Ribbon EdgeMarc 2900A configuration Cisco Unified Communication Manager CUCM

Table of Contents

Document Overview

This document provides a configuration guide for Ribbon EdgeMarc 2900A when connecting to Cisco Unified Communication Manager CUCM.

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

- For additional information on Cisco Unified Communication Manager, visit http://www.cisco.com.
- For additional information on Ribbon SBC, visit https://ribboncommunications.com/

Introduction

The interoperability compliance testing focuses on verifying inbound and outbound calls flows between Ribbbon EdgeMarc 2900A and the Cisco Unified Communication Manager CUCM platform.

Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring both the Ribbon SBCs and the third-party product. There will be steps that require navigating the third-party product as well as the Ribbon SBC Command Line Interface (CLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete 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 provided:

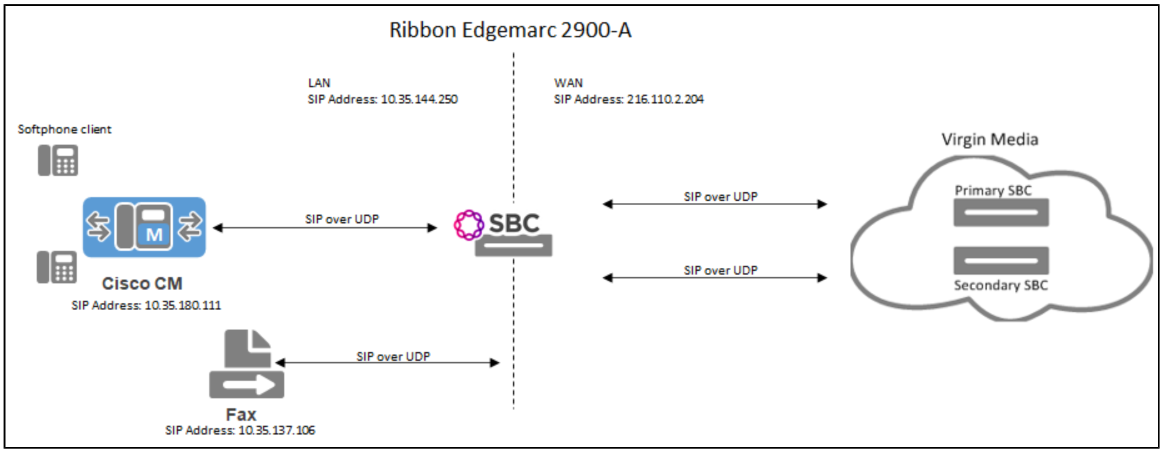
Table 1: Requirements

	Equipment	**Software Version**
Ribbon Communications	Ribbon EdgeMarc 2900A	V15.8.0
Third-party Equipment	Cisco Unified Communication Manager	12.0.1.21900-7
	Kapanga Softphone	1.00.2182d
	MicroSIP Softphone	3.19.29
	Cisco IP Communicator Softphone	8.6.2.0
	VentaFax Fax Machine	7.6.243.616
	NGT Lite	v.1.51

Reference Configuration

The following reference configuration shows connectivity between the third-party product and Ribbon EdgeMarc 2900A.

Figure 1: Reference Configuration



Support

For any questions regarding this document or the content herein, 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/outbound call
- Hold and Resume
- Call Forwarding
- FAX
- DTMF
- Conference Call
- Action on eSBC outage (restart of eSBC)
- Action on Loss of Virgin Media primary SBC

Cisco UCM 12 Configuration

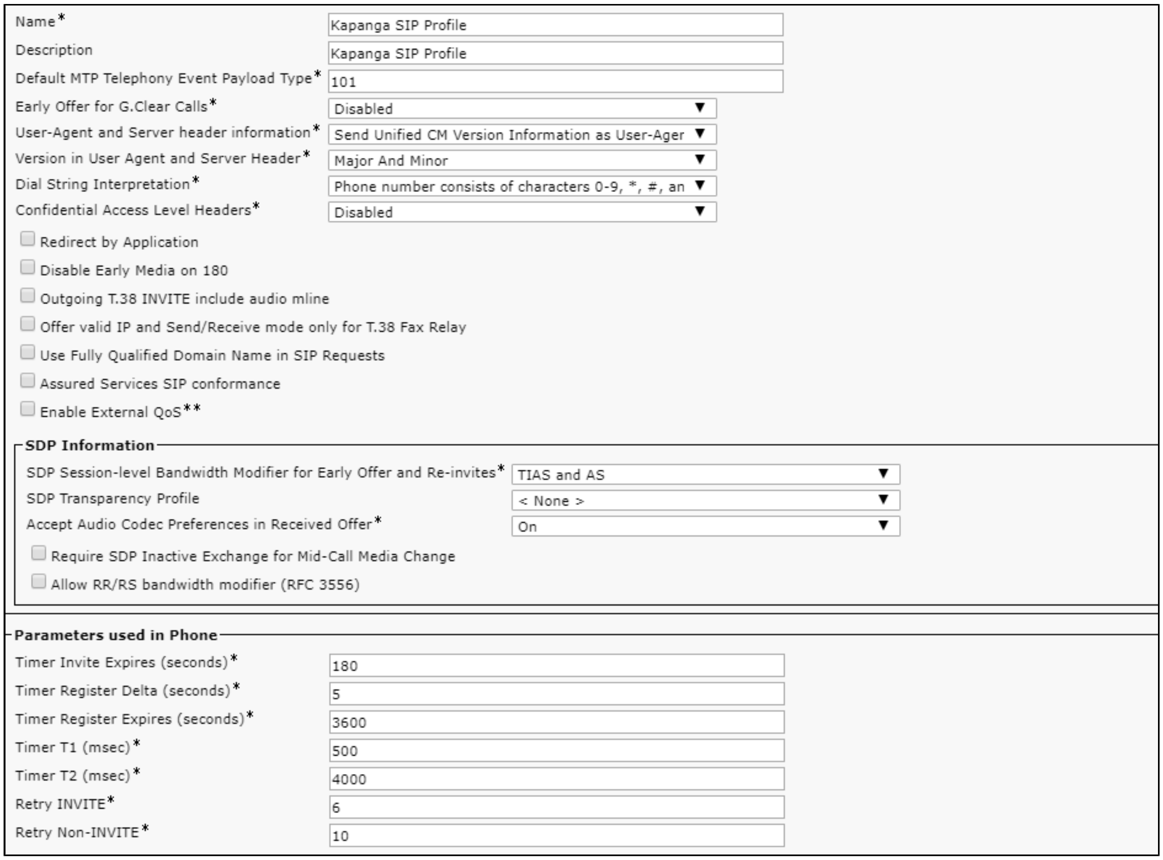
The following new configurations are included in this section:

1. SIP Profile
2. SIP Trunk Security Profile
3. Trunk
4. Route Group
5. Route List
6. Route Pattern

1. SIP Profile

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

Figure 2: SIP Profile



Name*	Kapanga SIP Profile
Description	Kapanga SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Ager
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, an
Confidential Access Level Headers*	Disabled

- [ ] Redirect by Application
- [ ] Disable Early Media on 180
- [ ] Outgoing T.38 INVITE include audio mline
- [ ] Offer valid IP and Send/Receive mode only for T.38 Fax Relay
- [ ] Use Fully Qualified Domain Name in SIP Requests
- [ ] Assured Services SIP conformance
- [ ] Enable External QoS**

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	< None >
Accept Audio Codec Preferences in Received Offer*	On

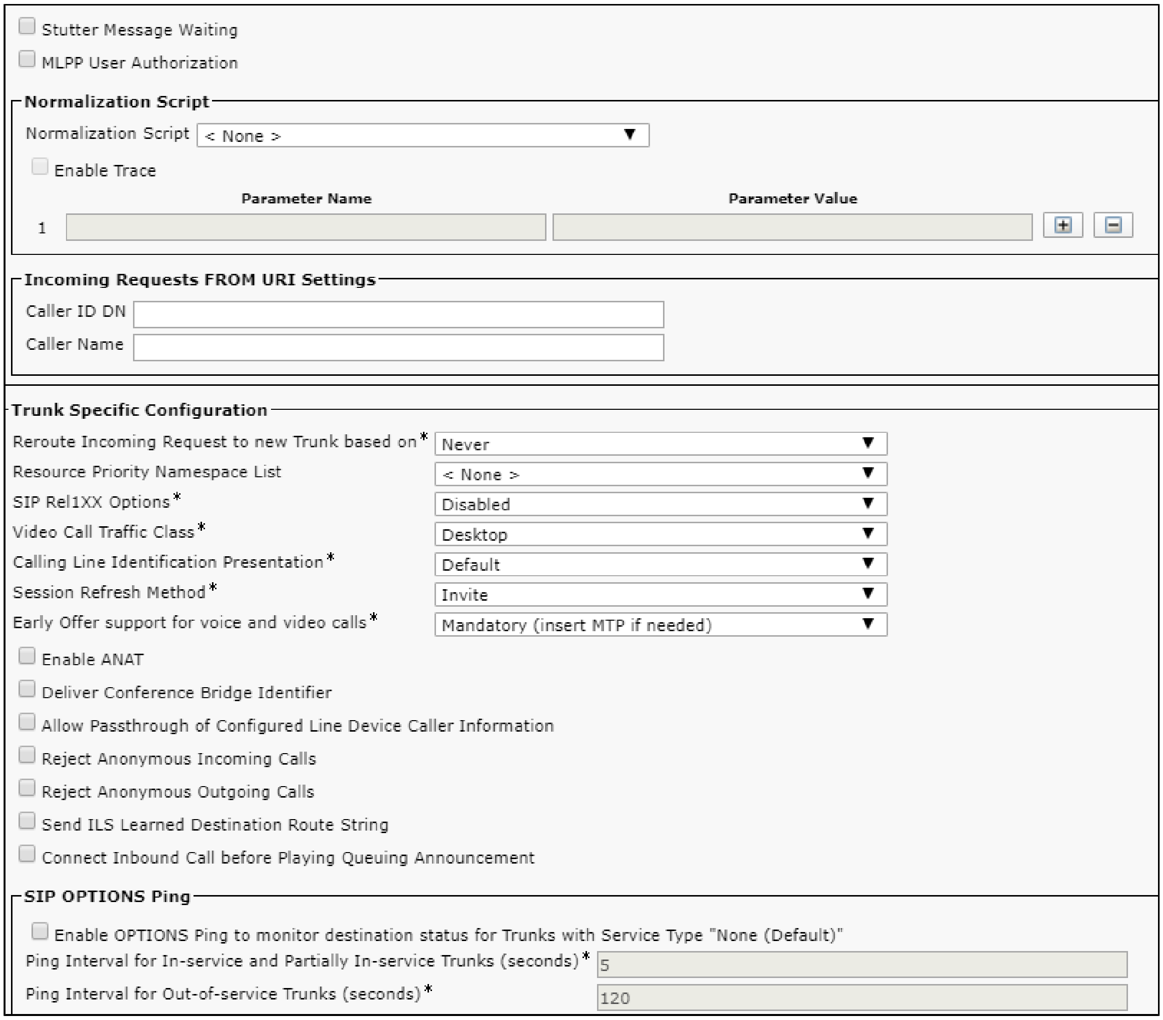
- [ ] Require SDP Inactive Exchange for Mid-Call Media Change
- [ ] Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10

Media Port Ranges	(•) Common Port Range for Audio and Video ( ) Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

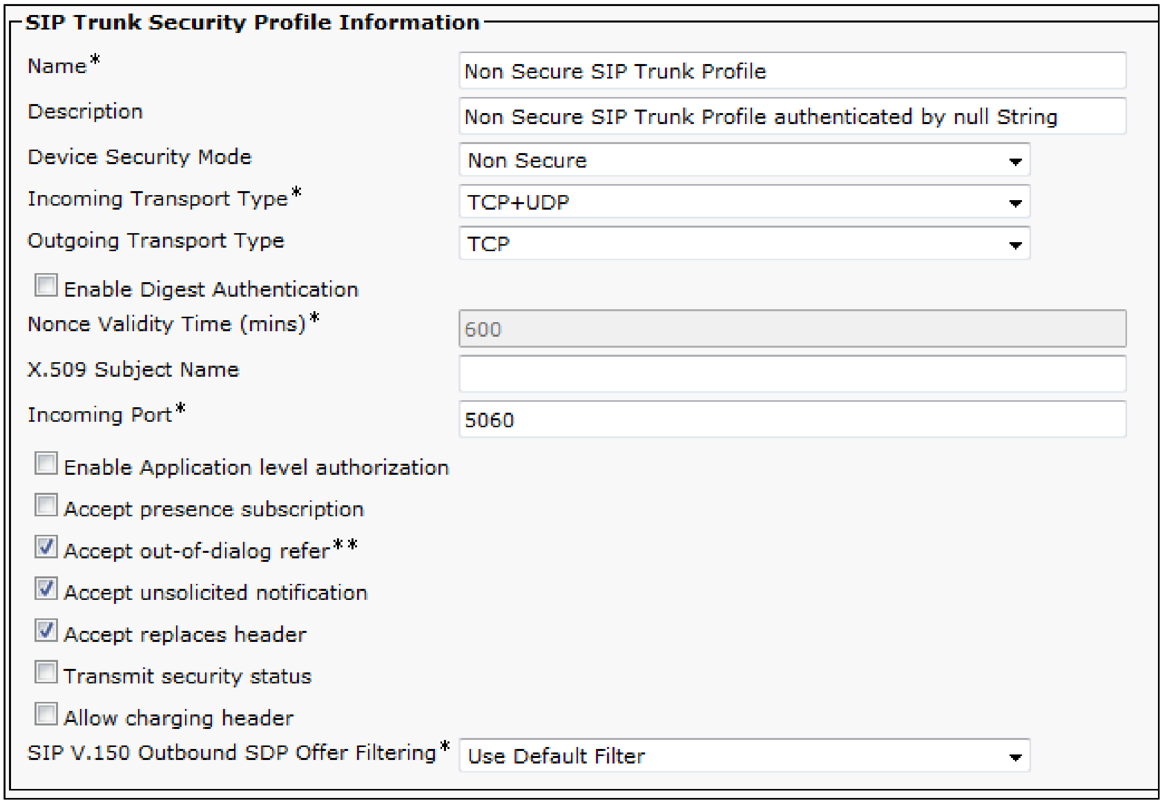
- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD



2. SIP Trunk Security Profile

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

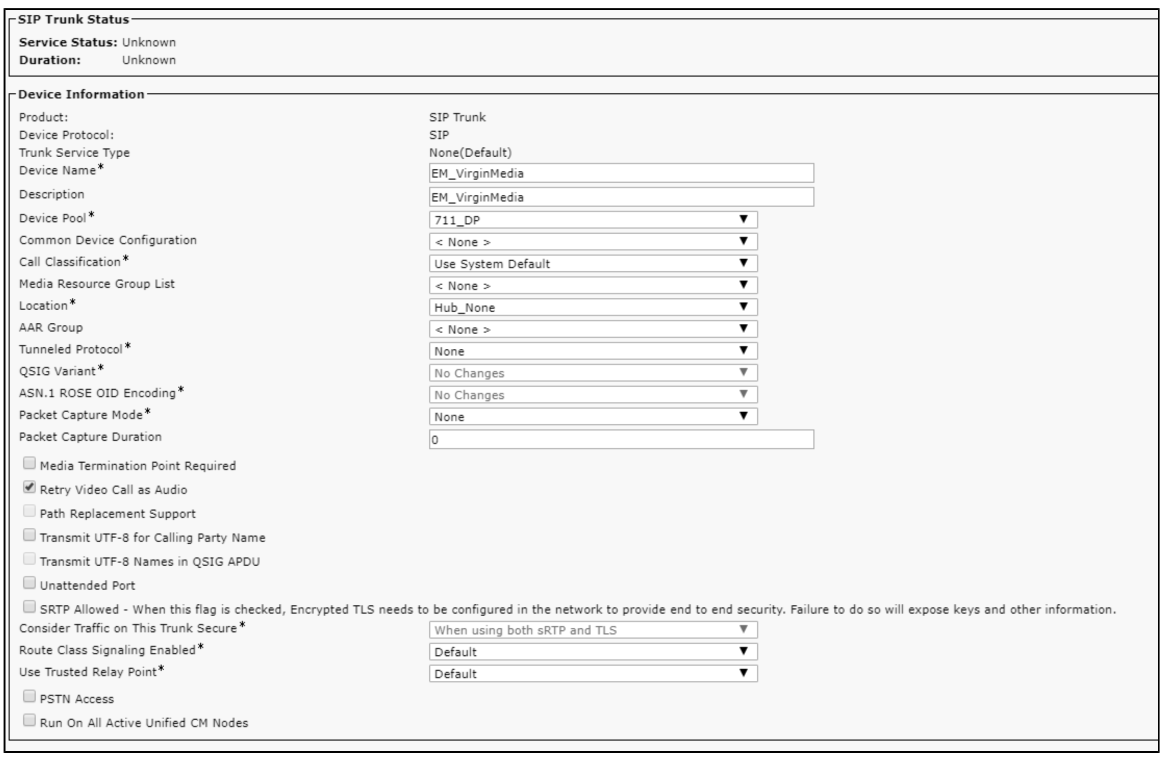
Figure 3: SIP Trunk Security Profile

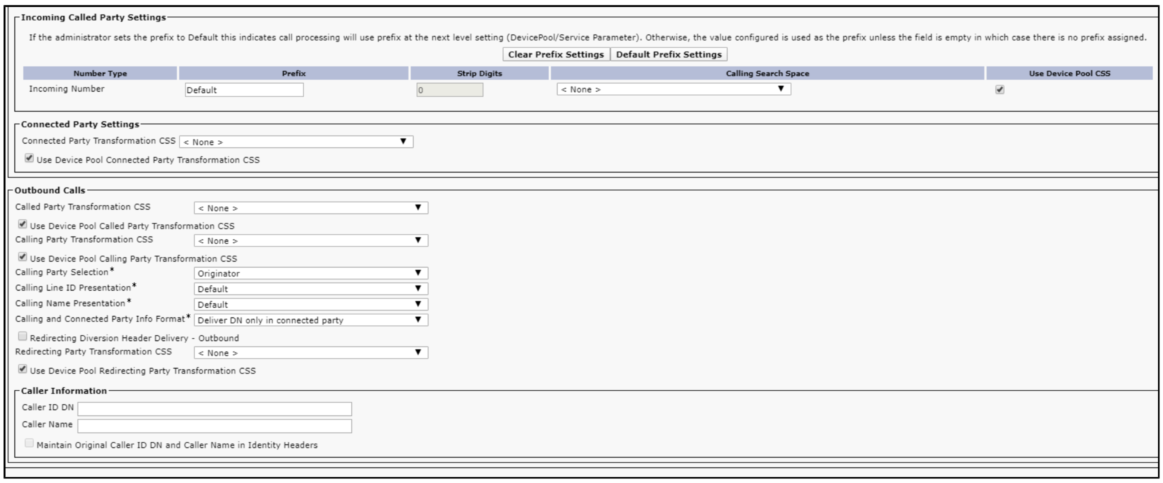


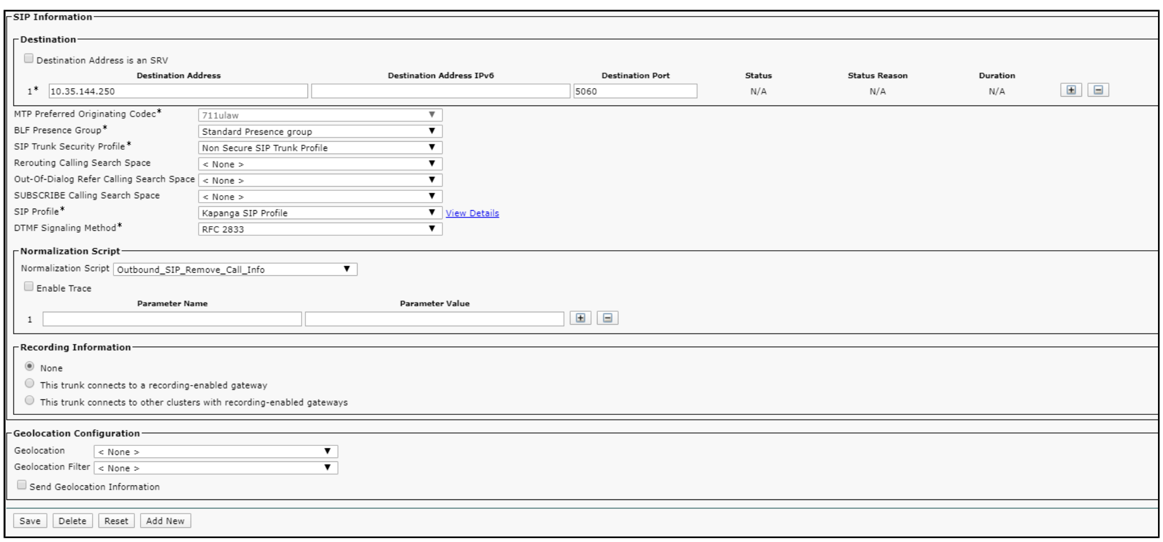
3. Trunk

Select **Device > Trunk**

Figure 4: Trunk



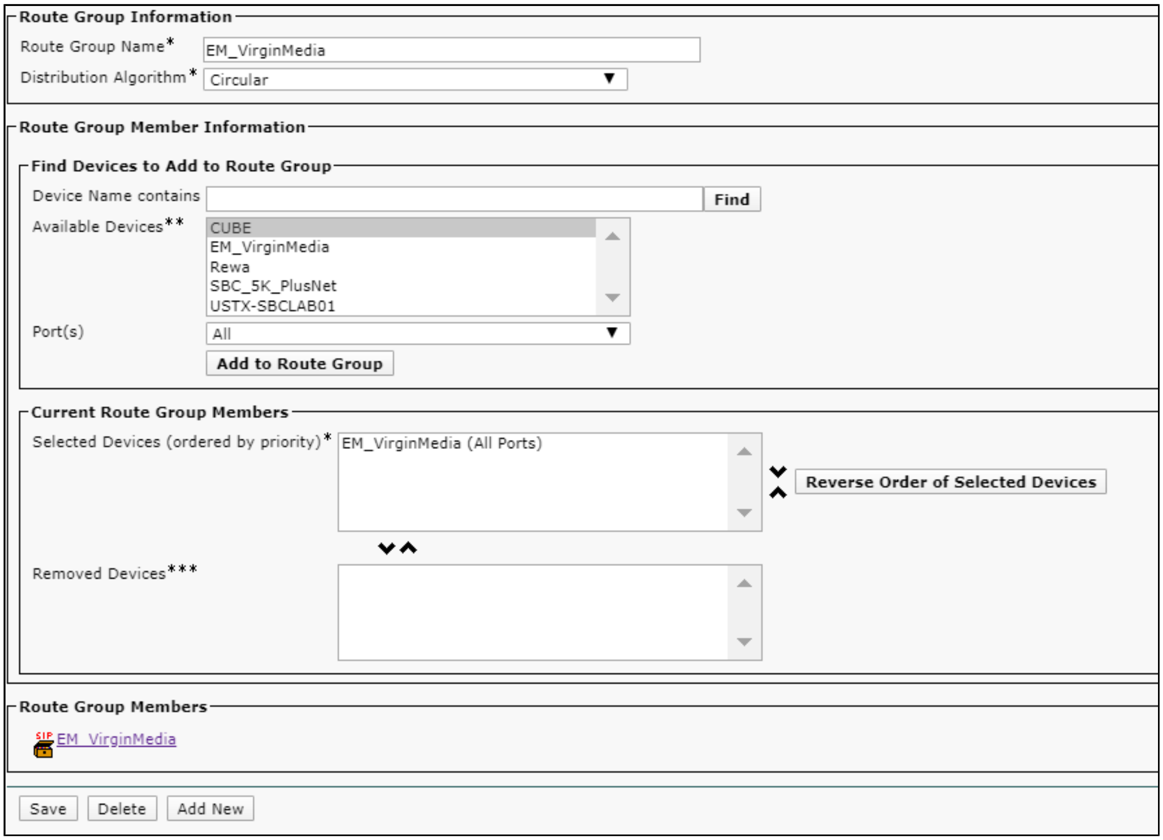




4. Route Group

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

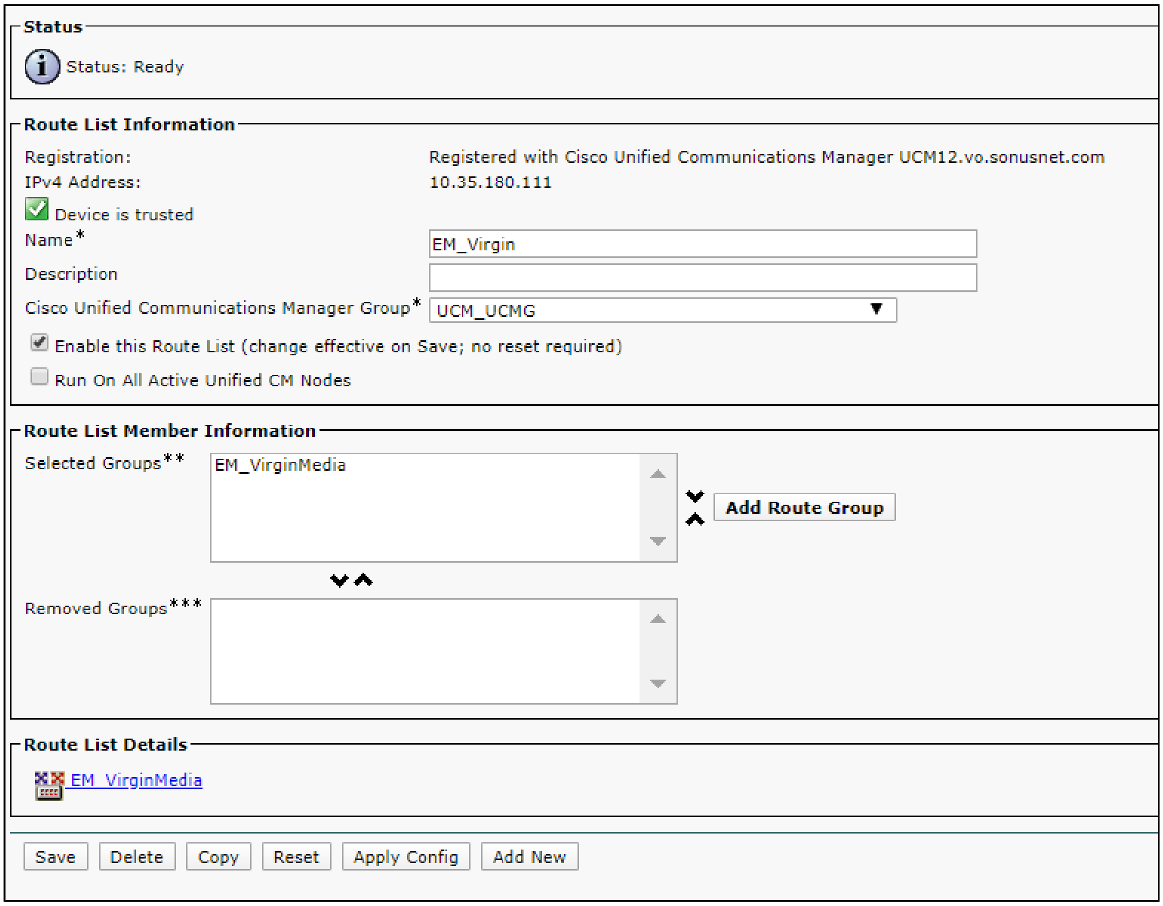
Figure 5: Route Group



5. Route List

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

Figure 6: Route List

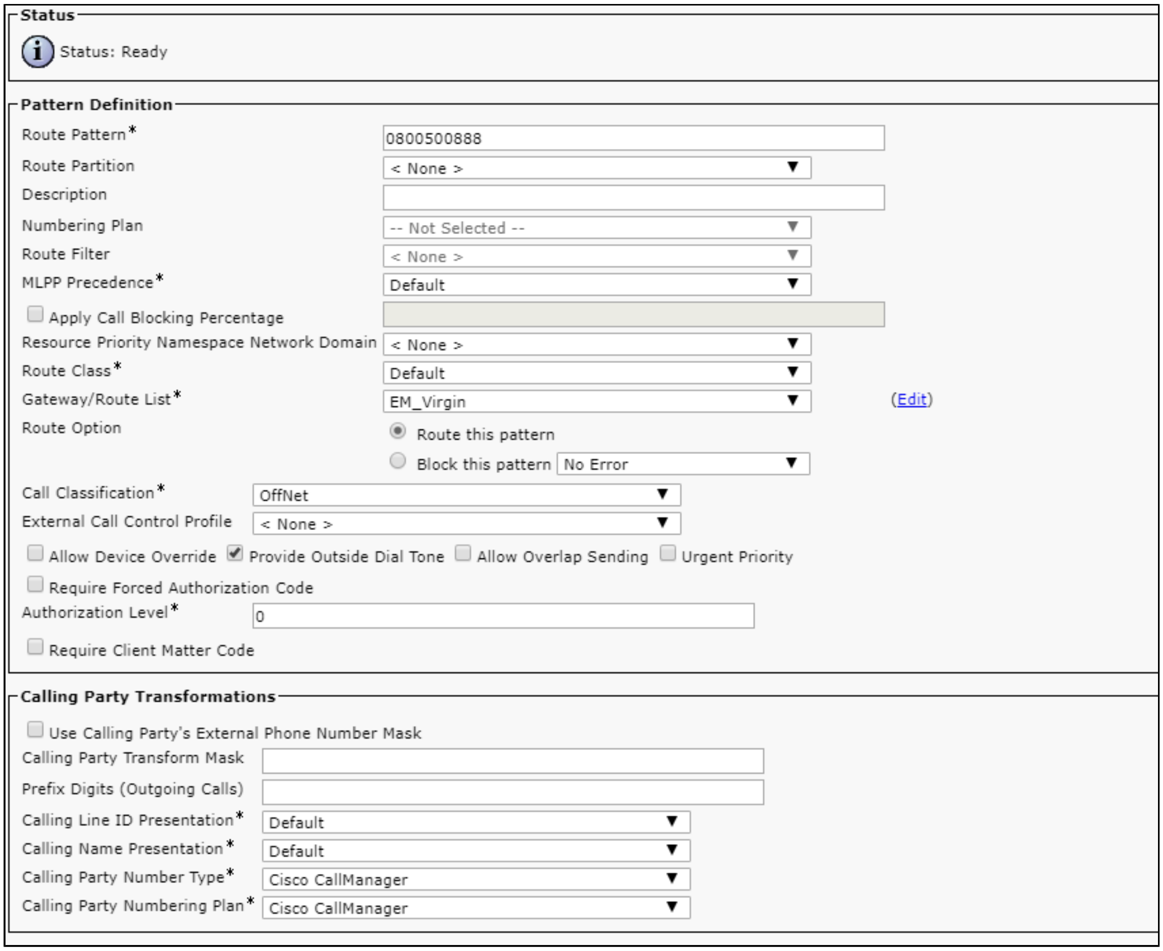


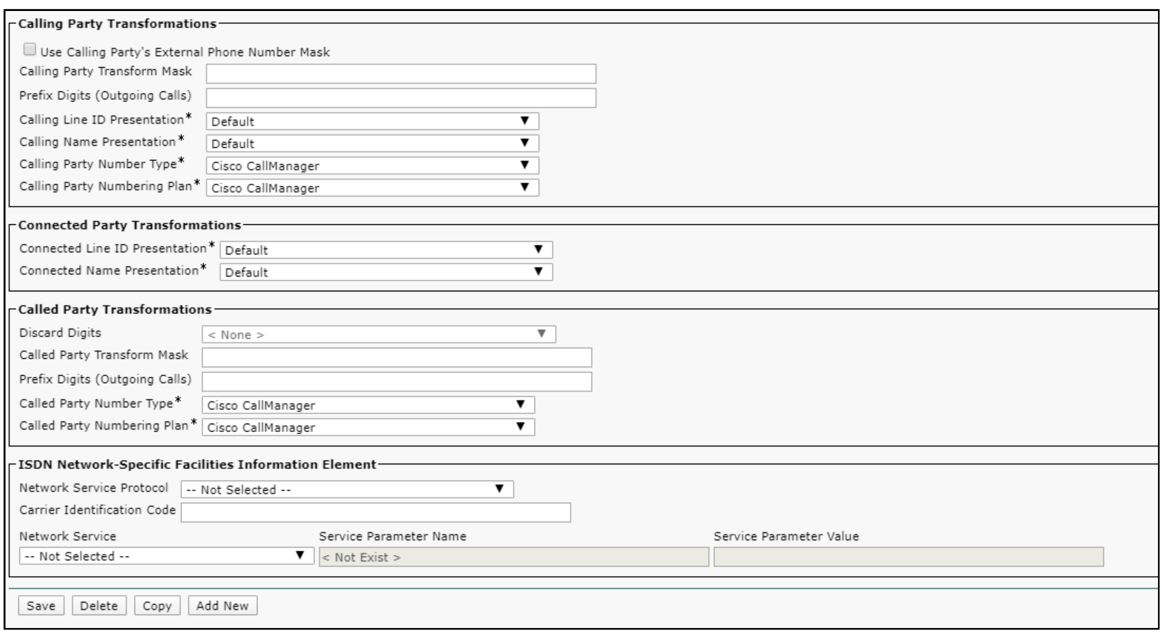
6. Route Pattern

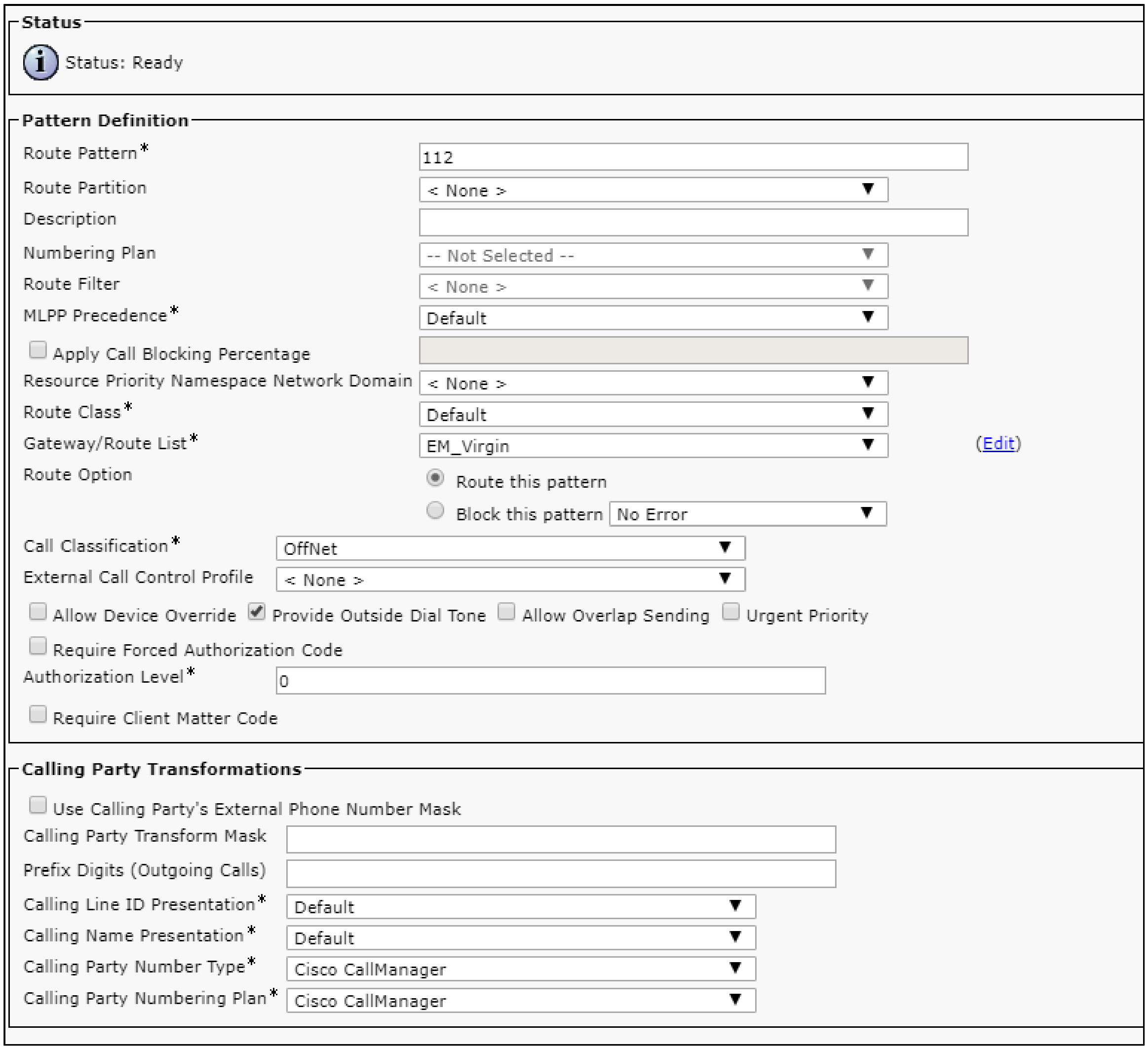
Select **Call Routing > Route/Hunt > Route Pattern**

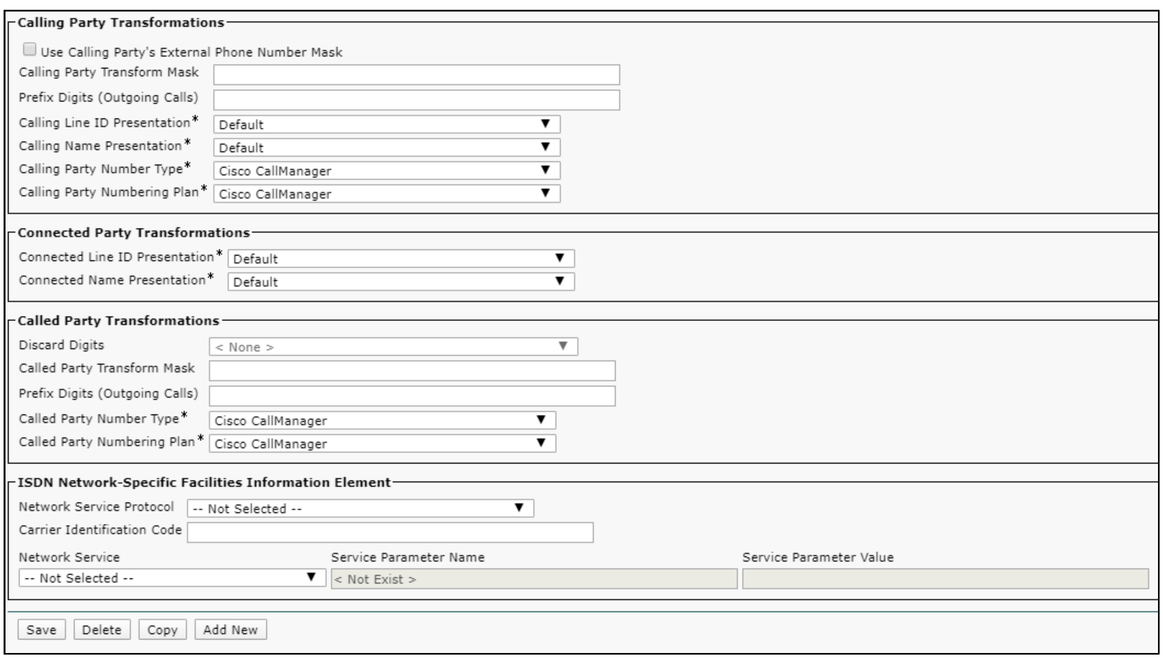
> **Note**
> Use this procedure to create any Route Pattern configuration.

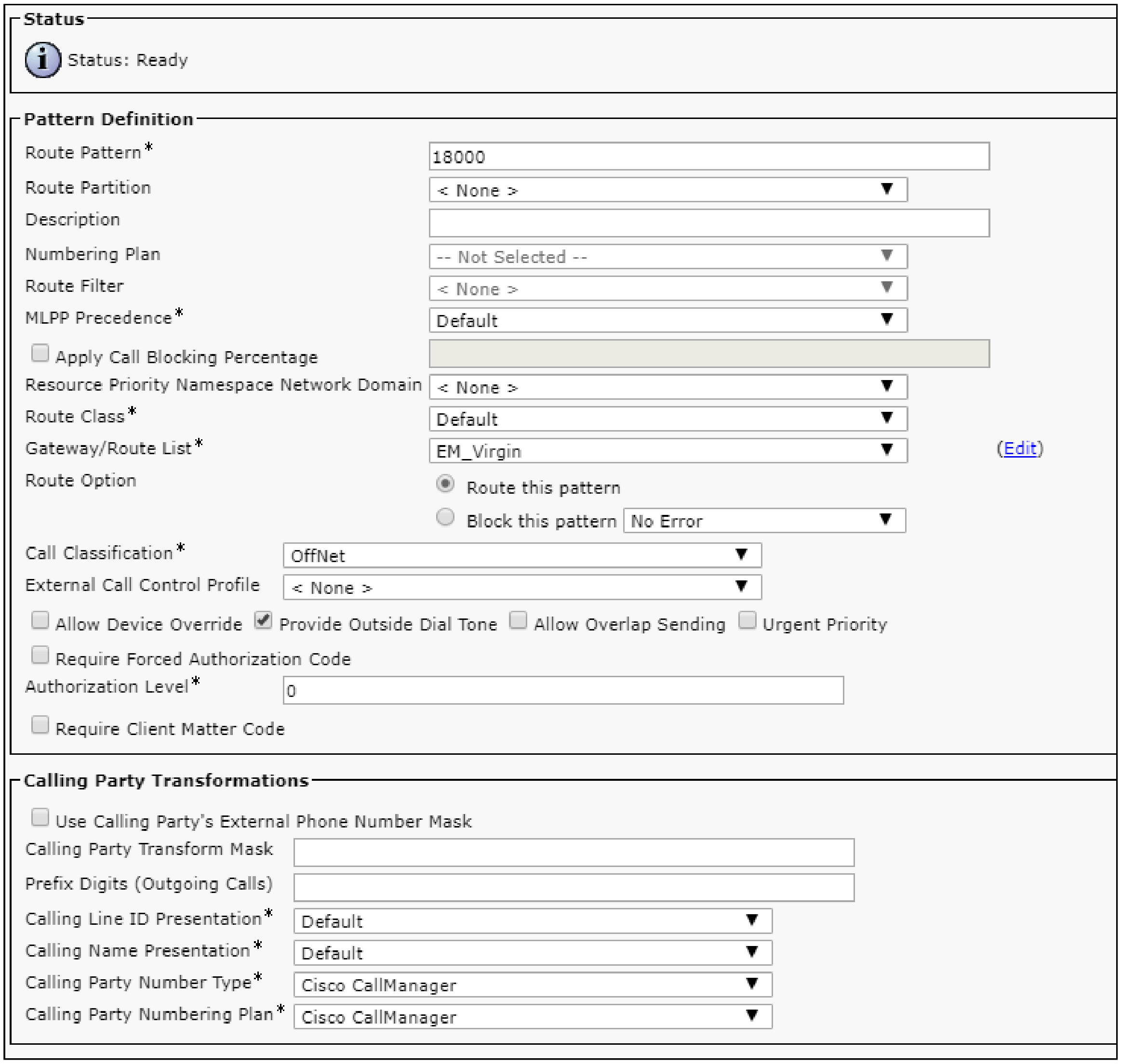
Figure 7: Route Pattern

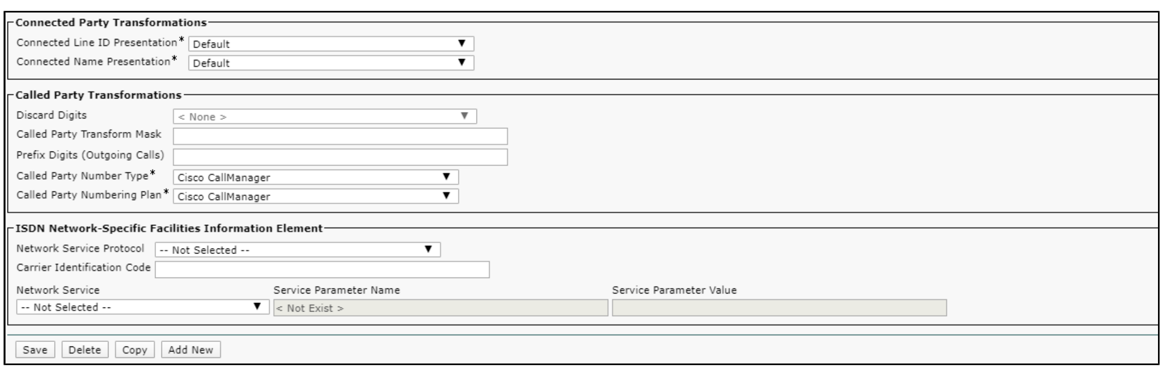

Status
Status: Ready
Pattern Definition
Route Pattern* 0800500888
Route Partition < None >
Description
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* EM_Virgin (Edit)
Route Option
Route this pattern
Block this pattern No Error
Call Classification* OffNet
External Call Control Profile < None >
Allow Device Override
Provide Outside Dial Tone
Allow Overlap Sending
Urgent Priority
Require Forced Authorization Code
Authorization Level* 0
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager

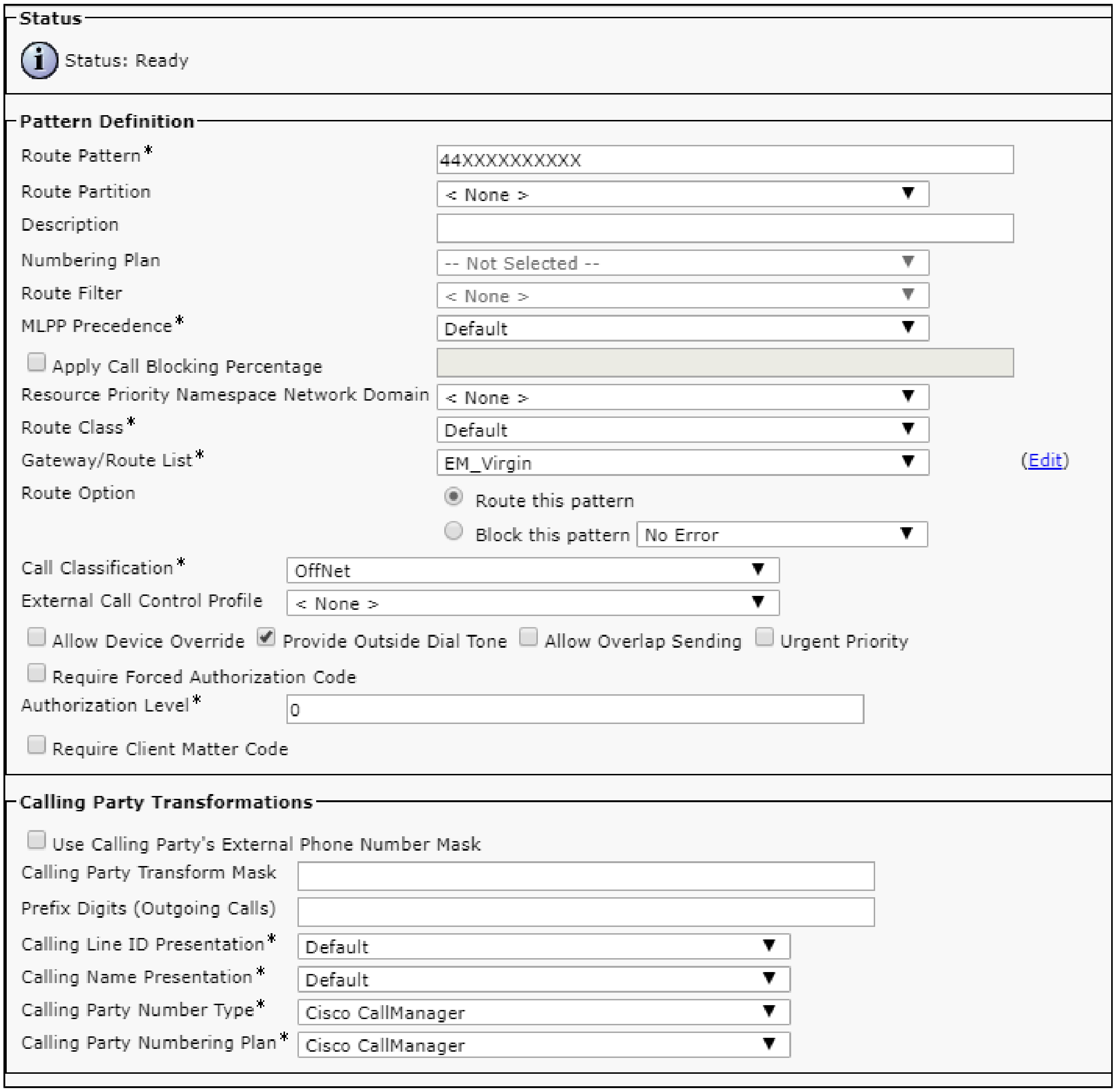

Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Called Party Transformations
Discard Digits < None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol -- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New

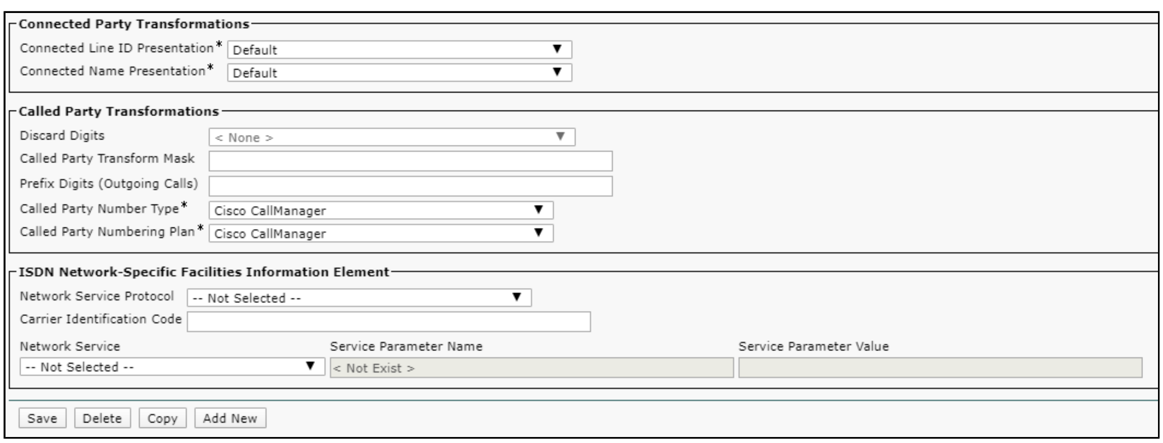

Status
Status: Ready
Pattern Definition
Route Pattern*
112
Route Partition
< None >
Description
Numbering Plan
-- Not Selected --
Route Filter
< None >
MLPP Precedence*
Default
Apply Call Blocking Percentage
Resource Priority Namespace Network Domain
< None >
Route Class*
Default
Gateway/Route List*
EM_Virgin
(Edit)
Route Option
Route this pattern
Block this pattern
No Error
Call Classification*
OffNet
External Call Control Profile
< None >
Allow Device Override
Provide Outside Dial Tone
Allow Overlap Sending
Urgent Priority
Require Forced Authorization Code
Authorization Level*
0
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager


Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Called Party Transformations
Discard Digits
< None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Cisco CallManager
Called Party Numbering Plan*
Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New


Status
Status: Ready
Pattern Definition
Route Pattern*
18000
Route Partition
< None >
Description
Numbering Plan
-- Not Selected --
Route Filter
< None >
MLPP Precedence*
Default
Apply Call Blocking Percentage
Resource Priority Namespace Network Domain
< None >
Route Class*
Default
Gateway/Route List*
EM_Virgin
(Edit)
Route Option
Route this pattern
Block this pattern
No Error
Call Classification*
OffNet
External Call Control Profile
< None >
Allow Device Override
Provide Outside Dial Tone
Allow Overlap Sending
Urgent Priority
Require Forced Authorization Code
Authorization Level*
0
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling Party Number Type*
Cisco CallManager
Calling Party Numbering Plan*
Cisco CallManager


Connected Party Transformations
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Called Party Transformations
Discard Digits
< None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Cisco CallManager
Called Party Numbering Plan*
Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New





EdgeMarc Configuration

Network

- LAN and WAN Interfaces
- Static Routes

VoIP

- VoIP Settings
- SIP Settings
- B2BUA

Network

LAN and WAN Interfaces

Login to EdgeMarc as root user and go to **Network** to configure the LAN and WAN interfaces.

Figure 8: EdgeMarc Network LAN Interface

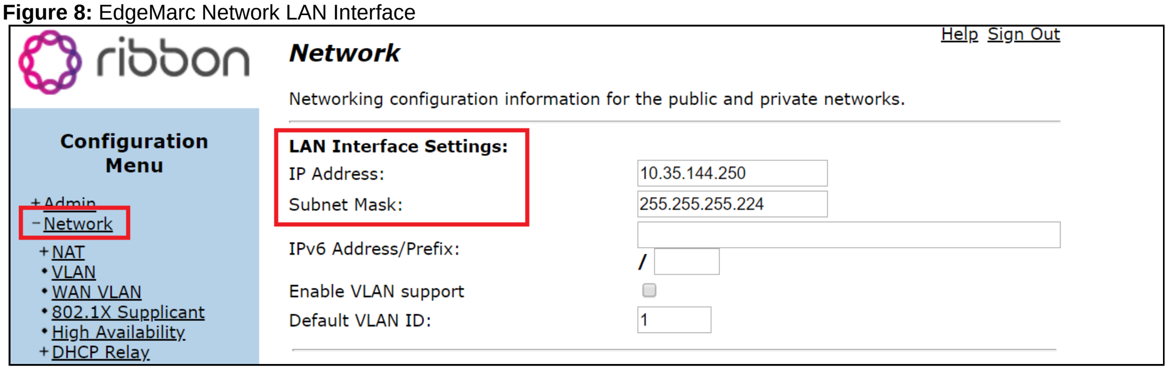
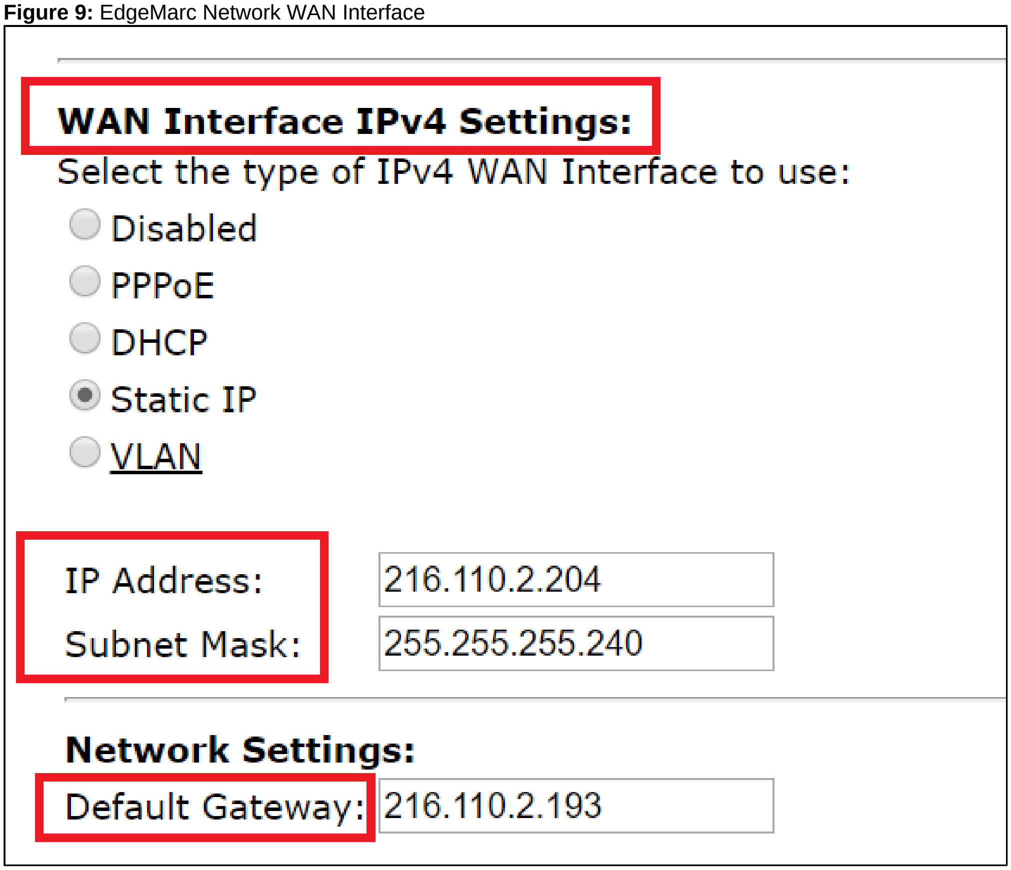


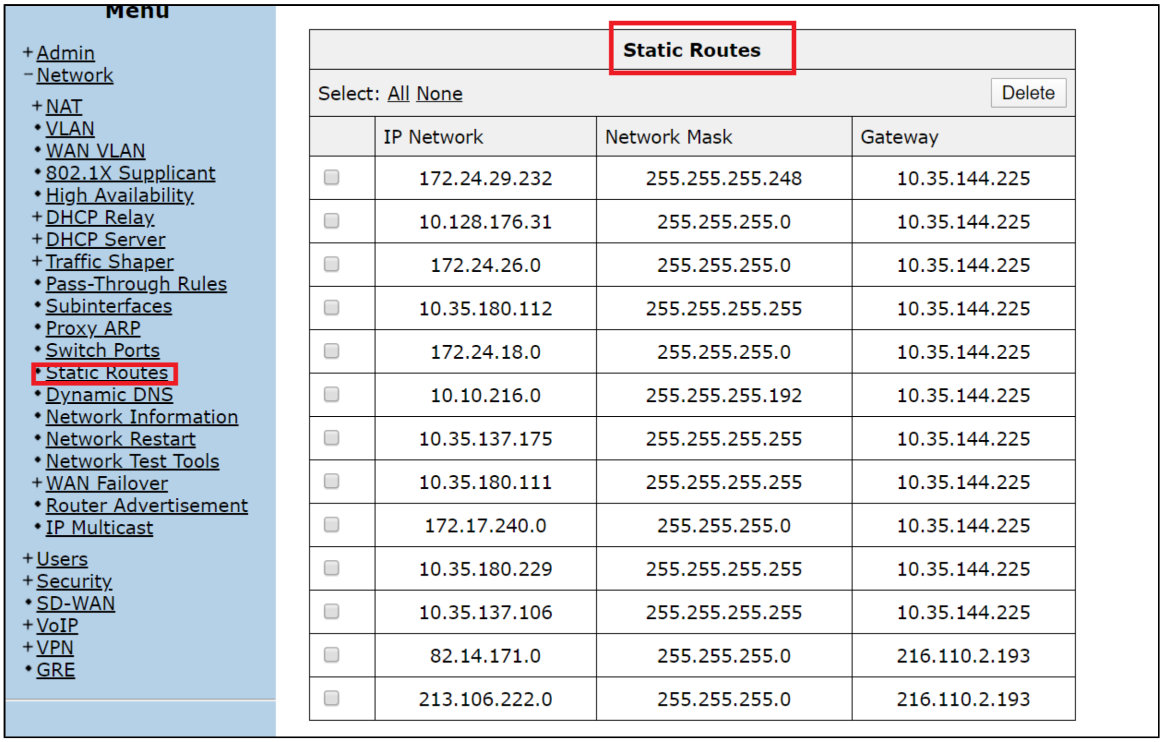
Figure 9: EdgeMarc Network WAN Interface



Static Routes

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

Figure 10: Static Routes

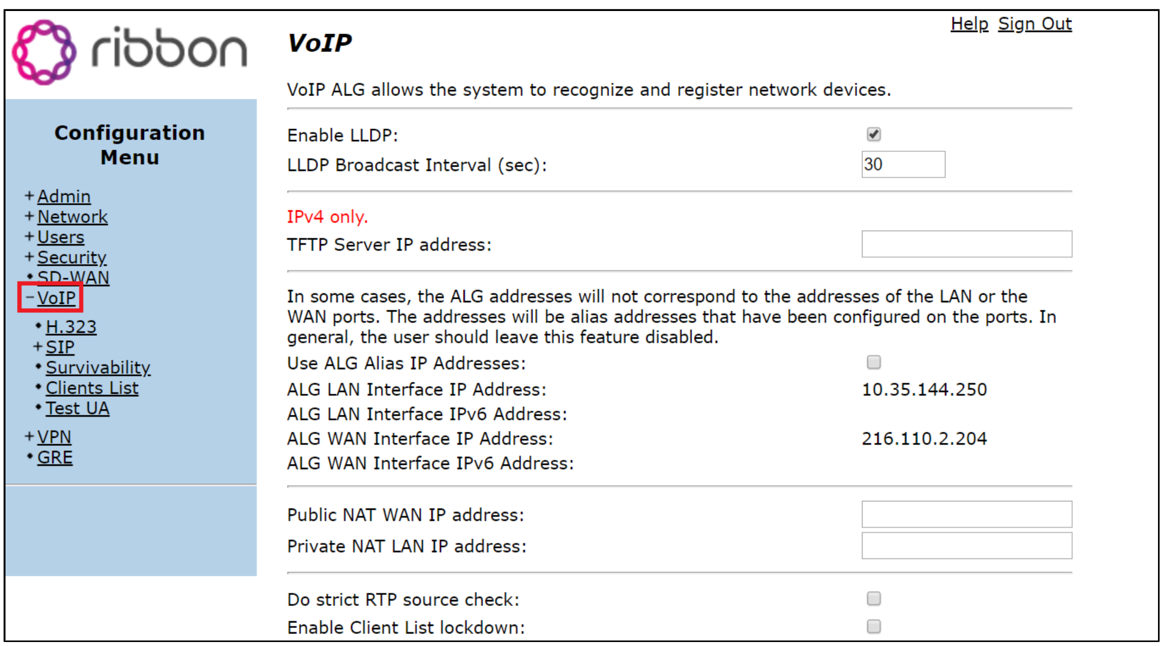


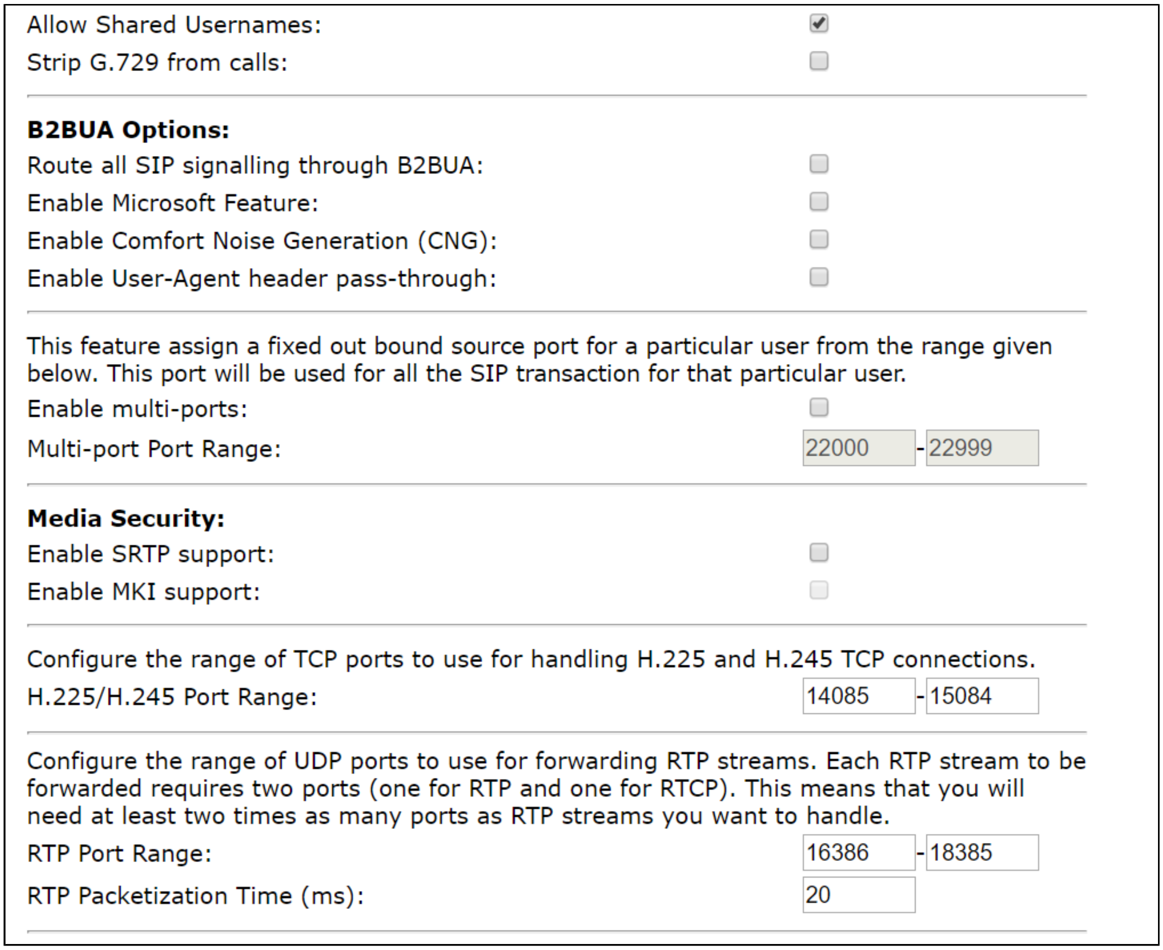
VoIP

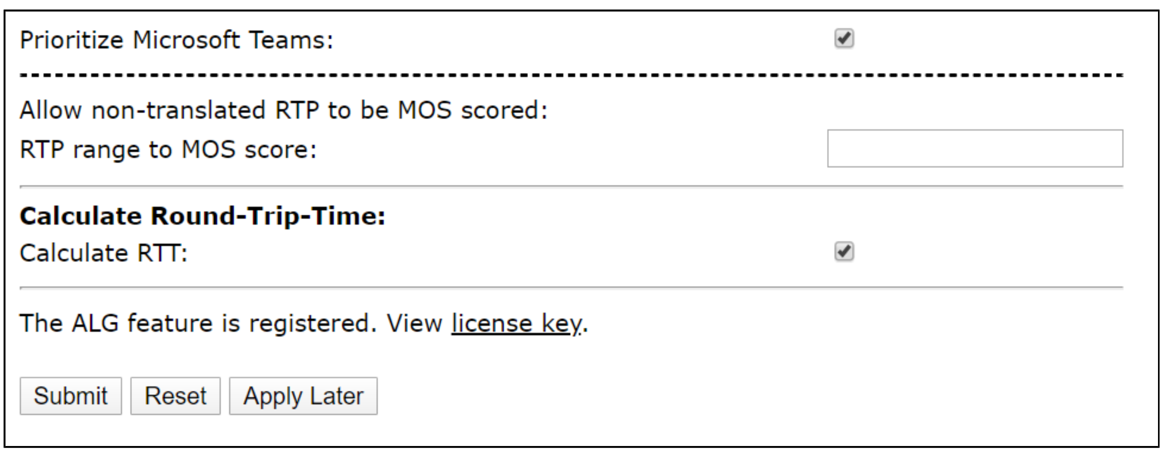
VoIP Settings

1. Login as root user and navigate to **VoIP** to configure the VoIP features.

Figure 11: VoIP



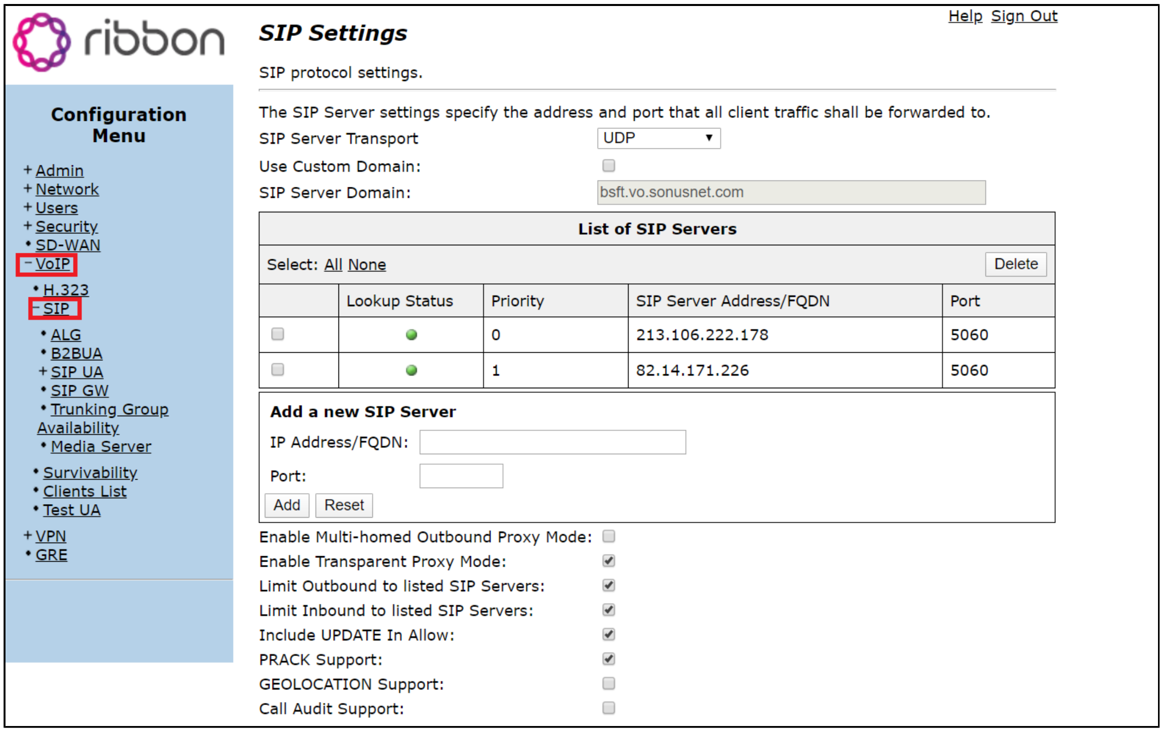


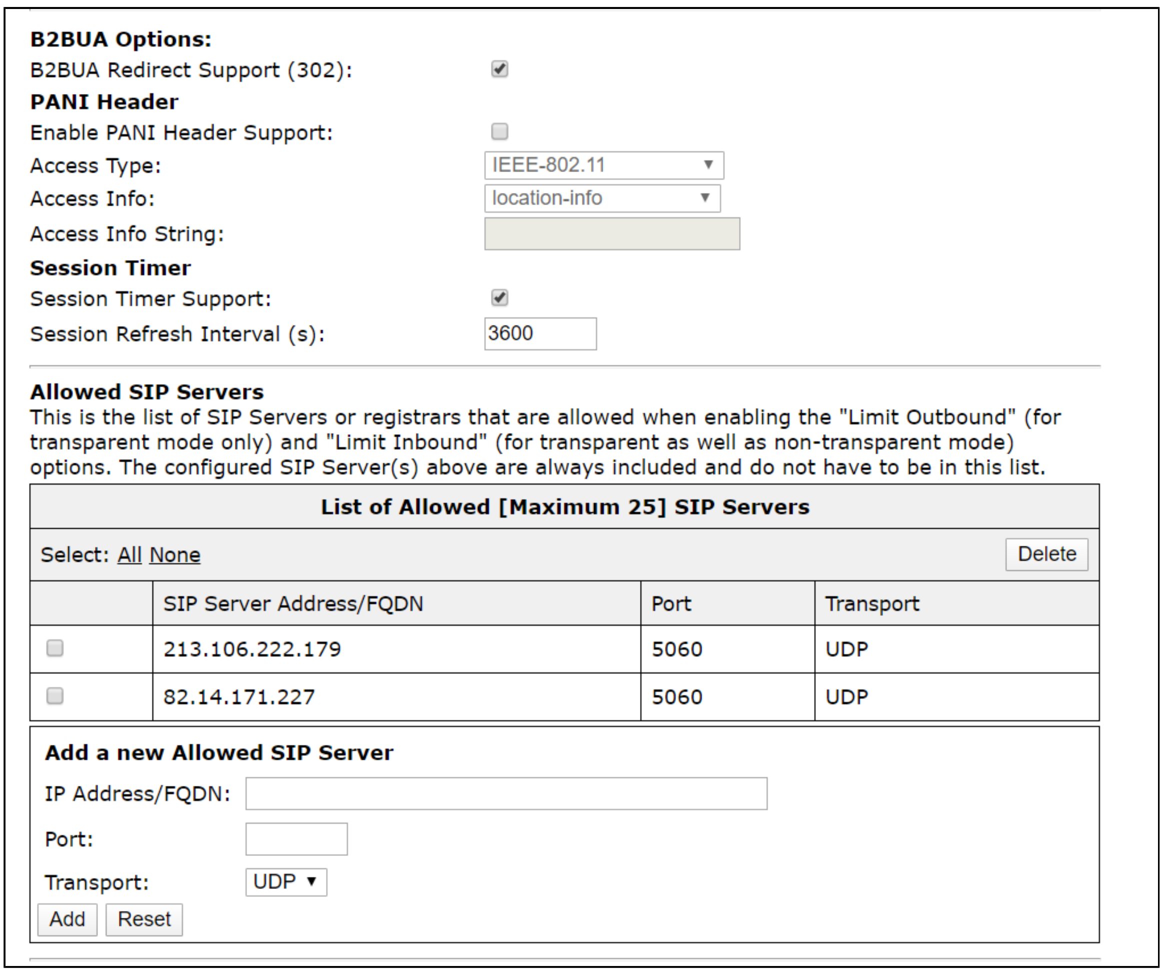


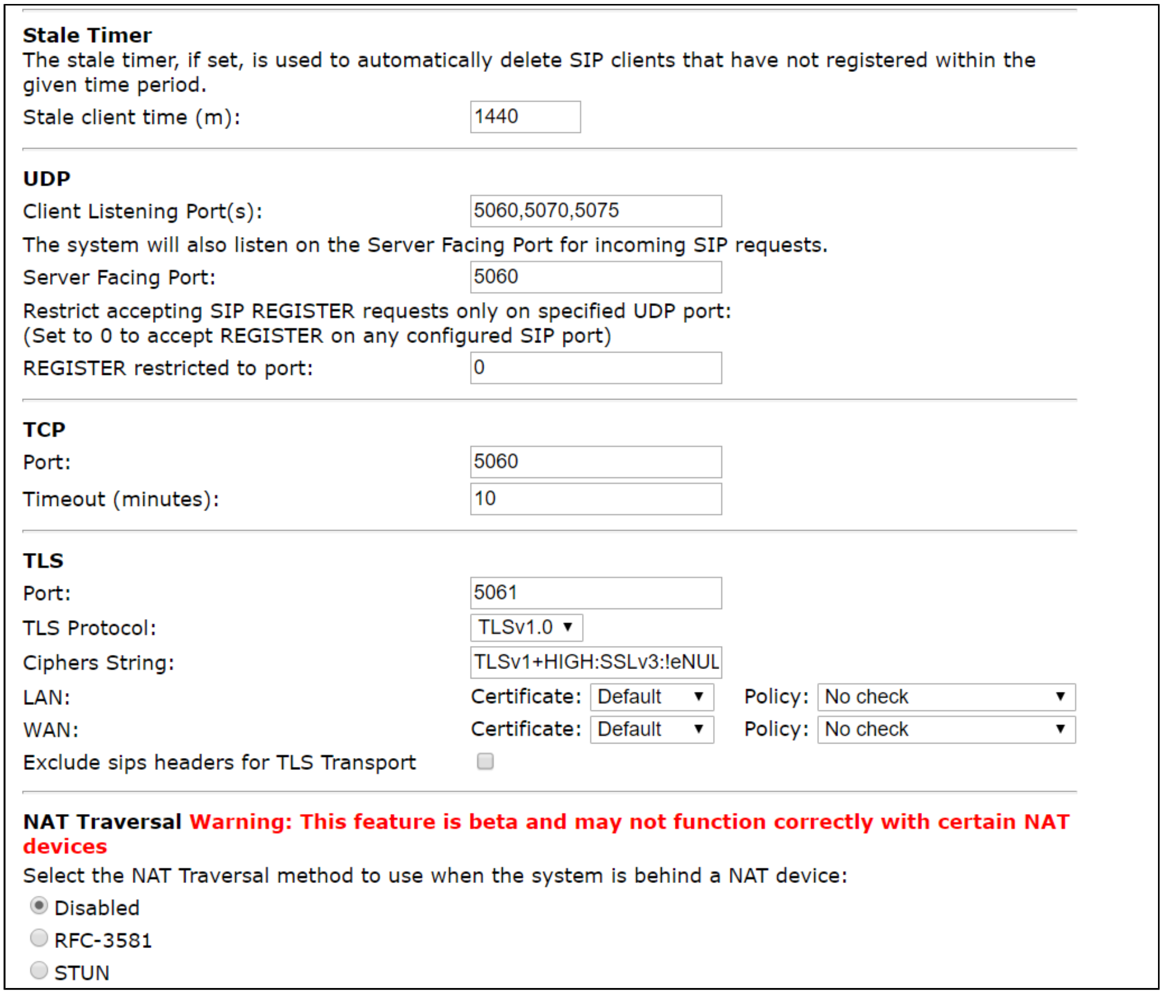
SIP Settings

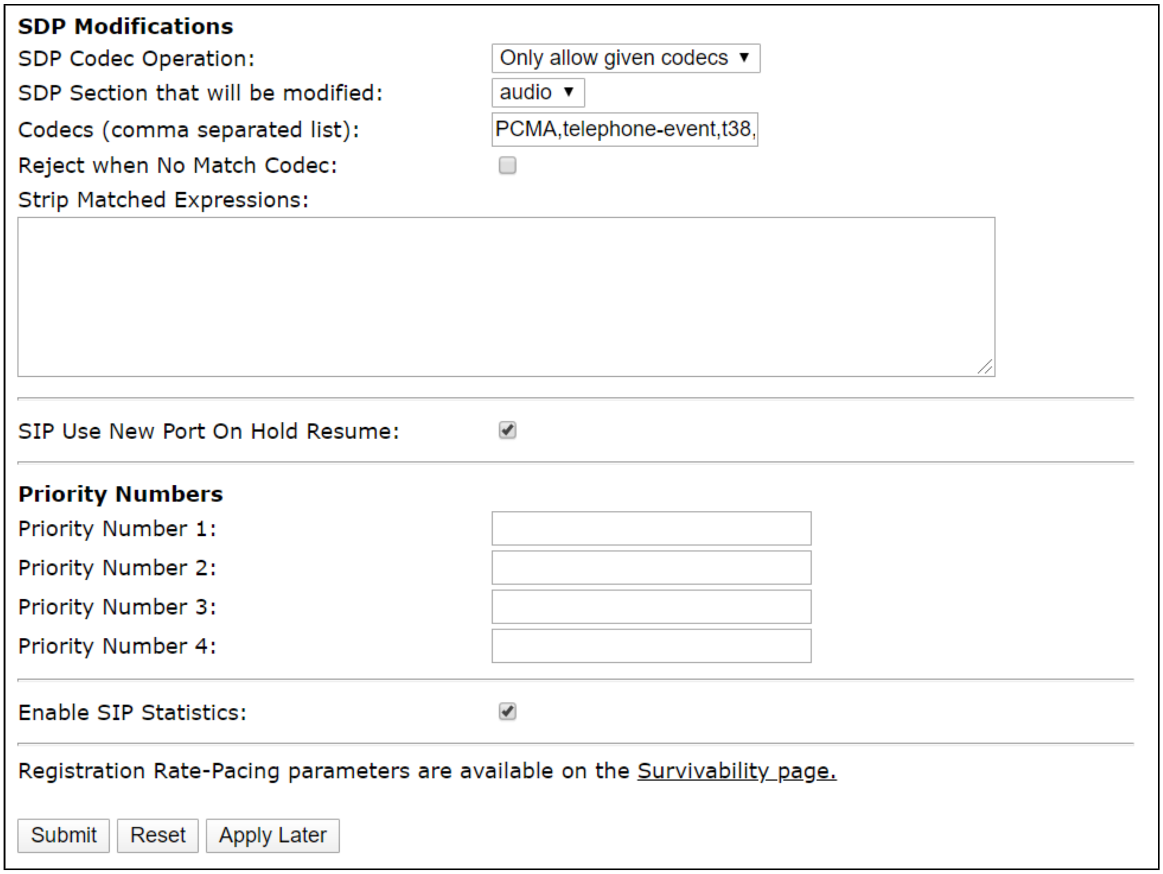
1. Navigate to **VoIP** > **SIP** to configure the SIP settings.
2. Configure the SIP servers.

Figure 12: SIP


ribbon
SIP Settings
Help Sign Out
SIP protocol settings.
Configuration Menu
+Admin
+Network
+Users
+Security
•SD-WAN
−VoIP
•H.323
−SIP
•ALG
•B2BUA
+SIP UA
•SIP GW
•Trunking Group Availability
•Media Server
•Survivability
•Clients List
•Test UA
+VPN
•GRE
The SIP Server settings specify the address and port that all client traffic shall be forwarded to.
SIP Server Transport UDP
Use Custom Domain:
SIP Server Domain: bsft.vo.sonusnet.com
List of SIP Servers
Select: All None
Delete
Lookup Status
Priority
SIP Server Address/FQDN
Port
0
213.106.222.178
5060
1
82.14.171.226
5060
Add a new SIP Server
IP Address/FQDN:
Port:
Add
Reset
Enable Multi-homed Outbound Proxy Mode:
Enable Transparent Proxy Mode:
Limit Outbound to listed SIP Servers:
Limit Inbound to listed SIP Servers:
Include UPDATE In Allow:
PRACK Support:
GEOLOCATION Support:
Call Audit Support:


B2BUA Options:
B2BUA Redirect Support (302):
PANI Header
Enable PANI Header Support:
Access Type: IEEE-802.11
Access Info: location-info
Access Info String:
Session Timer
Session Timer Support:
Session Refresh Interval (s): 3600
Allowed SIP Servers
This is the list of SIP Servers or registrars that are allowed when enabling the "Limit Outbound" (for transparent mode only) and "Limit Inbound" (for transparent as well as non-transparent mode) options. The configured SIP Server(s) above are always included and do not have to be in this list.
List of Allowed [Maximum 25] SIP Servers
Select: All None
Delete
SIP Server Address/FQDN
Port
Transport
213.106.222.179
5060
UDP
82.14.171.227
5060
UDP
Add a new Allowed SIP Server
IP Address/FQDN:
Port:
Transport: UDP
Add
Reset

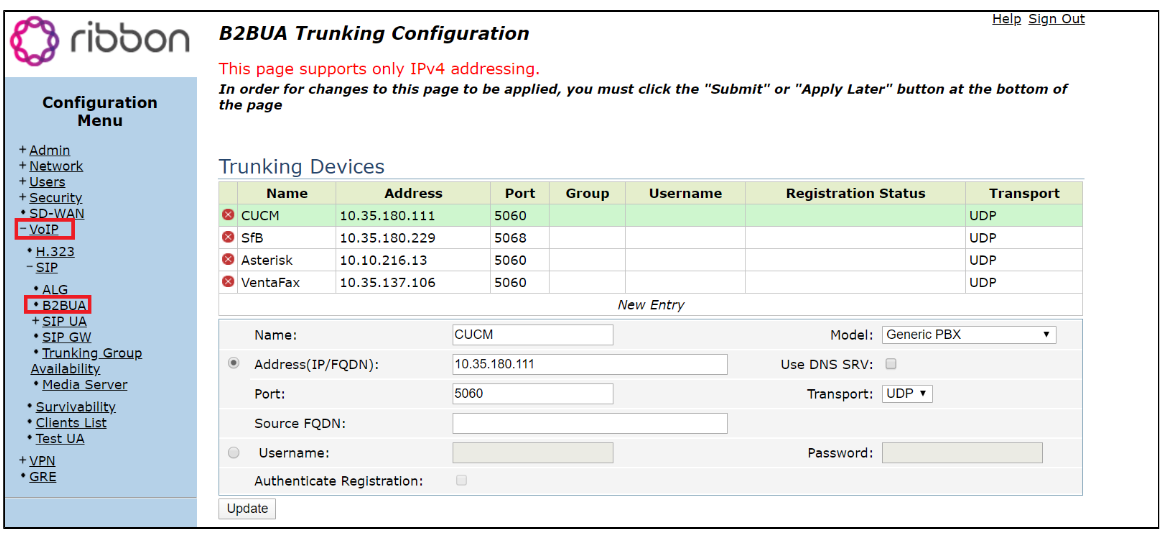

Stale Timer
The stale timer, if set, is used to automatically delete SIP clients that have not registered within the given time period.
Stale client time (m):
1440
UDP
Client Listening Port(s):
5060,5070,5075
The system will also listen on the Server Facing Port for incoming SIP requests.
Server Facing Port:
5060
Restrict accepting SIP REGISTER requests only on specified UDP port:
(Set to 0 to accept REGISTER on any configured SIP port)
REGISTER restricted to port:
0
TCP
Port:
5060
Timeout (minutes):
10
TLS
Port:
5061
TLS Protocol:
TLSv1.0
Ciphers String:
TLSv1+HIGH:SSLv3:!eNUL
LAN:
Certificate:
Default
Policy:
No check
WAN:
Certificate:
Default
Policy:
No check
Exclude sips headers for TLS Transport
NAT Traversal Warning: This feature is beta and may not function correctly with certain NAT devices
Select the NAT Traversal method to use when the system is behind a NAT device:
Disabled
RFC-3581
STUN

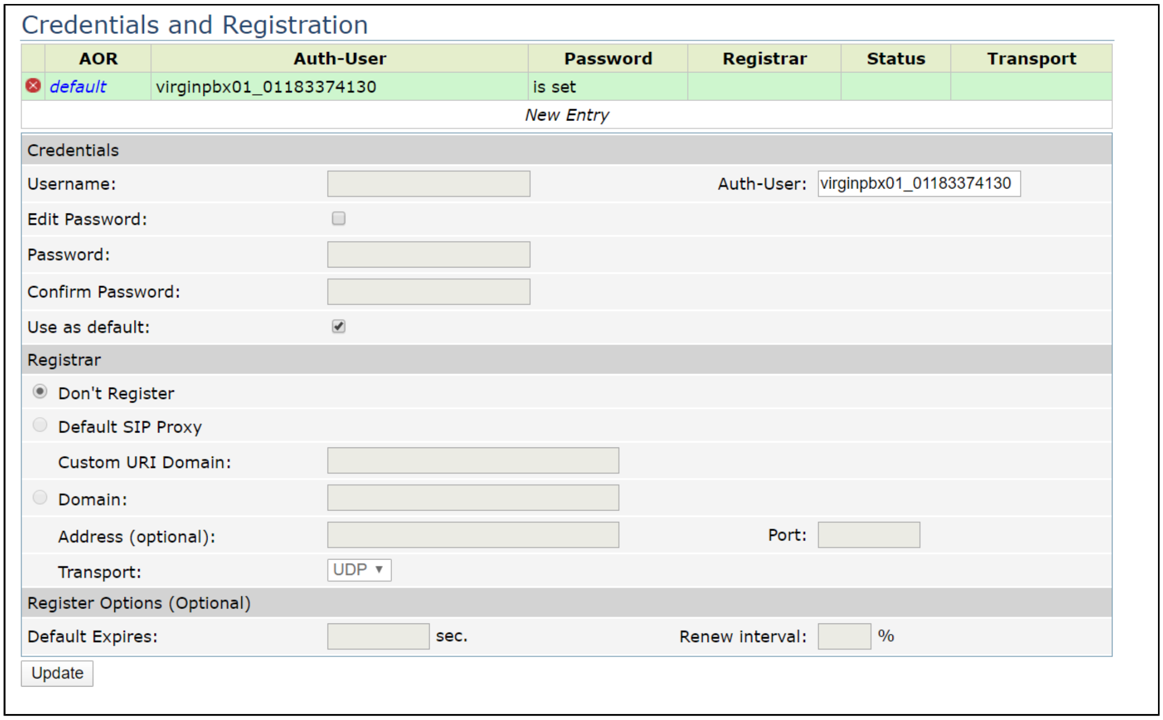


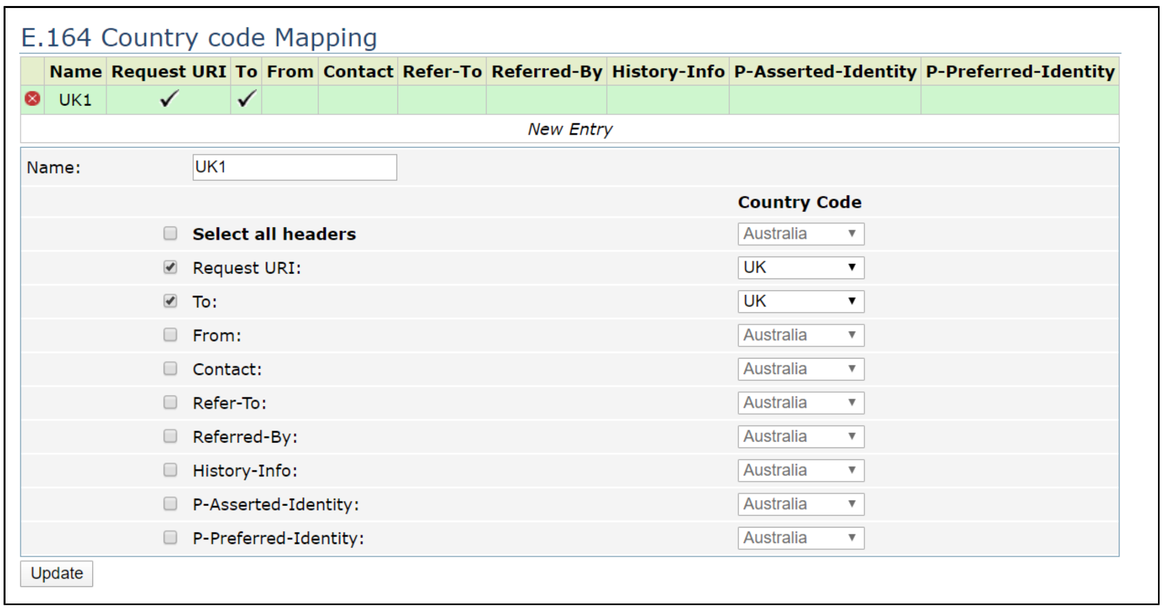
B2BUA

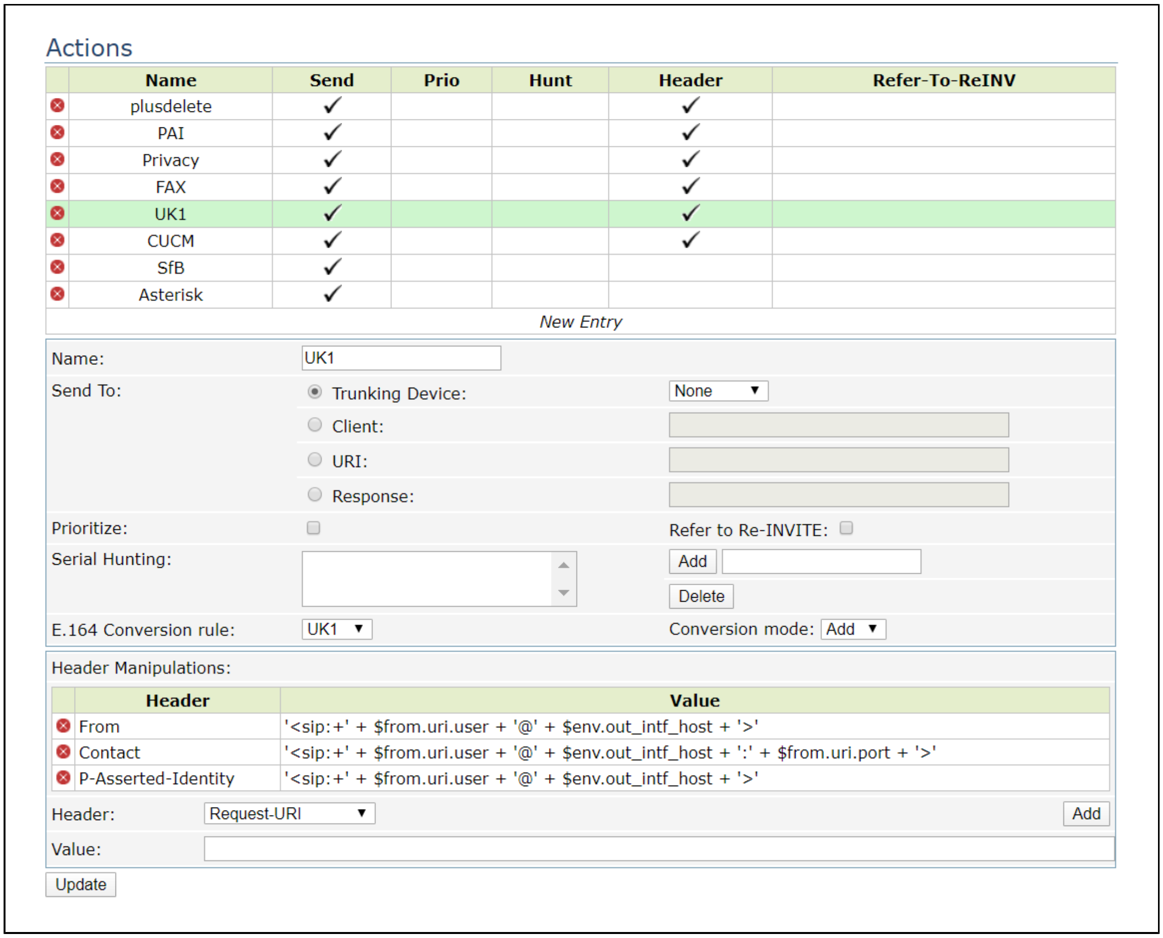
1. Navigate to **VoIP** > **B2BUA**
2. Configure LAN Part with the next form.

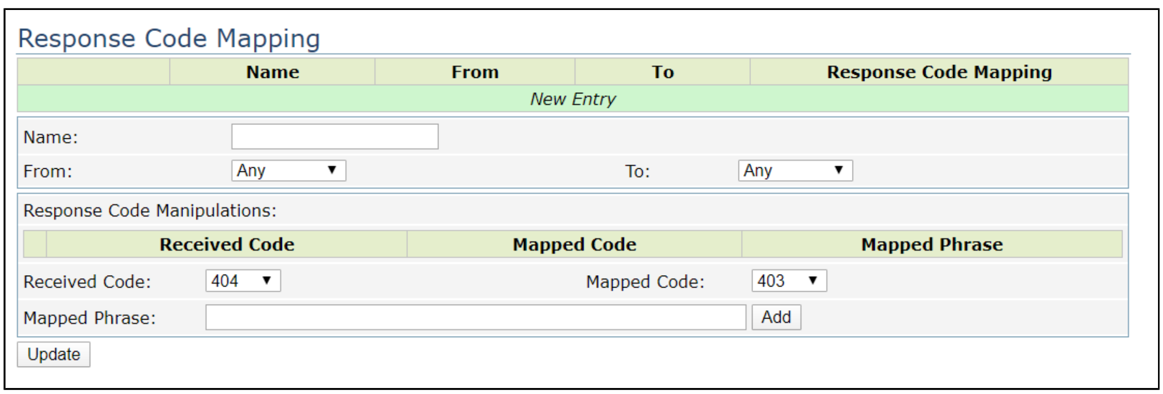
Figure 13: B2BUA

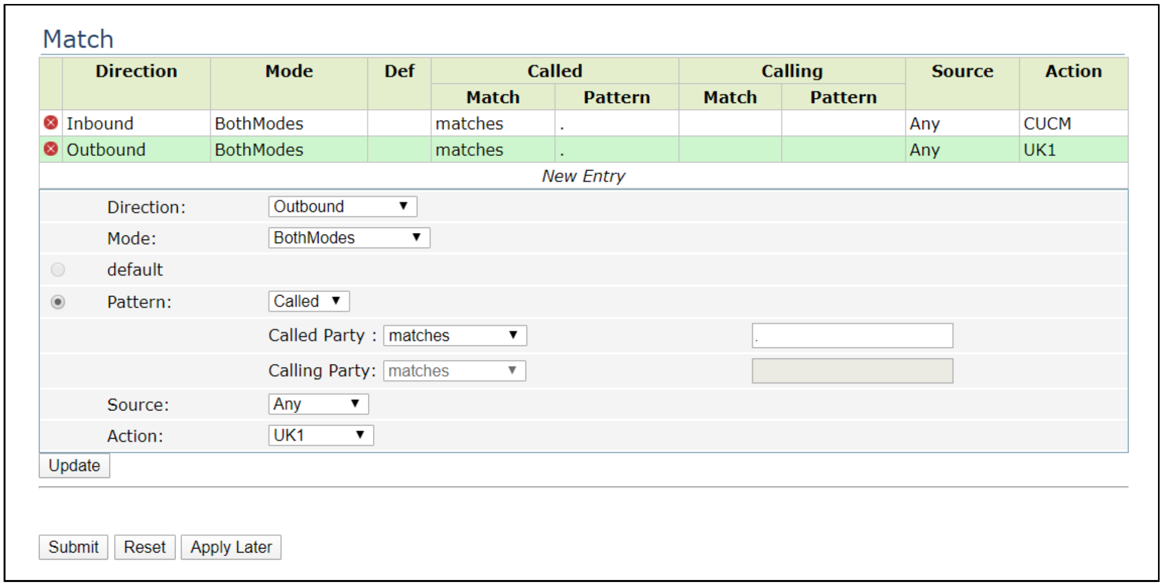



Credentials and Registration
AOR
Auth-User
Password
Registrar
Status
Transport
default
virginpbx01_01183374130
is set
New Entry
Credentials
Username:
Auth-User:
virginpbx01_01183374130
Edit Password:
Password:
Confirm Password:
Use as default:
Registrar
Don't Register
Default SIP Proxy
Custom URI Domain:
Domain:
Address (optional):
Port:
Transport:
UDP
Register Options (Optional)
Default Expires:
sec.
Renew interval:
%
Update


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:
UK1
Country Code
Select all headers
Australia
Request URI:
UK
To:
UK
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
plusdelete
PAI
Privacy
FAX
UK1
CUCM
SfB
Asterisk
New Entry
Name:
UK1
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 + '>'
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



Test Results

The following table provides information about the tests that Ribbon performed to complete all scenarios that Virgin Media needs for customers.

S. No	Procedure	Observation	Result	Comment
IOP1	Vendors eSBC response to SIP OPTIONS messages from SBC	No calls are required for this test. SIP trace to be captured for approximately 60 seconds and checked for correct signaling. For each eSBC, the SBC periodically sends an OPTIONS request to the vendors eSBC to check if its SIP stack is reachable. If a SIP response 200 OK is received from the IP-PBX, the SIP trunk is placed or remains in an In-Service state. e.g. 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. SIP trace to be captured for approximately 60 seconds (depending on agreement) and checked for correct signaling. Vendors eSBC setup for Solution IP.Addr Mode eSBC configured to send OPTIONS messages to the SBC on a periodic basis. The SBC responds with SIP response 200OK, for example: "OPTIONS sip:ping@192.168.1.10:5060 SIP/2.0" Verify that the eSBC can simultaneously send SIP OPTIONS messages to both the solution SBC addresses.	Pass	
IOP4	Basic test call from IP-PBX to PSTN line through SBC-A (using SBC-A IPV4 ip address).	IP-PBX line initiates call, Call is answered, IP-PBX line terminates call. Vendors eSBC setup for Solution IP.Addr Mode Call from the IP-PBX. Invite seen from eSBC to SBC-A, proxy authentication challenge returned to eSBC, re-invite with correct credentials from eSBC and call progresses as expected. For 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 and confirm that G.711 A law codec with 10 or 20ms packetisation is used. Also check to see if INVITE contains the Session-Expires header and that the INVITE is syntactically correct. Check for Supported Header to see if 'timer' is supported. Ensure that the response in the 200 OK is compatible with the INVITE and verify that the Required Header contains 'timer'. (x-ref IOP9)	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 test completed eSBC to be configure to use Primary SBC-A for calls to route to.	IP-PBX line initiates call, Call is answered, IP-PBX line terminates call. Vendors eSBC setup for Solution IP.Addr Mode Call from the IP-PBX. Invite seen from eSBC to SBC-B, proxy authentication challenge returned to eSBC, re-invite with correct credentials from eSBC and call progresses as expected. e.g. 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 and confirm that G.711 A law codec with 10ms or 20ms packetisation is being used.	Pass	
IOP7b	Called Number format - vendors 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)	SBC to be configured for Global calling plan. IP-PBX line initiates call to PSTN line, Call is answered. IP-PBX line terminates call. Configure the eSBC to present the called number in the user part of the Request & To URIs to be sent 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 - vendors eSBC to soft switch number normalization - Global Dial Plan Test eSBC capability to send 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)	SBC to be configured for Global calling plan. IP-PBX line initiates call to PSTN line, Call is answered. IP-PBX terminates call. Configure the eSBC to present the calling number in the user part of the From & PAI URIs to be sent 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)	SBC to be configured for Global calling plan. PSTN line initiates call to IP-PBX line, Call is answered. PSTN line terminates call. 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) Also check to see that the INVITE contains Session-Expires header and that it is syntactically correct. Check for Supported Header and ensure 'timer' is supported. Ensure response in 200 OK is compatible with INVITE and check for Required Header and if it contains '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)	SBC to be configured for Global calling plan. PSTN line initiates call to IP-PBX line, Call is answered. PSTN line terminates call. 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	Call made from IP-PBX line to the Emergency services using 999. Call answered. Either party terminates 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	Call made from IP-PBX line to the Emergency services using 112. Call answered, Either party terminates 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	Call made from IP-PBX line using a text direct set to the Emergency services using 18000. Call answered. Either party terminates 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	Call made from IP-PBX line to PSTN line, Answer Call. IP-PBX line terminates call.	Pass	
IOP15	PSTN calls SIP #1, SIP #1 conferences in SIP #2	Call made from IP-PBX line to PSTN line, Answer Call. PSTN line terminates call	Pass	
IOP16	IP-PBX Line to PSTN - Busy subscriber	Call made from IP-PBX line to a busy PSTN line (without divert on busy) Wait for soft switch to return busy response. Ensure that eSBC does not recurse and setup call via secondary SIP trunk.	Pass	
IOP17	IP-PBX Line to PSTN - No answer timeout test	Call made from IP-PBX line to a PSTN line (without divert on no answer) Do not answer call. Wait for soft switch to return no answer timeout response. Ensure that eSBC does not recurse and setup call via secondary SIP trunk.	Pass With Caveat	Cancel message is sent by SfB 2015 server and there is not an option to change the timer for this.
IOP18	IP-PBX Line to PSTN - Subscriber not reachable Vendor to call 01189111111	Call made from IP-PBX line to an invalid number. Wait for soft switch to return response. Ensure that eSBC does not recurse and setup call via secondary SIP trunk.	Pass	
IOP19	PSTN Line to IP-PBX - call answer - Originator disconnect.	Call made from a PSTN line to an IP-PBX line, Answer Call. Originator disconnects call.	Pass	.
IOP20	PSTN Line to IP-PBX - call answer - Terminator disconnect	Call made from a PSTN line to an IP-PBX line, Answer Call. IP-PBX line terminates call.	Pass	
IOP21	PSTN Line to IP-PBX - busy subscriber	Call made from PSTN line to a busy IP-PBX line (without divert on busy) Wait for IP-PBX to return busy response.	De-Scoped	SfB 2015/Lync does not support Busy line due to a permanent call waiting service. If a UM/Voicemail service is activated call goes there.
IOP22	PSTN Line to IP-PBX - No answer timeout test, Invoked by PBX	Call made from a PSTN line to an IP-PBX line (without divert on no answer). Wait for the IP-PBX to return no answer timeout response	De-Scoped	SfB2015/Lync does not support No answer time out. If a UM /Voicemail service is activated call goes there.
IOP23	PSTN Line to IP-PBX - subscriber not reachable	Call made from a PSTN line to an invalid number/unprogrammed DDI on the IP-PBX. Wait for IP-PBX to return response.	Pass	
IOP24	Verify CLIP service on IP-PBX line (incoming call from PSTN)	Call made from PSTN line to IP-PBX line. PSTN line is set to allow CLI presentation. Check that CLI is delivered as expected. Either party terminates call.	Pass	
IOP25	Verify CLIR service on IP-PBX line (incoming call from PSTN)	Call made from PSTN line to IP-PBX line. PSTN line is set to restrict CLI presentation. Check that CLI is not delivered as expected. Either party terminates call.	Pass	
IOP26	Verify CLIP service on PSTN line (outgoing call from IP-PBX, From)	Ensure number used in From header is agreed with Virgin Media and entered into the soft switch database for screening purposes. Call made from an IP-PBX line to a PSTN line. Ensure that the eSBC is configured such that the IP-PBX line sends From header containing Calling Line ID (CLI) in the INVITE. Ensure that the eSBC allows presentation of its CLI using privacy-header (Privacy: none or privacy-header not present) Ensure that the expected CLI is presented to the PSTN line. Either party terminates 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	Ensure number used in PAI/PPI header is agreed with Virgin Media and entered into the soft switch database for screening purposes. Call made from an IP-PBX line to a PSTN line. Ensure that the eSBC is configured such that the IP-PBX line sends PAI /PPI header containing Calling Line ID (CLI) in the INVITE. If PAI header is populated this will be used in preference to the From header. Ensure that the eSBC allows presentation of its CLI using privacy-header (Privacy: none or privacy-header not present) Ensure that the expected CLI is presented to the PSTN line. Either party terminates call.	Pass	
IOP28	Verify CLIR service on PSTN line (outgoing call from IP-PBX)	Ensure number used in From/PAI header is agreed with Virgin Media and entered into the soft switch database for screening purposes. Call made from an IP-PBX line to a PSTN line. Ensure that the eSBC is configured such that the IP-PBX line sends From and/or PAI header containing either the Calling Line ID or obscured information in the INVITE. e.g. From: "user751000" <sip:+441256751000@192.168.1.10>;tag=12345 From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=12345 Ensure that the eSBC restricts presentation of its CLI using privacy-header (Privacy: id or Privacy: user or Privacy: user;id) Ensure that CLI is NOT presented to the PSTN line. Either party terminates call.	Pass	
IOP29	Verify Call Forward Immediate (unconditional) on a IP-PBX line (Incoming call from PSTN, call forward terminates within IP-PBX)	Call made from a PSTN line to an IP-PBX line with call forward to a line within the same IP-PBX, Answer Call. Either party terminates call. Does the IP-PBX have configuration settings to send SIP status 181 messages to the soft switch?	Pass	
IOP30	Verify Call Forward Immediate (unconditional) on a IP-PBX line (Incoming call from PSTN, call forward terminates PSTN)	Call made from a PSTN line to an IP-PBX line with call forward to a line in the PSTN, answer call. Either party terminates call.	Pass	
IOP31	Verify Call Forward Busy on IP-PBX line (Incoming call from PSTN, call forward terminates within IP-PBX)	Call made from a PSTN line to an IP-PBX line with Call Forward Busy (or equivalent) to a line within the IP-PBX, answer call. Either party terminates call.	Not-Exec	
IOP32	Verify Call Forward No-answer on IP-PBX line (Incoming call from PSTN, call forward terminates within IP-PBX)	Call made from a PSTN line to an IP-PBX line with Call Forward No-answer (or equivalent) to a line within the IP-PBX, Answer Call. Either party terminates call.	Pass	
IOP33	Verify Call Hold Service on IP-PBX (Incoming call from PSTN)	Call made from a PSTN line to an IP-PBX line with Call Hold, answer call. IP-PBX line puts the call on hold. Leave call on hold for 30 seconds and then retrieve call. Ensure speech path is re-established in both directions. Either party terminates call.	Pass	
IOP34	Verify 3-party conference service on IP-PBX (Incoming call from PSTN, 3rd party within IP-PBX)	Call made from a PSTN line to an IP-PBX line with 3-party conference, answer call. IP-PBX line uses the 3-party conference facility to put the PSTN line on hold while dialing 3rd party. (another IP-PBX line) Once the 3rd party has answered the call, place the three parties in a conference. Ensure that all parties have a two-way speech path. Keep the speech path open for at least 20 seconds. Either party terminates call.	Pass	Conference is created on the SfB 2015 server as another room /place where all other users are connected. All users must release the call to be disconnected from this conference call.
IOP35	Verify 3-party conference service on IP-PBX (Incoming call from PSTN, 3rd party PSTN)	Call made from a PSTN line to an IP-PBX line with 3-party conference, answer call. IP-PBX line uses the 3-party conference facility to put PSTN line on hold whilst dialing 3rd party. (another PSTN line) Once the 3rd party has answered the call, place the 3 parties in a conference. Ensure that all parties have a two-way speech path. Keep the speech path open for at least 20 seconds. Either party terminates call.	Pass	Conference is created on the SfB 2015 server as another room /place where all other users are connected. All users must release the call to be disconnected from this conference call.
IOP36	Verify do-not-disturb service on IP-PBX line (Incoming call from PSTN)	Does not ring. PSTN line receives an appropriate announcement or tone. Record the SIP status received from IP-PBX.	Pass	

IOP37	Verify Call park service on IP-PBX line (Incoming call from PSTN)	Call made from a PSTN line to IP-PBX line A with Call Park (or equivalent) feature active, answer call. Place the call in the Park condition. After 10 seconds, retrieve call from IP-PBX line B using the Call Park pick-up code. Ensure speech path is re-established in both directions. Either party terminates call.	Pass	
IOP38	Verify Call Waiting on an IP-PBX line, involving a PSTN line	Call made from PSTN line A to an IP-PBX line with Call Waiting active, answer call. Call made from PSTN line B to the same IP-PBX line which should receive an indication that a second call is waiting. PSTN line B receives ringback tone. IP-PBX line answers the call from PSTN line B. PSTN line A should receive an appropriate indication that they are now on hold. IP-PBX line toggles the call back to PSTN line A Ensure speech path is re-established in both directions and that PSTN line B should receive an appropriate indication that they are now on hold. Either party terminates call.	Pass	
IOP39	Verify DTMF transmission from/to IP-PBX - Inband	Configure the IP-PBX/eSBC to send DTMF transmission in-band. Call made from IP-PBX line to a PSTN line, answer call. PSTN line presses each of the keys on the number pad in turn. Note the far end experience. IP-PBX line presses each of the keys on the number pad in turn. Note the far end experience. Was the received DTMF tone reflective the length of time the key was pressed?	Pass	
IOP40	Verify DTMF transmission from/to IP-PBX - RFC 2833 - telephone-event	Configure the IP-PBX/eSBC to send DTMF transmission using RFC 2833 - telephone-event. Call made from IP-PBX line to a PSTN line, Answer call. PSTN line presses each of the keys on the number pad in turn. Note the far end experience. IP-PBX line presses each of the keys on the number pad in turn. Note the far end experience. Was the received DTMF tone reflective the length of time the key was pressed?	Pass	
IOP41	T.38 Fax transmission mode - PSTN to IP-PBX origination	Configure the ATA/IP-PBX/eSBC such that Fax transmission is sent using T.38 Version 0 Fax transmission mode. Call made from PSTN line to an IP-PBX line, answer call. Fax transmission is completed and call is terminated by either of the end terminal devices. Ensure Wireshark trace shows that T.38 Fax Transmission is used. Check that the fax is transmitted and received as expected.	Pass	
IOP42	T.38 Fax transmission mode - IP-PBX to PSTN origination	Configure the ATA/IP-PBX/eSBC such that Fax transmission is sent using T.38 Version 0 Fax transmission mode. Call made from IP-PBX line to a PSTN line, answer call. Fax transmission is completed and call is terminated by either of the end terminal devices. Ensure Wireshark trace shows that T.38 Fax Transmission is used. Check that the fax is transmitted and received as expected.	Pass	
IOP43	In-band G.711 Fax transmission mode - PSTN to IP-PBX origination	Configure the ATA/IP-PBX/eSBC such that Fax transmission is sent using in-band G.711 Fax transmission mode. Call made from PSTN line to an IP-PBX line, answer call. Fax transmission is completed and call is terminated by either of the end terminal devices. Ensure Wireshark trace shows that in-band G.711 Fax Transmission is used. Check that the fax is transmitted and received as expected.	Pass	
IOP44	In-band G.711 Fax transmission mode - IP-PBX to PSTN origination	Configure the ATA/IP-PBX/eSBC such that Fax transmission is sent using in-band G.711 Fax transmission mode. Call made from IP-PBX line to a PSTN line, Answer call. Fax transmission is completed and call is terminated by either of the end terminal devices. Ensure Wireshark trace shows that in-band G.711 Fax Transmission is used. Check that the fax is transmitted and received as expected.	Pass	
IOP45	Test of Call in progress audit function (response to in-call OPTIONS from soft switch to eSBC) & session refresh & response to UPDATE messages.	Call made from an IP-PBX line to a PSTN line, answer call. Leave the two parties in conversation for 35 minutes. Ensure Session-expires setting is 3600 or less. Ensure both parties have two-way speech at beginning and end of call. Either party terminates call. Check the Wireshark trace to ensure that in-call OPTIONS are sent by the soft switch and that the eSBC responds with status 200OK. Check to see if the eSBC sends any in-call audit SIP messages. Check for session refresh Update or Re-Invite and correct response.	Pass	

IOP46	Test of 4 simultaneous calls, 2 inbound, 2 outbound calls Vendor to configure eSBC for Round robin to ensure calls go to both Primary and secondary SBC	Configure the eSBC so that successive calls route to alternate SBCs (round robin, cyclic, etc.) Make 4 simultaneous calls 2 inbound, 2 outbound calls. Answer calls and ensure two-way speech path for each call.	Pass	
IOP47	Test of eSBC endpoint restart-recovery	Restart the eSBC and ensure that after recovery, inbound and outbound calls are successful.	Pass	
IOP48	Test of eSBC loss of Ethernet link and reconnection	Remove the Ethernet link between the eSBC and CE router. Leave in this condition for at least 3 minutes. Reconnect the Ethernet link and ensure that after approximately 2 minutes inbound and outbound calls are successful.	Pass	
IOP49	Test of Primary SBC loss	** Contact MSL engineer to carry out the following ** On the Primary SBC carry out the ALLSTOP command to disable the SBC. Call made from IP-PBX line to a PSTN Line. Call should attempt to route to Primary SBC. On non-response to INVITE, eSBC re-routes the call to the Secondary SBC. Wait for call answer. Either party terminates call. ** Contact MSL engineer to carry out the following ** Restart the Primary SBC	Pass	
IOP51	Test of verify call forward Internal Busy	Additional test to cover when vendors are using Microsoft Skype for Business 2015. PBX Subscriber 1 to make call to another PBX Subscriber 2 so that PSTN to call PBX subscriber 1 is Busy. PSTN call PBX user 1. The call should automatically go to voicemail after 10 seconds when forwarding is off. VM is on another PBX Internal Line call should go to Voice Mail. If voicemail PSTN to listen VM announcement if another PBX user, check speech is clear in both directions. If forwarded to voicemail PSTN terminated call after hearing VM announcement. If forwarded to another user either party terminate the call after checking speech is clear in both directions.	Pass	
IOP52	Test of Call forward internal on No Answer	Additional test to cover when vendors are using Microsoft Skype for Business 2015. PSTN call PBX user 1. PBX User 1 is not to answer the call. The call should automatically go to voicemail (VM) which is in another internal PBX line if call forwarding is turned off. Call automatically goes to voicemail after 10 seconds. PSTN terminated call after hearing VM announcement. If forwarded is ON call is forwarded to another PBX user internal. Check speech quality, terminate the call after checking speech is clear in both directions.	Pass	
IOP53	Test Call from PBX to PSTN	1. Configure eSBC to offer T.38 in addition to G711A-law and G711-U law. 2. Call made from PBX to PSTN. 3. Call established and two dialogs for 10 minutes. 4. Check Wireshark output. You should not see T.38 being reflected in the protocol column after the call has been established for 7 minutes. 5. If T.38 is reflected in the protocol column, make a note of this.	Pass	