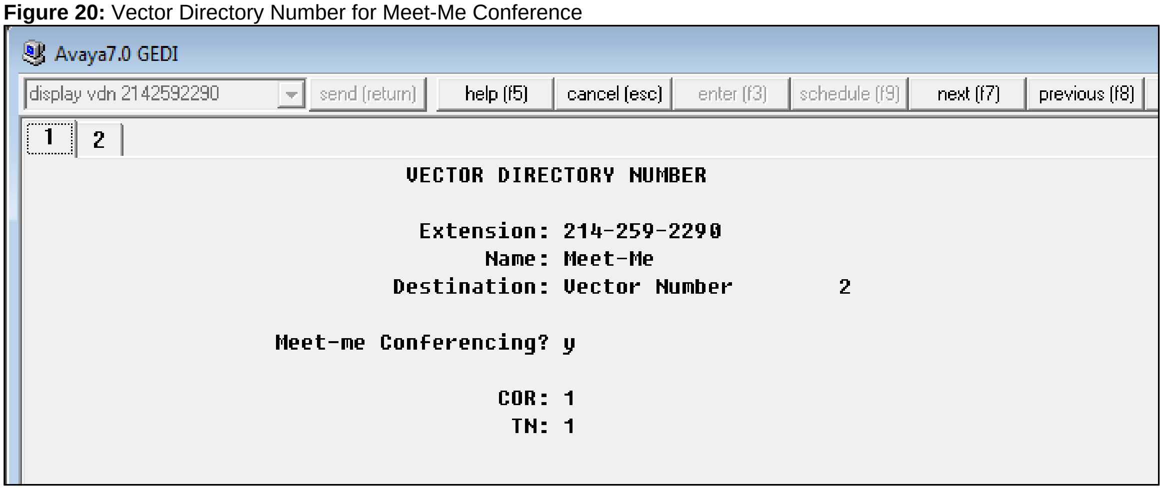


Figure 20: Vector Directory Number for Meet-Me Conference



8. Vector

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

Figure 21: Vector for Auto Attendant

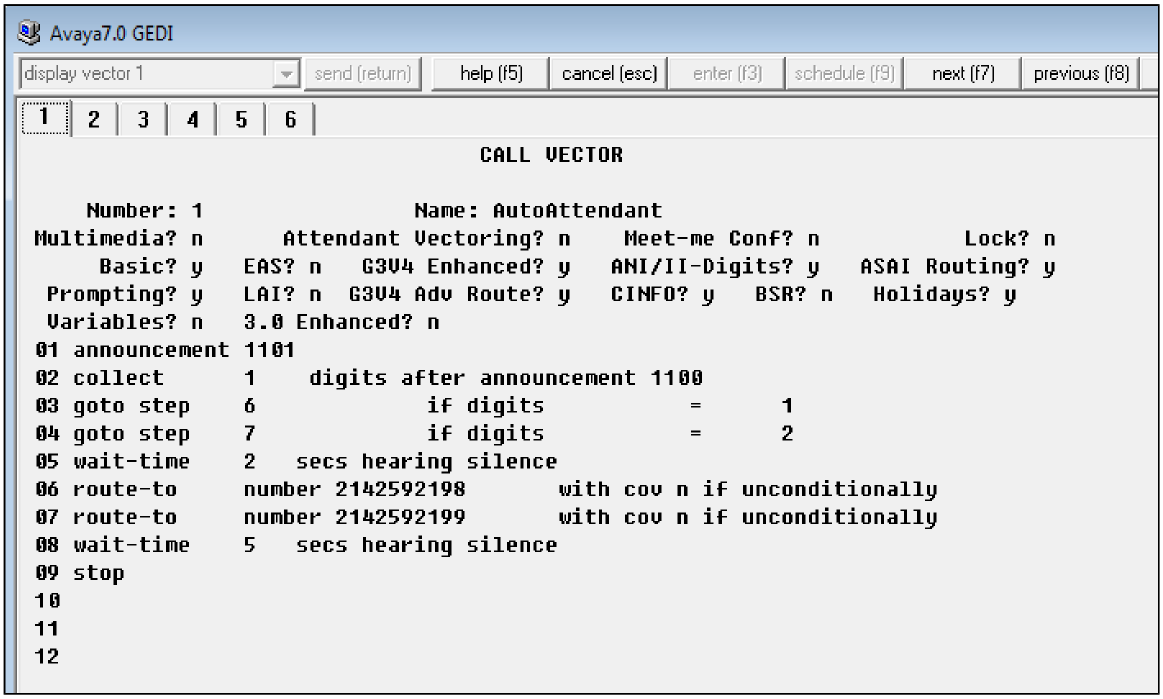
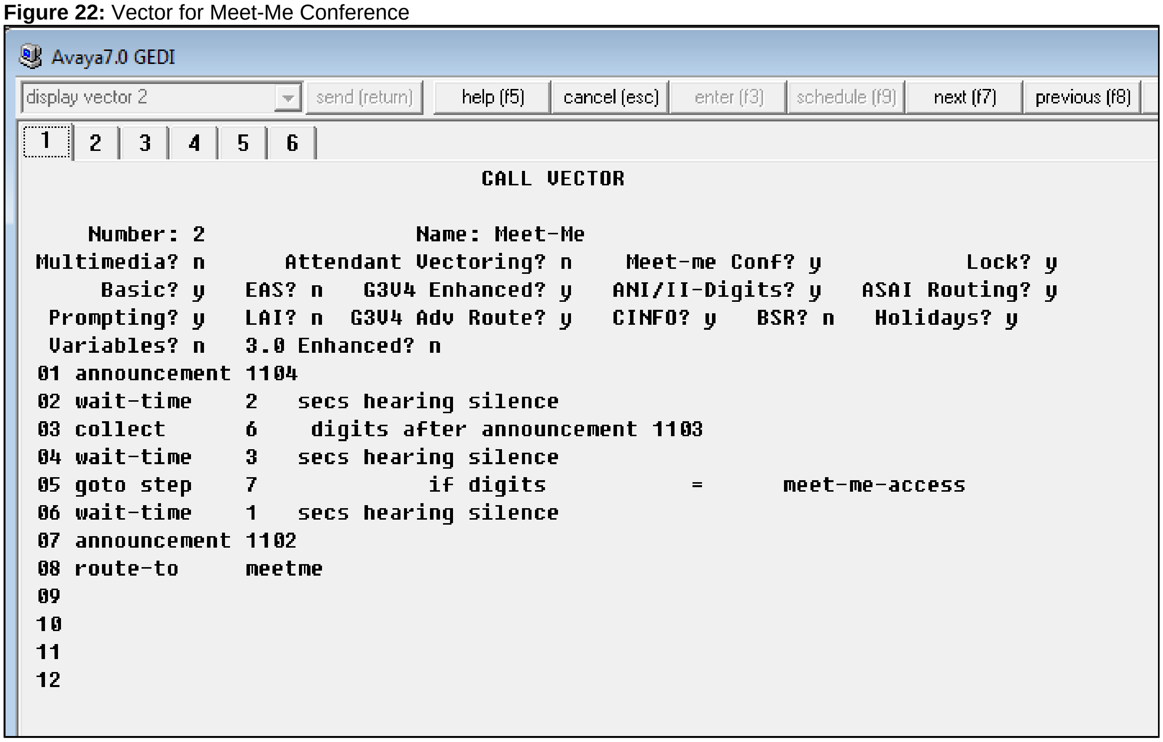


Figure 22: Vector for Meet-Me Conference



Ribbon SBC Edge Series Configuration

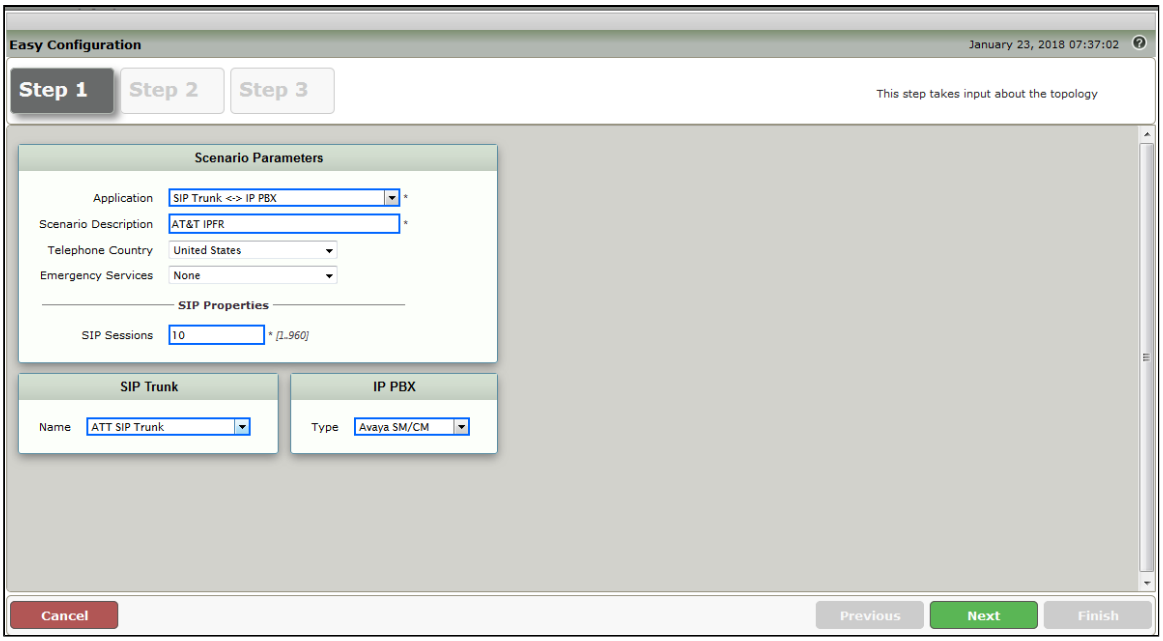
The following steps provide an example of how to configure Ribbon SBC Edge.

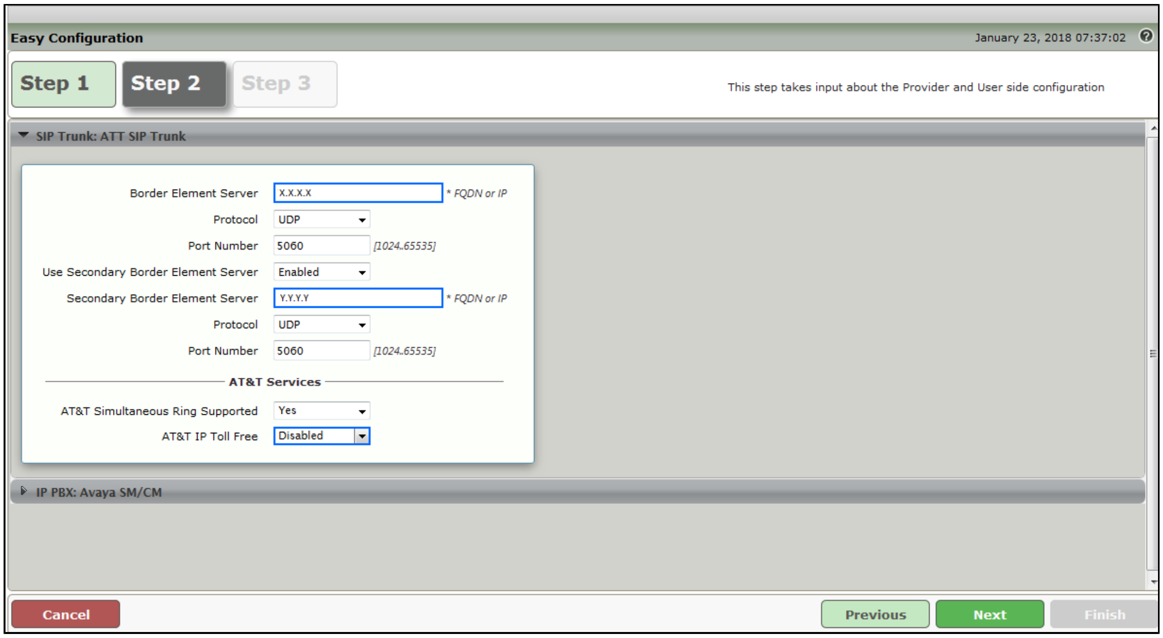
1. Easy Configuration Wizard
2. SIP Profile
3. Q.850 Cause Code to SIP Override Tables
4. Tone Table
5. Media Profile

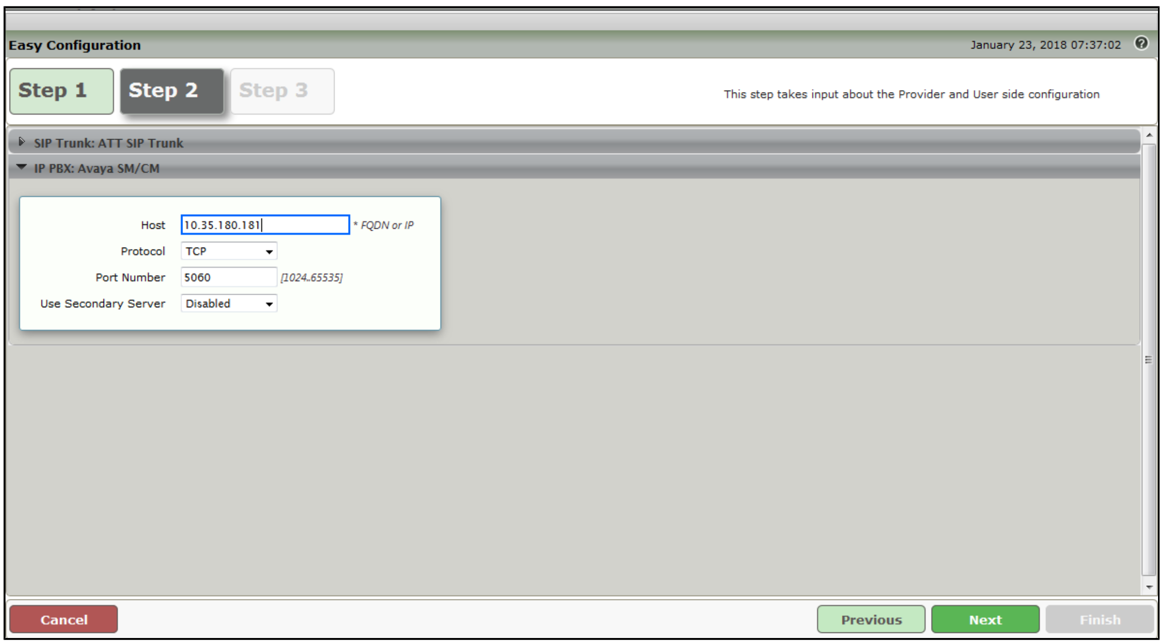
1. Easy Configuration Wizard

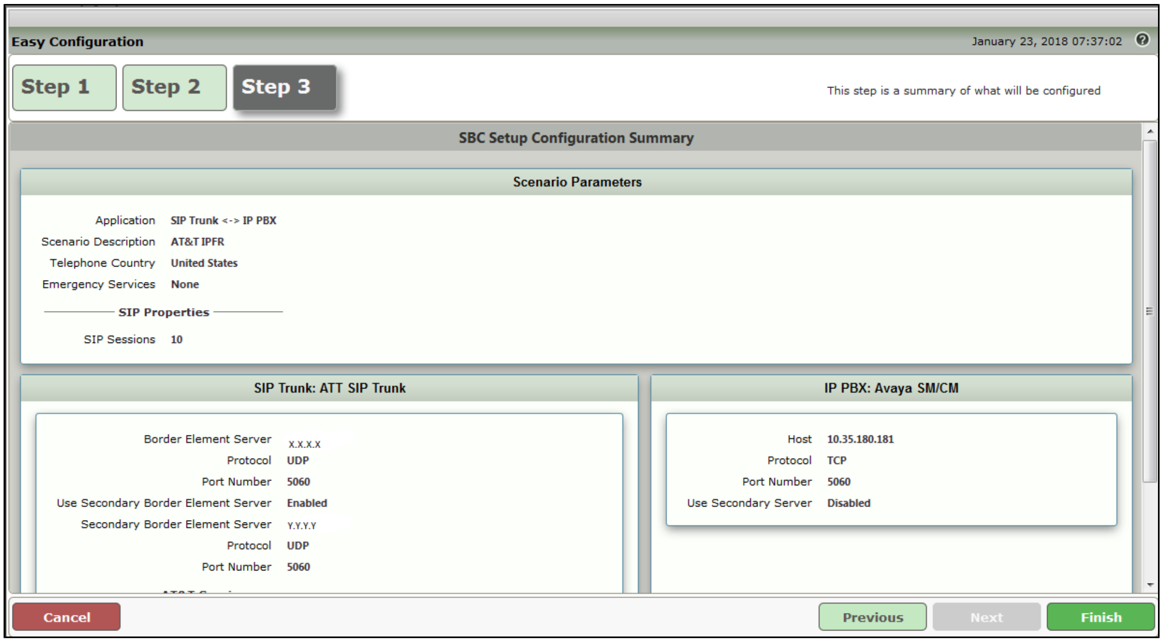
The SBCinterface includes an Easy Configuration Wizard, which enables end users to quickly deploy SBC. Based on a template, you can configure items such as endpoint (define user and provider), routing (routing configuration applied to scenario), and country (tone table parameters and emergency numbers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

Figure 23: Easy Configuration Wizard




Easy Configuration
January 23, 2018 07:37:02
Step 1
Step 2
Step 3
This step takes input about the Provider and User side configuration
SIP Trunk: ATT SIP Trunk
Border Element Server
X.X.X.X
* FQDN or IP
Protocol
UDP
Port Number
5060
[1024..65535]
Use Secondary Border Element Server
Enabled
Secondary Border Element Server
Y.Y.Y.Y
* FQDN or IP
Protocol
UDP
Port Number
5060
[1024..65535]
AT&T Services
AT&T Simultaneous Ring Supported
Yes
AT&T IP Toll Free
Disabled
IP PBX: Avaya SM/CM
Cancel
Previous
Next
Finish


Easy Configuration
January 23, 2018 07:37:02
Step 1
Step 2
Step 3
This step takes input about the Provider and User side configuration
SIP Trunk: ATT SIP Trunk
IP PBX: Avaya SM/CM
Host
10.35.180.181
* FQDN or IP
Protocol
TCP
Port Number
5060
[1024..65535]
Use Secondary Server
Disabled
Cancel
Previous
Next
Finish



2.SIP Profile

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

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

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

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

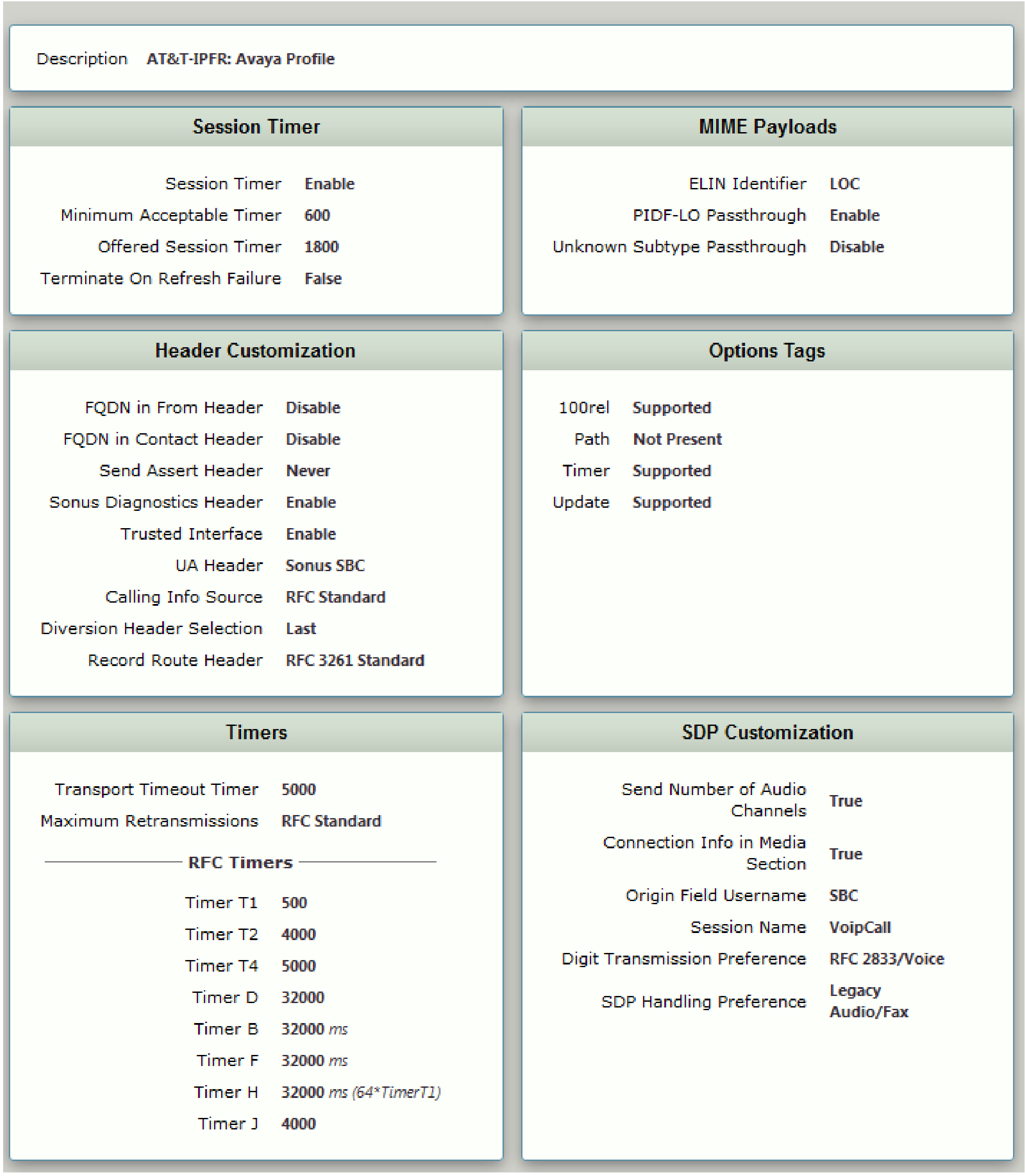
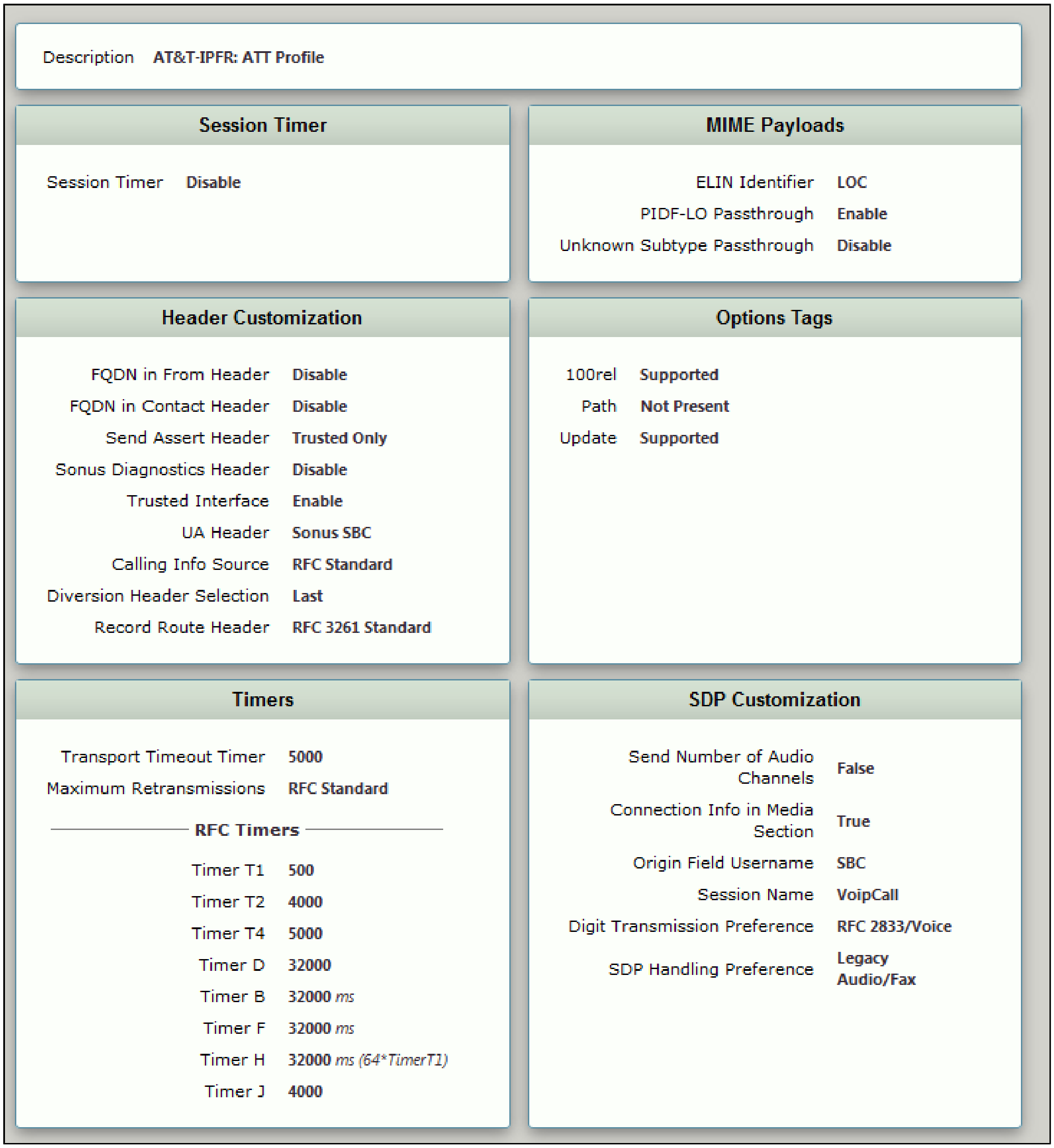


Figure 25: AT&T-IPFR: ATT SIP Profile



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

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

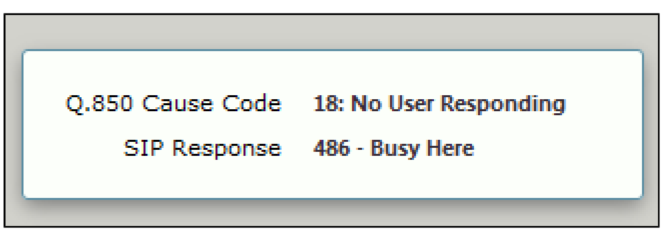
Send Number of Audio Channels False
Connection Info in Media Section True
Origin Field Username SBC
Session Name VoipCall
Digit Transmission Preference RFC 2833/Voice
SDP Handling Preference Legacy Audio/Fax

3. Q.850 Cause Code to SIP Override Table

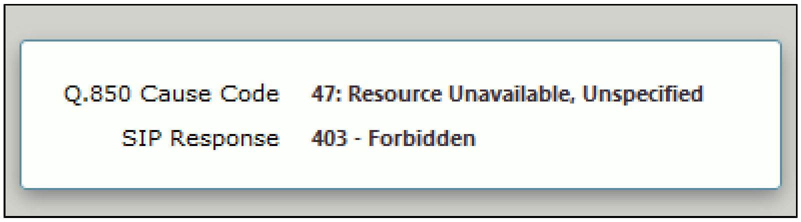
By default, the SBC Edge uses RFC 3398 cause code mappings.Q.850 Cause Code to SIP Override Table allows you to define other mappings for cause codes.

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

Figure 26: 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 Edgeadministrator to customize the tones a user hears when placing a call. You can modify the tone to match your local PSTN or PBX. The default tone table configures the following categories with the United States' values:

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

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

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

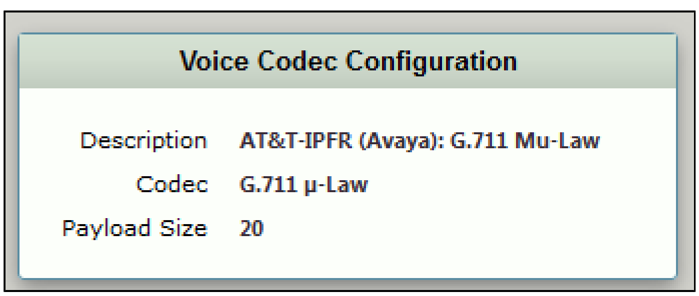
Tone Type	Frequency 1 (Hz)	Amplitude 1 (dBm)	Frequency 2 (Hz)	Amplitude 2 (dBm)
Ringback	440	-19	480	-19
Dial	350	-13	440	-13
Busy	480	-24	620	-24
Congestion	480	-24	620	-24
Call Waiting	440	-13	0	0
Disconnect	480	-24	620	-24
Confirmation	350	-13	440	-13

5. Media Profile

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

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

Figure 28: AT&T-IPFR (Avaya)



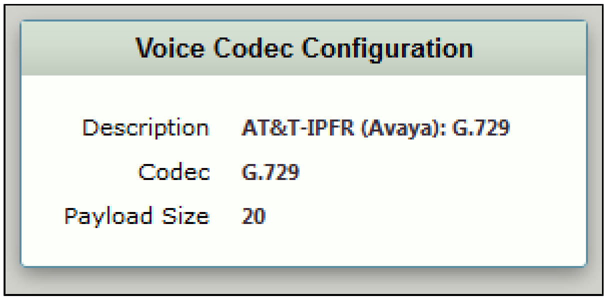
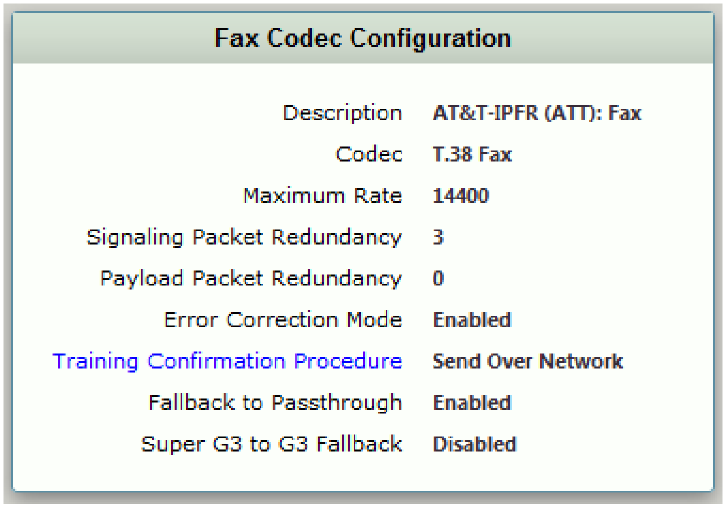
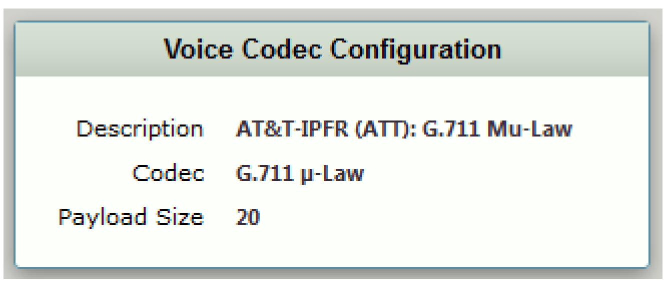
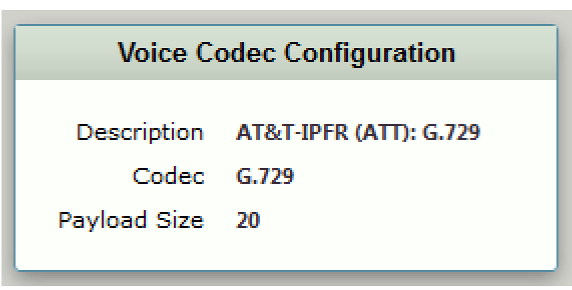


Figure 29: AT&T-IPFR (ATT)





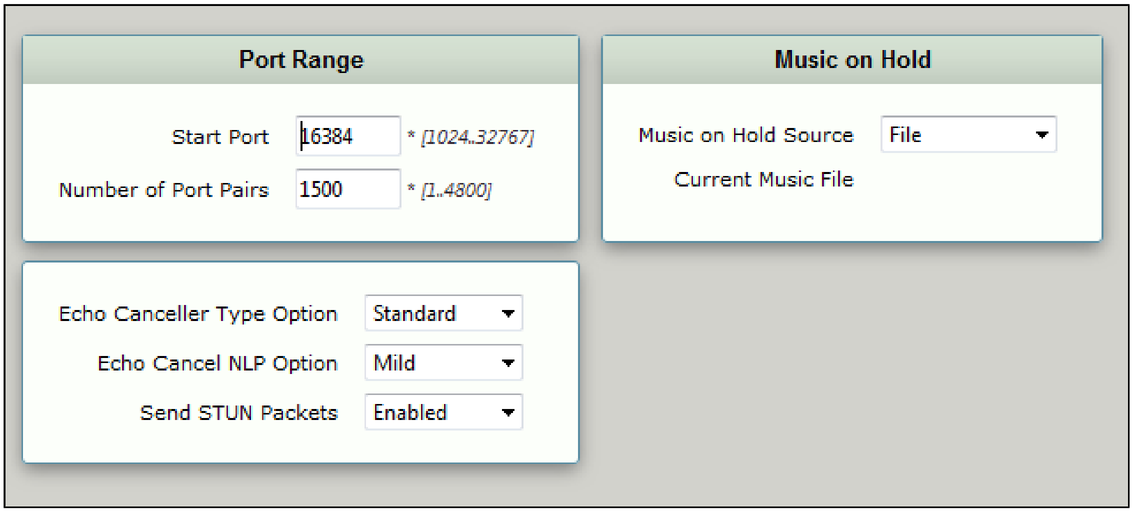


6. Media System Configuration

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

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

Figure 30: Media System Configuration



7. Media List

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

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

Figure 31: AT&T-IPFR: Avaya List

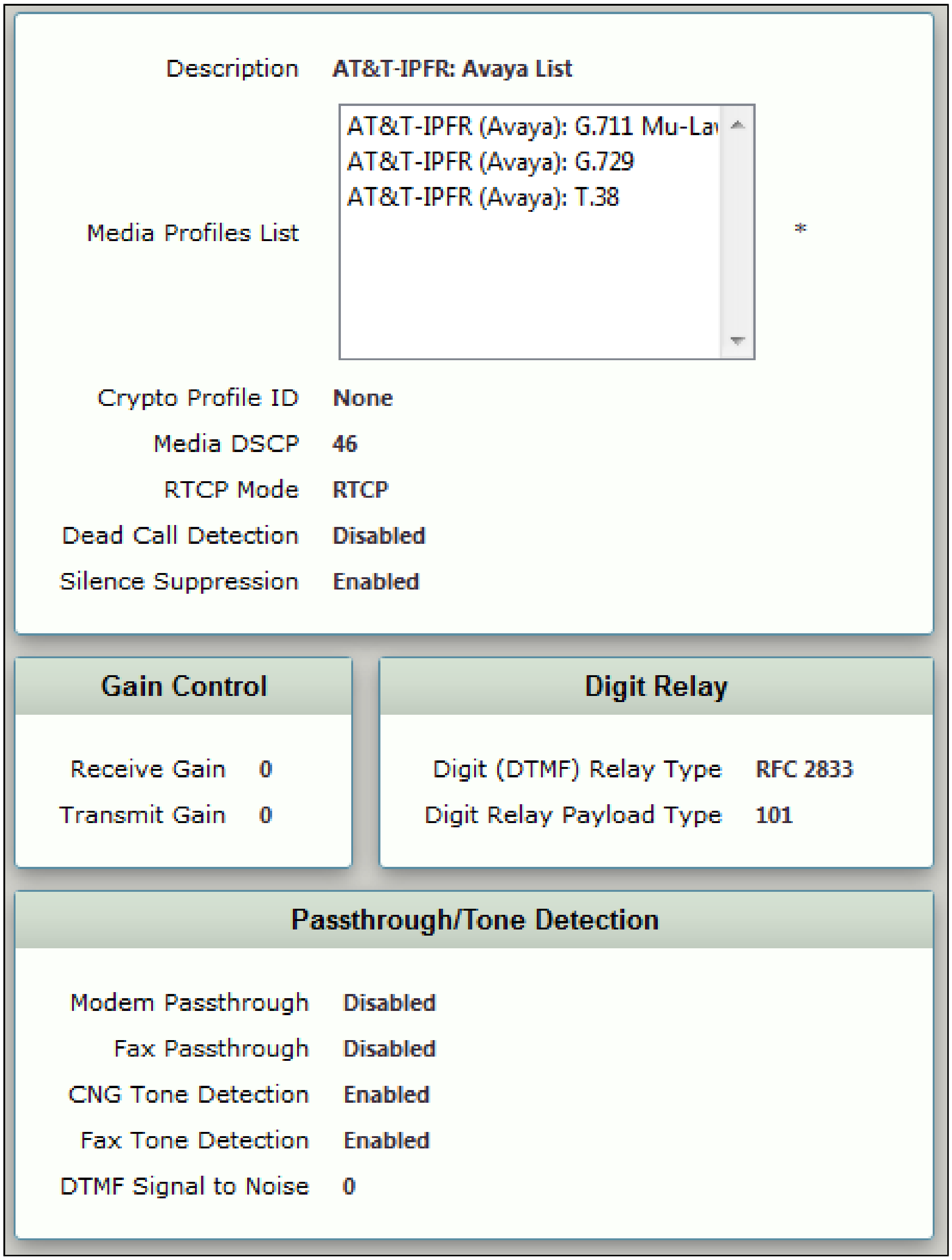
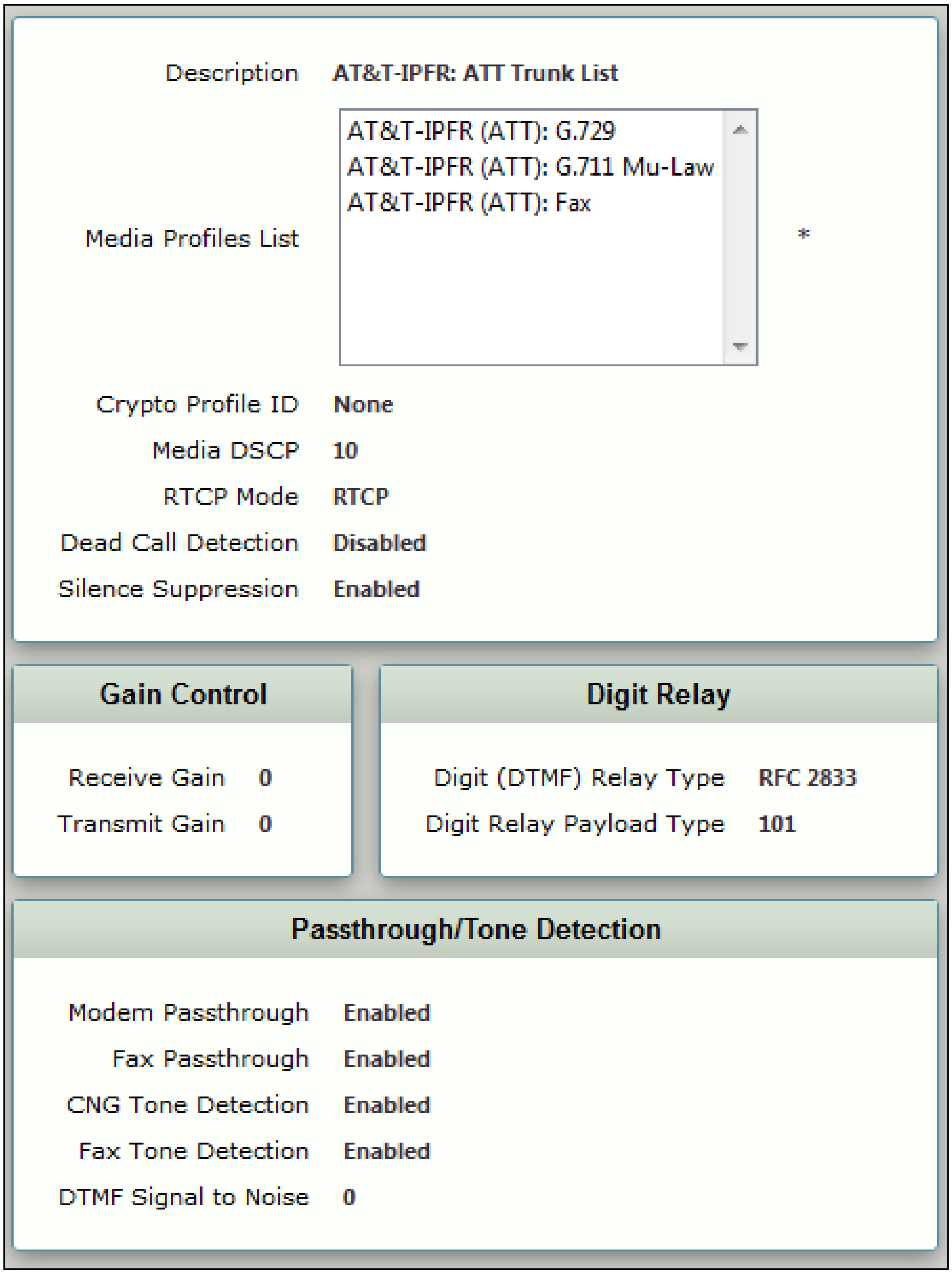


Figure 32: AT&T-IPFR: ATT Trunk List



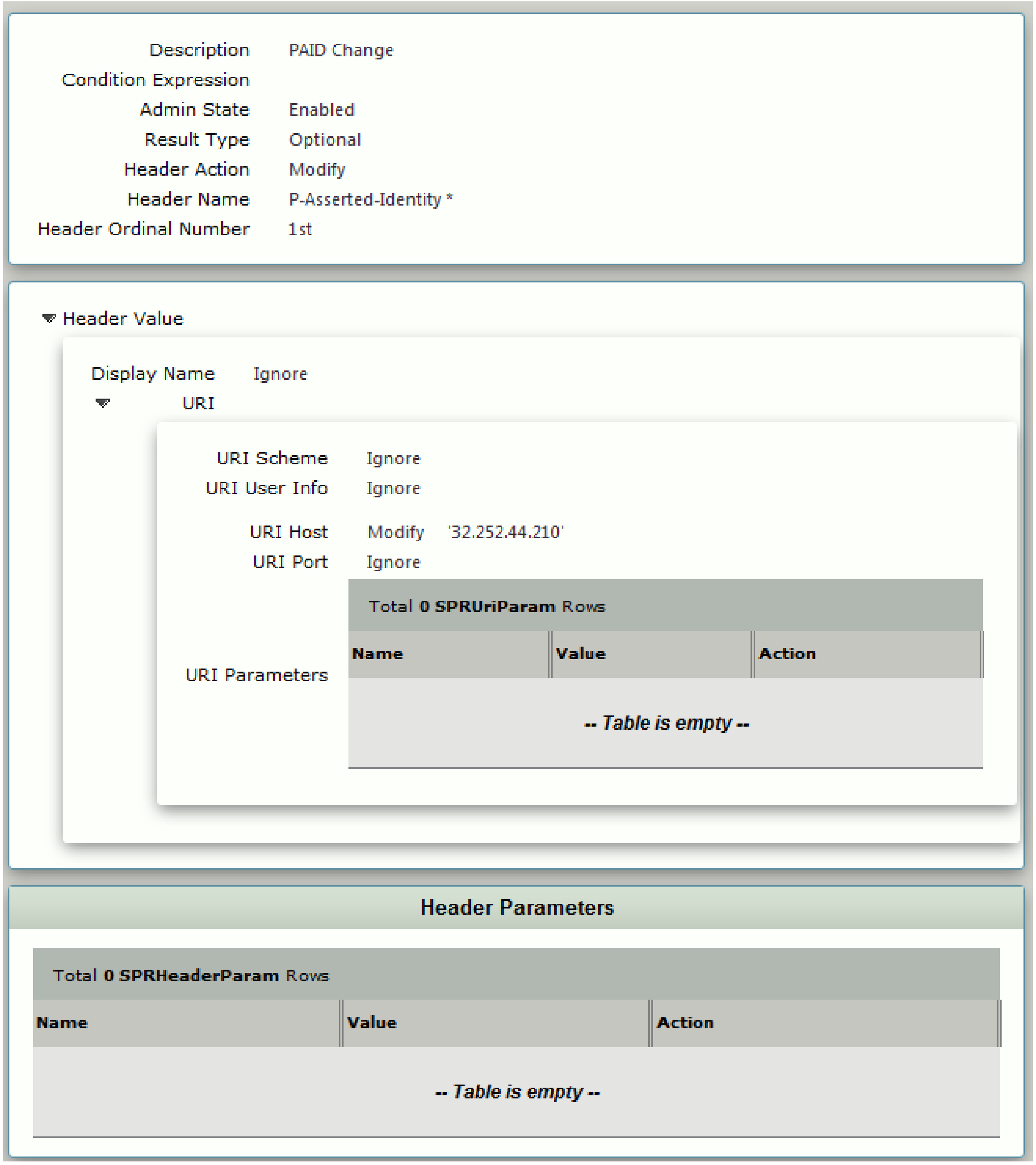
8. Message Manipulation

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

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

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

Figure 33: SMM TO ATT



9. SIP Server

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

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

Figure 34: AT&T-IPFR: Avaya CM

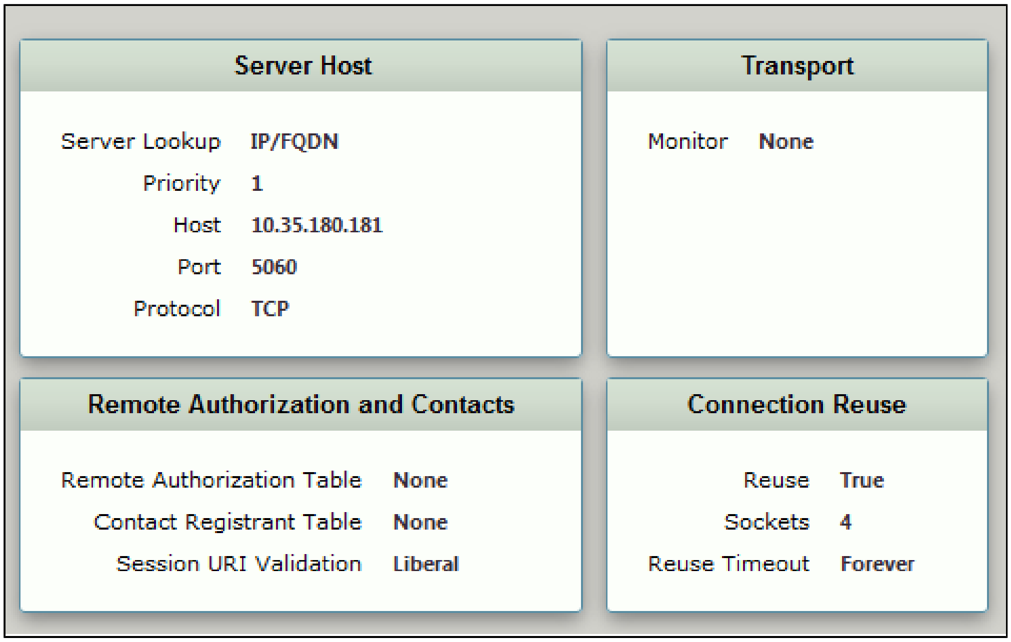


Figure 35: AT&T-IPFR: Border Element

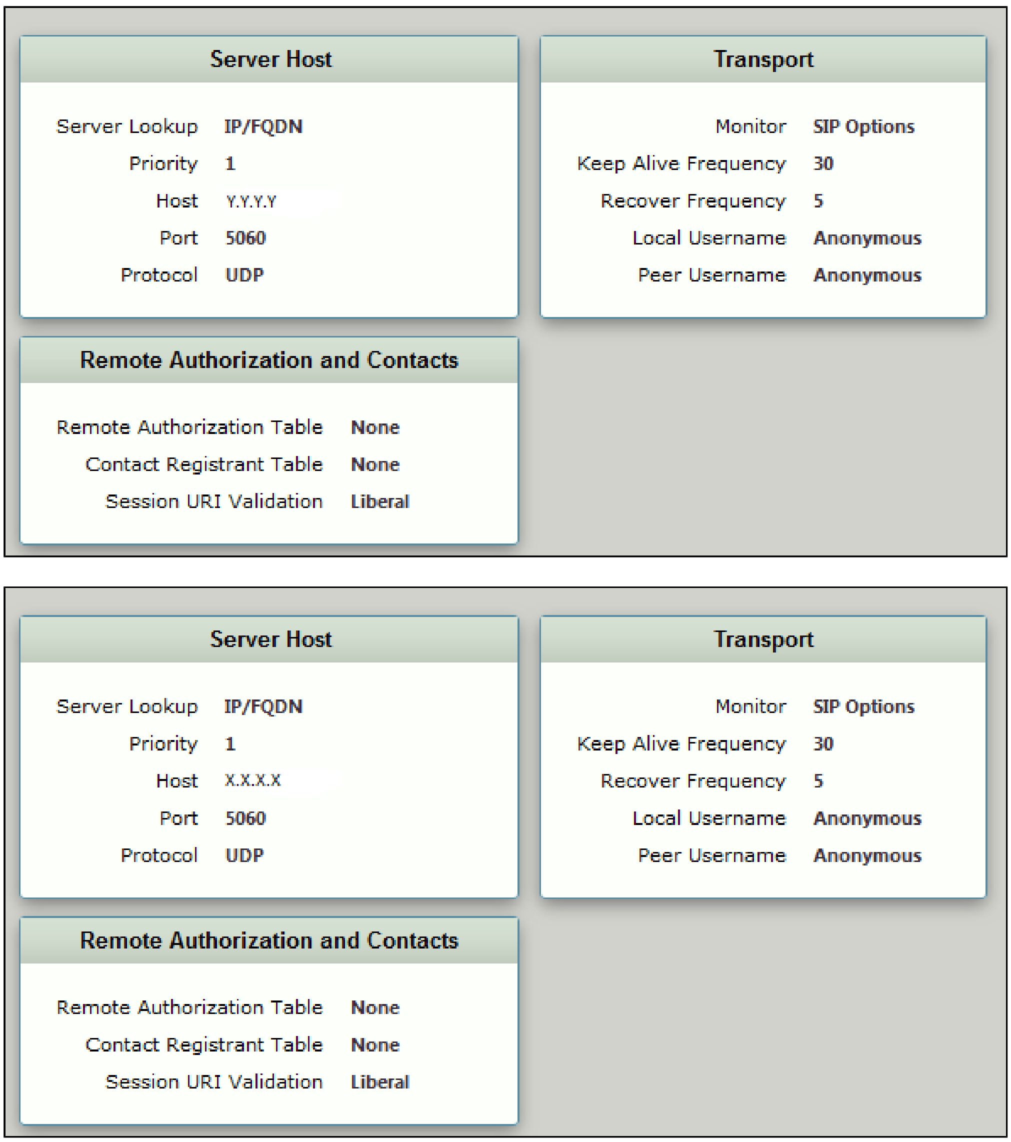
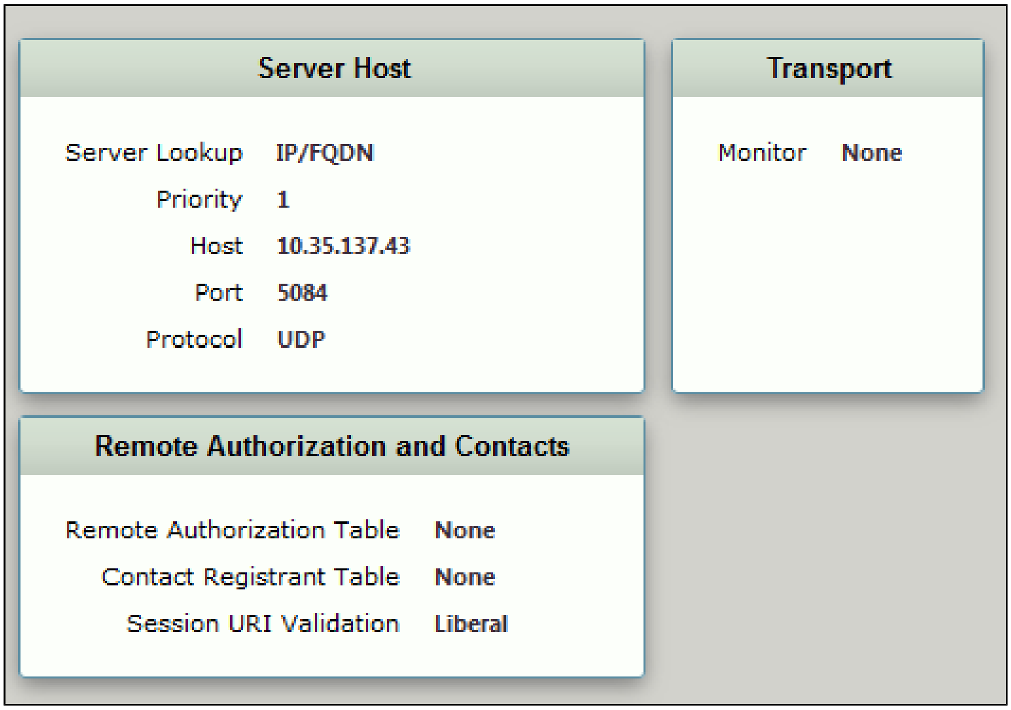


Figure 36: Fax-TenorGW



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 37: AT&T-IPFR: Avaya CM

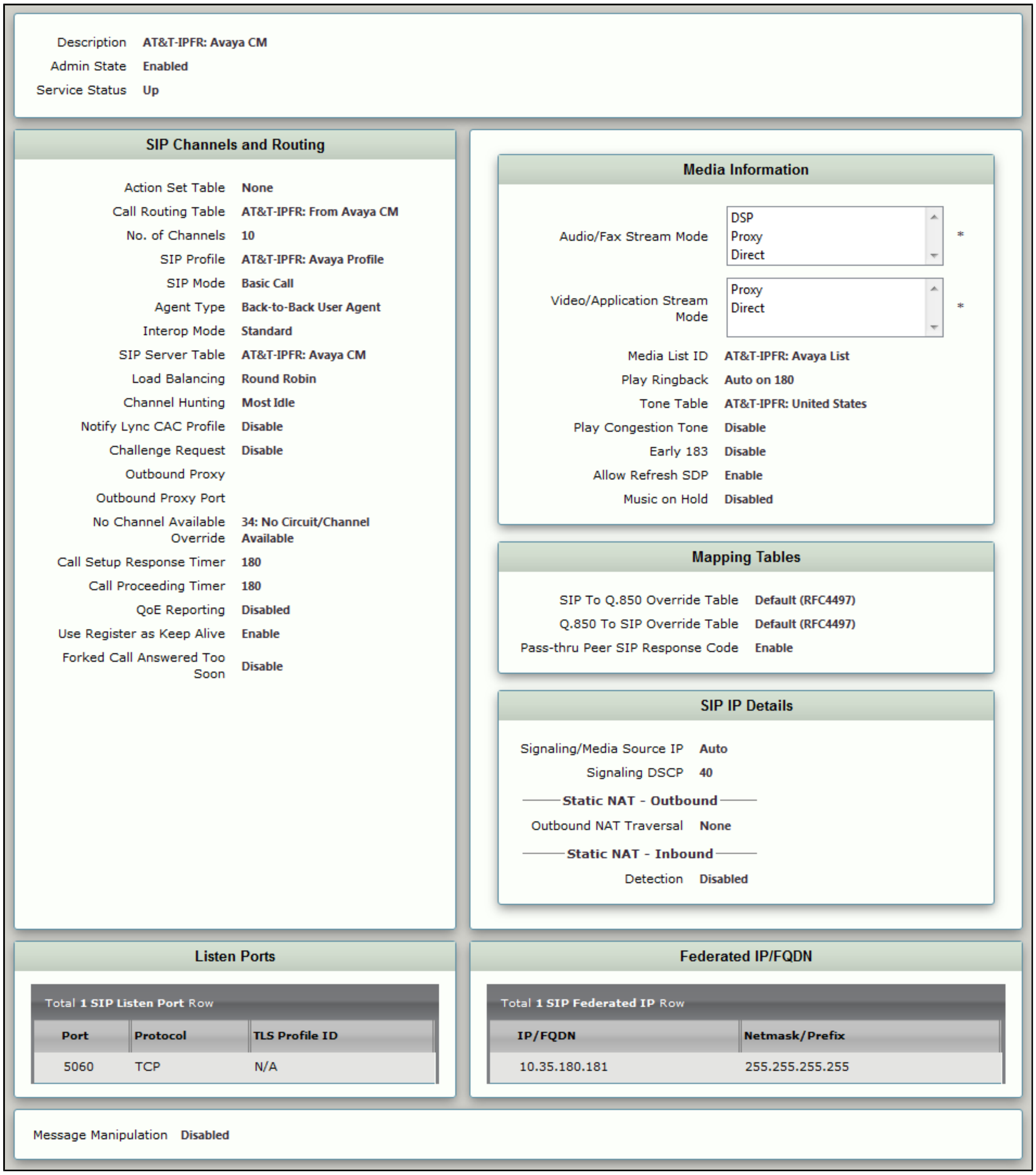


Figure 38: AT&T-IPFR: ATT Border Element

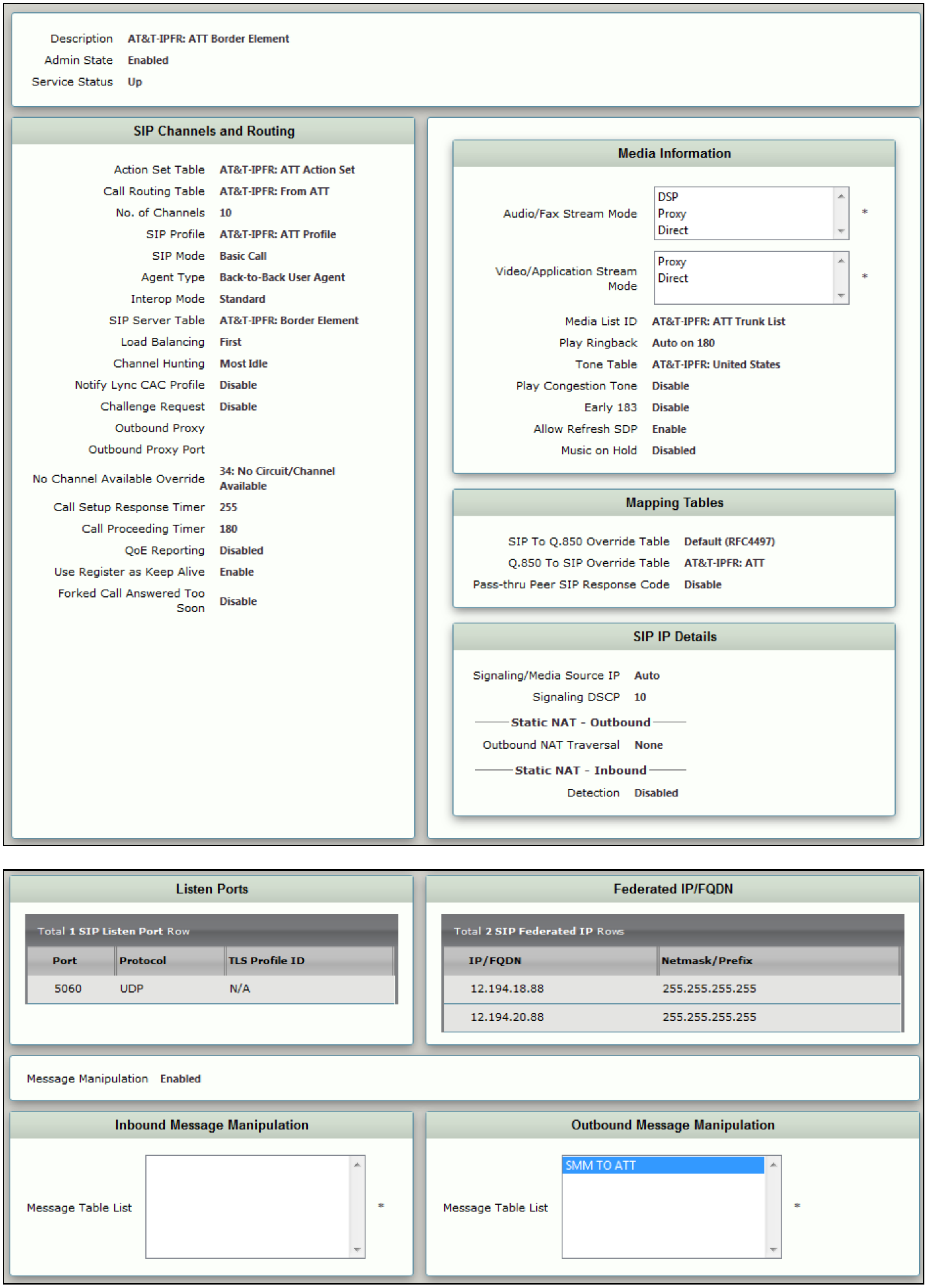
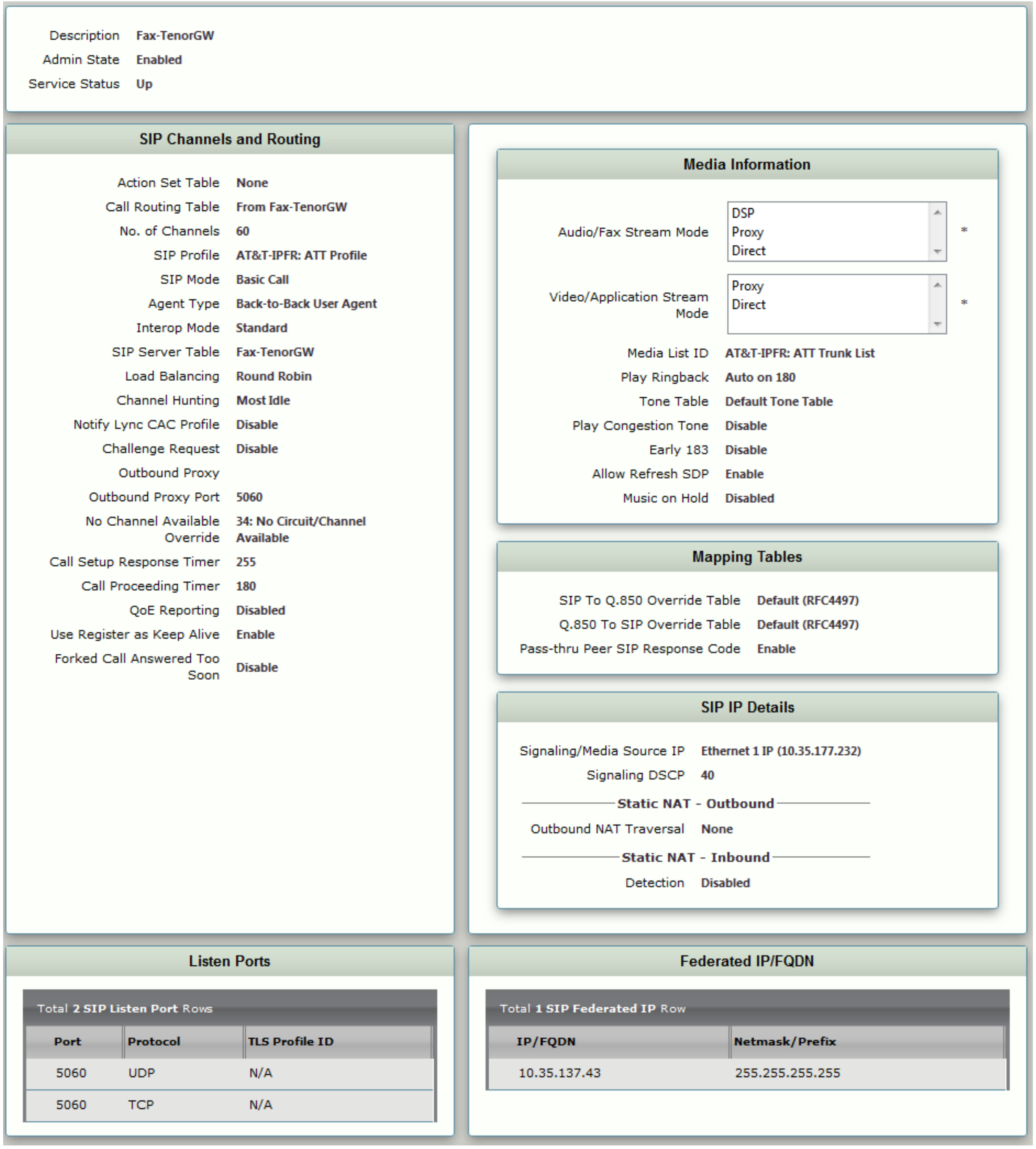


Figure 39: Fax-TenorGW



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 setscan reference.

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

Figure 40: AT&T-IPFR: From ATT

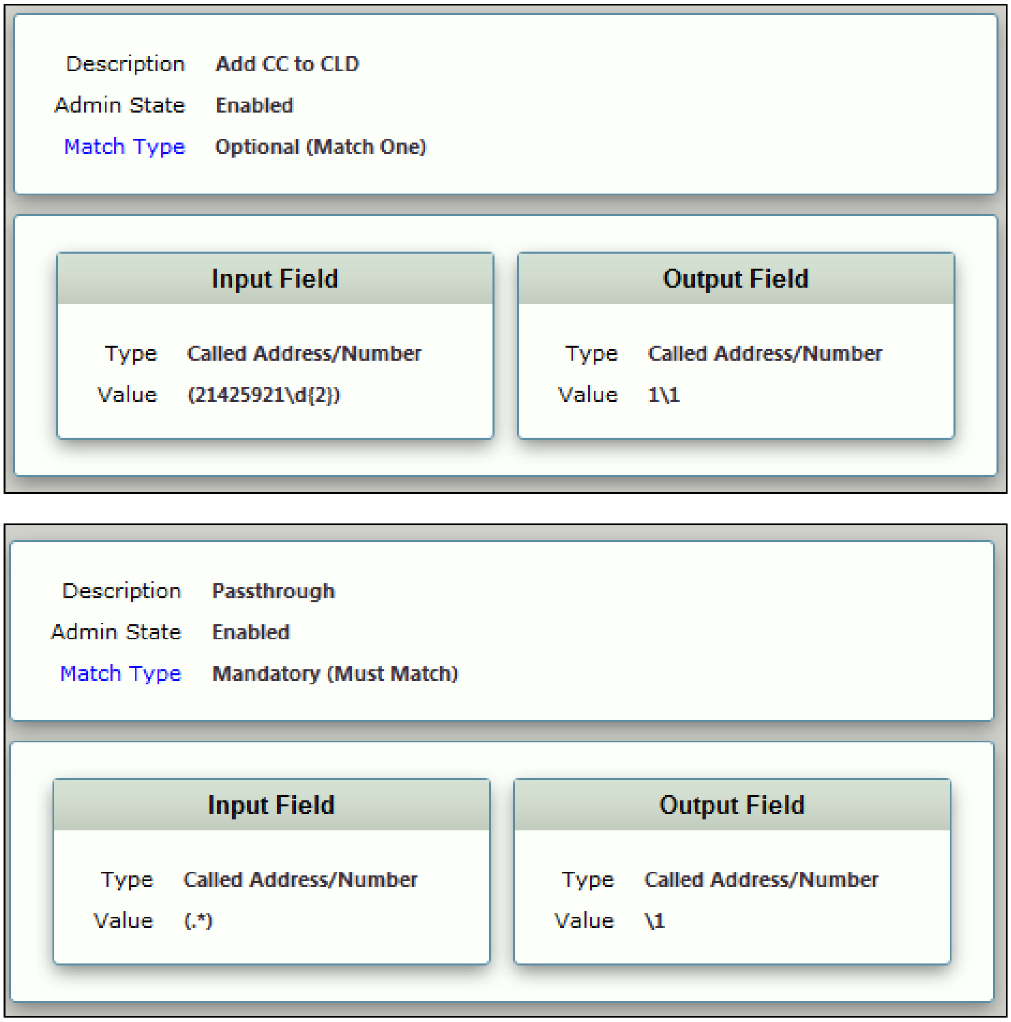
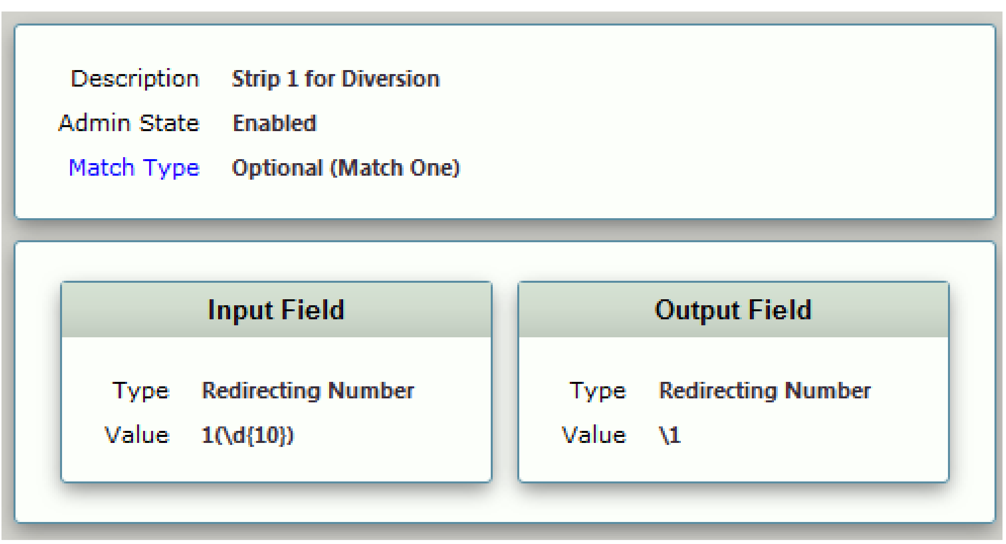
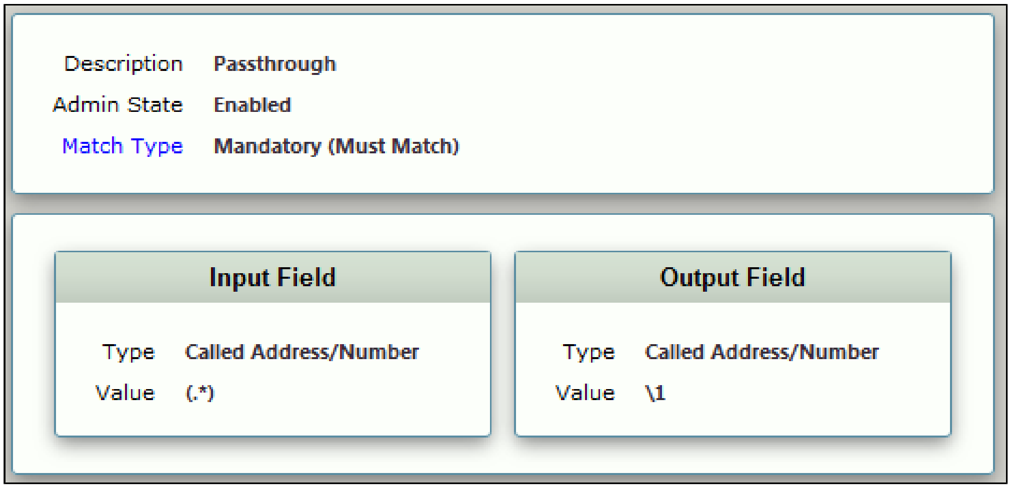


Figure 41: AT&T-IPFR: From Avaya CM





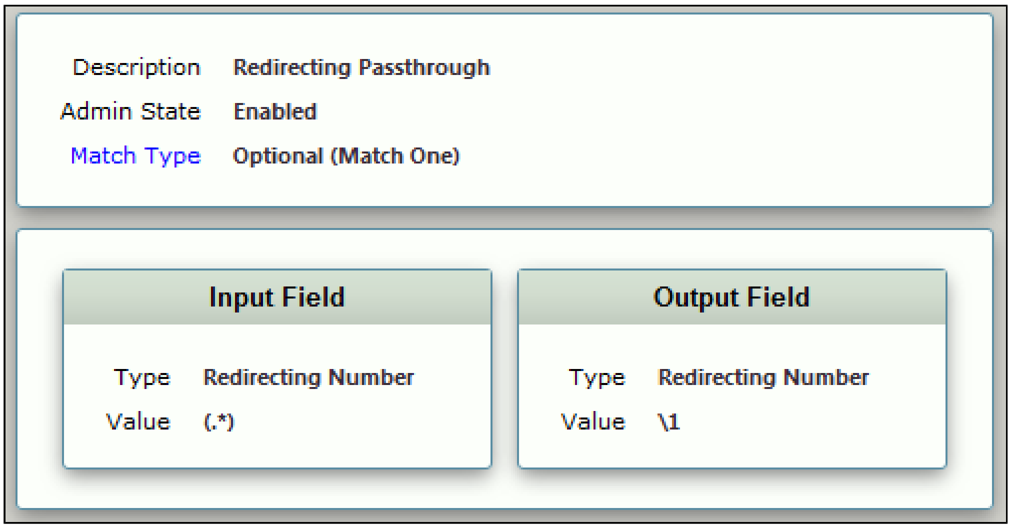


Figure 42: From Fax-TenorGW

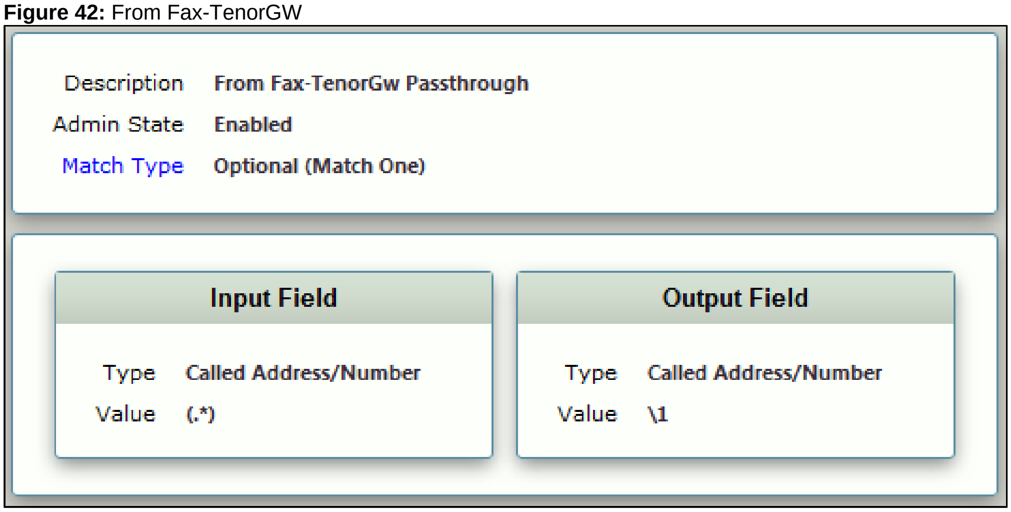
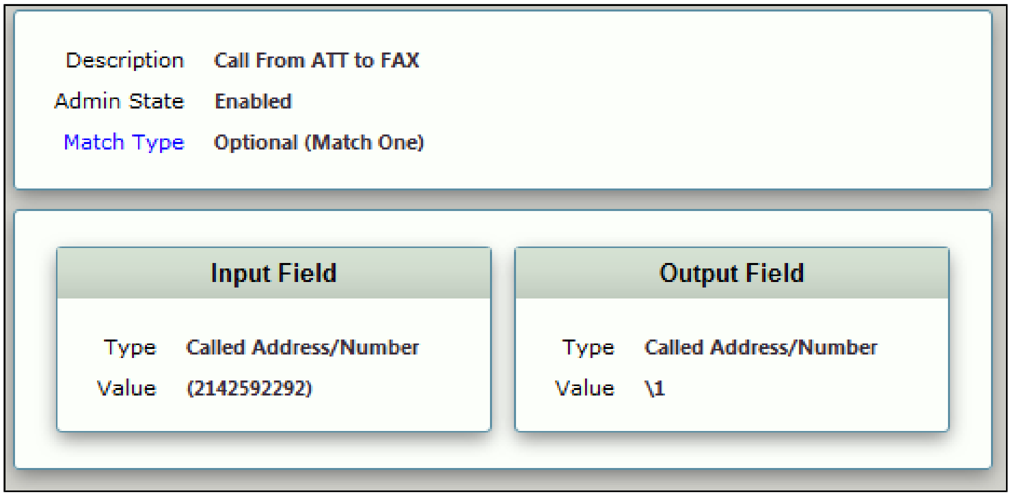


Figure 43: From ATT to FAX



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 44: AT&T-IPFR: From Avaya CM

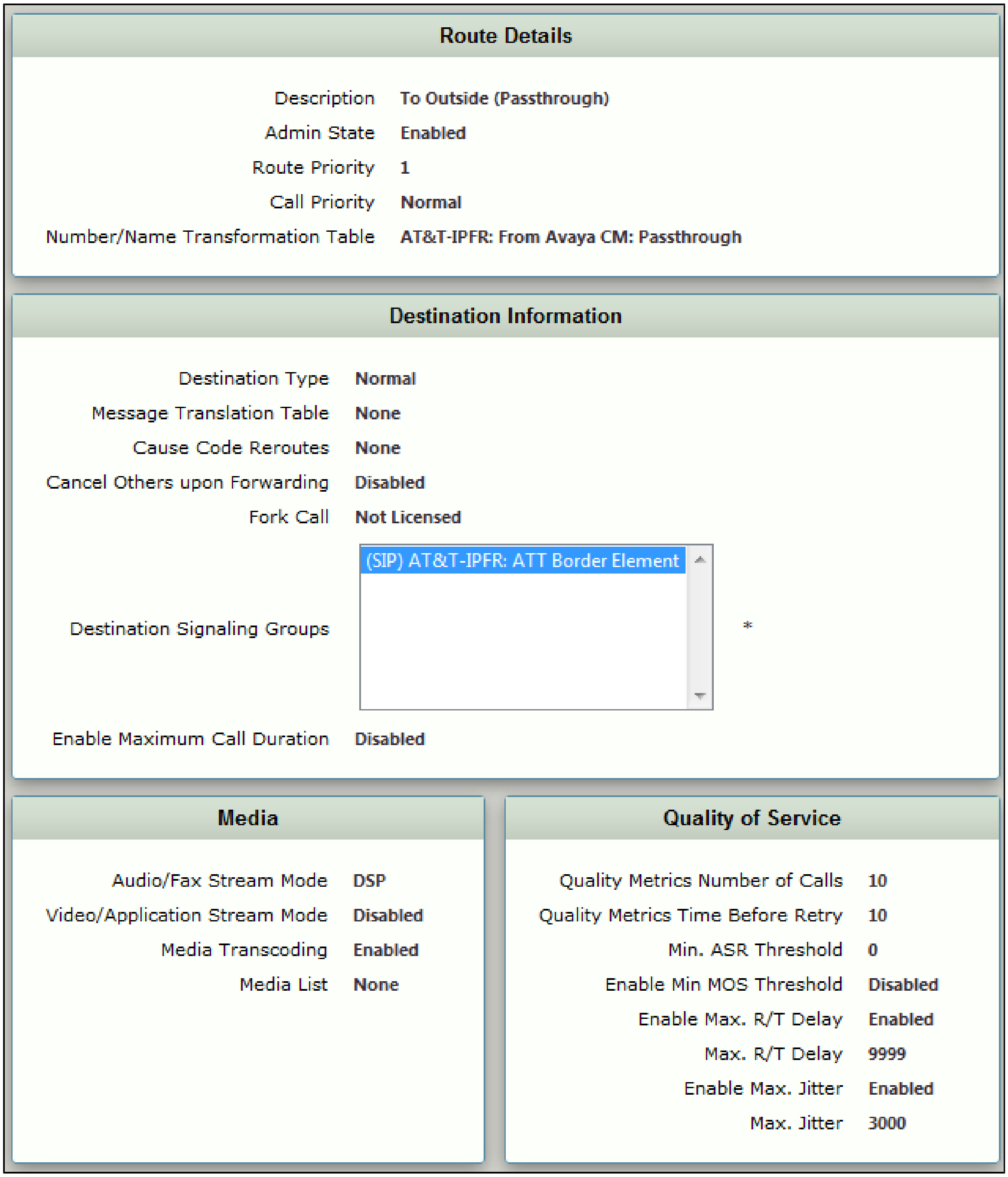


Figure 45: AT&T-IPFR: From ATT

Route Details

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

Destination Information

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

Media

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

Quality of Service

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

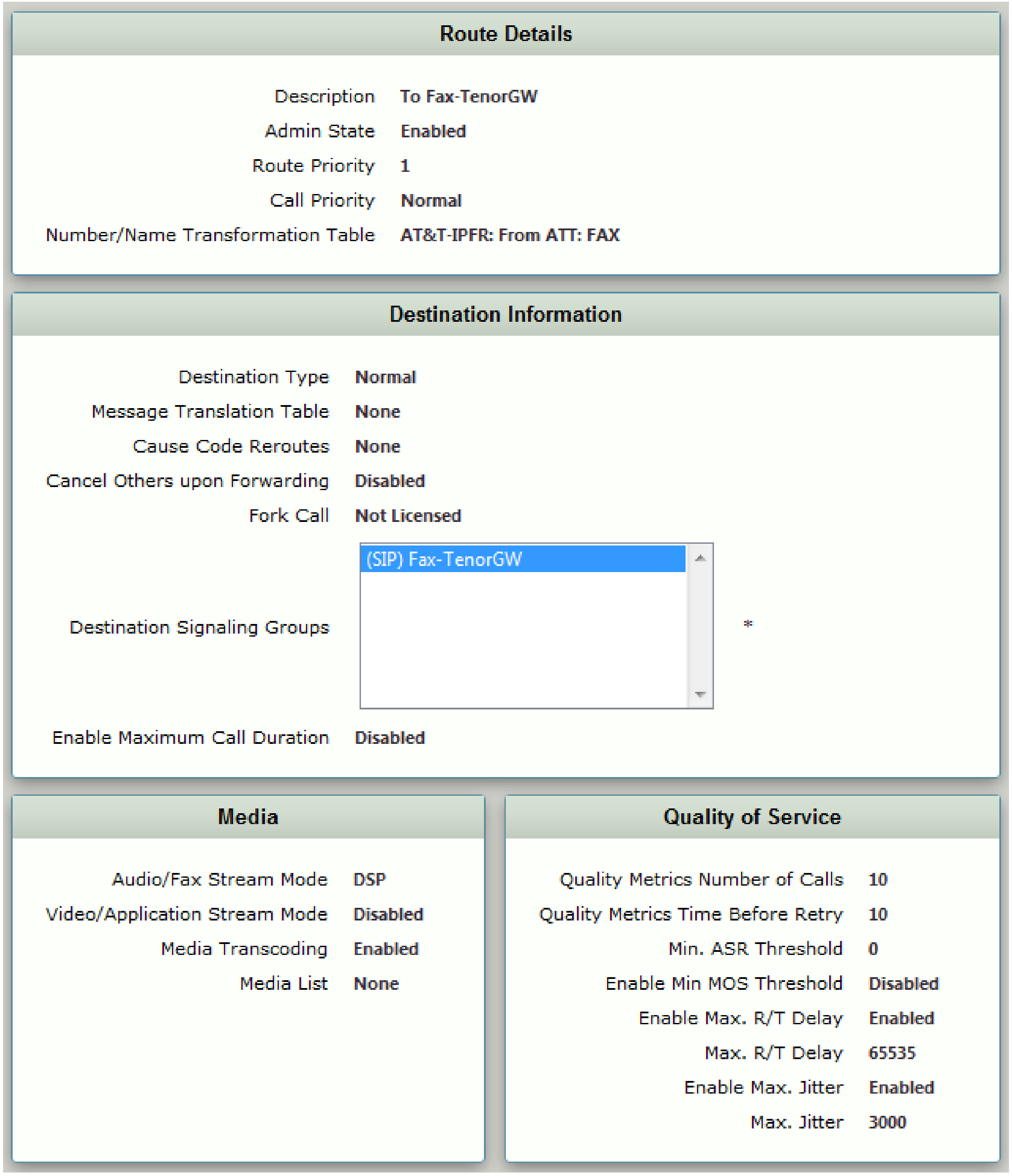
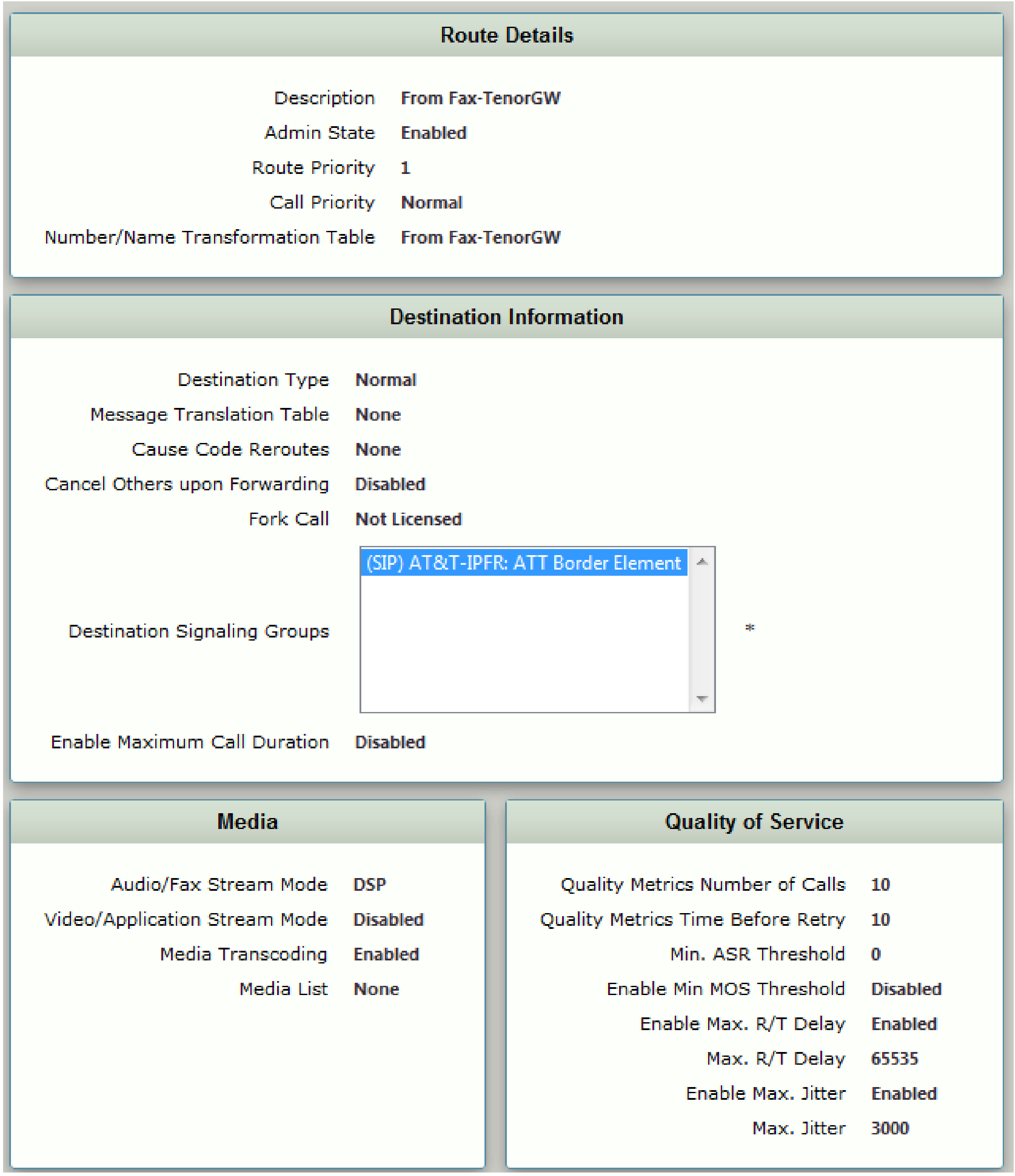


Figure 46: From Fax-TenorGW



Conclusion

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