
Ribbon EdgeMarc 2900A configuration Cisco Unified Communication Manager CUCM

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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 Ribbon 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:

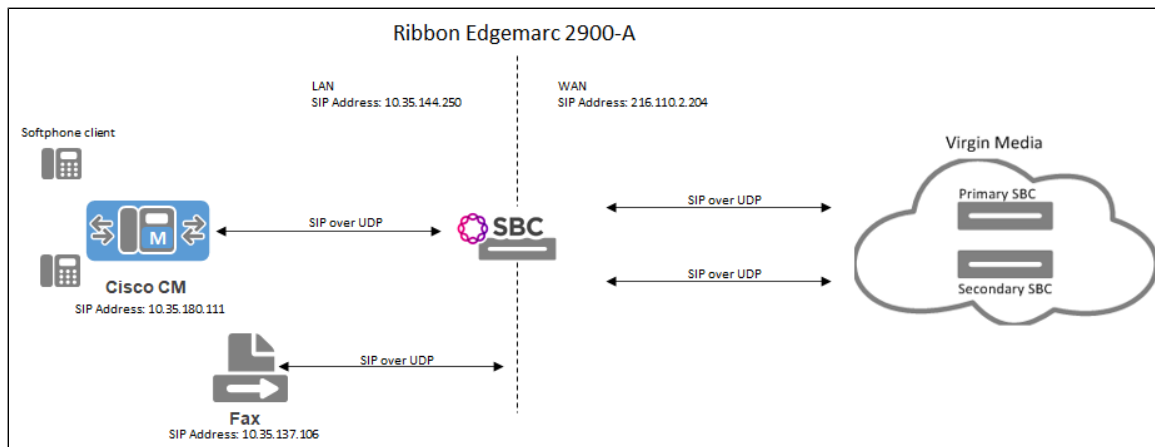
Table 1: Requirements

| | 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 ▼ |
| <input type="checkbox"/> Redirect by Application <input type="checkbox"/> Disable Early Media on 180 <input type="checkbox"/> Outgoing T.38 INVITE include audio mline <input type="checkbox"/> Offer valid IP and Send/Receive mode only for T.38 Fax Relay <input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests <input type="checkbox"/> Assured Services SIP conformance <input type="checkbox"/> 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 ▼ |
| <input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change <input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556) | |
| Parameters used in Phone | |
| Timer Invite Expires (seconds)* | 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 | <input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> 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 |
| <input checked="" type="checkbox"/> Conference Join Enabled | |
| <input type="checkbox"/> RFC 2543 Hold | |
| <input checked="" type="checkbox"/> Semi Attended Transfer | |
| <input type="checkbox"/> Enable VAD | |

Stutter Message Waiting
 MLPP User Authorization

Normalization Script

Normalization Script

Enable Trace

| | Parameter Name | Parameter Value | | |
|---|----------------------|----------------------|----------------------------------|----------------------------------|
| 1 | <input type="text"/> | <input type="text"/> | <input type="button" value="+"/> | <input type="button" value="-"/> |

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Resource Priority Namespace List

SIP Rel1XX Options*

Video Call Traffic Class*

Calling Line Identification Presentation*

Session Refresh Method*

Early Offer support for voice and video calls*

Enable ANAT

Deliver Conference Bridge Identifier

Allow Passthrough of Configured Line Device Caller Information

Reject Anonymous Incoming Calls

Reject Anonymous Outgoing Calls

Send ILS Learned Destination Route String

Connect Inbound Call before Playing Queuing Announcement

SIP OPTIONS Ping

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)*

Ping Interval for Out-of-service Trunks (seconds)*

Ping Retry Timer (milliseconds)*

Ping Retry Count*

SDP Information

Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow multiple codecs in answer SDP

2. SIP Trunk Security Profile

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

Figure 3: SIP Trunk Security Profile

SIP Trunk Security Profile Information

Name*

Description

Device Security Mode

Incoming Transport Type*

Outgoing Transport Type

Enable Digest Authentication

Nonce Validity Time (mins)*

X.509 Subject Name

Incoming Port*

Enable Application level authorization

Accept presence subscription

Accept out-of-dialog refer**

Accept unsolicited notification

Accept replaces header

Transmit security status

Allow charging header

SIP V.150 Outbound SDP Offer Filtering*

3. Trunk

Select **Device > Trunk**

Figure 4: Trunk

SIP Trunk Status

Service Status: Unknown
Duration: Unknown

Device Information

Product: SIP Trunk

Device Protocol: SIP

Trunk Service Type: None(Default)

Device Name*

Description

Device Pool*

Common Device Configuration

Call Classification*

Media Resource Group List

Location*

AAR Group

Tunneled Protocol*

QSIG Variant*

ASN.1 ROSE OID Encoding*

Packet Capture Mode*

Packet Capture Duration

Media Termination Point Required

Retry Video Call as Audio

Path Replacement Support

Transmit UTF-8 for Calling Party Name

Transmit UTF-8 Names in QSIG APDU

Unattended Port

SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*

Route Class Signaling Enabled*

Use Trusted Relay Point*

PSTN Access

Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)
E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information
MLPP Domain < None >
Confidential Access Mode < None >
Confidential Access Level < None >

Call Routing Information
 Remote-Party-Id
 Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default
Trust Received Identity* Trust All (Default)

Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
 Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings | Default Prefix Settings

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|---------|--------------|----------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <input checked="" type="checkbox"/> |

Incoming Called Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings | Default Prefix Settings

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|---------|--------------|----------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <input checked="" type="checkbox"/> |

Connected Party Settings
Connected Party Transformation CSS < None >
 Use Device Pool Connected Party Transformation CSS

Outbound Calls
Called Party Transformation CSS < None >
 Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS < None >
 Use Device Pool Calling Party Transformation CSS
Calling Party Selection* Originator
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling and Connected Party Info Format* Deliver DN only in connected party
 Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS < None >
 Use Device Pool Redirecting Party Transformation CSS

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

SIP Information

Destination
 Destination Address is an SRV

| Destination Address | Destination Address IPv6 | Destination Port | Status | Status Reason | Duration |
|---------------------|--------------------------|------------------|--------|---------------|----------|
| 1* 10.35.144.250 | | 5060 | N/A | N/A | N/A |

MTP Preferred Originating Codec* 711ulaw
BLF Presence Group* Standard Presence group
SIP Trunk Security Profile* Non Secure SIP Trunk Profile
Rerouting Calling Search Space < None >
Out-Of-Dialog Refer Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* Kapanga SIP Profile [View Details](#)
DTMF Signaling Method* RFC 2833

Normalization Script
Normalization Script Outbound_SIP_Remove_Call_Info
 Enable Trace

| Parameter Name | Parameter Value |
|----------------|-----------------|
| 1 | |

Recording Information
 None
 This trunk connects to a recording-enabled gateway
 This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration
Geolocation < None >
Geolocation Filter < None >
 Send Geolocation Information

Save | Delete | Reset | Add New

4. Route Group

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

Figure 5: Route Group

Route Group Information

Route Group Name*

Distribution Algorithm*

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains

Available Devices**

CUBE

EM_VirginMedia

Rewa

SBC_5K_PlusNet

USTX-SBCLAB01

Port(s)

Current Route Group Members

Selected Devices (ordered by priority)*

EM_VirginMedia (All Ports)

Removed Devices***

Route Group Members


[EM_VirginMedia](#)

5. Route List

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

Figure 6: Route List

Status

 Status: Ready

Route List Information

Registration: Registered with Cisco Unified Communications Manager UCM12.vo.sonusnet.com
 IPv4 Address: 10.35.180.111

Device is trusted

Name*

Description

Cisco Unified Communications Manager Group*

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

Run On All Active Unified CM Nodes


Route List Member Information

Selected Groups**

▼ ▲

Removed Groups***

Route List Details

 [EM_VirginMedia](#)

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*
 Route Partition
 Description
 Numbering Plan
 Route Filter
 MLPP Precedence*
 Apply Call Blocking Percentage
 Resource Priority Namespace Network Domain
 Route Class*
 Gateway/Route List* [\(Edit\)](#)
 Route Option
 Route this pattern
 Block this pattern
 Call Classification*
 External Call Control Profile
 Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
 Require Forced Authorization Code
 Authorization Level*
 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*
 Calling Name Presentation*
 Calling Party Number Type*
 Calling Party Numbering Plan*

Calling Party Transformations

Use Calling Party's External Phone Number Mask
 Calling Party Transform Mask
 Prefix Digits (Outgoing Calls)
 Calling Line ID Presentation*
 Calling Name Presentation*
 Calling Party Number Type*
 Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*
 Connected Name Presentation*

Called Party Transformations

Discard Digits
 Called Party Transform Mask
 Prefix Digits (Outgoing Calls)
 Called Party Number Type*
 Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol
 Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|---|--|-------------------------|
| <input type="text" value="-- Not Selected --"/> | <input type="text" value="< Not Exist >"/> | <input type="text"/> |

Status

 Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

External Call Control Profile

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

Require Forced Authorization Code

Authorization Level*

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*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*


ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--|---|-------------------------|
| <input type="text" value=" -- Not Selected --"/> | <input type="text" value=" < Not Exist >"/> | <input type="text"/> |

Status

 Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class*

Gateway/Route List* [\(Edit\)](#)

Route Option
 Route this pattern
 Block this pattern

Call Classification*

External Call Control Profile

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

Require Forced Authorization Code

Authorization Level*

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*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*


ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|---|--|-------------------------|
| <input type="text" value="-- Not Selected --"/> | <input type="text" value="< Not Exist >"/> | <input type="text"/> |

Status

 Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

External Call Control Profile

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

Require Forced Authorization Code

Authorization Level*

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*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|---|--|-------------------------|
| <input type="text" value="-- Not Selected --"/> | <input type="text" value="< Not Exist >"/> | <input type="text"/> |

EdgeMarc Configuration

Network

- LAN and WAN Interfaces
- Static Routes

VoIP

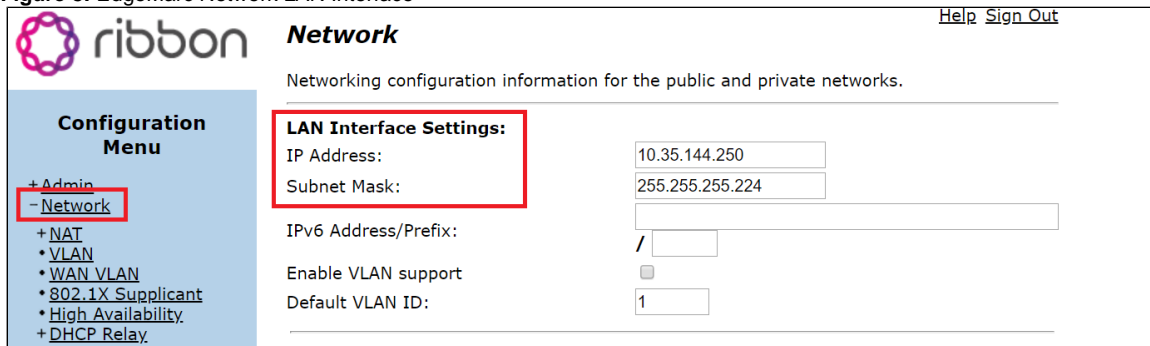
- VoIP Settings
- SIP Settings
- B2BUA

Network

LAN and WAN Interfaces

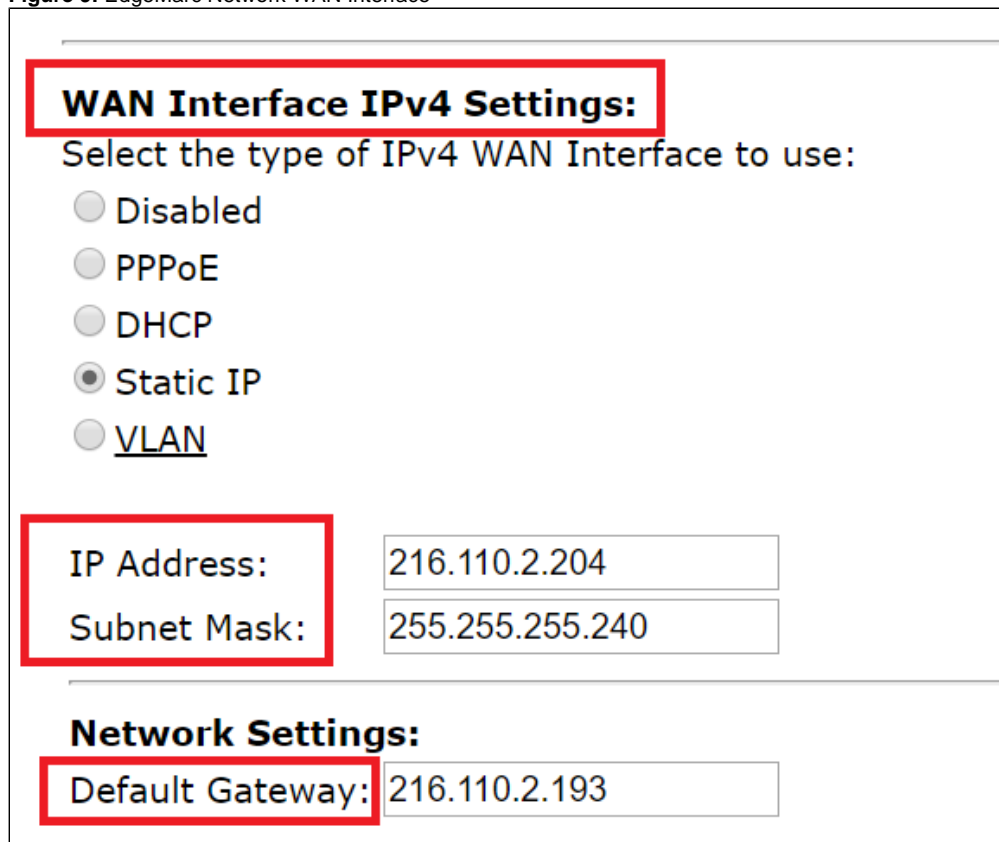
Login to EdgeMarc as root user and go to **Network** to configure the LAN and WAN interfaces.

Figure 8: EdgeMarc Network LAN Interface



The screenshot shows the EdgeMarc Network configuration page for the LAN interface. The ribbon logo is in the top left, and 'Network' is in the top center. A 'Help Sign Out' link is in the top right. Below the ribbon logo is a 'Configuration Menu' with options: + Admin, - Network (highlighted), + NAT, + VLAN, + WAN VLAN, + 802.1X Supplicant, + High Availability, and + DHCP Relay. The main content area is titled 'LAN Interface Settings:' and contains the following fields: IP Address (10.35.144.250), Subnet Mask (255.255.255.224), IPv6 Address/Prefix (empty), Enable VLAN support (checkbox), and Default VLAN ID (1).

Figure 9: EdgeMarc Network WAN Interface



The screenshot shows the EdgeMarc Network configuration page for the WAN interface. The main content area is titled 'WAN Interface IPv4 Settings:' and contains the following options: Disabled, PPPoE, DHCP, Static IP (selected), and VLAN. Below these options are the following fields: IP Address (216.110.2.204), Subnet Mask (255.255.255.240), and Network Settings: Default Gateway (216.110.2.193).

Static Routes

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

Figure 10: Static Routes

Menu

- + Admin
- Network
 - + NAT
 - VLAN
 - WAN VLAN
 - 802.1X Supplicant
 - High Availability
 - + DHCP Relay
 - + DHCP Server
 - + Traffic Shaper
 - Pass-Through Rules
 - Subinterfaces
 - Proxy ARP
 - Switch Ports
 - **Static Routes**
 - Dynamic DNS
 - Network Information
 - Network Restart
 - Network Test Tools
- + WAN Failover
- Router Advertisement
- IP Multicast

- + Users
- + Security
- SD-WAN
- + VoIP
- + VPN
- GRE

Static Routes

Select: All [None](#)
Delete

| | IP Network | Network Mask | Gateway |
|--------------------------|---------------|-----------------|---------------|
| <input type="checkbox"/> | 172.24.29.232 | 255.255.255.248 | 10.35.144.225 |
| <input type="checkbox"/> | 10.128.176.31 | 255.255.255.0 | 10.35.144.225 |
| <input type="checkbox"/> | 172.24.26.0 | 255.255.255.0 | 10.35.144.225 |
| <input type="checkbox"/> | 10.35.180.112 | 255.255.255.255 | 10.35.144.225 |
| <input type="checkbox"/> | 172.24.18.0 | 255.255.255.0 | 10.35.144.225 |
| <input type="checkbox"/> | 10.10.216.0 | 255.255.255.192 | 10.35.144.225 |
| <input type="checkbox"/> | 10.35.137.175 | 255.255.255.255 | 10.35.144.225 |
| <input type="checkbox"/> | 10.35.180.111 | 255.255.255.255 | 10.35.144.225 |
| <input type="checkbox"/> | 172.17.240.0 | 255.255.255.0 | 10.35.144.225 |
| <input type="checkbox"/> | 10.35.180.229 | 255.255.255.255 | 10.35.144.225 |
| <input type="checkbox"/> | 10.35.137.106 | 255.255.255.255 | 10.35.144.225 |
| <input type="checkbox"/> | 82.14.171.0 | 255.255.255.0 | 216.110.2.193 |
| <input type="checkbox"/> | 213.106.222.0 | 255.255.255.0 | 216.110.2.193 |

VoIP

VoIP Settings

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

Figure 11: VoIP

Configuration Menu

- + Admin
- + Network
- + Users
- + Security
- **SD-WAN**
- **VoIP**
 - H.323
 - + SIP
 - Survivability
 - Clients List
 - Test UA
- + VPN
- GRE

[Help](#) [Sign Out](#)

VoIP

VoIP ALG allows the system to recognize and register network devices.

Enable LLDP:

LLDP Broadcast Interval (sec):

IPv4 only.

TFTP Server IP address:

In some cases, the ALG addresses will not correspond to the addresses of the LAN or the WAN ports. The addresses will be alias addresses that have been configured on the ports. In general, the user should leave this feature disabled.

Use ALG Alias IP Addresses:

ALG LAN Interface IP Address: 10.35.144.250

ALG LAN Interface IPv6 Address:

ALG WAN Interface IP Address: 216.110.2.204

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:

B2BUA Options:

Route all SIP signalling through B2BUA:

Enable Microsoft Feature:

Enable Comfort Noise Generation (CNG):

Enable User-Agent header pass-through:

This feature assign a fixed out bound source port for a particular user from the range given below. This port will be used for all the SIP transaction for that particular user.

Enable multi-ports:

Multi-port Port Range: -

Media Security:

Enable SRTP support:

Enable MKI support:

Configure the range of TCP ports to use for handling H.225 and H.245 TCP connections.

H.225/H.245 Port Range: -

Configure the range of UDP ports to use for forwarding RTP streams. Each RTP stream to be forwarded requires two ports (one for RTP and one for RTCP). This means that you will need at least two times as many ports as RTP streams you want to handle.

RTP Port Range: -

RTP Packetization Time (ms):

Prioritize Microsoft Teams:

Allow non-translated RTP to be MOS scored:

RTP range to MOS score:

Calculate Round-Trip-Time:

Calculate RTT:

The ALG feature is registered. View [license key](#).

SIP Settings

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

Figure 12: SIP



SIP Settings

[Help](#) [Sign Out](#)

SIP protocol settings.

The SIP Server settings specify the address and port that all client traffic shall be forwarded to.

SIP Server Transport:

Use Custom Domain:

SIP Server Domain:

List of SIP Servers

Select: [All](#) [None](#)

| | Lookup Status | Priority | SIP Server Address/FQDN | Port |
|--------------------------|--------------------------------------|----------|-------------------------|------|
| <input type="checkbox"/> | ● | 0 | 213.106.222.178 | 5060 |
| <input type="checkbox"/> | ● | 1 | 82.14.171.226 | 5060 |

Add a new SIP Server

IP Address/FQDN:

Port:

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:

Configuration Menu

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+ [H.323](#)

+ [SIP](#)

+ [ALG](#)

+ [B2BUA](#)

+ [SIP UA](#)

+ [SIP GW](#)

+ [Trunking Group](#)

+ [Availability](#)

+ [Media Server](#)

+ [Survivability](#)

+ [Clients List](#)

+ [Test UA](#)

+ [VPN](#)

+ [GRE](#)

B2BUA Options:

B2BUA Redirect Support (302):

PANI Header

Enable PANI Header Support:

Access Type:

Access Info:

Access Info String:

Session Timer

Session Timer Support:

Session Refresh Interval (s):

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](#)

| | SIP Server Address/FQDN | Port | Transport |
|--------------------------|-------------------------|------|-----------|
| <input type="checkbox"/> | 213.106.222.179 | 5060 | UDP |
| <input type="checkbox"/> | 82.14.171.227 | 5060 | UDP |

Add a new Allowed SIP Server

IP Address/FQDN:

Port:

Transport:

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):

UDP

Client Listening Port(s):

The system will also listen on the Server Facing Port for incoming SIP requests.

Server Facing Port:

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:

TCP

Port:

Timeout (minutes):

TLS

Port:

TLS Protocol:

Ciphers String:

LAN: Certificate: Policy:

WAN: Certificate: Policy:

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

SDP Modifications

SDP Codec Operation:

SDP Section that will be modified:

Codecs (comma separated list):

Reject when No Match Codec:

Strip Matched Expressions:

SIP Use New Port On Hold Resume:

Priority Numbers

Priority Number 1:

Priority Number 2:

Priority Number 3:

Priority Number 4:


Enable SIP Statistics:

Registration Rate-Pacing parameters are available on the [Survivability page](#).

B2BUA

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

Figure 13: B2BUA



B2BUA Trunking Configuration

[Help](#) [Sign Out](#)

This page supports only IPv4 addressing.

In order for changes to this page to be applied, you must click the "Submit" or "Apply Later" button at the bottom of the page

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

Trunking Devices

| Name | Address | Port | Group | Username | Registration Status | Transport |
|------------|---------------|------|-------|----------|---------------------|-----------|
| ✗ CUCM | 10.35.180.111 | 5060 | | | | UDP |
| ✗ SFB | 10.35.180.229 | 5068 | | | | UDP |
| ✗ Asterisk | 10.10.216.13 | 5060 | | | | UDP |
| ✗ VentaFax | 10.35.137.106 | 5060 | | | | UDP |

New Entry

Name: Model:

Address(IP/FQDN):
 Use DNS SRV:

Port: Transport:

Source FQDN:

Username: Password:

Authenticate Registration:

Credentials and Registration

| AOR | Auth-User | Password | Registrar | Status | Transport |
|--|-------------------------------------|------------|--|----------------------|-----------|
| default | virginpbx01_01183374130 | is set | | | |
| <i>New Entry</i> | | | | | |
| Credentials | | | | | |
| Username: | <input type="text"/> | Auth-User: | <input type="text" value="virginpbx01_01183374130"/> | | |
| Edit Password: | <input type="checkbox"/> | | | | |
| Password: | <input type="text"/> | | | | |
| Confirm Password: | <input type="text"/> | | | | |
| Use as default: | <input checked="" type="checkbox"/> | | | | |
| Registrar | | | | | |
| <input checked="" type="radio"/> Don't Register <input type="radio"/> Default SIP Proxy | | | | | |
| Custom URI Domain: | <input type="text"/> | | | | |
| Domain: | <input type="text"/> | | | | |
| Address (optional): | <input type="text"/> | Port: | <input type="text"/> | | |
| Transport: | <input type="text" value="UDP"/> | | | | |
| Register Options (Optional) | | | | | |
| Default Expires: | <input type="text"/> | sec. | Renew interval: | <input type="text"/> | % |
| <input type="button" value="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 | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | | | | | | | |
| <i>New Entry</i> | | | | | | | | | |
| Name: | <input type="text" value="UK1"/> | | | | | | | | |
| Country Code | | | | | | | | | |
| <input type="checkbox"/> Select all headers | <input type="text" value="Australia"/> | | | | | | | | |
| <input checked="" type="checkbox"/> Request URI: | <input type="text" value="UK"/> | | | | | | | | |
| <input checked="" type="checkbox"/> To: | <input type="text" value="UK"/> | | | | | | | | |
| <input type="checkbox"/> From: | <input type="text" value="Australia"/> | | | | | | | | |
| <input type="checkbox"/> Contact: | <input type="text" value="Australia"/> | | | | | | | | |
| <input type="checkbox"/> Refer-To: | <input type="text" value="Australia"/> | | | | | | | | |
| <input type="checkbox"/> Referred-By: | <input type="text" value="Australia"/> | | | | | | | | |
| <input type="checkbox"/> History-Info: | <input type="text" value="Australia"/> | | | | | | | | |
| <input type="checkbox"/> P-Asserted-Identity: | <input type="text" value="Australia"/> | | | | | | | | |
| <input type="checkbox"/> P-Preferred-Identity: | <input type="text" value="Australia"/> | | | | | | | | |
| <input type="button" value="Update"/> | | | | | | | | | |

Actions

| | Name | Send | Prio | Hunt | Header | Refer-To-ReINV |
|---|------------|------|------|------|--------|----------------|
| ✖ | plusdelete | ✓ | | | ✓ | |
| ✖ | PAI | ✓ | | | ✓ | |
| ✖ | Privacy | ✓ | | | ✓ | |
| ✖ | FAX | ✓ | | | ✓ | |
| ✖ | UK1 | ✓ | | | ✓ | |
| ✖ | CUCM | ✓ | | | ✓ | |
| ✖ | SfB | ✓ | | | | |
| ✖ | Asterisk | ✓ | | | | |

New Entry

Name:

Send To: Trunking Device:
 Client:
 URI:
 Response:

Prioritize: Refer to Re-INVITE:

Serial Hunting:

E.164 Conversion rule: Conversion mode:

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:

Value:

Response Code Mapping

| Name | From | To | Response Code Mapping |
|--|--|---|------------------------------------|
| <i>New Entry</i> | | | |
| Name: <input type="text"/> | From: <input type="text" value="Any"/> <input type="button" value="v"/> | To: <input type="text" value="Any"/> <input type="button" value="v"/> | |
| Response Code Manipulations: | | | |
| Received Code | Mapped Code | Mapped Phrase | |
| Received Code: <input type="text" value="404"/> <input type="button" value="v"/> | Mapped Code: <input type="text" value="403"/> <input type="button" value="v"/> | Mapped Phrase: <input type="text"/> | <input type="button" value="Add"/> |
| <input type="button" value="Update"/> | | | |

Match

| | Direction | Mode | Def | Called | | Calling | | Source | Action |
|-------------------------------------|-----------|-----------|-----|---------|---------|---------|---------|--------|--------|
| | | | | Match | Pattern | Match | Pattern | | |
| <input checked="" type="checkbox"/> | Inbound | BothModes | | matches | . | | | Any | CUCM |
| <input checked="" type="checkbox"/> | Outbound | BothModes | | matches | . | | | Any | UK1 |

New Entry

Direction:

Mode:

default

Pattern:

Called Party:

Calling Party:

Source:

Action:

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 | <p>No calls are required for this test. SIP trace to be captured for approximately 60 seconds and checked for correct signaling.</p> <p>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.</p> <p>e.g. OPTIONS sip:ping@<ip-pbx_IP_Addr>;5060 SIP/2.0</p> | Pass | |
| IOP2 | SBC response to SIP OPTIONS messages from vendor eSBC | <p>No calls are required for this test. SIP trace to be captured for approximately 60 seconds (depending on agreement) and checked for correct signaling.</p> <p>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"</p> <p>Verify that the eSBC can simultaneously send SIP OPTIONS messages to both the solution SBC addresses.</p> | Pass | |
| IOP4 | Basic test call from IP-PBX to PSTN line through SBC-A (using SBC-A IPV4 ip address). | <p>IP-PBX line initiates call, Call is answered, IP-PBX line terminates call.</p> <p>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></p> <p>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)</p> | Pass | |

| | | | | |
|--------|--|--|------|--|
| IOP5 | <p>Basic test call from IP-PBX to PSTN line through SBC-B (using SBC-B IPv4 ip address)</p> <p>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.</p> | <p>IP-PBX line initiates call, Call is answered, IP-PBX line terminates call.</p> <p>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></p> <p>Check the wireshark trace and confirm that G.711 A law codec with 10ms or 20ms packetisation is being used.</p> | Pass | |
| IOP7b | <p>Called Number format - vendors eSBC to soft switch number normalization - Global Dial Plan</p> <p>Test eSBC capability to send the called number in one of the following Global number formats (user part of Request & To URIs)</p> <p>0yyyyyyyyy (where y refers to any number, calling party = national) +44yyyyyyyyy (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)</p> | <p>SBC to be configured for Global calling plan.</p> <p>IP-PBX line initiates call to PSTN line, Call is answered. IP-PBX line terminates call.</p> <p>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</p> <p>0yyyyyyyyy (where y refers to any number, calling party = national) +44yyyyyyyyy (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)</p> | Pass | |
| IOP8b | <p>Calling Number format - vendors eSBC to soft switch number normalization - Global Dial Plan</p> <p>Test eSBC capability to send calling number in one of the following Global number formats (user part of FROM & PAI URIs)</p> <p>0yyyyyyyyy (where y refers to any number, calling party = national) +44yyyyyyyyy (where y refers to any number, calling party = national) 00yyyyyyyyy (where y refers to any number, calling party = international) yyyyyyyyyy (where y refers to any number, calling party = unknown)</p> | <p>SBC to be configured for Global calling plan.</p> <p>IP-PBX line initiates call to PSTN line, Call is answered. IP-PBX terminates call.</p> <p>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</p> <p>0yyyyyyyyy (where y refers to any number, calling party = national) +44yyyyyyyyy (where y refers to any number, calling party = national) 00yyyyyyyyy (where y refers to any number, calling party = international) yyyyyyyyyy (where y refers to any number, calling party = unknown)</p> | Pass | |
| IOP9b | <p>Called Number format - soft switch to eSBC number normalization - Global Dial Plan</p> <p>Test eSBC capability of accepting the called number in one of the following Global number formats (user part of Request & To URIs)</p> <p>+44yyyyyyyyy (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)</p> | <p>SBC to be configured for Global calling plan.</p> <p>PSTN line initiates call to IP-PBX line, Call is answered. PSTN line terminates call.</p> <p>Configure the eSBC to accept the called number in the user part of the Request & To URIs in one of the following formats</p> <p>+44yyyyyyyyy (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)</p> <p>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'.</p> | Pass | |
| IOP10b | <p>Calling Number format - soft switch to eSBC number normalization - Global Dial Plan</p> <p>Test eSBC capability of accepting the calling number in one of the following Global number formats (user part of FROM & PAI URIs)</p> <p>+44yyyyyyyyy (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)</p> | <p>SBC to be configured for Global calling plan.</p> <p>PSTN line initiates call to IP-PBX line, Call is answered. PSTN line terminates call.</p> <p>Configure the eSBC to accept the calling number in the user part of the Request & To URIs in one of the following formats</p> <p>+44yyyyyyyyy (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)</p> | 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 0118911111 | 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 | |

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| 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 | <ol style="list-style-type: none"> 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 | |