Ribbon EdgeMarc 6000 ISDN Configuration to Avaya Communication Manager 7.1

Table of Contents

Document Overview

This document provides a configuration guide for Ribbon EdgeMarc 6000 when connecting to the Avaya Communication Manager (CM) 7.1.

This configuration guide supports features in the Virgin Media SIP Trunk Application.

- For additional information on Avaya CM 7.1, refer to https://support.avaya.com/products/P0001/avaya-aura-communication-manager/.
- For additional information on the Ribbon SBC, refer to https://ribboncommunications.com/.

Introduction

The interoperability compliance testing focuses on verifying inbound and outbound call flows between the Ribbon EdgeMarc 6000 and the Avaya CM 7.1 platform.

Audience

This is a technical document intended for telecommunications engineers for configuring both the Ribbon SBCs and the third-party product. Users will perform steps to navigate the third-party product as well as the Ribbon SBC Command Line Interface (CLI). Understanding the basic concepts of TCP /UDP/TLS, IP/Routing, and SIP/RTP is also necessary for completing the configuration and for troubleshooting, if necessary.

Note
This configuration guide is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided “AS IS”. Users must take full responsibility for the application of the specifications and information in this guide.

Requirements

The following equipment and software were used for the sample configuration:

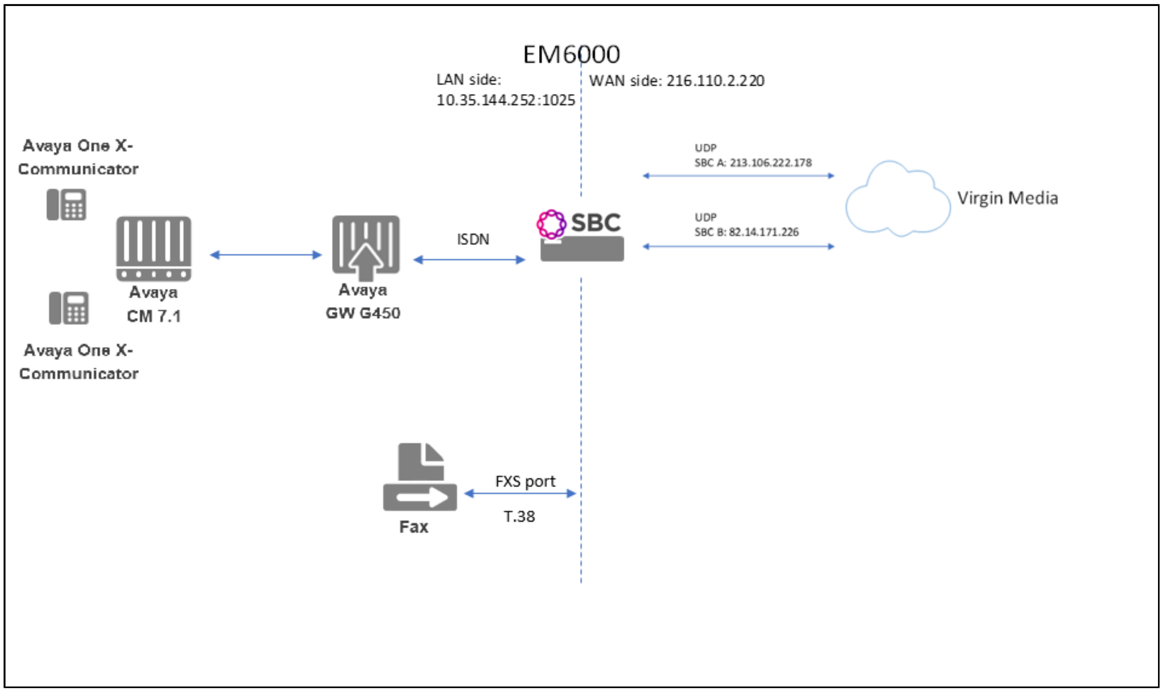
Table 1: Requirements

	Equipment	Software Version
Ribbon Communications	Ribbon EdgeMarc 6000	V16.0.0
Third-party Equipment	Avaya Communication Manager	7.1
	Avaya one-X Communicator	6.2
	NGT Lite	v.1.51

Reference Configuration

The following reference configuration shows the connectivity between the third-party and the Ribbon EdgeMarc 6000.

Figure 1: Reference Configuration



Support

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

Third-Party Product Features

Ribbon supports the following third-party product features:

- Basic originated and terminated calls
- Basic inbound and outbound calls
- Hold and Resume
- Call Forwarding
- DTMF
- Conference calls
- Action on eSBC outage (restart of eSBC)
- Action on Loss of Virgin Media primary SBC

Configure Avaya Communication Manager 7.1

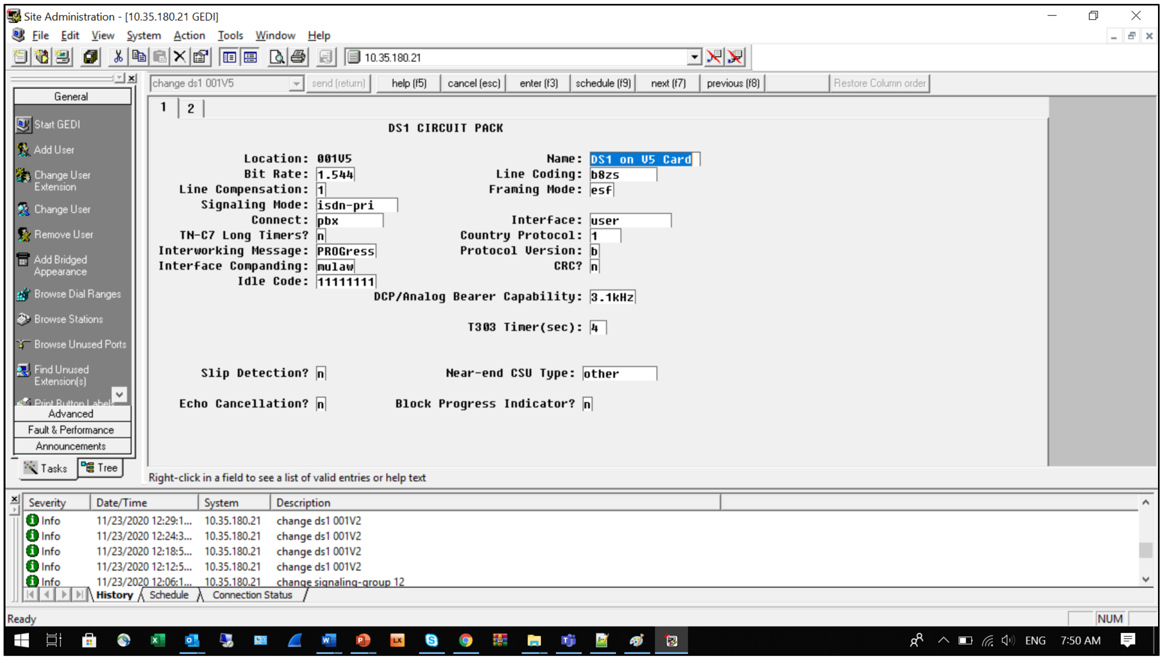
The following new configurations are included in this section:

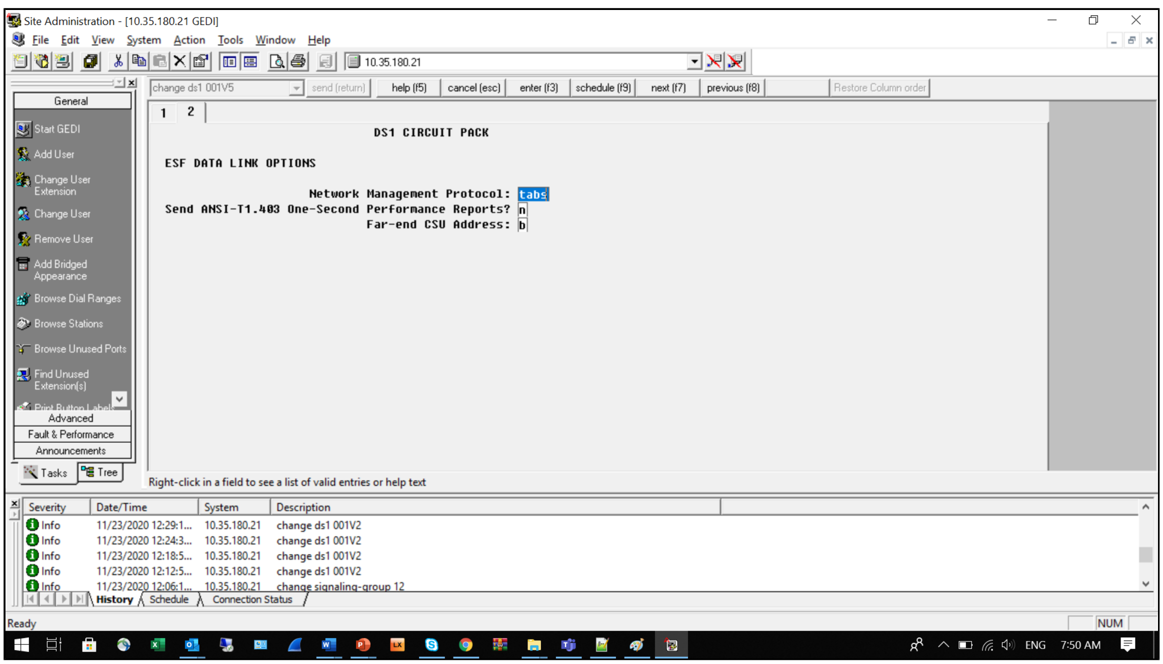
1. DS1 Circuit Pack
2. Signaling Group
3. Trunk Group
4. Route Pattern
5. ARS Digit Analysis Table
6. Station
7. Change public-unknown-numbering

1. DS1 Circuit Pack

1. Use the Site Administration to log into the Avaya CM 7.1.
2. Type **change ds1 001V5** in the command line.
3. Configure the Signaling Mode field with **isdn-pri**.
4. Configure the Connect field with **pbx**.
5. Configure the Interface field with **user**.

Figure 2: DS1 Circuit Pack





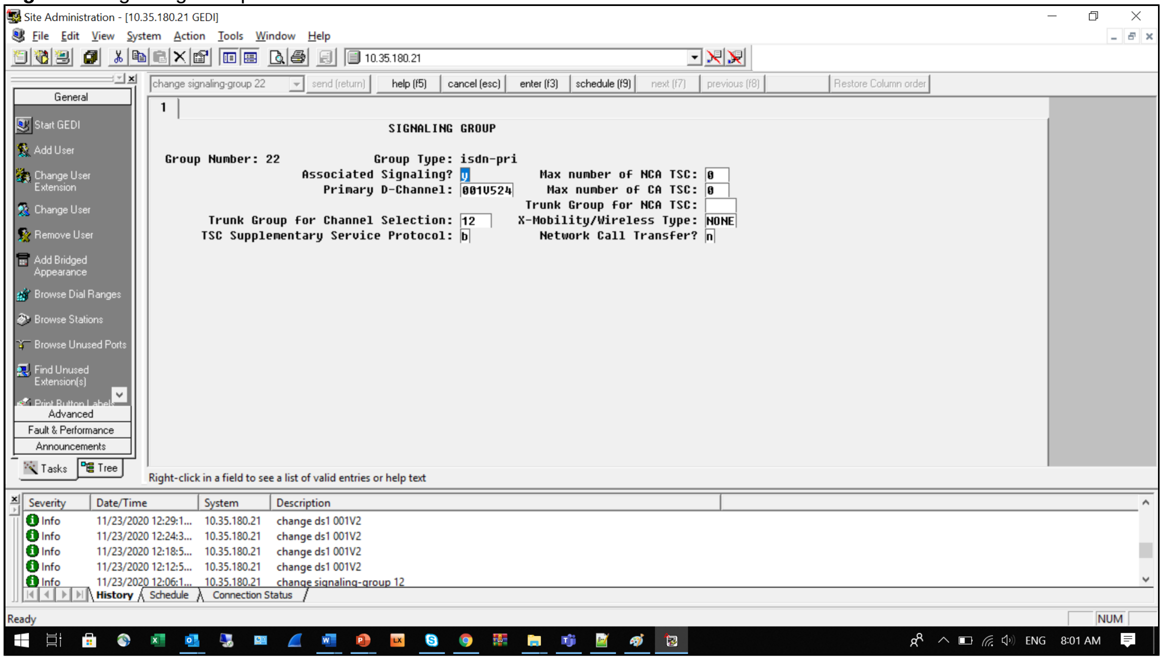
2. Signaling Group

1. Use the Site Administration to log into the Avaya CM 7.1.
2. Type **add signaling-group next** in the command line.

> **Note**
> The next switch auto-generates the next available group number for the Signaling Group, which is the most efficient method for creating a new Signaling Group.

3. Confirm the next available Signaling Group information and press **F3** to save the changes.

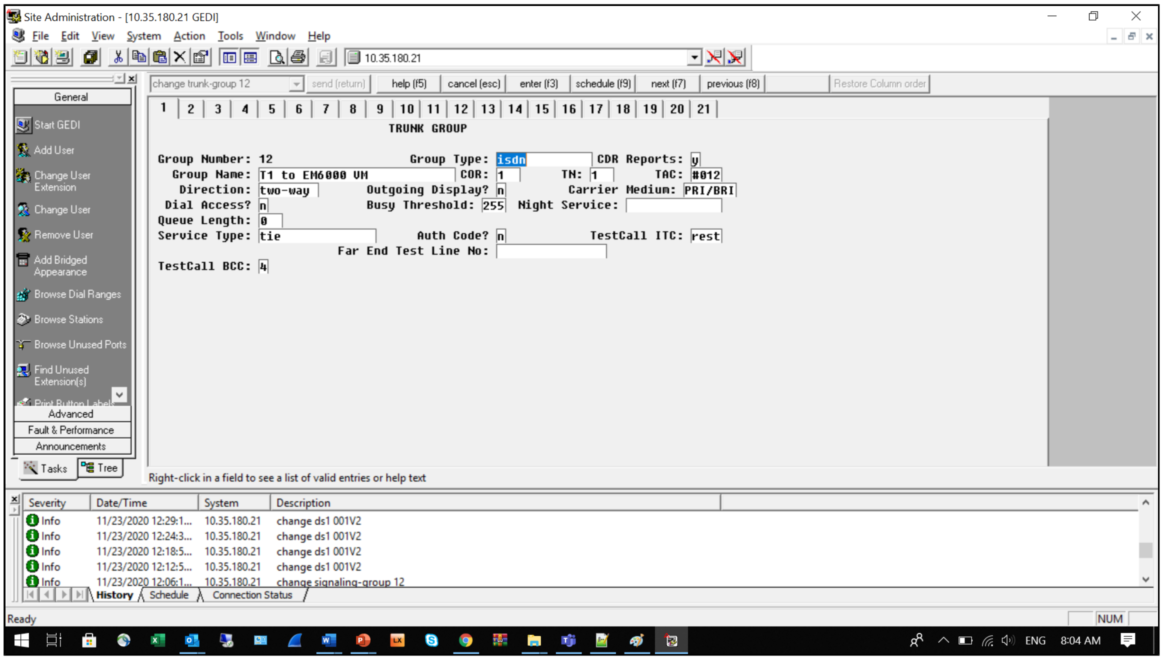
Figure 3: Signaling Group

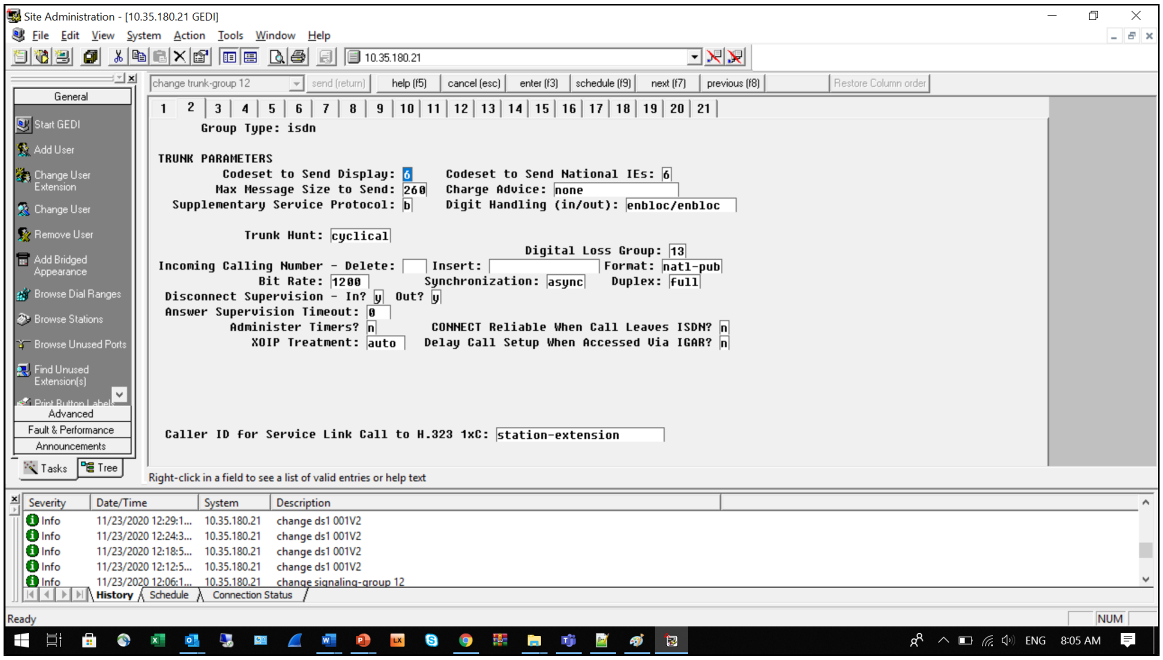


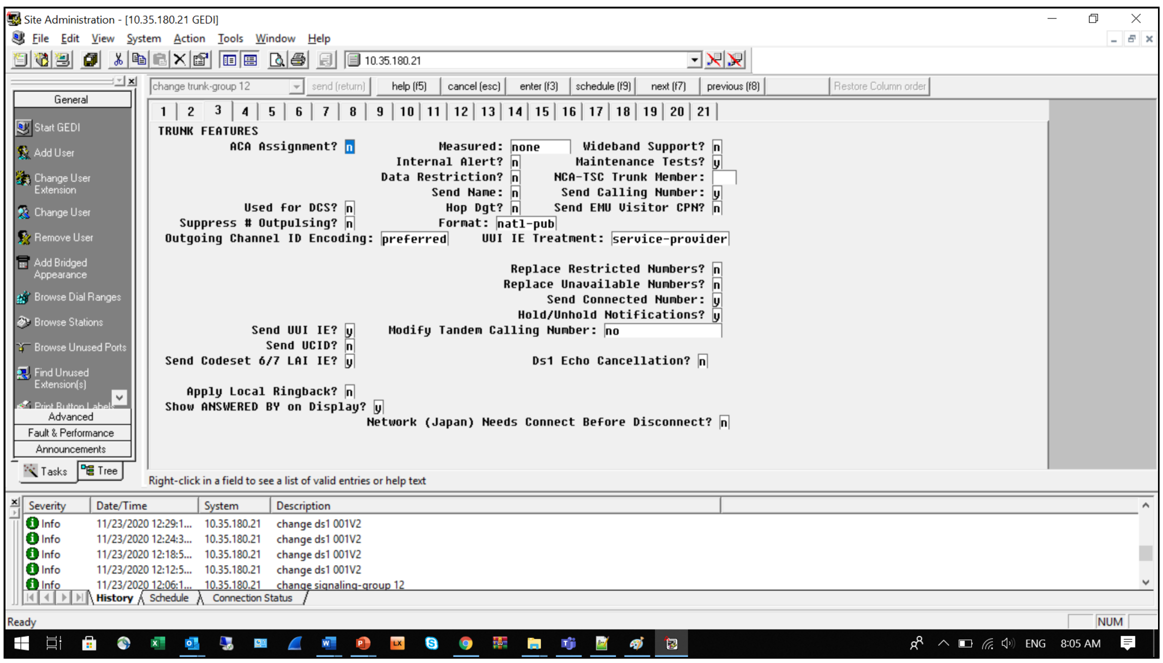
3. Trunk Group

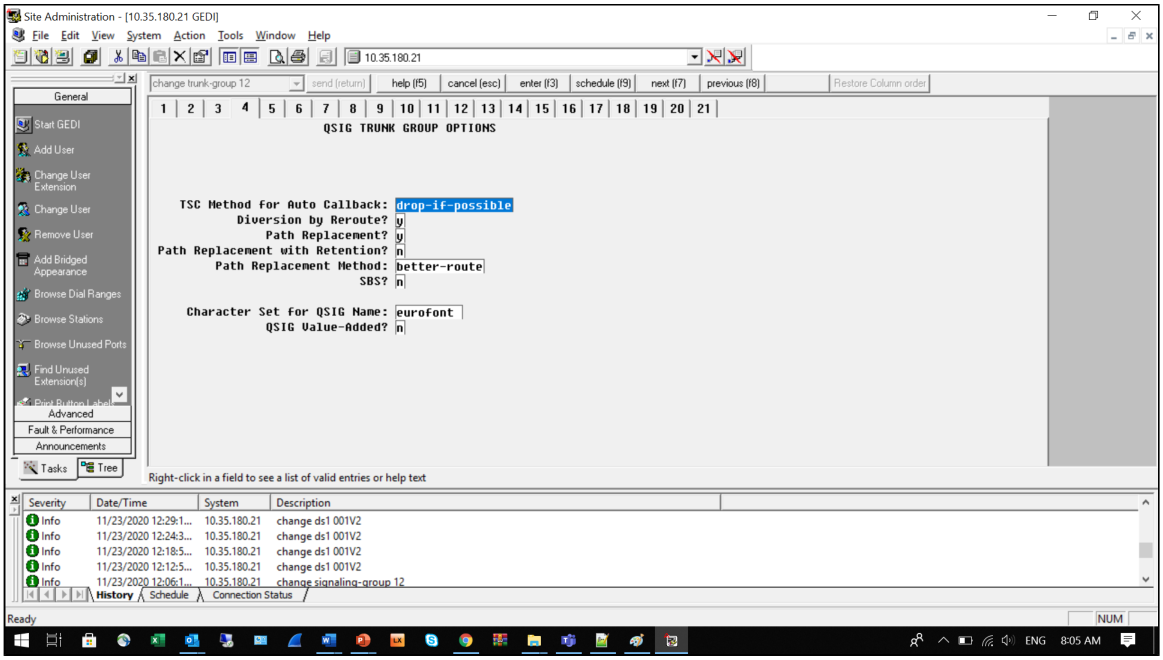
1. Use the Site Administration to log into the Avaya CM 7.1.
2. Type **add trunk-group next** in the command line.
3. Enter the trunk group information and press F3 to save the changes.

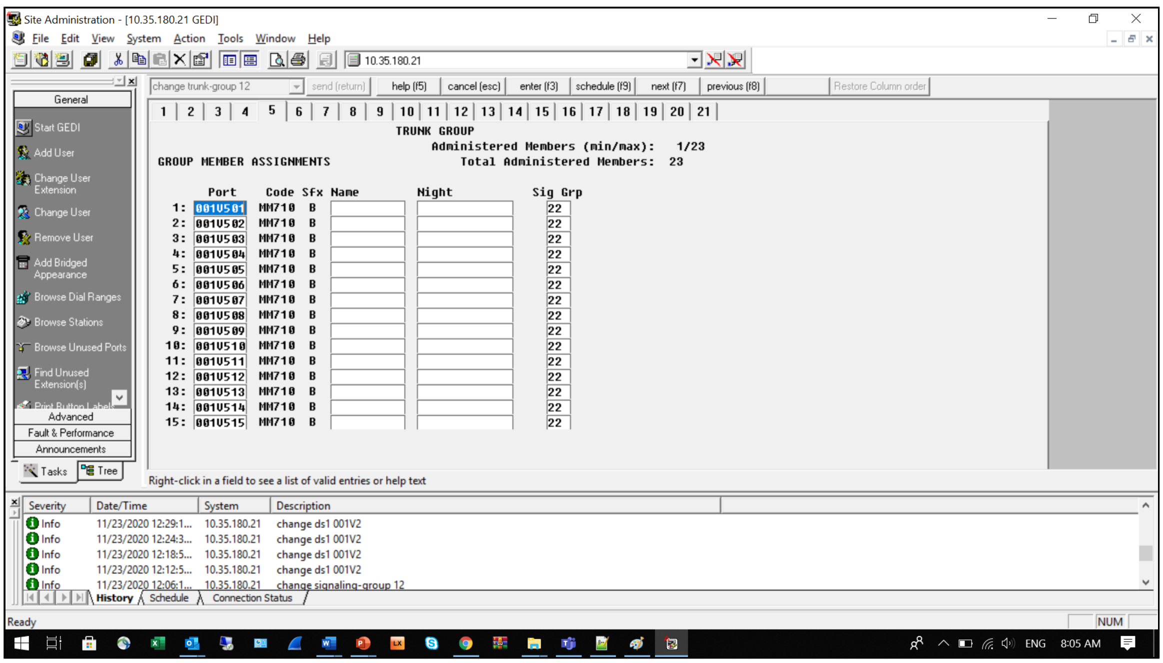
Figure 4: Trunk Group

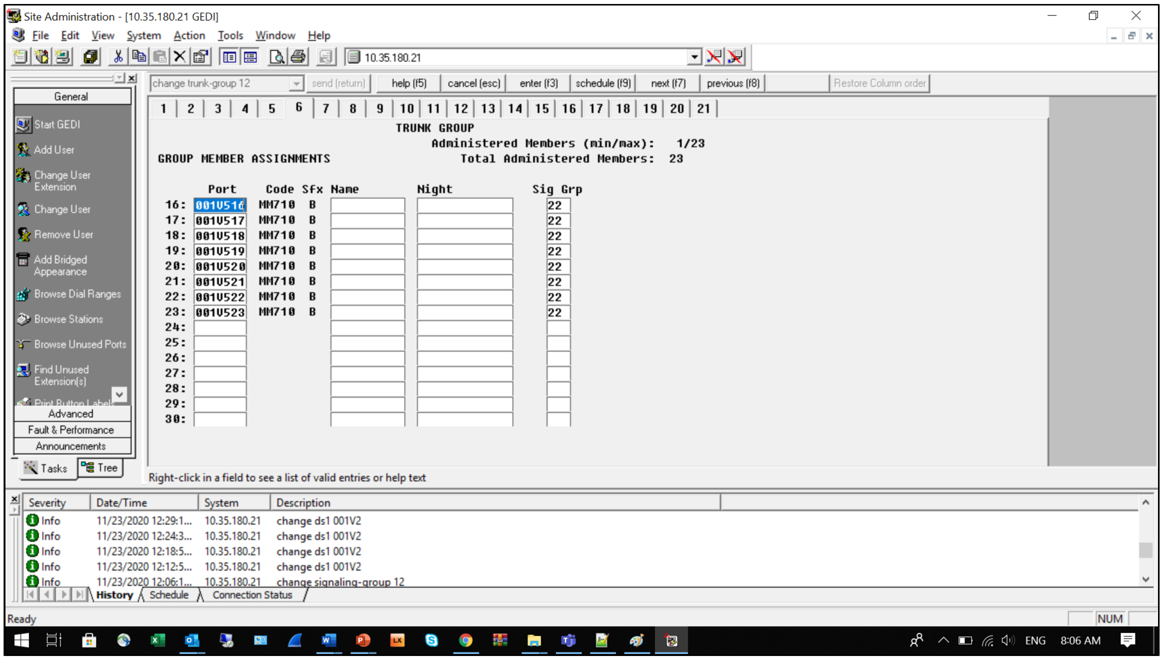



Site Administration - [10.35.180.21 GEDI]
change trunk-group 12
Group Type: isdn
TRUNK PARAMETERS
Codeset to Send Display: 6
Codeset to Send National IEs: 6
Max Message Size to Send: 260
Charge Advice: none
Supplementary Service Protocol: b
Digit Handling (in/out): enbloc/enbloc
Trunk Hunt: cyclical
Digital Loss Group: 13
Incoming Calling Number - Delete:
Insert:
Format: natl-pub
Bit Rate: 1200
Synchronization: async
Duplex: full
Disconnect Supervision - In? y
Out? y
Answer Supervision Timeout: 0
Administer Timers? n
CONNECT Reliable When Call Leaves ISDN? n
XOIP Treatment: auto
Delay Call Setup When Accessed Via IGAR? n
Caller ID for Service Link Call to H.323 1xC: station-extension
Right-click in a field to see a list of valid entries or help text


Site Administration - [10.35.180.21 GEDI]
change trunk-group 12
TRUNK FEATURES
ACA Assignment? n
Measured: none
Wideband Support? n
Internal Alert? n
Maintenance Tests? y
Data Restriction? n
NCA-TSC Trunk Member:
Send Name: n
Send Calling Number: y
Used for DCS? n
Hop Dgt? n
Send EMU Visitor CPN? n
Suppress # Outpulsing? n
Format: natl-pub
Outgoing Channel ID Encoding: preferred
UUI IE Treatment: service-provider
Replace Restricted Numbers? n
Replace Unavailable Numbers? n
Send Connected Number: y
Hold/Unhold Notifications? y
Send UUI IE? y
Modify Tandem Calling Number: no
Send UCID? n
Send Codeset 6/7 LAI IE? y
Ds1 Echo Cancellation? n
Apply Local Ringback? n
Show ANSWERED BY on Display? y
Network (Japan) Needs Connect Before Disconnect? n
Right-click in a field to see a list of valid entries or help text


Site Administration - [10.35.180.21 GEDI]
change trunk-group 12
QSIG TRUNK GROUP OPTIONS
TSC Method for Auto Callback: drop-if-possible
Diversion by Reroute? y
Path Replacement? y
Path Replacement with Retention? n
Path Replacement Method: better-route
SBS? n
Character Set for QSIG Name: eurofont
QSIG Value-Added? n
Right-click in a field to see a list of valid entries or help text

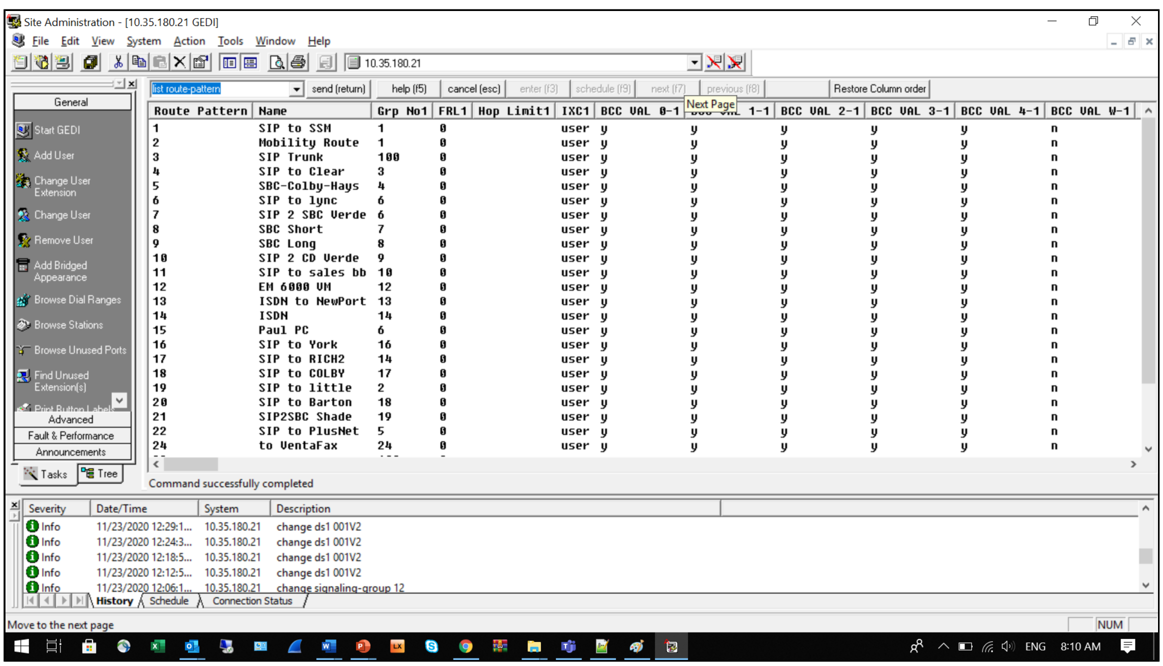

Site Administration - [10.35.180.21 GEDI]
change trunk-group 12
TRUNK GROUP
Administered Members (min/max): 1/23
GROUP MEMBER ASSIGNMENTS
Total Administered Members: 23
Port Code Sfx Name Night Sig Grp
1: 001V501 MM710 B 22
2: 001V502 MM710 B 22
3: 001V503 MM710 B 22
4: 001V504 MM710 B 22
5: 001V505 MM710 B 22
6: 001V506 MM710 B 22
7: 001V507 MM710 B 22
8: 001V508 MM710 B 22
9: 001V509 MM710 B 22
10: 001V510 MM710 B 22
11: 001V511 MM710 B 22
12: 001V512 MM710 B 22
13: 001V513 MM710 B 22
14: 001V514 MM710 B 22
15: 001V515 MM710 B 22
Right-click in a field to see a list of valid entries or help text

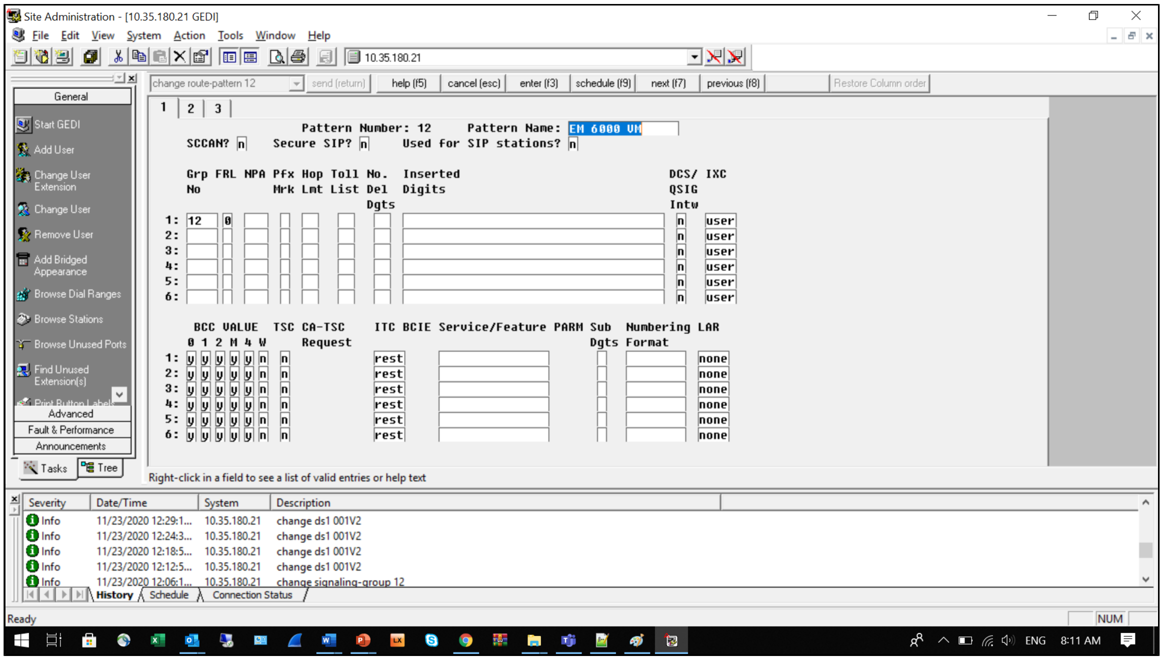


4. Route Pattern

1. Use the Site Administration to log into the Avaya CM 7.1.
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 type the available route pattern number. Press **F3** to save the changes.

Figure 5: Route Pattern

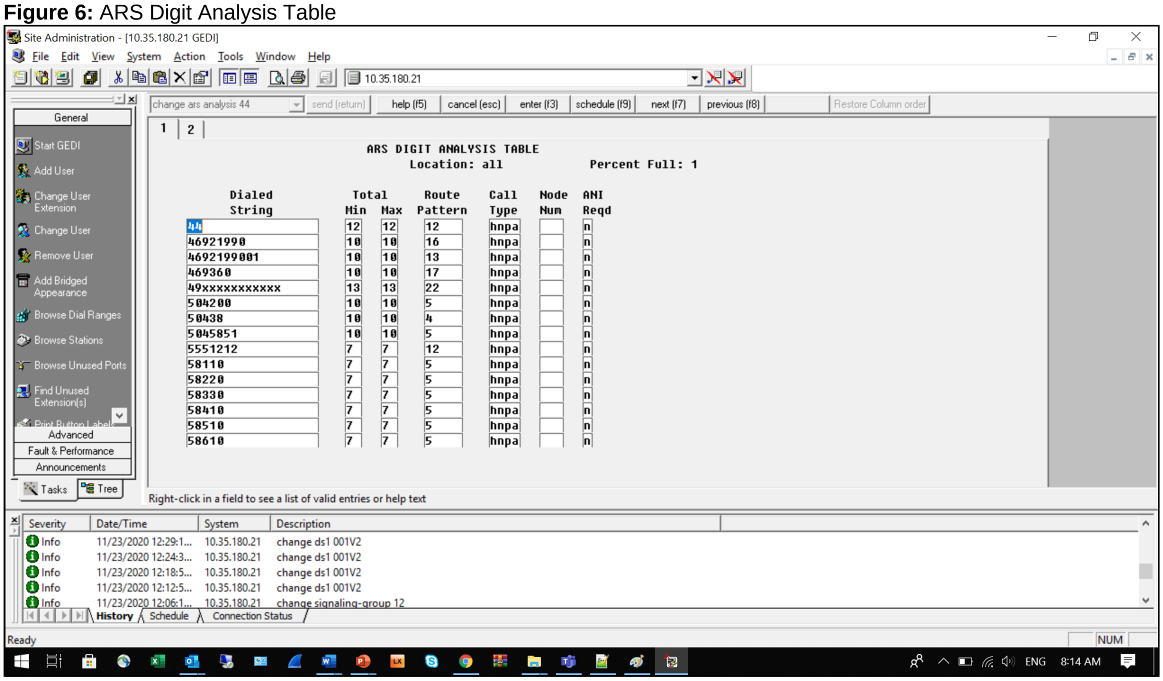




5. ARS Digit Analysis Table

1. Use the Site Administration to log into the Avaya CM 7.1.
2. Type **change ars analysis <dialed_number>** to add or change the handling for a specified called number.
3. Confirm the changes and press **F3** to save.

Figure 6: ARS Digit Analysis Table

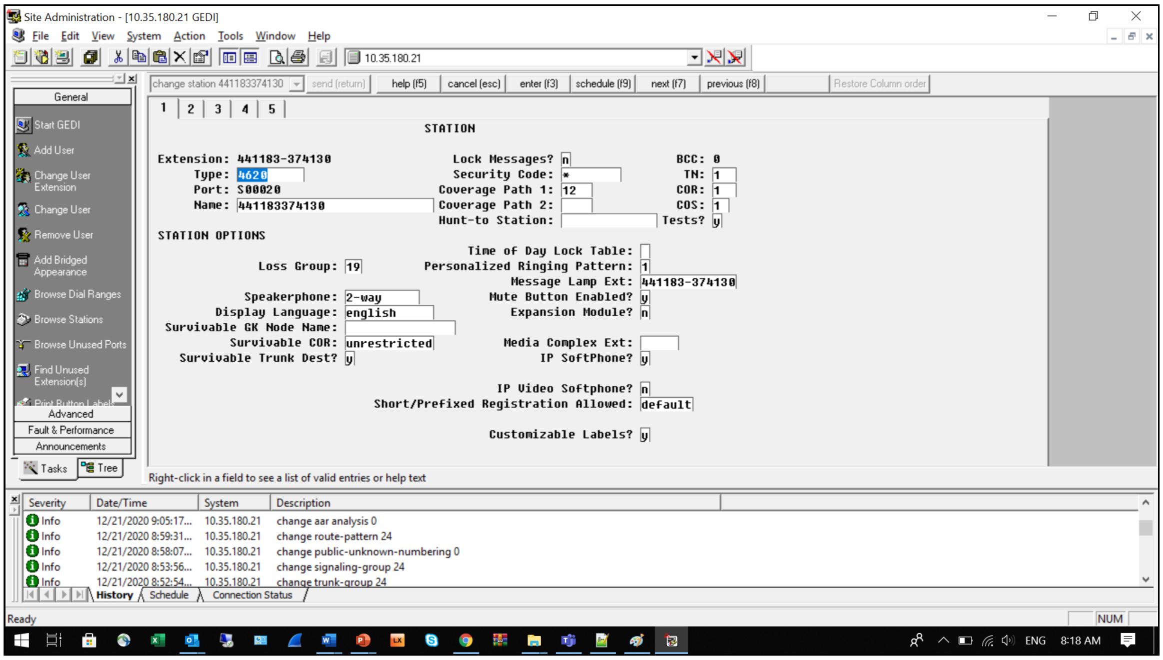


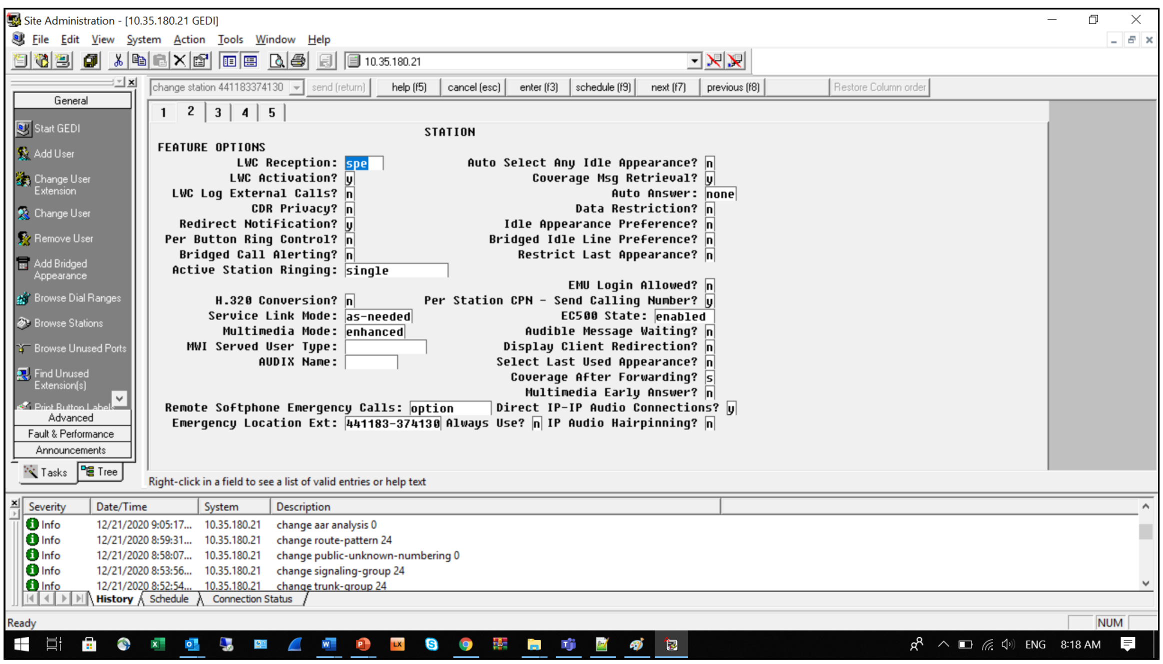
6. Station

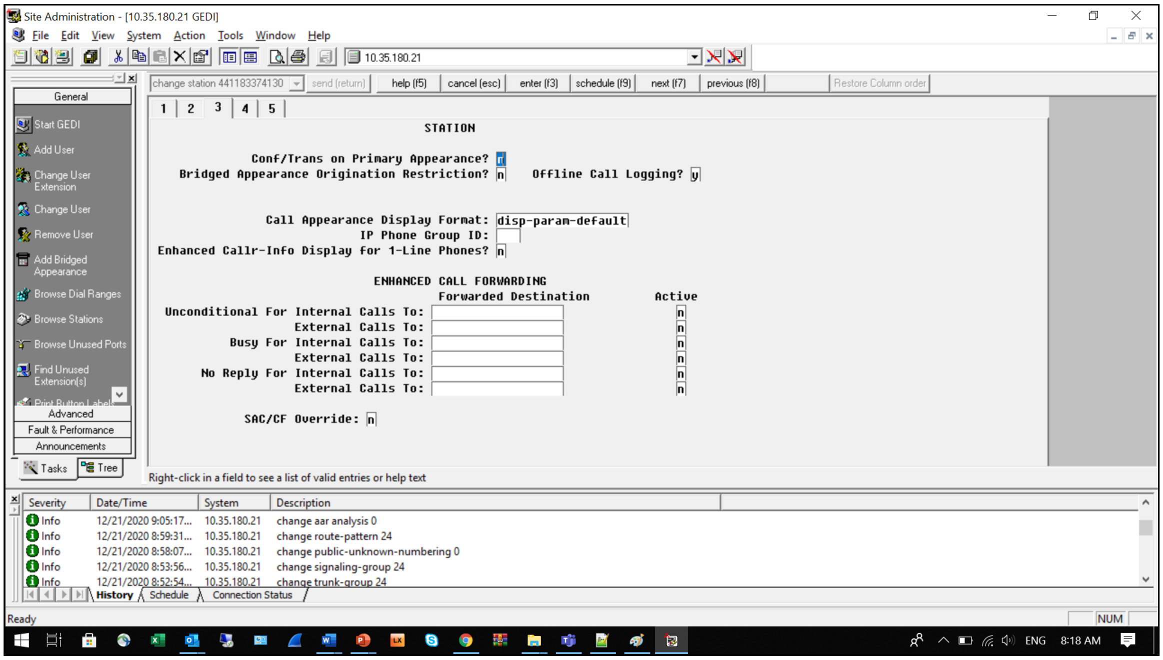
1. Use the Site Administration to log into the Avaya CM 7.1.
2. Type **add station next** to add a new station.

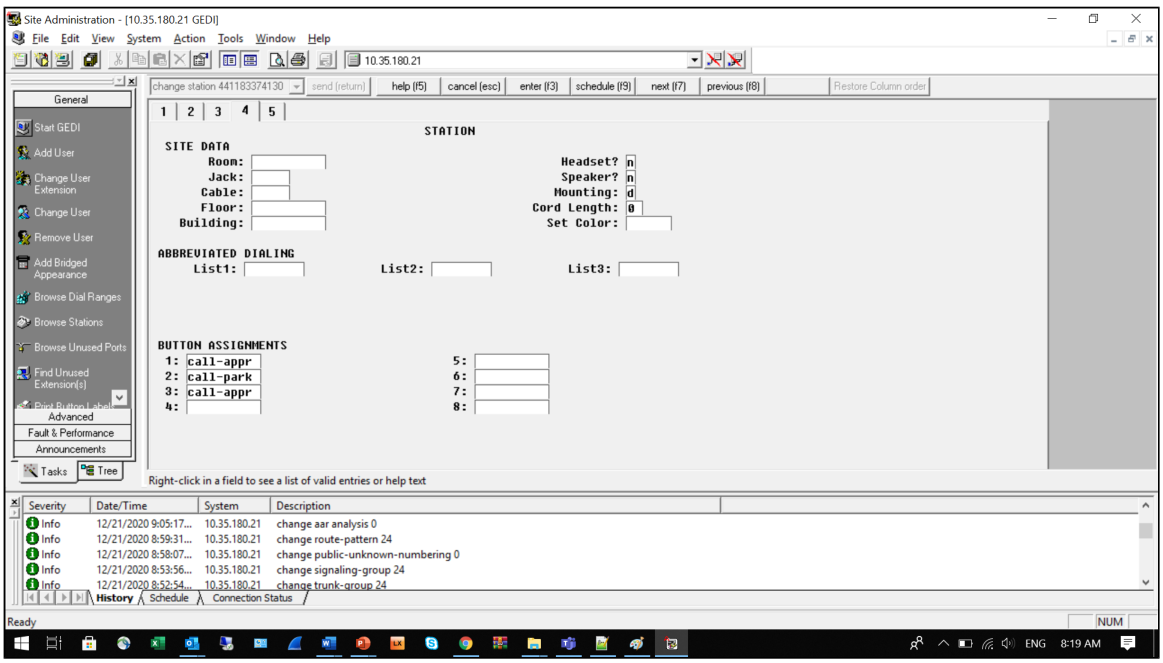
3. Confirm the changes and press **F3** to save.

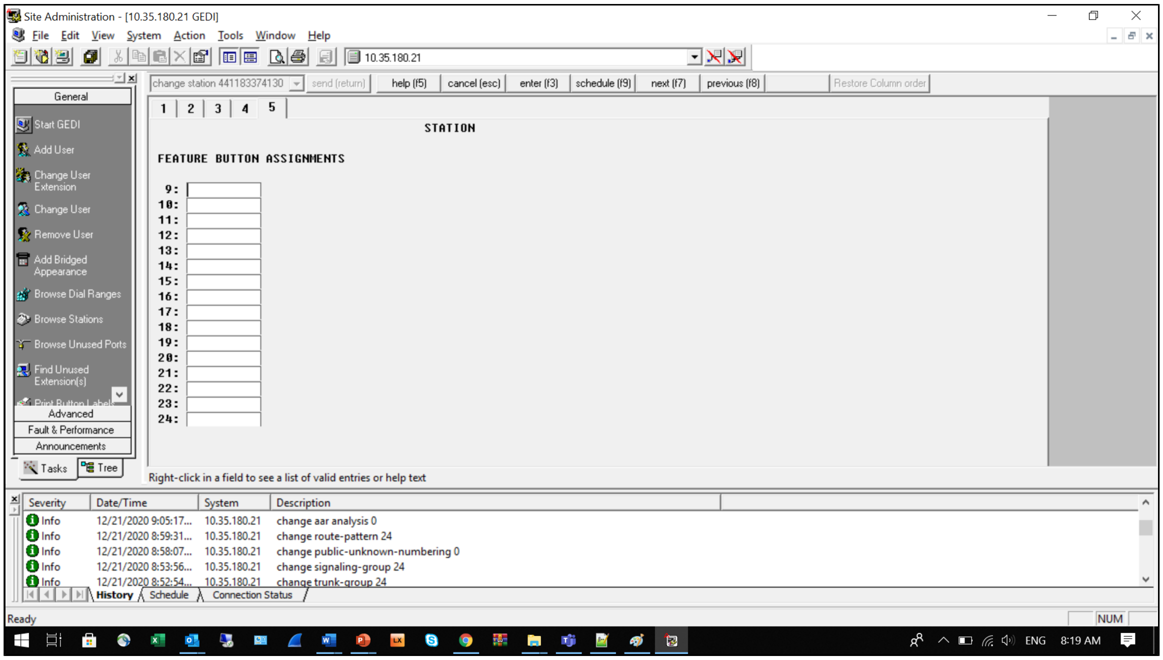
Figure 7: Station






Site Administration - [10.35.180.21 GEDI]
File Edit View System Action Tools Window Help
10.35.180.21
change station 441183374130
send (return)
help (f5)
cancel (esc)
enter (f3)
schedule (f9)
next (f7)
previous (f8)
Restore Column order
General
Start GEDI
Add User
Change User Extension
Change User
Remove User
Add Bridged Appearance
Browse Dial Ranges
Browse Stations
Browse Unused Ports
Find Unused Extension(s)
Advanced
Fault & Performance
Announcements
Tasks
Tree
1 2 3 4 5
STATION
Conf/Trans on Primary Appearance? n
Bridged Appearance Origination Restriction? n
Offline Call Logging? y
Call Appearance Display Format: disp-param-default
IP Phone Group ID:
Enhanced Callr-Info Display for 1-Line Phones? n
ENHANCED CALL FORWARDING
Forwarded Destination
Active
Unconditional For Internal Calls To: n
External Calls To: n
Busy For Internal Calls To: n
External Calls To: n
No Reply For Internal Calls To: n
External Calls To: n
SAC/CF Override: n
Right-click in a field to see a list of valid entries or help text
Severity
Date/Time
System
Description
Info 12/21/2020 9:05:17... 10.35.180.21 change aar analysis 0
Info 12/21/2020 8:59:31... 10.35.180.21 change route-pattern 24
Info 12/21/2020 8:58:07... 10.35.180.21 change public-unknown-numbering 0
Info 12/21/2020 8:53:56... 10.35.180.21 change signaling-group 24
Info 12/21/2020 8:52:54... 10.35.180.21 change trunk-group 24
History
Schedule
Connection Status
Ready
NUM
ENG
8:18 AM

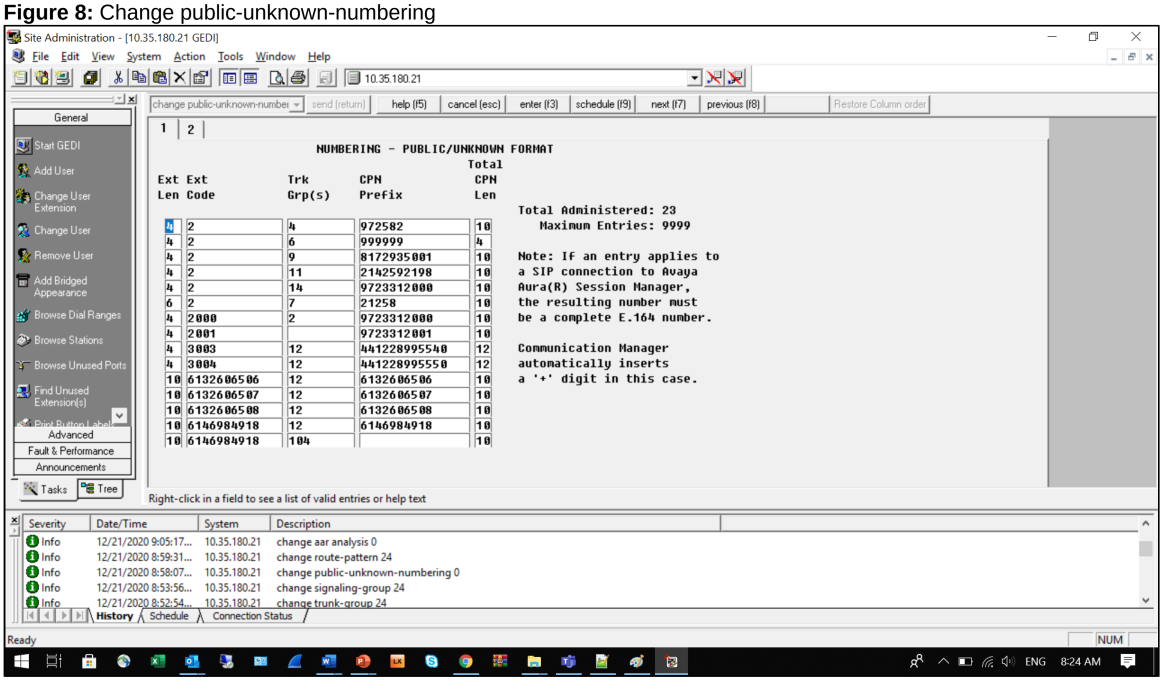

Site Administration - [10.35.180.21 GEDI]
File Edit View System Action Tools Window Help
10.35.180.21
change station 441183374130
send (return)
help (f5)
cancel (esc)
enter (f3)
schedule (f9)
next (f7)
previous (f8)
Restore Column order
General
Start GEDI
Add User
Change User Extension
Change User
Remove User
Add Bridged Appearance
Browse Dial Ranges
Browse Stations
Browse Unused Ports
Find Unused Extension(s)
Advanced
Fault & Performance
Announcements
Tasks
Tree
1 2 3 4 5
STATION
SITE DATA
Room:
Headset? n
Jack:
Speaker? n
Cable:
Mounting: d
Floor:
Cord Length: 0
Building:
Set Color:
ABBREVIATED DIALING
List1:
List2:
List3:
BUTTON ASSIGNMENTS
1: call-appr
5:
2: call-park
6:
3: call-appr
7:
4:
8:
Right-click in a field to see a list of valid entries or help text
Severity
Date/Time
System
Description
Info 12/21/2020 9:05:17... 10.35.180.21 change aar analysis 0
Info 12/21/2020 8:59:31... 10.35.180.21 change route-pattern 24
Info 12/21/2020 8:58:07... 10.35.180.21 change public-unknown-numbering 0
Info 12/21/2020 8:53:56... 10.35.180.21 change signaling-group 24
Info 12/21/2020 8:52:54... 10.35.180.21 change trunk-group 24
History
Schedule
Connection Status
Ready
NUM
ENG
8:19 AM



7. Change public-unknown-numbering

1. Use the Site Administration to log into the Avaya CM 7.1.
2. Type **change public-unknown-numbering 0** and add the caller number station. With this command, the caller number will appear for outgoing calls in an ISDN interconnection.
3. Confirm the changes and press **F3** to save.

Figure 8: Change public-unknown-numbering



EdgeMarc Configuration

Network

- LAN and WAN Interfaces

- Static Routes
- T1/E1 Configuration
- ISDN - Network

VoIP

- VoIP Settings
- SIP Settings
- B2BUA
- Survivability

Network

LAN and WAN Interfaces

1. Log into the EdgeMarc as a **root** user.
2. Click **Network** to configure the LAN and WAN interfaces.

Figure 9: EdgeMarc Network LAN Interface

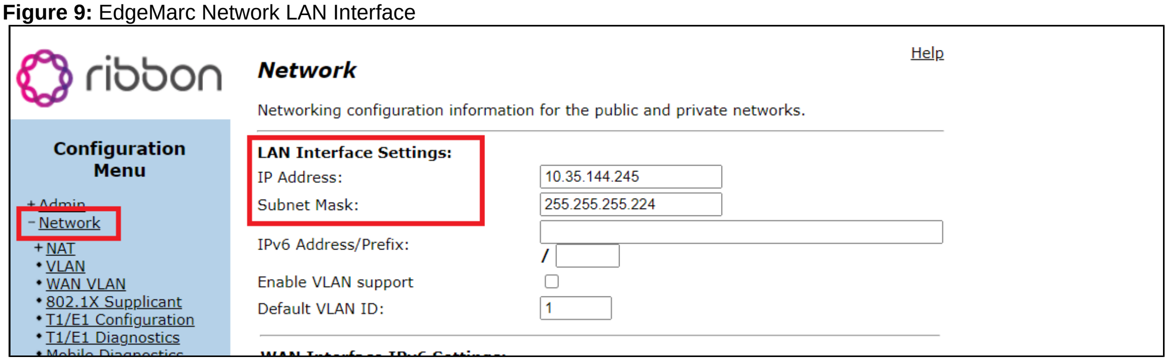
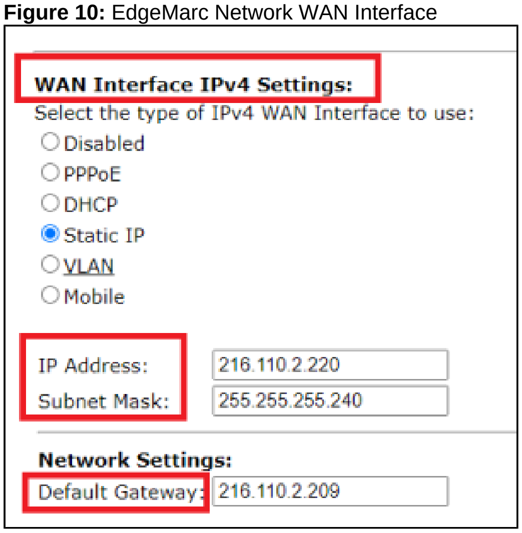


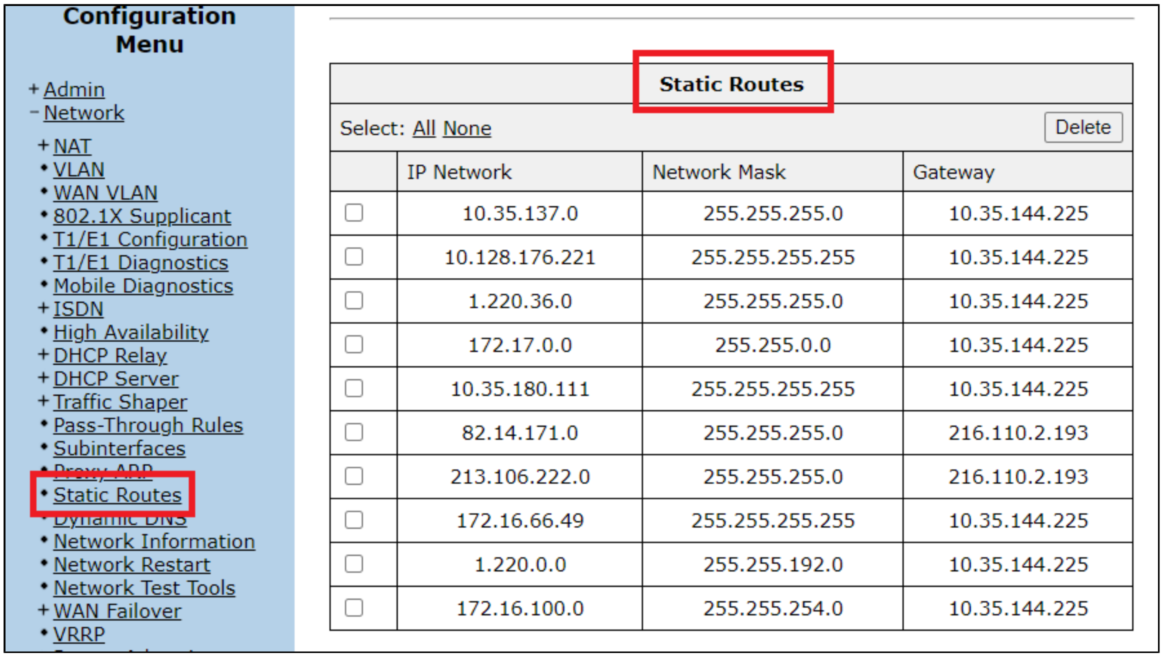
Figure 10: EdgeMarc Network WAN Interface



Static Routes

Select **Network > Static Routes** to configure the routes.

Figure 11: Static Routes

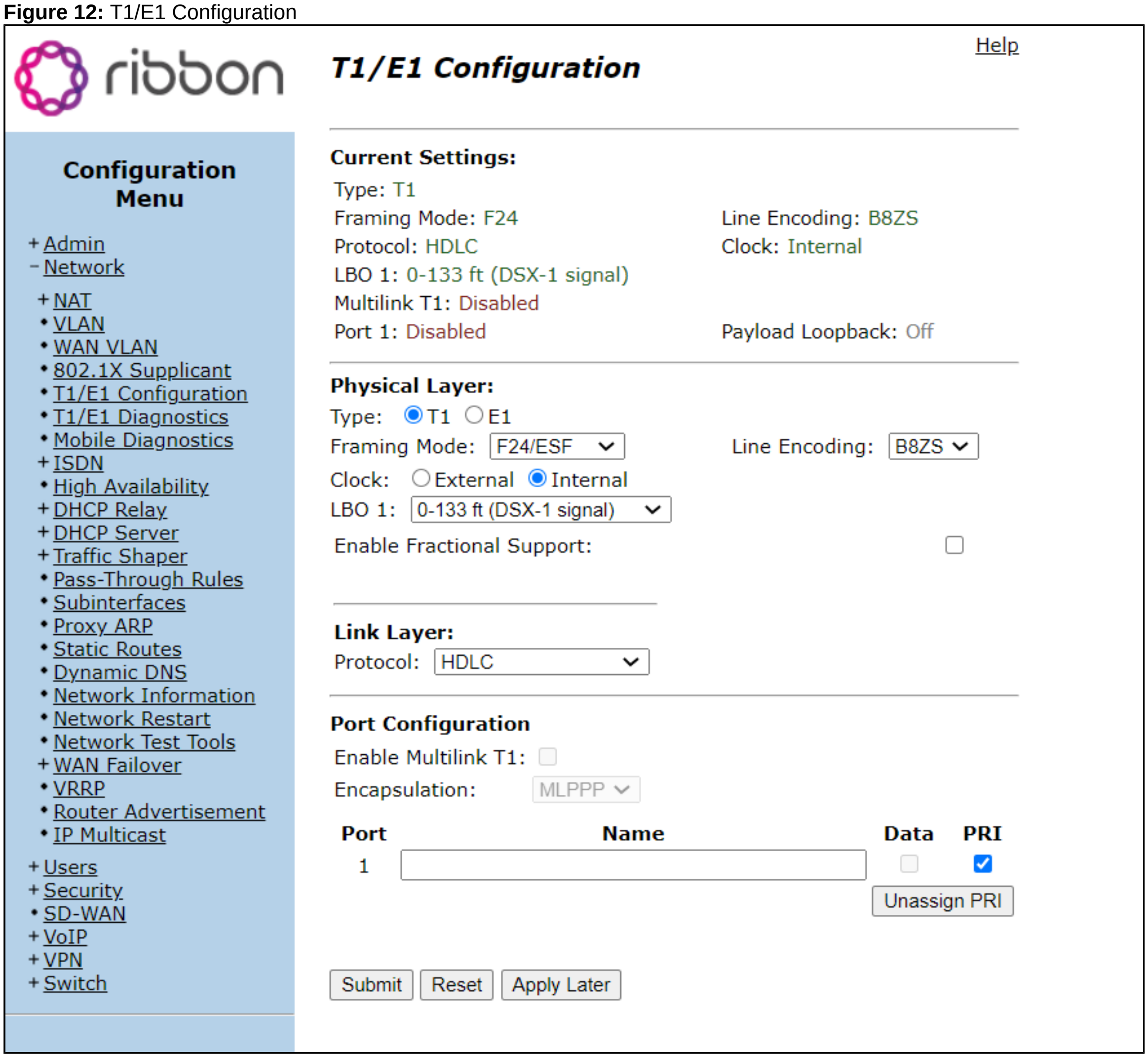


	IP Network	Network Mask	Gateway
☐	10.35.137.0	255.255.255.0	10.35.144.225
☐	10.128.176.221	255.255.255.255	10.35.144.225
☐	1.220.36.0	255.255.255.0	10.35.144.225
☐	172.17.0.0	255.255.0.0	10.35.144.225
☐	10.35.180.111	255.255.255.255	10.35.144.225
☐	82.14.171.0	255.255.255.0	216.110.2.193
☐	213.106.222.0	255.255.255.0	216.110.2.193
☐	172.16.66.49	255.255.255.255	10.35.144.225
☐	1.220.0.0	255.255.192.0	10.35.144.225
☐	172.16.100.0	255.255.254.0	10.35.144.225

T1/E1 Configuration

Select **Network > T1/E1 Configuration** to configure the T1.

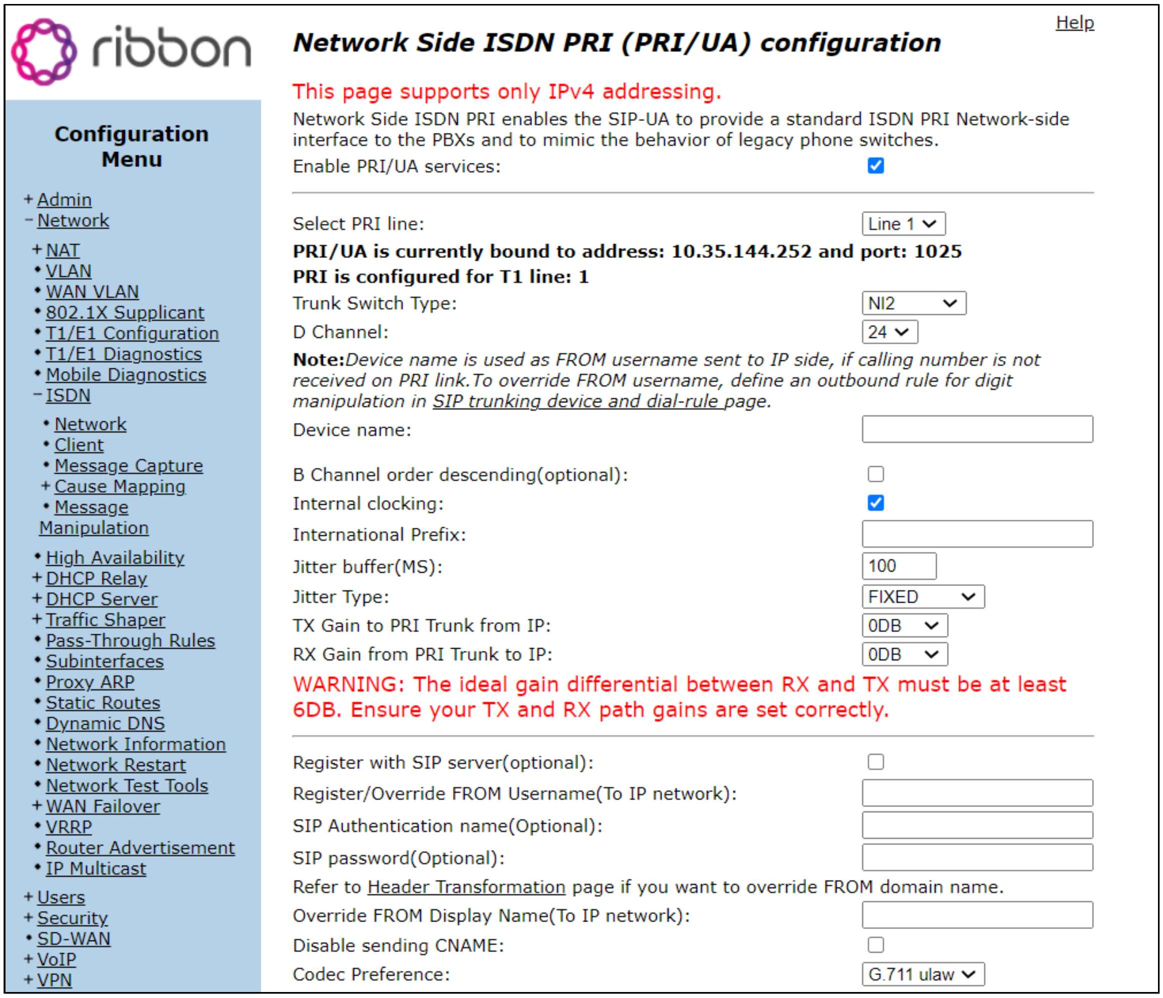
Figure 12: T1/E1 Configuration

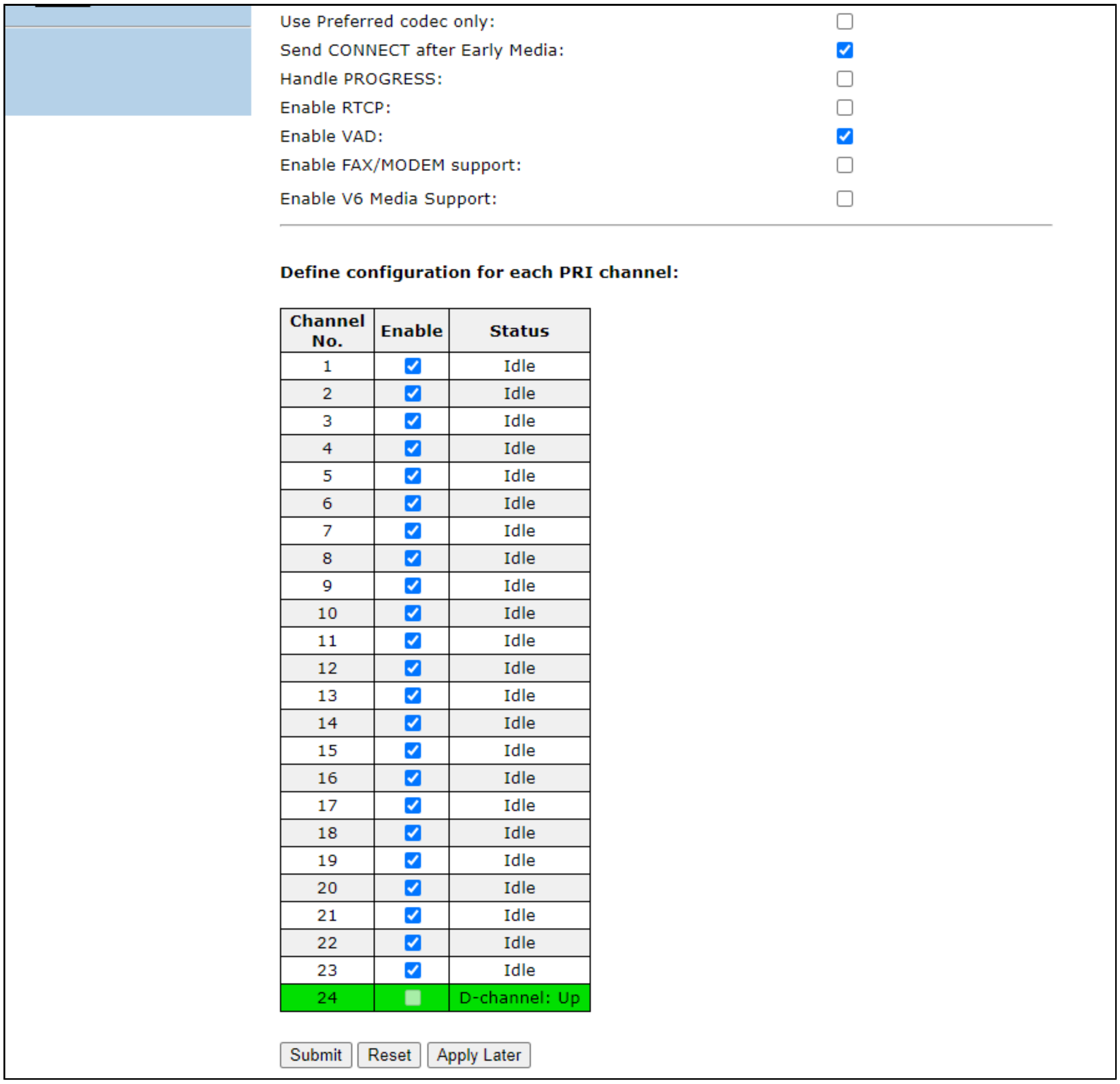


ISDN - Network

Select **Network > ISDN > Network** to configure the Network Side ISDN PRI configuration.

Figure 13: ISDN - Network



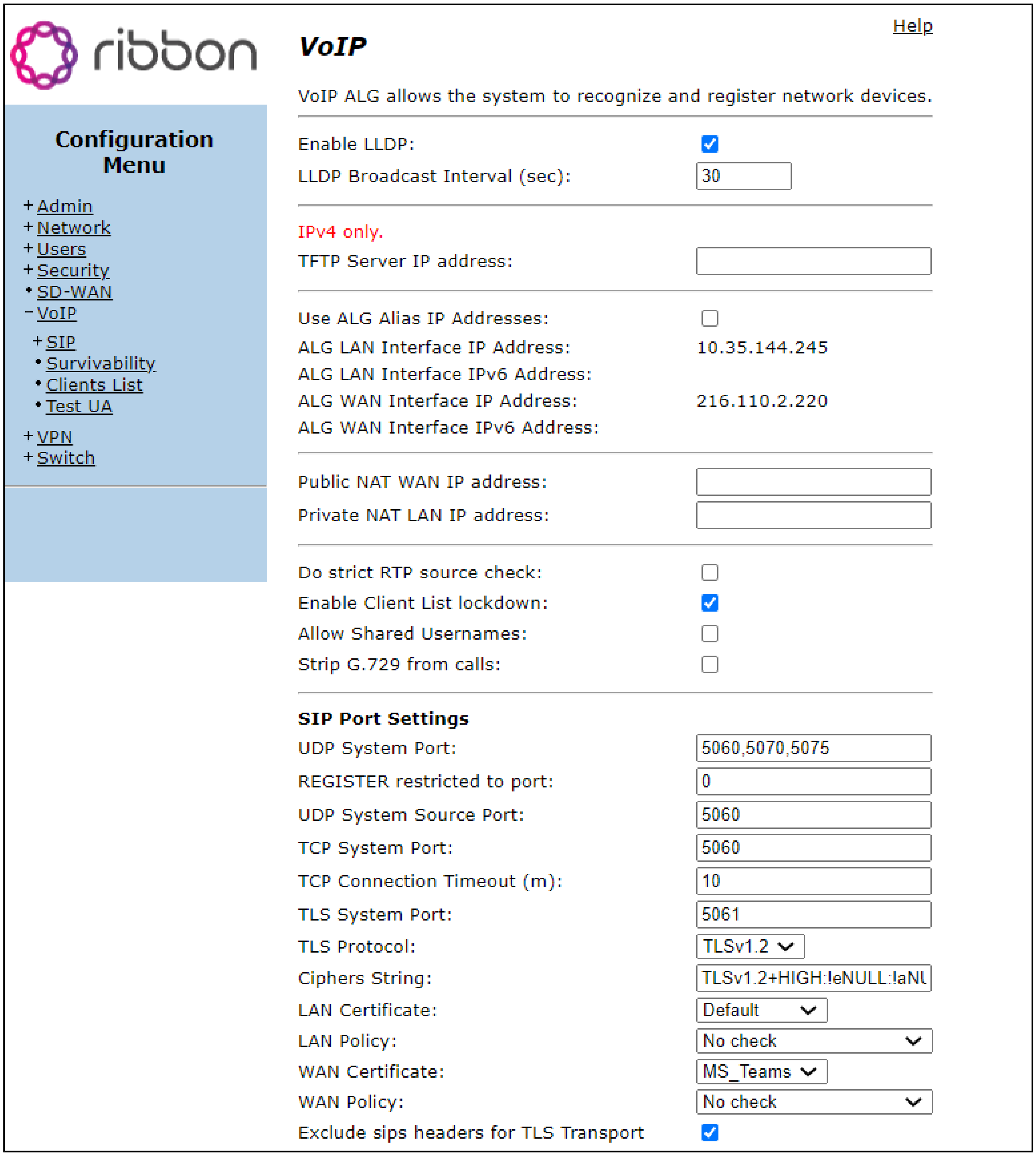


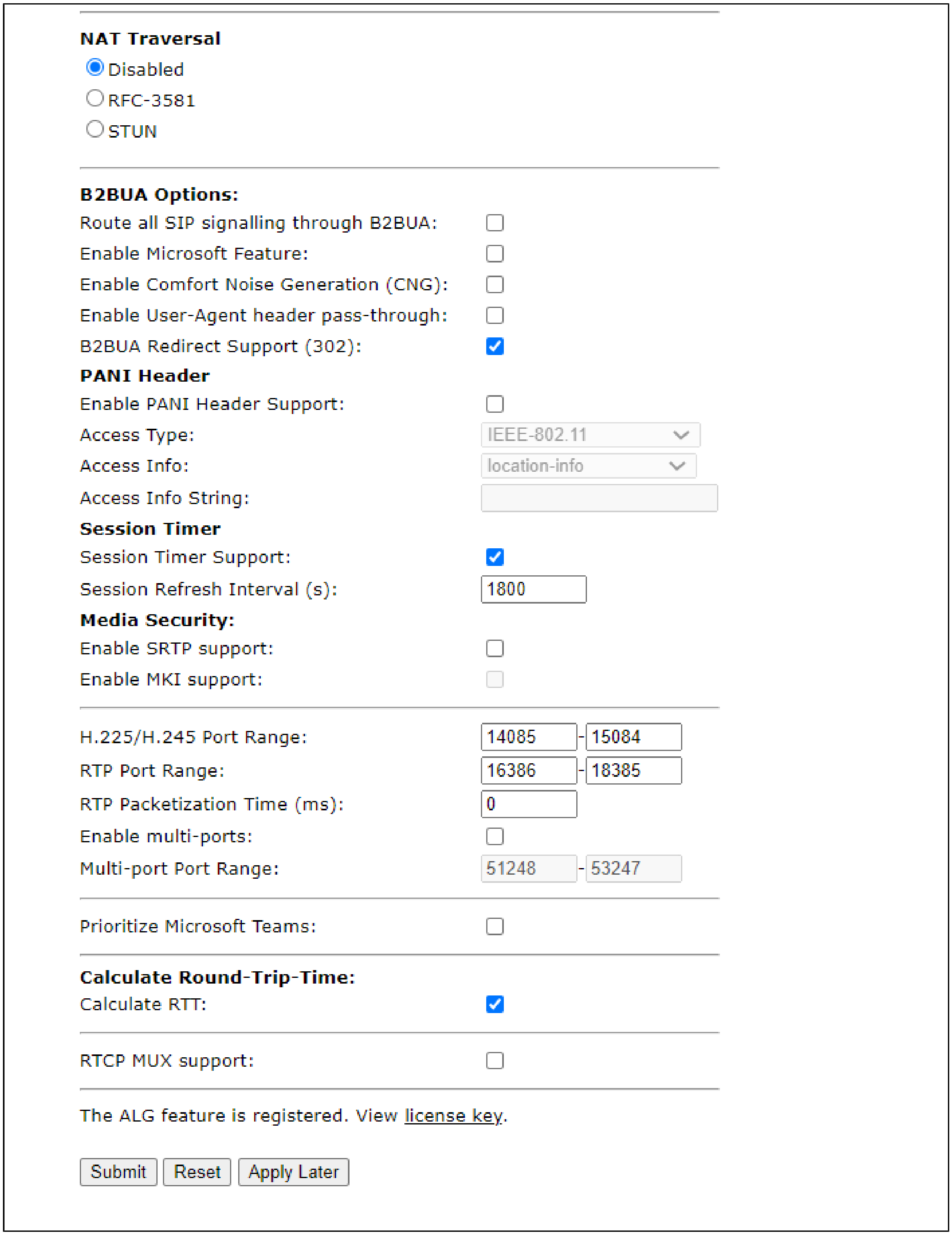
VoIP

VoIP Settings

1. Login as a **root** user.
2. Click **VoIP** to configure the VoIP features.

Figure 14: VoIP

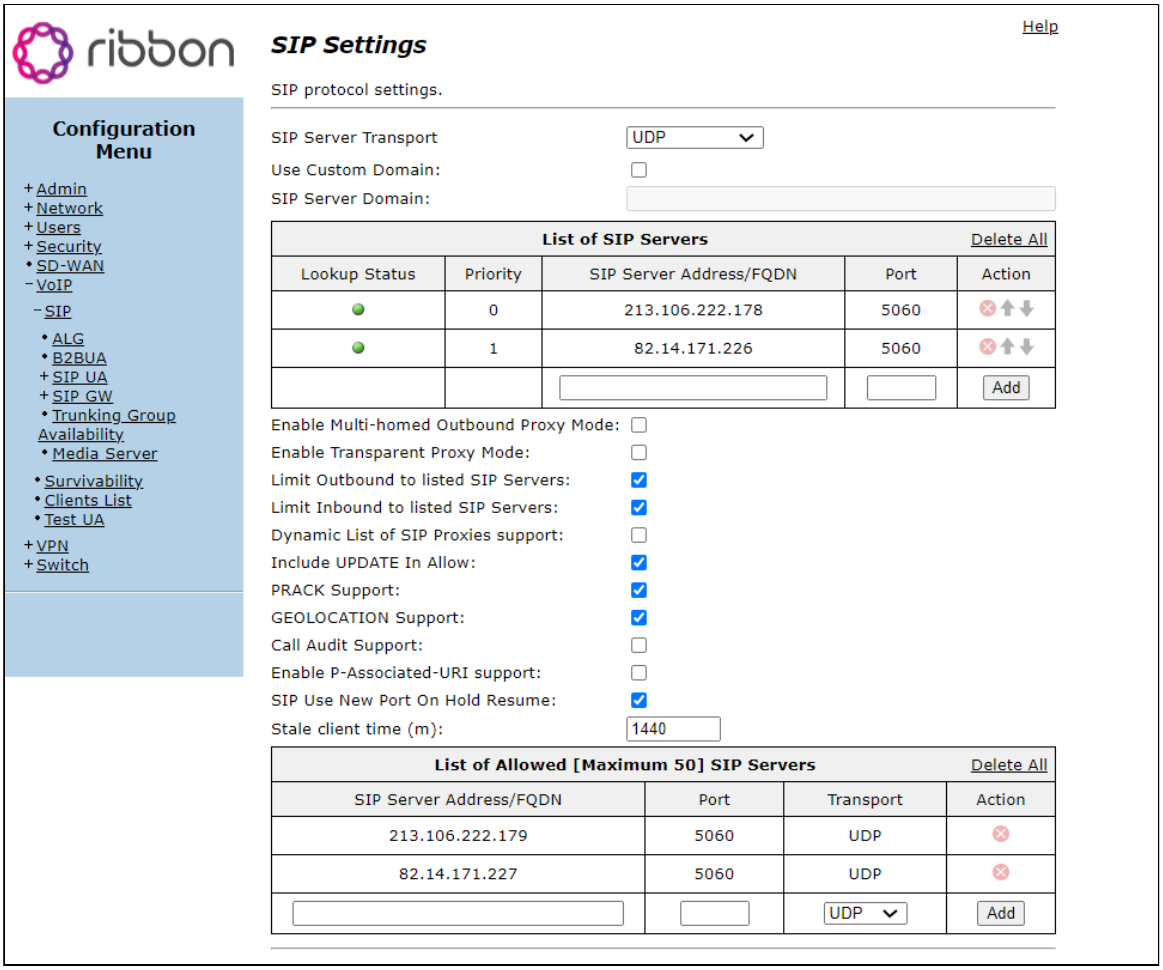

Help
VoIP
VoIP ALG allows the system to recognize and register network devices.
Enable LLDP:
LLDP Broadcast Interval (sec):
30
IPv4 only.
TFTP Server IP address:
Use ALG Alias IP Addresses:
ALG LAN Interface IP Address:
10.35.144.245
ALG LAN Interface IPv6 Address:
ALG WAN Interface IP Address:
216.110.2.220
ALG WAN Interface IPv6 Address:
Public NAT WAN IP address:
Private NAT LAN IP address:
Do strict RTP source check:
Enable Client List lockdown:
Allow Shared Usernames:
Strip G.729 from calls:
SIP Port Settings
UDP System Port:
5060,5070,5075
REGISTER restricted to port:
0
UDP System Source Port:
5060
TCP System Port:
5060
TCP Connection Timeout (m):
10
TLS System Port:
5061
TLS Protocol:
TLSv1.2
Ciphers String:
TLSv1.2+HIGH:!eNULL:!aNU
LAN Certificate:
Default
LAN Policy:
No check
WAN Certificate:
MS_Teams
WAN Policy:
No check
Exclude sips headers for TLS Transport
Configuration Menu
+Admin
+Network
+Users
+Security
• SD-WAN
−VoIP
+SIP
• Survivability
• Clients List
• Test UA
+VPN
+Switch

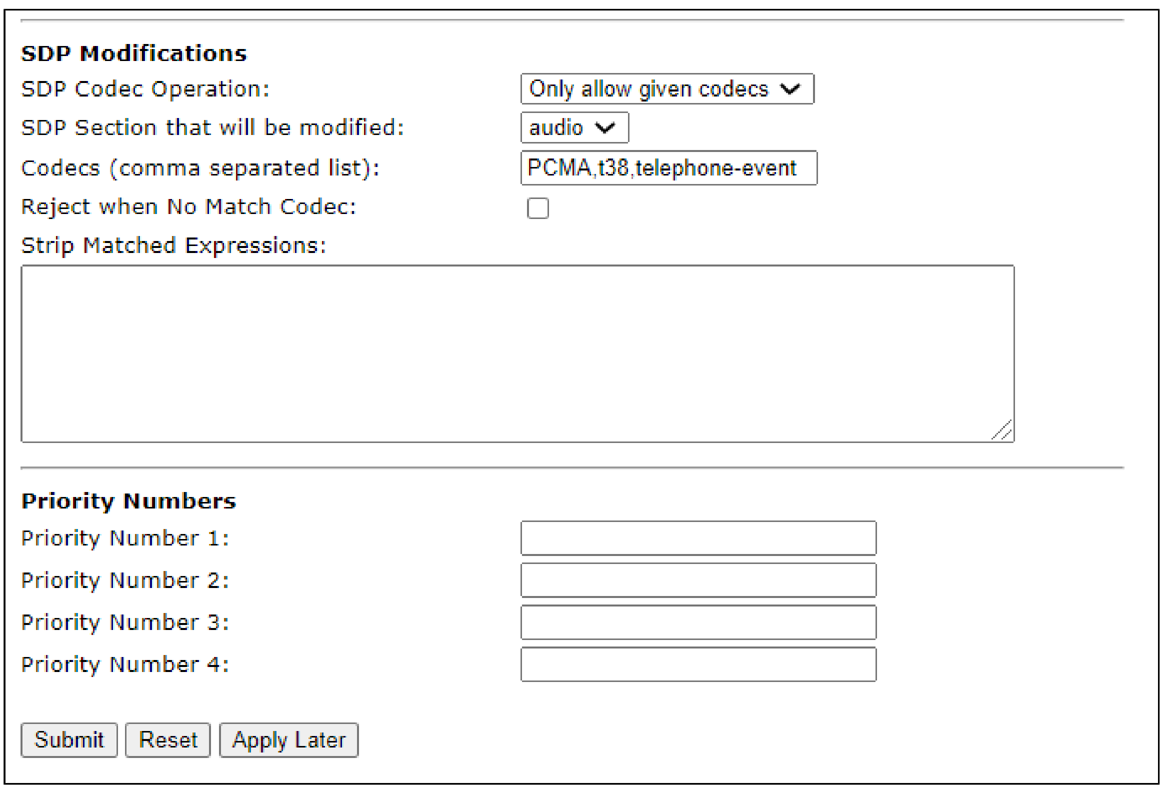


SIP Settings

1. Select **VoIP > SIP** to configure the SIP settings.
2. Configure the SIP servers.

Figure 15: SIP Settings

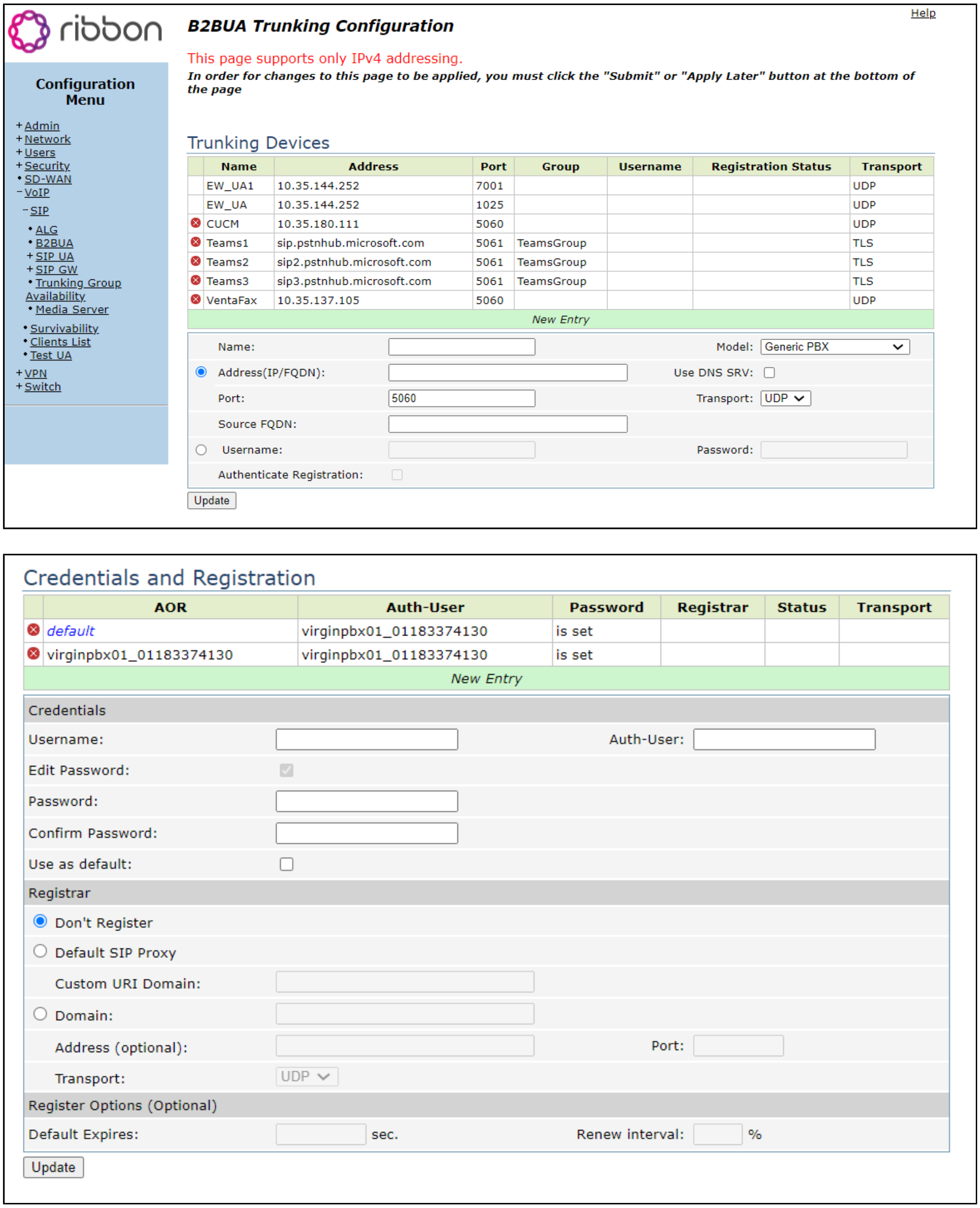

ribbon
SIP Settings
Help
SIP protocol settings.
Configuration Menu
+Admin
+Network
+Users
+Security
•SD-WAN
−VoIP
−SIP
•ALG
•B2BUA
+SIP UA
+SIP GW
•Trunking Group Availability
•Media Server
•Survivability
•Clients List
•Test UA
+VPN
+Switch
SIP Server Transport
UDP
Use Custom Domain:
SIP Server Domain:
List of SIP Servers
Delete All
Lookup Status
Priority
SIP Server Address/FQDN
Port
Action
0
213.106.222.178
5060
1
82.14.171.226
5060
Add
Enable Multi-homed Outbound Proxy Mode:
Enable Transparent Proxy Mode:
Limit Outbound to listed SIP Servers:
Limit Inbound to listed SIP Servers:
Dynamic List of SIP Proxies support:
Include UPDATE In Allow:
PRACK Support:
GEOLOCATION Support:
Call Audit Support:
Enable P-Associated-URI support:
SIP Use New Port On Hold Resume:
Stale client time (m):
1440
List of Allowed [Maximum 50] SIP Servers
Delete All
SIP Server Address/FQDN
Port
Transport
Action
213.106.222.179
5060
UDP
82.14.171.227
5060
UDP
UDP
Add

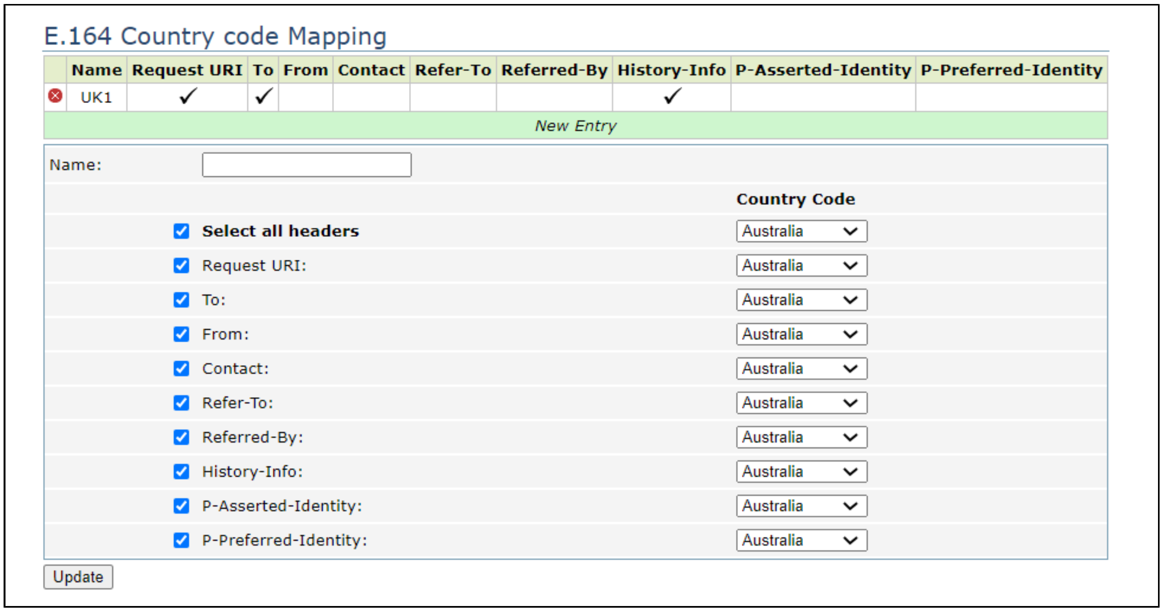

SDP Modifications
SDP Codec Operation:
Only allow given codecs
SDP Section that will be modified:
audio
Codecs (comma separated list):
PCMA,t38,telephone-event
Reject when No Match Codec:
Strip Matched Expressions:
Priority Numbers
Priority Number 1:
Priority Number 2:
Priority Number 3:
Priority Number 4:
Submit
Reset
Apply Later

B2BUA

1. Select **VoIP > B2BUA** to access the B2BUA trunking configuration.
2. Configure the next parameters.

Figure 16: B2BUA





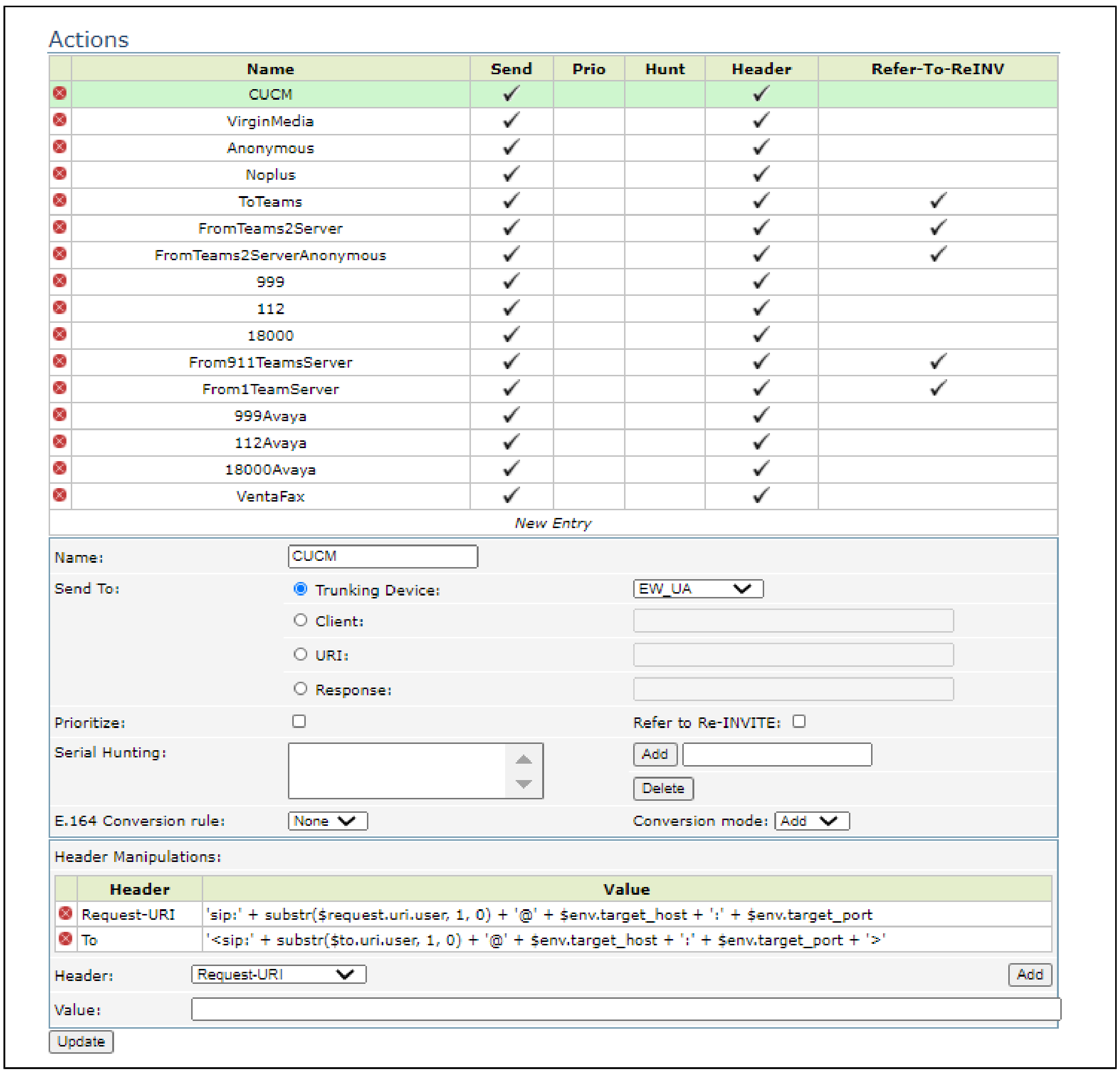
E.164 Country code Mapping

Name	Request URI	To	From	Contact	Refer-To	Referred-By	History-Info	P-Asserted-Identity	P-Preferred-Identity
UK1	✓	✓					✓		

New Entry

Name:		
		Country Code
☑	**Select all headers**	Australia
☑	Request URI:	Australia
☑	To:	Australia
☑	From:	Australia
☑	Contact:	Australia
☑	Refer-To:	Australia
☑	Referred-By:	Australia
☑	History-Info:	Australia
☑	P-Asserted-Identity:	Australia
☑	P-Preferred-Identity:	Australia

Update



Actions

Name	Send	Prio	Hunt	Header	Refer-To-ReINV
CUCM	✓			✓	
VirginMedia	✓			✓	
Anonymous	✓			✓	
Noplus	✓			✓	
ToTeams	✓			✓	✓
FromTeams2Server	✓			✓	✓
FromTeams2ServerAnonymous	✓			✓	✓
999	✓			✓	
112	✓			✓	
18000	✓			✓	
From911TeamsServer	✓			✓	✓
From1TeamServer	✓			✓	✓
999Avaya	✓			✓	
112Avaya	✓			✓	
18000Avaya	✓			✓	
VentaFax	✓			✓	

New Entry

Name: CUCM

Send To: ◉ Trunking Device: EW_UA

○ Client:

○ URI:

○ Response:

Prioritize: ☐ Refer to Re-INVITE: ☐

Serial Hunting: Add Delete

E.164 Conversion rule: None Conversion mode: Add

Header Manipulations:

Header	Value
Request-URI	'sip:' + substr($request.uri.user, 1, 0) + '@' + $env.target_host + ':' + $env.target_port
To	'<sip:' + substr($to.uri.user, 1, 0) + '@' + $env.target_host + ':' + $env.target_port + '>'

Header: Request-URI Add

Value:

Update

Actions

	Name	Send	Prio	Hunt	Header	Refer-To-ReINV
⊗	CUCM	✓			✓	
⊗	VirginMedia	✓			✓	
⊗	Anonymous	✓			✓	
⊗	Noplus	✓			✓	
⊗	ToTeams	✓			✓	✓
⊗	FromTeams2Server	✓			✓	✓
⊗	FromTeams2ServerAnonymous	✓			✓	✓
⊗	999	✓			✓	
⊗	112	✓			✓	
⊗	18000	✓			✓	
⊗	From911TeamsServer	✓			✓	✓
⊗	From1TeamServer	✓			✓	✓
⊗	999Avaya	✓			✓	
⊗	112Avaya	✓			✓	
⊗	18000Avaya	✓			✓	
⊗	VentaFax	✓			✓	

New Entry

Name: Anonymous

Send To: (•) Trunking Device: None

( ) Client:

( ) URI:

( ) Response:

Prioritize: ☐ Refer to Re-INVITE: ☐

Serial Hunting: [Add] [Delete]

E.164 Conversion rule: UK1 Conversion mode: Add

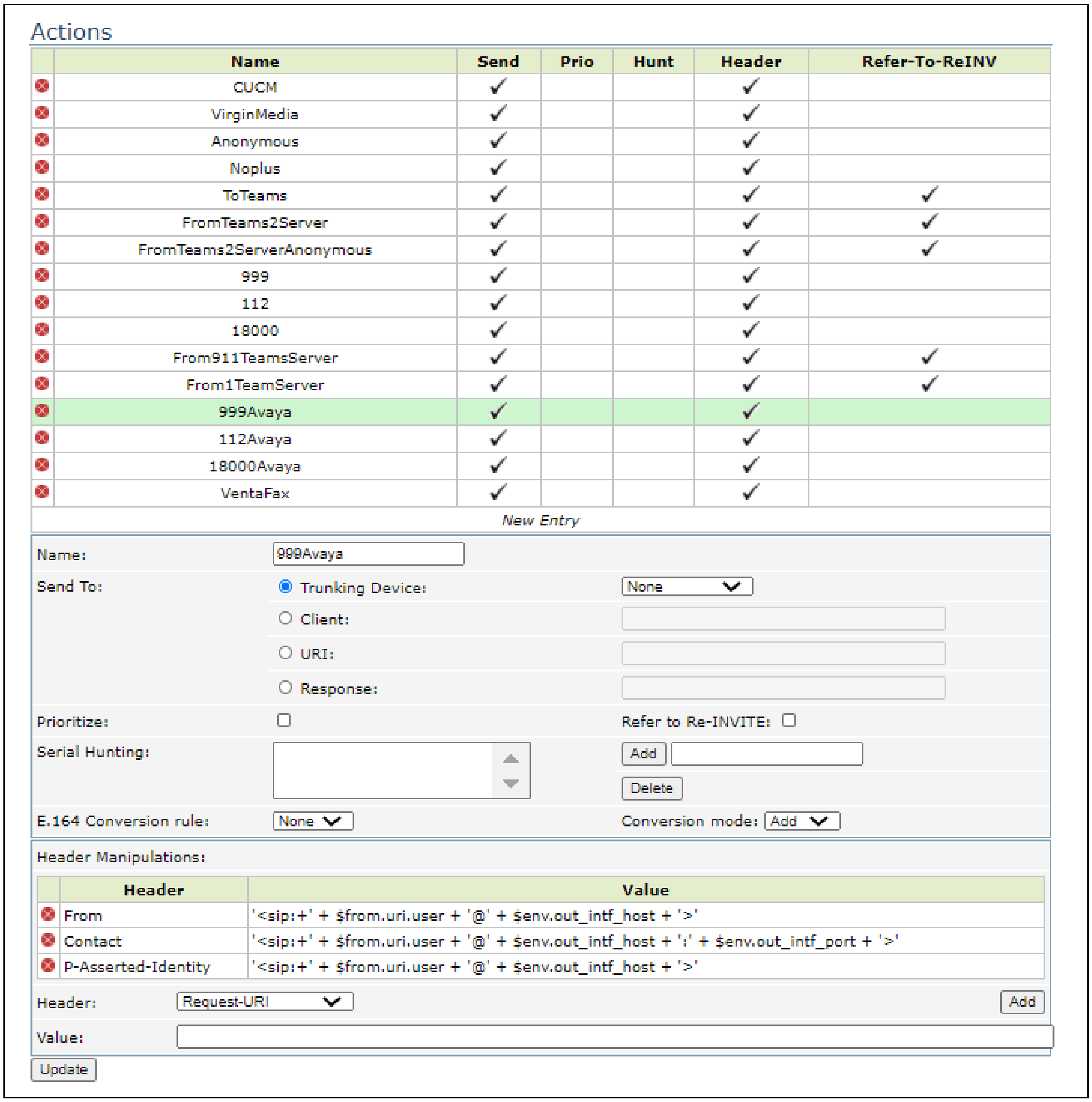
Header Manipulations:

	Header	Value
⊗	From	'<sip:' + $from.uri.user + '@' + $env.out_intf_host + '>'
⊗	Contact	'<sip:' + $from.uri.user + '@' + $env.out_intf_host + ':' + $from.uri.port + '>'
⊗	P-Asserted-Identity	'<sip:' + $from.uri.user + '@' + $env.out_intf_host + '>'
⊗	Privacy	$privacy.text

Header: Request-URI [Add]

Value:

[Update]



Actions

	Name	Send	Prio	Hunt	Header	Refer-To-ReINV
⊗	CUCM	✓			✓	
⊗	VirginMedia	✓			✓	
⊗	Anonymous	✓			✓	
⊗	Noplus	✓			✓	
⊗	ToTeams	✓			✓	✓
⊗	FromTeams2Server	✓			✓	✓
⊗	FromTeams2ServerAnonymous	✓			✓	✓
⊗	999	✓			✓	
⊗	112	✓			✓	
⊗	18000	✓			✓	
⊗	From911TeamsServer	✓			✓	✓
⊗	From1TeamServer	✓			✓	✓
⊗	999Avaya	✓			✓	
⊗	112Avaya	✓			✓	
⊗	18000Avaya	✓			✓	
⊗	VentaFax	✓			✓	

New Entry

Name: 999Avaya

Send To: Trunking Device: None

Client:

URI:

Response:

Prioritize: Refer to Re-INVITE:

Serial Hunting: Add Delete

E.164 Conversion rule: None Conversion mode: Add

Header Manipulations:

	Header	Value
⊗	From	'<sip:+' + $from.uri.user + '@' + $env.out_intf_host + '>'
⊗	Contact	'<sip:+' + $from.uri.user + '@' + $env.out_intf_host + ':' + $env.out_intf_port + '>'
⊗	P-Asserted-Identity	'<sip:+' + $from.uri.user + '@' + $env.out_intf_host + '>'

Header: Request-URI Add

Value:

Update

Actions

	Name	Send	Prio	Hunt	Header	Refer-To-ReINV
⊗	CUCM	✓			✓	
⊗	VirginMedia	✓			✓	
⊗	Anonymous	✓			✓	
⊗	Noplus	✓			✓	
⊗	ToTeams	✓			✓	✓
⊗	FromTeams2Server	✓			✓	✓
⊗	FromTeams2ServerAnonymous	✓			✓	✓
⊗	999	✓			✓	
⊗	112	✓			✓	
⊗	18000	✓			✓	
⊗	From911TeamsServer	✓			✓	✓
⊗	From1TeamServer	✓			✓	✓
⊗	999Avaya	✓			✓	
⊗	112Avaya	✓			✓	
⊗	18000Avaya	✓			✓	
⊗	VentaFax	✓			✓	

New Entry

Name: 112Avaya

Send To:
- (•) Trunking Device: None
- ( ) Client:
- ( ) URI:
- ( ) Response:

Prioritize: ☐ Refer to Re-INVITE: ☐

Serial Hunting: [Add] [Delete]

E.164 Conversion rule: None Conversion mode: Add

Header Manipulations:

	Header	Value
⊗	From	'<sip:+' + $from.uri.user + '@' + $env.out_intf_host + '>'
⊗	Contact	'<sip:+' + $from.uri.user + '@' + $env.out_intf_host + ':' + $env.out_intf_port + '>'
⊗	P-Asserted-Identity	'<sip:+' + $from.uri.user + '@' + $env.out_intf_host + '>'

Header: Request-URI [Add]

Value:

[Update]

Actions

	Name	Send	Prio	Hunt	Header	Refer-To-ReINV
	CUCM	✓			✓	
	VirginMedia	✓			✓	
	Anonymous	✓			✓	
	Noplus	✓			✓	
	ToTeams	✓			✓	✓
	FromTeams2Server	✓			✓	✓
	FromTeams2ServerAnonymous	✓			✓	✓
	999	✓			✓	
	112	✓			✓	
	18000	✓			✓	
	From911TeamsServer	✓			✓	✓
	From1TeamServer	✓			✓	✓
	999Avaya	✓			✓	
	112Avaya	✓			✓	
	18000Avaya	✓			✓	
	VentaFax	✓			✓	
	New Entry					

Name: 18000Avaya

Send To:
- (●) Trunking Device: None
- ( ) Client:
- ( ) URI:
- ( ) Response:

Prioritize: ☐ Refer to Re-INVITE: ☐

Serial Hunting: Add Delete

E.164 Conversion rule: None Conversion mode: Add

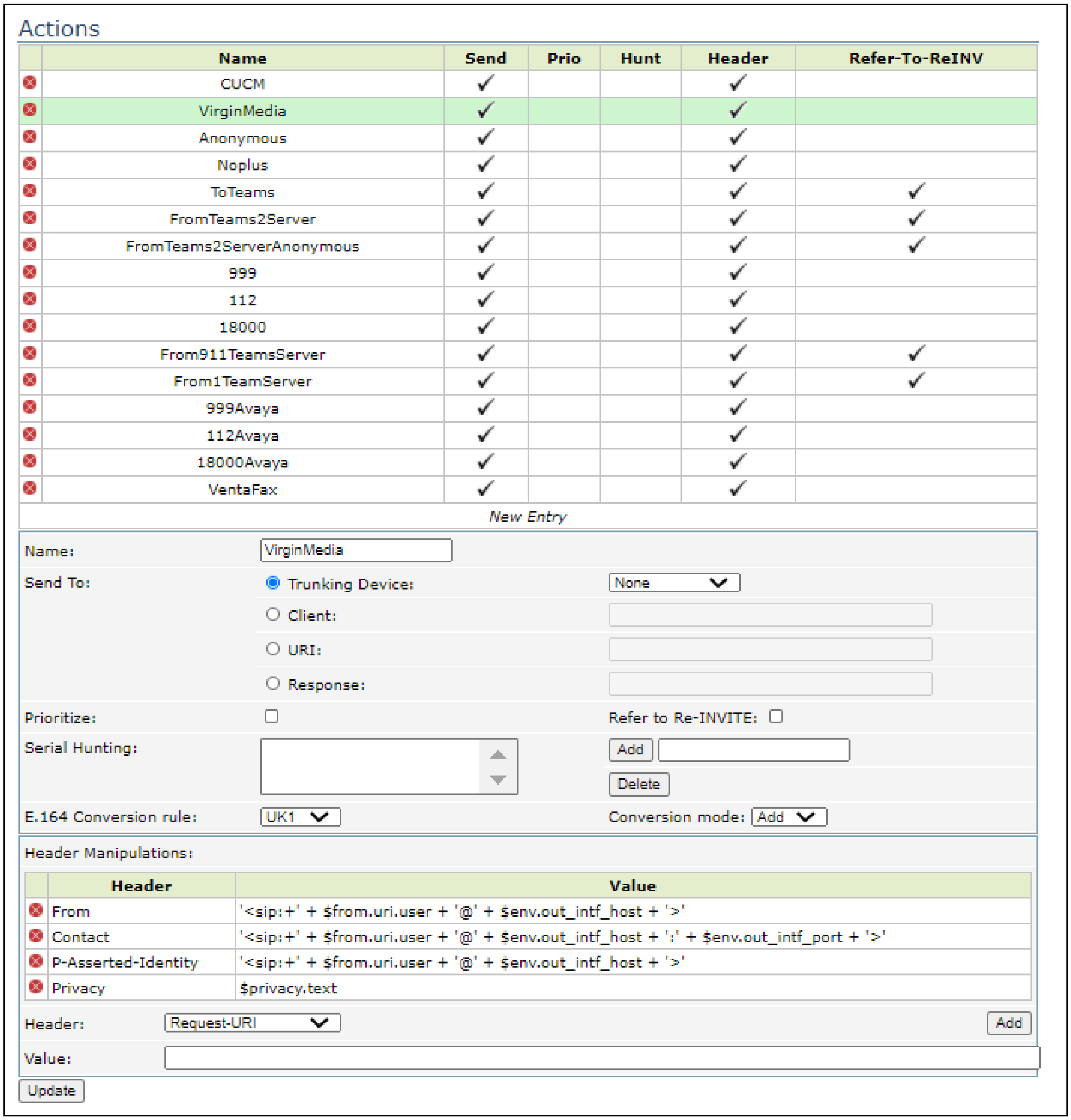
Header Manipulations:

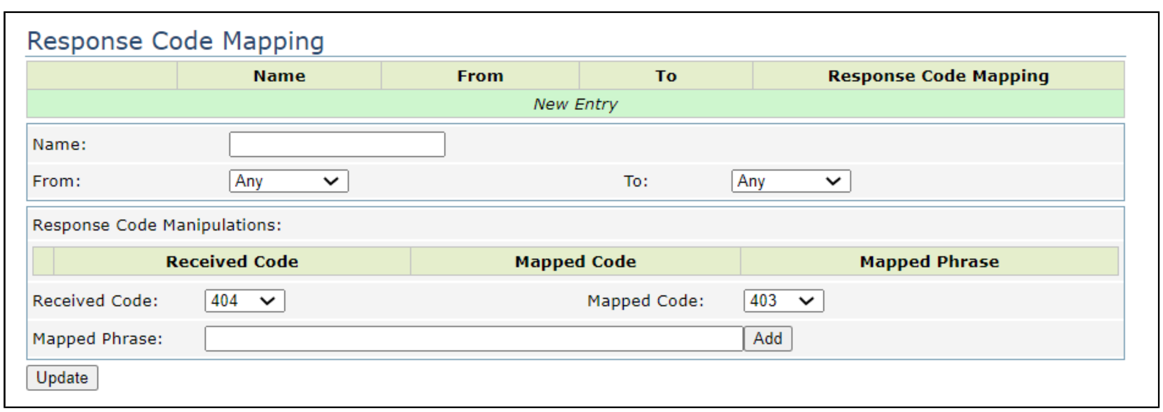
	Header	Value
	From	'<sip:+' + $from.uri.user + '@' + $env.out_intf_host + '>'
	Contact	'<sip:+' + $from.uri.user + '@' + $env.out_intf_host + ':' + $env.out_intf_port + '>'
	P-Asserted-Identity	'<sip:+' + $from.uri.user + '@' + $env.out_intf_host + '>'

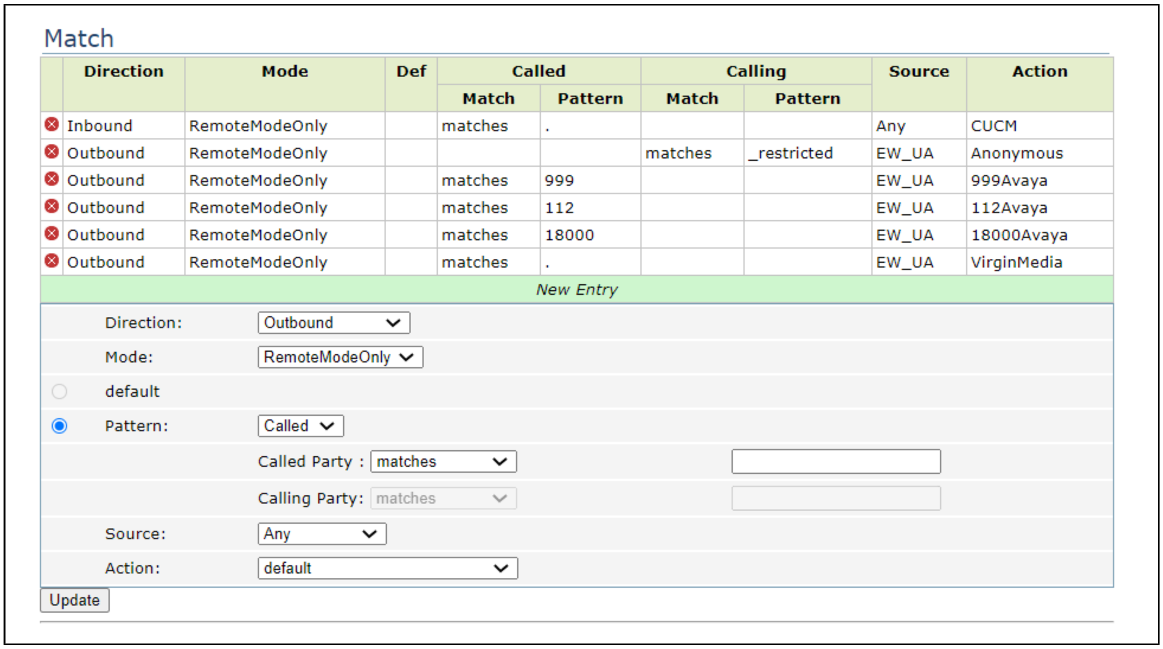
Header: Request-URI Add

Value:

Update


Actions
Name | Send | Prio | Hunt | Header | Refer-To-ReINV
CUCM
VirginMedia
Anonymous
Noplus
ToTeams
FromTeams2Server
FromTeams2ServerAnonymous
999
112
18000
From911TeamsServer
From1TeamServer
999Avaya
112Avaya
18000Avaya
VentaFax
New Entry
Name: VirginMedia
Send To:
Trunking Device: None
Client:
URI:
Response:
Prioritize:
Refer to Re-INVITE:
Serial Hunting:
Add
Delete
E.164 Conversion rule: UK1
Conversion mode: Add
Header Manipulations:
Header | Value
From | '<sip:+' + $from.uri.user + '@' + $env.out_intf_host + '>'
Contact | '<sip:+' + $from.uri.user + '@' + $env.out_intf_host + ':' + $env.out_intf_port + '>'
P-Asserted-Identity | '<sip:+' + $from.uri.user + '@' + $env.out_intf_host + '>'
Privacy | $privacy.text
Header: Request-URI
Add
Value:
Update

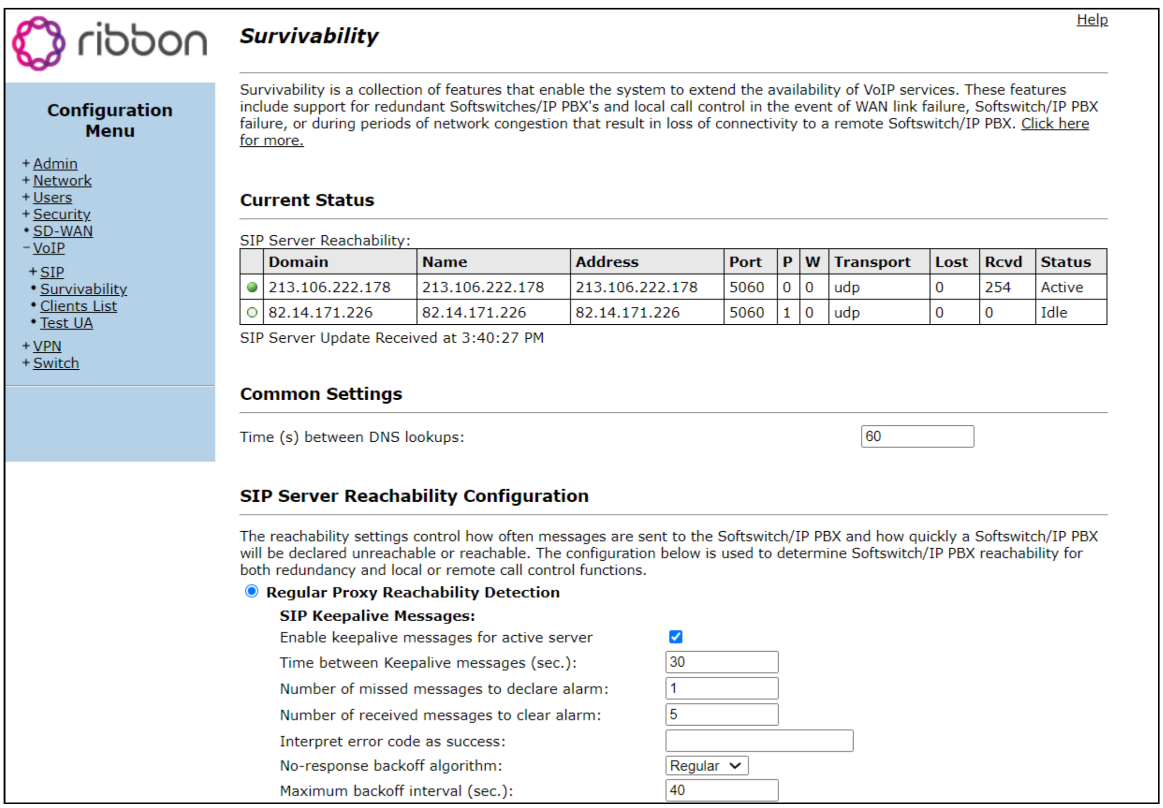

Response Code Mapping
Name | From | To | Response Code Mapping
New Entry
Name:
From: Any
To: Any
Response Code Manipulations:
Received Code | Mapped Code | Mapped Phrase
Received Code: 404
Mapped Code: 403
Mapped Phrase:
Add
Update

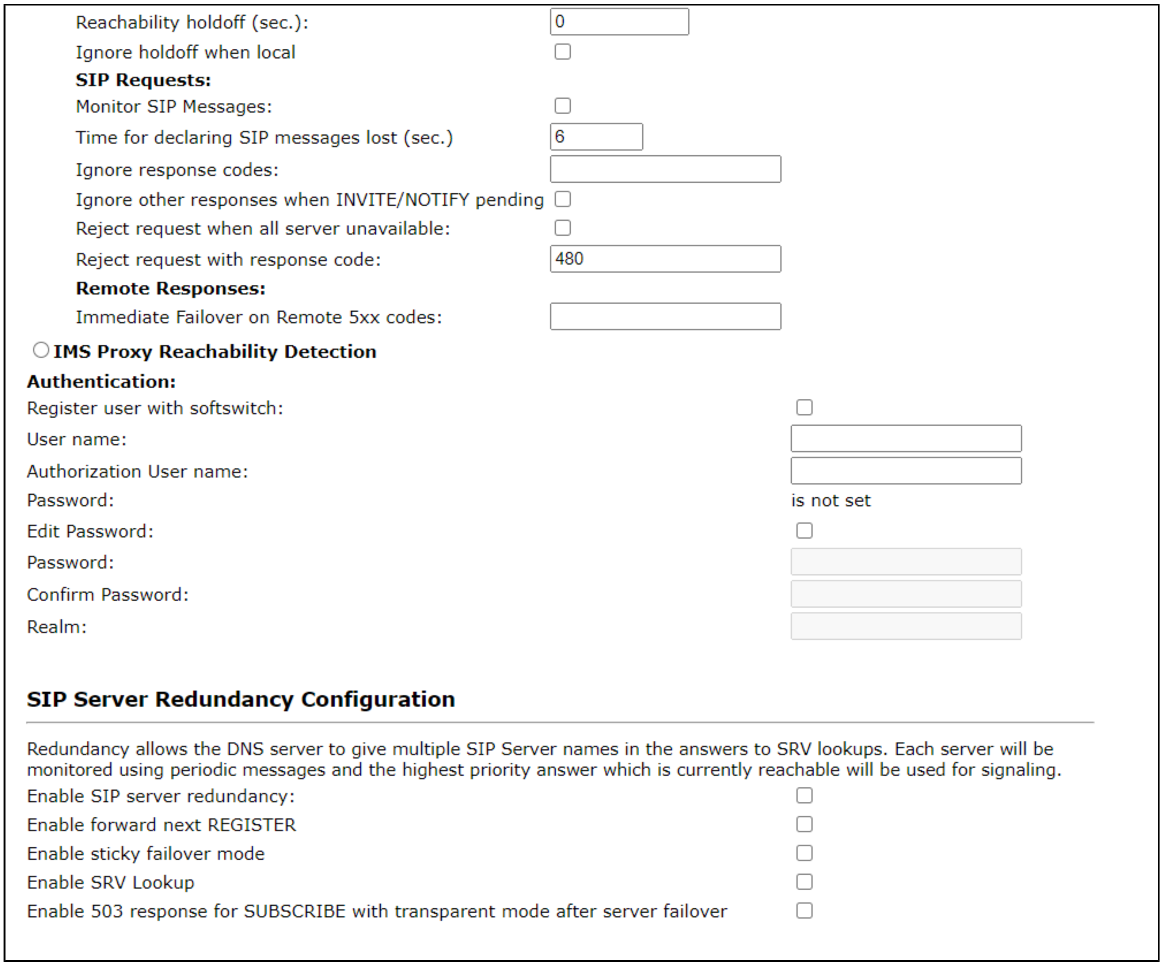


Survivability

1. Select **VoIP > Survivability**.
2. Configure the parameters.

Figure 17: Survivability

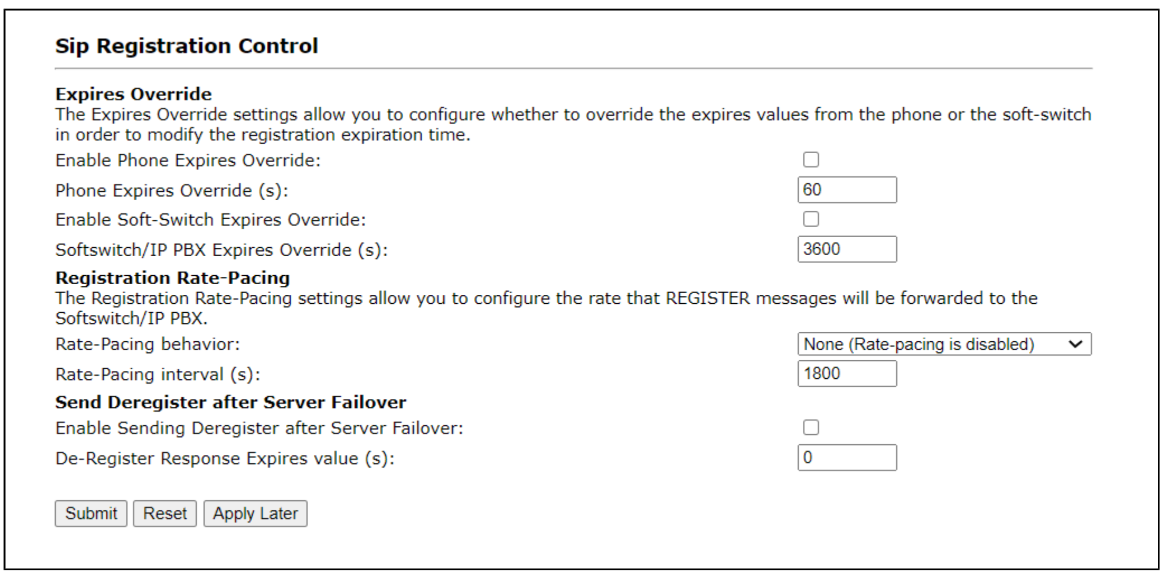




Reachability holdoff (sec.): 0
Ignore holdoff when local
SIP Requests:
Monitor SIP Messages:
Time for declaring SIP messages lost (sec.) 6
Ignore response codes:
Ignore other responses when INVITE/NOTIFY pending
Reject request when all server unavailable:
Reject request with response code: 480
Remote Responses:
Immediate Failover on Remote 5xx codes:
IMS Proxy Reachability Detection
Authentication:
Register user with softswitch:
User name:
Authorization User name:
Password: is not set
Edit Password:
Password:
Confirm Password:
Realm:

SIP Server Redundancy Configuration

Redundancy allows the DNS server to give multiple SIP Server names in the answers to SRV lookups. Each server will be monitored using periodic messages and the highest priority answer which is currently reachable will be used for signaling.
Enable SIP server redundancy:
Enable forward next REGISTER
Enable sticky failover mode
Enable SRV Lookup
Enable 503 response for SUBSCRIBE with transparent mode after server failover



Sip Registration Control

Expires Override
The Expires Override settings allow you to configure whether to override the expires values from the phone or the soft-switch in order to modify the registration expiration time.
Enable Phone Expires Override:
Phone Expires Override (s): 60
Enable Soft-Switch Expires Override:
Softswitch/IP PBX Expires Override (s): 3600
Registration Rate-Pacing
The Registration Rate-Pacing settings allow you to configure the rate that REGISTER messages will be forwarded to the Softswitch/IP PBX.
Rate-Pacing behavior: None (Rate-pacing is disabled)
Rate-Pacing interval (s): 1800
Send Deregister after Server Failover
Enable Sending Deregister after Server Failover:
De-Register Response Expires value (s): 0

Submit Reset Apply Later

Test Results

The following table provides Ribbon's test results of the scenarios that Virgin Media requires for its customers.

S. No	Procedure	Observation	Result	Comment

IOP1	Vendor's eSBC response to SIP OPTIONS messages from SBC	No calls are required for this test. Capture the SIP trace for approximately 60 seconds and check for correct signaling. For each eSBC, the SBC periodically sends an OPTIONS request to the vendor's eSBC to check if its SIP stack is reachable. If the IP-PBX sends a SIP response 200 OK, the SIP trunk is placed or remains in an In-Service state. **Example:** OPTIONS sip:ping@<ip-pbx_IP_Addr>:5060 SIP/2.0	Pass	
IOP2	SBC response to SIP OPTIONS messages from vendor eSBC	No calls are required for this test. Capture SIP trace for approximately 60 seconds (depending on agreement) and check for correct signaling. **Vendor's eSBC Setup for Solution IP.Addr Mode** 1. The eSBC is configured to send OPTIONS messages to the SBC periodically. 2. The SBC responds with SIP response 200 OK. **Example:** "OPTIONS sip:ping@192.168.1.10:5060 SIP/2.0" • Verify that the eSBC can simultaneously send SIP OPTIONS messages to both solution SBC addresses.	Pass	
IOP4	Basic test call from IP-PBX to PSTN line through SBC-A (using SBC-A IPV4 IP address)	1. The IP-PBX line initiates the call. 2. The call is answered. 3. The IP-PBX line terminates the call. **Vendor's eSBC Setup for Solution IP.Addr Mode** A call progresses successfully when: • A call is received from the IP-PBX. • An Invite is seen from the eSBC to SBC-A. • Proxy authentication challenge is returned to the eSBC. • A Re-invite with correct credentials is received from the eSBC. **Example:** Request-Line: INVITE sip:<B-party>@<SBC-A ip.addr TBD>:5060 SIP/2.0 To: sip:<B-Party>@<SBC-A ip.addr TBD> • Check the Wireshark trace to confirm that the G.711 A law codec with 10 or 20ms packetization is used. • Verify that the INVITE contains the Session-Expires header, and the INVITE is syntactically correct. • Check the Supported Header to ensure that it supports the timer. Ensure that the response in the 200 OK is compatible with the INVITE. Also, verify that the Required Header contains the timer.	Pass	

IOP5	Basic test call from IP-PBX to PSTN line through SBC-B (using SBC-B IPV4 ip address) Vendor to configure eSBC so that it used secondary SBC (SBC_B) for this test Once the test completes, configure eSBC to use Primary SBC-A for calls to route to.	1. The IP-PBX line initiates the call. 2. The call is answered. 3. The IP-PBX line terminates the call. **Vendor's eSBC Setup for Solution IP.Addr Mode** A call progresses successfully when: • A call is received from the IP-PBX. • An Invite is seen from the eSBC to SBC-B. • Proxy authentication challenge is returned to the eSBC. • A re-invite with correct credentials is received from the eSBC. **Example:** Request-Line: INVITE sip:<B-party>@<SBC-B ip.addr TBD>: 5060 SIP/2.0 To: sip:<B-Party>@<SBC-B ip.addr TBD> Check the Wireshark trace to confirm that the G.711 A law codec with 10ms or 20ms packetization is used.	Pass	
IOP7b	Called Number format - vendor's eSBC to soft switch number normalization - Global Dial Plan Test eSBC capability to send the called number in one of the following Global number formats (user part of Request & To URIs): 0yyyyyyyyyy (where y refers to any number, calling party = national) +44yyyyyyyyyy (where y refers to any number, calling party = national) +yyyyyyyyyy (where y refers to any number, calling party = international) yyyyyyyyyy (where y refers to any number, calling party = unknown)	Configure the SBC for the Global calling plan. 1. The IP-PBX line initiates a call to the PSTN line. 2. The call is answered. 3. The IP-PBX line terminates the call. 4. Configure the eSBC to present the called number in the user part of the **Request & To URIs** and send it in one of the following formats: • 0yyyyyyyyyy (where y refers to any number, calling party = national) • +44yyyyyyyyyy (where y refers to any number, calling party = national) • +yyyyyyyyyy (where y refers to any number, calling party = international) • yyyyyyyyyy (where y refers to any number, calling party = unknown)	Pass	
IOP8b	Calling Number format - vendor's eSBC to soft switch number normalization - Global Dial Plan Test eSBC capability to send the calling number in one of the following Global number formats (user part of From & PAI URIs): 0yyyyyyyyyy (where y refers to any number, calling party = national) +44yyyyyyyyyy (where y refers to any number, calling party = national) 00yyyyyyyyyy (where y refers to any number, calling party = international) yyyyyyyyyy (where y refers to any number, calling party = unknown)	Configure the SBC for the Global calling plan. 1. The IP-PBX line initiates a call to the PSTN line. 2. The call is answered. 3. The IP-PBX terminates the call. 4. Configure the eSBC to present the calling number in the user part of the **From & PAI URIs** and send it in one of the following formats: • 0yyyyyyyyyy (where y refers to any number, calling party = national) • +44yyyyyyyyyy (where y refers to any number, calling party = national) • 00yyyyyyyyyy (where y refers to any number, calling party = international) • yyyyyyyyyy (where y refers to any number, calling party = unknown)	Pass	

IOP9b	Called Number format - soft switch to eSBC number normalization - Global Dial Plan Test eSBC capability of accepting the called number in one of the following Global number formats (user part of Request & To URIs): +44yyyyyyyyy (where y refers to any number, calling party = national) +yyyyyyyyy (where y refers to any number, calling party = international) yyyyyyyyyy (where y refers to any number, calling party = unknown)	Configure the SBC for the Global calling plan. 1. The PSTN line initiates a call to the IP-PBX line. 2. The call is answered. 3. The PSTN line terminates the call. 4. Configure the eSBC to accept the called number in the user part of the **Request & To URIs** in one of the following formats: • +44yyyyyyyyy (where y refers to any number, calling party = national) • +yyyyyyyyy (where y refers to any number, calling party = international) • yyyyyyyyyy (where y refers to any number, calling party = unknown) 5. Verify that the INVITE contains the Session-Expires header and the INVITE is syntactically correct. 6. Check the Supported Header to ensure that it supports the timer. Ensure that the response in the 200 OK is compatible with the INVITE. Also, verify that the Required Header contains the timer.	Pass	
IOP10b	Calling Number format - soft switch to eSBC number normalization - Global Dial Plan Test eSBC capability of accepting the calling number in one of the following Global number formats (user part of From & PAI URIs): +44yyyyyyyyy (where y refers to any number, calling party = national) +yyyyyyyyy (where y refers to any number, calling party = international) yyyyyyyyyy (where y refers to any number, calling party = unknown)	Configure the SBC for the Global calling plan. 1. The PSTN line initiates a call to the IP-PBX line. 2. The call is answered. 3. The PSTN line terminates the call. 4. Configure the eSBC to accept the calling number in the user part of the **Request & To URIs** in one of the following formats: • +44yyyyyyyyy (where y refers to any number, calling party = national) • +yyyyyyyyy (where y refers to any number, calling party = international) • yyyyyyyyyy (where y refers to any number, calling party = unknown)	Pass	
IOP11	Emergency Call Handling -IP-PBX Line to PSTN - UK Emergency call 999	1. Make a call from the IP-PBX line to the Emergency services using 999. 2. The call is answered. 3. Either party terminates the call. **Example:** Request-Line: INVITE sip:999@<SBC-A ip.addr TBD>:5060 SIP/2.0 To: <sip:999@<SBC-A ip.addr TBD>> From: <sip:<A-party>@<IP-PBX IP.Addr>	Pass	

IOP12	Emergency Call Handling -IP-PBX Line to PSTN - UK Emergency call 112	1. Make a call from the IP-PBX line to the Emergency services using 112. 2. The call is answered. 3. Either party terminates the call. **Example:** Request-Line: INVITE sip:112@<SBC-A ip.addr TBD>:5060 SIP/2.0 To: <sip:112@<SBC-A ip.addr TBD>> From: <sip:<A-party>@<IP-PBX IP.Addr>	Pass	
IOP13	Emergency Call Handling -IP-PBX Line to PSTN - UK Emergency call 18000 - Text Direct	1. Make a call from the IP-PBX line using a text direct set to the Emergency services using 18000. 2. The call is answered. 3. Either party terminates the call. **Example:** Request-Line: INVITE sip:18000@<SBC-A ip.addr TBD>:5060 SIP/2.0 To: <sip:18000@<SBC-A ip.addr TBD>> From: <sip:<A-party>@<IP-PBX IP.Addr>	Pass	
IOP14	IP-PBX Line to PSTN - Call answer - Originator disconnect	1. Make a call from the IP-PBX line to the PSTN line. 2. Answer the call. 3. The IP-PBX line terminates the call.	Pass	
IOP15	PSTN calls SIP #1, SIP #1 conferences in SIP #2	1. Make a call from the IP-PBX line to the PSTN line. 2. Answer the call. 3. The PSTN line terminates the call.	Pass	
IOP16	IP-PBX Line to PSTN - Busy subscriber	1. Make a call from the IP-PBX line to a busy PSTN line (without divert on busy). 2. Wait for the soft switch to return the busy response. 3. Ensure that the eSBC is not recursive. 4. Set up the call via the secondary SIP trunk.	Pass	
IOP17	IP-PBX Line to PSTN - No answer timeout test	1. Make a call from the IP-PBX line to a PSTN line (without divert on no answer). 2. Do not answer the call. 3. Wait for the soft switch to return the no answer timeout response. 4. Ensure that the eSBC is not recursive. 5. Set up the call via the secondary SIP trunk.	Pass with Caveat	It is not possible to modify the timer in Avaya CM 7.1 (3 minutes by default) because we use the terminal Avaya One-X Communicator. At the same time Virgin Media sent a CANCEL. Since we have two CANCELs at the same time, VM sent us 481 Call Leg does not Exist.

IOP18	IP-PBX Line to PSTN - Subscriber not reachable Vendor to call 01189111111	1. Make a call from the IP-PBX line to an invalid number. 2. Wait for the soft switch to return a response. 3. Ensure that the eSBC is not recursive. 4. Set up the call via the secondary SIP trunk.	Pass	
IOP19	PSTN Line to IP-PBX - Call answer - Originator disconnect	1. Make a call from a PSTN line to an IP-PBX line. 2. Answer the call. 3. The originator disconnects the call.	Pass	.
IOP20	PSTN Line to IP-PBX - Call answer - Terminator disconnect	1. Make a call from a PSTN line to an IP-PBX line. 2. Answer the call. 3. The IP-PBX line terminates the call.	Pass with Caveat	All calls from PSTN to EdgeMarc show a duplication ptime attribute in SDP responses. This does not appear to impact on call (passed with a caveat) but is bad practice.
IOP21	PSTN Line to IP-PBX - Busy subscriber	1. Make a call from a PSTN line to a busy IP-PBX line (without divert on busy). 2. Wait for the IP-PBX to return the busy response.	Pass	
IOP22	PSTN Line to IP-PBX - No answer timeout test, Invoked by PBX	1. Make a call from a PSTN line to an IP-PBX line (without divert on no answer). 2. Wait for the IP-PBX to return the no answer timeout response.	Pass	
IOP23	PSTN Line to IP-PBX - Subscriber not reachable	1. Make a call from a PSTN line to an invalid number/unprogrammed DDI on the IP-PBX. 2. Wait for the IP-PBX to return a response.	Pass	
IOP24	Verify CLIP service on IP-PBX line (incoming call from PSTN)	1. Make a call from the PSTN line to the IP-PBX line. 2. The PSTN line is set to allow the CLI presentation. Check that the CLI is delivered as expected. 3. Either party terminates the call.	Pass	
IOP25	Verify CLIR service on IP-PBX line (incoming call from PSTN)	1. Make a call from the PSTN line to the IP-PBX line. 2. PSTN line is set to restrict the CLI presentation. Check that CLI is not delivered as expected. 3. Either party terminates the call.	Pass	

IOP26	Verify CLIP service on PSTN line (outgoing call from IP-PBX, From)	1. Ensure Virgin Media agrees with the number used in the From header, and enter the number into the soft switch database for screening. 2. Make a call from an IP-PBX line to a PSTN line. 3. Ensure the eSBC configuration enables the IP-PBX line to send the From header containing the Calling Line ID (CLI) in the INVITE. 4. Ensure that the eSBC allows the presentation of its CLI, using the privacy-header (Privacy: none or privacy-header not present). 5. Ensure that the expected CLI is presented to the PSTN line. 6. Either party terminates the call.	Pass	
IOP27	Verify CLIP service on PSTN line (outgoing call from IP-PBX, PAI/PPI) Vendor to ensure PAI number is different to that from which the call originates.	1. Ensure Virgin Media agrees with the number used in the PAI/PPI header, and enter the number into the soft switch database for screening. 2. Make a call from an IP-PBX line to a PSTN line. 3. Ensure that the eSBC configuration enables the IP-PBX line to send the PAI /PPI header containing the Calling Line ID (CLI) in the INVITE. Note that if the PAI header is populated, it will be used in preference to the From header. 4. Ensure that the eSBC allows the presentation of its CLI, using the privacy-header (Privacy: none or privacy-header not present). 5. Ensure that the expected CLI is presented to the PSTN line. 6. Either party terminates the call.	Pass	

IOP28	Verify CLIR service on PSTN line (outgoing call from IP-PBX)	1. Ensure Virgin Media agrees with the number used in the From/PAI header, and enter the number into the soft switch database for screening. 2. Make a call from an IP-PBX line to a PSTN line. 3. Ensure that the eSBC configuration enables the IP-PBX line to send the From and/or PAI header, containing either the Calling Line ID or obscured information in the INVITE. **Example:** From: "user751000" <sip:+441256751000@192.168.1.10>; tag=12345 From: "Anonymous" <sip:anonymous@anonymous.invalid>; tag=12345 • Ensure that the eSBC restricts the presentation of its CLI, using the privacy-header (Privacy: id or Privacy: user or Privacy: user;id). • Ensure that CLI is not presented to the PSTN line. • Either party terminates the call.	Pass	
IOP29	Verify Call Forward Immediate (unconditional) on an IP-PBX line (Incoming call from PSTN, call forward terminates within IP-PBX)	1. Make a call from a PSTN line to an IP-PBX line with Call Forward to a line within the same IP-PBX. 2. Answer the call. 3. Either party terminates the call. The IP-PBX does not have configuration settings to send SIP status 181 messages to the soft switch.	Pass	
IOP30	Verify Call Forward Immediate (unconditional) on an IP-PBX line (Incoming call from PSTN, call forward terminates PSTN)	1. Make a call from a PSTN line to an IP-PBX line with Call Forward to a line in the PSTN. 2. Answer the call. 3. Either party terminates the call.	Pass	
IOP31	Verify Call Forward Busy on IP-PBX line (Incoming call from PSTN, call forward terminates within IP-PBX)	1. Make a call from a PSTN line to an IP-PBX line with Call Forward Busy (or equivalent) to a line within the IP-PBX. 2. Answer the call. 3. Either party terminates the call.	Pass	
IOP32	Verify Call Forward No-answer on IP-PBX line (Incoming call from PSTN, call forward terminates within IP-PBX)	1. Make a call from a PSTN line to an IP-PBX line with Call Forward No-answer (or equivalent) to a line within the IP-PBX. 2. Answer the call. 3. Either party terminates the call.	Pass	

IOP33	Verify Call Hold Service on IP-PBX (Incoming call from PSTN)	1. Make a call from a PSTN line to an IP-PBX line with Call Hold. 2. Answer the call. 3. IP-PBX line places the call on hold. 4. Leave the call on hold for 30 seconds and then retrieve the call. 5. Ensure speech path is re-established in both directions. 6. Either party terminates the call.	Pass	
IOP34	Verify three-party conference service on IP-PBX (Incoming call from PSTN, third party within IP-PBX)	1. Make a call from a PSTN line to an IP-PBX line with a three-party conference. 2. Answer the call. 3. IP-PBX line uses the three-party conference facility to place the PSTN line on hold while dialing the third party (on another IP-PBX line). 4. Once the third party answers the call, place the three parties in a conference. 5. Ensure that all parties have a two-way speech path. Keep the speech path open for at least 20 seconds. 6. Either party terminates the call.	Pass	
IOP35	Verify three-party conference service on IP-PBX (Incoming call from PSTN, third party PSTN)	1. Make a call from a PSTN line to an IP-PBX line with a three-party conference. 2. Answer the call. 3. IP-PBX line uses the three-party conference facility to place the PSTN line on hold while dialing the third party (on another IP-PBX line). 4. Once the third party has answered the call, place the three parties in a conference. 5. Ensure that all parties have a two-way speech path. Keep the speech path open for at least 20 seconds. 6. Either party terminates the call.	Pass	
IOP36	Verify do-not-disturb service on IP-PBX line (Incoming call from PSTN)	1. The call does not ring. 2. PSTN line receives an appropriate announcement or tone. 3. Record the SIP status received from the IP-PBX.	Pass	

IOP37	Verify Call park service on IP-PBX line (Incoming call from PSTN)	1. Make a call from a PSTN line to an IP-PBX line A with the Call Park (or equivalent) feature active. 2. Answer the call. 3. Place the call in the Park condition. 4. After 10 seconds, retrieve the call from the IP-PBX line B, using the Call Park pick-up code. 5. Ensure the speech path is re-established in both directions. 6. Either party terminates the call.	Pass	
IOP38	Verify Call Waiting on an IP-PBX line, involving a PSTN line	1. Make a call from PSTN line A to an IP-PBX line with Call Waiting active. 2. Answer the call. 3. Make a call from the PSTN line B to the same IP-PBX line, which should receive an indication that a second call is waiting. 4. PSTN line B receives the ringback tone. 5. IP-PBX line answers the call from the PSTN line B. 6. PSTN line A should receive an appropriate indication that they are now on hold. 7. IP-PBX line toggles the call back to PSTN line A. 8. Ensure the speech path is re-established in both directions and that PSTN line B receives an appropriate indication that they are now on hold. 9. Either party terminates the call.	Pass	
IOP39	Verify DTMF transmission from /to IP-PBX - Inband	1. Configure the IP-PBX/eSBC to send the DTMF transmission in-band. 2. Make a call from an IP-PBX line to a PSTN line. 3. Answer the call. 4. PSTN line presses each of the keys on the number pad in turn. Note the far-end experience. 5. IP-PBX line presses each of the keys on the number pad in turn. Note the far-end experience. The received DTMF tone is reflective of the length of time the key was pressed.	Not executed	The SBC team from VM changed this to not executed as IN Band DTMF tones are not currently supported and will require a feature to be added to the code.

IOP40	Verify DTMF transmission from /to IP-PBX - RFC 2833 - Telephone-event	1. Configure the IP-PBX/eSBC to send the DTMF transmission, using RFC 2833 - telephone-event. 2. Make a call from an IP-PBX line to a PSTN line. 3. Answer the call. 4. PSTN line presses each of the keys on the number pad in turn. Note the far-end experience. 5. IP-PBX line presses each of the keys on the number pad in turn. Note the far-end experience. The received DTMF tone is reflective the length of time the key was pressed.	Pass	
IOP41	T.38 Fax transmission mode - PSTN to IP-PBX origination	1. Configure the ATA/IP-PBX/eSBC so that the Fax transmission is sent using the T.38 Version 0 Fax transmission mode. 2. Make a call from an PSTN line to an IP-PBX line. 3. Answer the call. 4. Fax transmission is completed and the call is terminated by either of the end terminal devices. 5. Ensure the Wireshark trace shows it is using the T.38 Fax Transmission. 6. Check that the fax is transmitted and received as expected.	Failed	The EM6000 only sent a Re-INVITE when the fax machine was connected directly to FXS ports to negotiate the T.38.
IOP42	T.38 Fax transmission mode - IP-PBX to PSTN origination	1. Configure the ATA/IP-PBX/eSBC so that the Fax transmission is sent using the T.38 Version 0 Fax transmission mode. 2. Make a call from an IP-PBX line to a PSTN line. 3. Answer the call. 4. Fax transmission is completed and the call is terminated by either of the end terminal devices. 5. Ensure Wireshark trace shows that the T. 38 Fax Transmission is used. 6. Check that the fax is transmitted and received as expected.	Failed	The EM6000 only sent a Re-INVITE when the fax machine was connected directly to FXS ports to negotiate the T.38.

IOP43	In-band G.711 Fax transmission mode - PSTN to IP-PBX origination	1. Configure the ATA/IP-PBX/eSBC so that Fax transmission is sent using the in-band G.711 Fax transmission mode. 2. Make a call from a PSTN line to an IP-PBX line. 3. Answer the call. 4. Fax transmission is completed and the call is terminated by either of the end terminal devices. 5. Ensure the Wireshark trace shows that the in-band G.711 Fax Transmission is used. 6. Check that the fax is transmitted and received as expected.	Pass	
IOP44	In-band G.711 Fax transmission mode - IP-PBX to PSTN origination	1. Configure the ATA/IP-PBX/eSBC so that the Fax transmission is sent using the in-band G.711 Fax transmission mode. 2. Make a call from an IP-PBX line to a PSTN line. 3. Answer the call. 4. Fax transmission is completed and the call is terminated by either of the end terminal devices. 5. Ensure the Wireshark trace shows that the in-band G.711 Fax Transmission is used. 6. Check that the fax is transmitted and received as expected.	Pass	

IOP45	Test for call in progress audit function (response to in-call OPTIONS from soft switch to eSBC) and session refresh and response to UPDATE messages	1. Make a call from an IP-PBX line to a PSTN line. 2. Answer the call. 3. Leave the two parties in conversation for 35 minutes. 4. Ensure the Session-expires setting is 3600 or less. 5. Ensure both parties have two-way speech at the beginning and end of call. 6. Either party terminates the call. 7. Check the Wireshark trace to ensure that the in-call OPTIONS are sent by the soft switch and that the eSBC responds with the status 200OK. 8. Check if the eSBC sends any in-call audit SIP messages. 9. Check for session refresh Update or Re-Invite and correct response.	Pass	
IOP46	Test for four simultaneous calls: two inbound, two outbound calls Vendor to configure eSBC for Round robin to ensure calls go to both primary and secondary SBC.	1. Configure the eSBC so that successive calls route to alternate SBCs (round robin, cyclic, and so on). 2. Make four simultaneous calls: two inbound and two outbound calls. 3. Answer the calls and ensure two-way speech path for each call.	Not executed	Round robin outbound calls is not a feature of the EdgeMarc platform. Adding this feature requires a change in the EdgeMarc code.
IOP47	Test for eSBC endpoint restart-recovery	1. Restart the eSBC. 2. Ensure that, after recovery, inbound and outbound calls are successful.	Pass	
IOP48	Test for eSBC loss of Ethernet link and reconnection	1. Remove the Ethernet link between the eSBC and CE router. Leave it in this condition for at least 3 minutes. 2. Reconnect the Ethernet link and ensure that after approximately 2 minutes inbound and outbound calls are successful.	Pass	

IOP49	Test for the Primary SBC loss	**Note:** Contact an MSL engineer to carry out the following. 1. On the Primary SBC, carry out the ALLSTOP command to disable the SBC. 2. Make a call from the IP-PBX line to a PSTN Line. 3. Make sure that the call tries to route to the Primary SBC. On a non-response to the INVITE, the eSBC re-routes the call to the Secondary SBC. 4. Wait for call answer. 5. Either party terminates the call. **Note:** Contact an MSL engineer to carry out the following. 1. Restart the Primary SBC.	Pass	
IOP51	Test for Call forward Internal Busy	The following is an additional test to cover when vendors are using the Avaya Communication Manager 7.1: 1. PBX Subscriber 1 makes a call to PBX Subscriber 2, so that the PSTN to call the PBX subscriber 1 is Busy. 2. PSTN calls PBX user 1. The call should automatically go to voicemail after 10 seconds when call forwarding is off. 3. If VM is on another PBX Internal Line, the call should go to voicemail. 4. PSTN user listens to the voiceMail announcement, and leaves a clear message for PBX Subscriber 1 in VM. 5. If forwarded to voicemail, the PSTN terminates the call after hearing the VM announcement. 6. If forwarded to another user, either party terminates the call after checking that speech is clear in both directions.	Not executed	

IOP52	Test for Call forward internal on No Answer	The following is an additional test to cover when vendors are using the Avaya Communication Manager 7.1: 1. PSTN calls PBX user 1. 2. The PBX User 1 should not answer the call. 3. The call should automatically go to voicemail (VM), which is in another internal PBX line if call forwarding is turned off. 4. The call automatically goes to voicemail after 10 seconds. 5. The PSTN terminates the call after hearing the VM announcement. 6. If call forwarding is ON, the call is forwarded to another PBX user internal. 7. Check the speech quality and terminate the call after checking that speech is clear in both directions.	Not executed	
IOP53	Test for making a call from a PBX to a PSTN	1. Configure the eSBC to offer the T.38 in addition to G711A-law and G711-U law. 2. Make a call from the PBX to a PSTN. 3. Ensure the call is connected and dialog takes place for 10 minutes. 4. Check the Wireshark output. Confirm that the T.38 is not reflected in the protocol column after the call is connected for 7 minutes. 5. If the T.38 is reflected in the protocol column, take a note.	Pass	