
RIBBON SBC Edge V7.0.0 IOT Skype for Business 2015 Airtel MUX Application Notes

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Document Overview

This document describes configuring the Ribbon **SBC 1000 / 2000 Series (Session Border Controller)** when connecting to **Skype for Business 2015**. For additional information about Ribbon SBCs, visit <https://ribboncommunications.com/>

Introduction


The interoperability compliance testing focuses on verifying inbound and outbound call flows between the Ribbon SBC 1000 / 2000 and Skype for Business 2015.

Document History

| Date | Name | Comment |
|------|------|---------------|
| | | Initial Draft |
| | | |

Audience

This technical document is provided for use by telecommunications engineers and network administrators with understanding of networking concepts such as TCP/UDP, IP/Routing, and SIP/RTP, along with experience using industry-standard utilities and tools. The information in this guide describes configuring and operating Ribbon SBCs. Some information describes using third-party products when administering and troubleshooting SBC operation.

 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

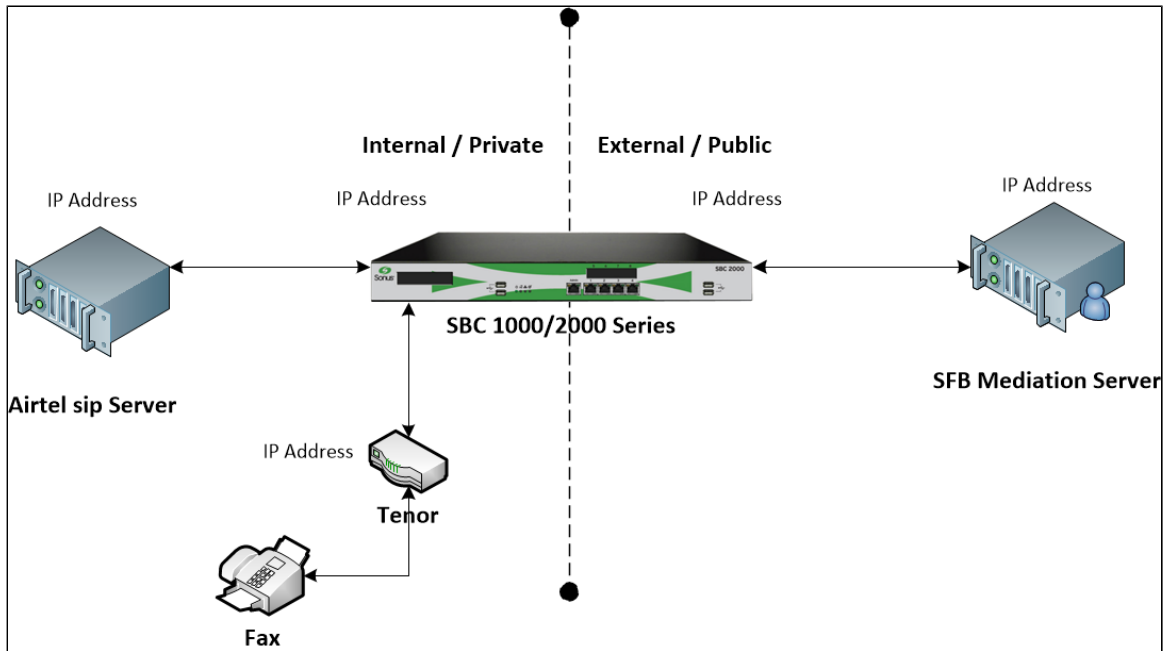
The following equipment and software are used in the reference configuration:

| | Equipment | Software Version |
|------------------------------|-----------------------|------------------|
| RIBBON Networks | RIBBON SBC 2000 Tenor | V7.0 |
| Third-party Equipment | SFB Mediation Server | |
| OS | | |
| Other software | | |

Reference Configuration

The following reference diagram shows connectivity between the Ribbon SBC 1000 / 2000 and third-party equipment that interoperates with the SBC.

Figure 1: Topology



Support

For questions about information in this document, contact Ribbon Support in either of the following ways:

- Global Support Assistance Center +1-978-614-8589 or +1-888-391-3434 (English language Support)
- Web: <https://ribboncommunications.com/services/ribbon-support-portal-login>

Ribbon SBC Edge Configuration

The following steps provide an example of how to configure the Ribbon SBC 1000/2000.

1. [Media Profile](#)
2. [Tone Table](#)
3. [Transformation Table](#)
4. [Sip Profile](#)
5. [Sip Remote Authorization Entry](#)
6. [Sip Contact Registration](#)
7. [Server Table](#)
8. [Signaling Group](#)

1. Media Profile

Select **Settings > Media > Media Profiles**

Media Profiles specify the individual voice and fax compression codecs and their associated settings for inclusion into a Media List. Different codecs provide varying levels of compression, allowing the reduction of bandwidth requirements. Listed below are the media profiles of the voice codecs used for testing the SBC 2000:

Note: The Digit Relay Payload Type must be set to 97.

Figure 2: SFB Media List

Media List Details: SFB Media List

Description **SFB Media List**

Media Profiles List **Default G711A** *

Crypto Profile ID **None**

Media DSCP **46**

RTCP Mode **RTCP**

Dead Call Detection **Disabled**

Silence Suppression **Disabled**

| Gain Control | | Digit Relay | |
|---------------|---|--------------------------|----------|
| Receive Gain | 0 | Digit (DTMF) Relay Type | RFC 2833 |
| Transmit Gain | 0 | Digit Relay Payload Type | 97 |

Passthrough/Tone Detection

Modem Passthrough **Disabled**

Fax Passthrough **Disabled**

CNG Tone Detection **Disabled**

Fax Tone Detection **Disabled**

DTMF Signal to Noise **0**

DTMF Minimum Level **-38**

Figure 3: Airtel Media List

Media List Details: Airtel Media List

Description **Airtel Media List**

Media Profiles List **Default G711A** *

Crypto Profile ID **None**

Media DSCP **46**

RTCP Mode **RTCP**

Dead Call Detection **Disabled**

Silence Suppression **Disabled**

| Gain Control | | Digit Relay | |
|---------------|---|--------------------------|----------|
| Receive Gain | 0 | Digit (DTMF) Relay Type | RFC 2833 |
| Transmit Gain | 0 | Digit Relay Payload Type | 97 |

Passthrough/Tone Detection

Modem Passthrough **Disabled**

Fax Passthrough **Disabled**

CNG Tone Detection **Disabled**

Fax Tone Detection **Disabled**

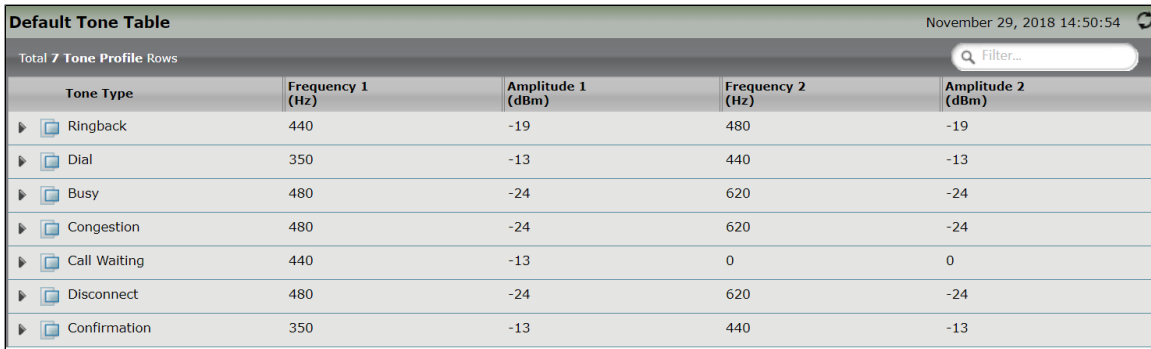
DTMF Signal to Noise **0**

DTMF Minimum Level **-38**

2. Tone Table

Select **Settings > Tone Tables**. Use default settings to specify a tone table.

Figure 4: Tone Tables



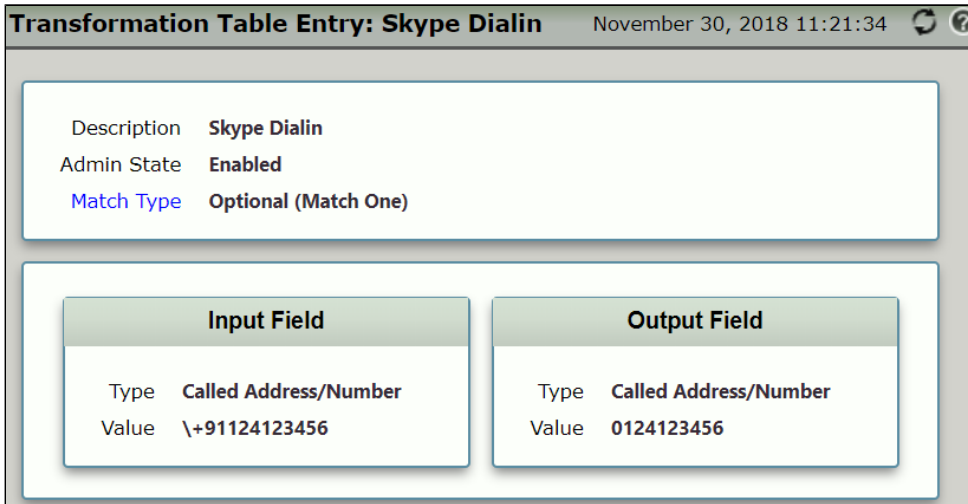
| Tone Type | Frequency 1 (Hz) | Amplitude 1 (dBm) | Frequency 2 (Hz) | Amplitude 2 (dBm) |
|----------------|------------------|-------------------|------------------|-------------------|
| ▶ Ringback | 440 | -19 | 480 | -19 |
| ▶ Dial | 350 | -13 | 440 | -13 |
| ▶ Busy | 480 | -24 | 620 | -24 |
| ▶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

3. Transformation Table

Select **Settings > Transformation**

Transformation Tables facilitate the conversion of names, numbers, and other fields when routing a call. They can, for example, convert a PSTN number (public) into a private extension number, or into a SIP address (URI). Each entry in a Call Routing Table requires a Transformation Table.

Figure 5: Inbound Transformation Table



Transformation Table Entry: Skype Dialin November 30, 2018 11:21:34

Description **Skype Dialin**
Admin State **Enabled**
Match Type **Optional (Match One)**

| Input Field | Output Field |
|---|--|
| Type Called Address/Number Value \+91124123456 | Type Called Address/Number Value 0124123456 |

Figure 6: Outbound Transformation Table

Transformation Table Entry: Caller-ID Outbound November 30, 2018 11:27:18

| | |
|-------------|-----------------------------|
| Description | Caller-ID Outbound |
| Admin State | Enabled |
| Match Type | Optional (Match One) |

| Input Field | Output Field |
|-----------------------------------|-----------------------------------|
| Type Called Address/Number | Type Called Address/Number |
| Value (.*) | Value \1 |

4. Sip Profile

Select Settings > SIP > SIP Profiles.

SIP Profiles control how the SBC Edge communicates with SIP devices. The SIP Profile controls important characteristics such as the following: session timers, SIP header customization, SIP timers, MIME payloads, and option tags.

Figure 7: SFB SIP Profile

SIP Profile Entry: SFB Profile

Description SFB Profile

| Session Timer | | MIME Payloads | |
|------------------------------|--------|-----------------------------|---------|
| Session Timer | Enable | ELIN Identifier | LOC |
| Minimum Acceptable Timer | 600 | PIDF-LO Passthrough | Enable |
| Offered Session Timer | 3600 | Unknown Subtype Passthrough | Disable |
| Terminate On Refresh Failure | False | | |

| Header Customization | | Options Tags | |
|----------------------------|-------------------|--------------|-------------|
| FQDN in From Header | Disable | 100rel | Supported |
| FQDN in Contact Header | Disable | Path | Not Present |
| Send Assert Header | Trusted Only | Timer | Supported |
| Sonus Diagnostics Header | Enable | Update | Supported |
| Trusted Interface | Enable | | |
| UA Header | Sonus SBC | | |
| Calling Info Source | RFC Standard | | |
| Diversion Header Selection | Last | | |
| Record Route Header | RFC 3261 Standard | | |

| Timers | | SDP Customization | |
|-------------------------|-----------------------|----------------------------------|------------------|
| Transport Timeout Timer | 5000 | Send Number of Audio Channels | False |
| Maximum Retransmissions | RFC Standard | Connection Info in Media Section | True |
| ————— RFC Timers ————— | | | |
| Timer T1 | 500 | Origin Field Username | SBC |
| Timer T2 | 4000 | Session Name | VoipCall |
| Timer T4 | 5000 | Digit Transmission Preference | RFC 2833/Voice |
| Timer D | 32000 | SDP Handling Preference | Legacy Audio/Fax |
| Timer B | 32000 ms | | |
| Timer F | 32000 ms | | |
| Timer H | 32000 ms (64*TimerT1) | | |
| Timer J | 4000 | | |

Figure 8: Airtel SIP Profile

SIP Profile Entry: Airtel Profile

Description Airtel Profile

| Session Timer | |
|------------------------------|--------|
| Session Timer | Enable |
| Minimum Acceptable Timer | 600 |
| Offered Session Timer | 3600 |
| Terminate On Refresh Failure | True |

| MIME Payloads | |
|-----------------------------|---------|
| ELIN Identifier | LOC |
| PIDF-LO Passthrough | Enable |
| Unknown Subtype Passthrough | Disable |

| Header Customization | |
|----------------------------|-------------------|
| FQDN in From Header | Static |
| FQDN in Contact Header | Sonus SBC FQDN |
| Static Host | ims.airtel.in |
| Send Assert Header | Trusted Only |
| Sonus Diagnostics Header | Enable |
| Trusted Interface | Enable |
| UA Header | Sonus SBC |
| Calling Info Source | RFC Standard |
| Diversion Header Selection | Last |
| Record Route Header | RFC 3261 Standard |

| Options Tags | |
|--------------|-------------|
| 100rel | Supported |
| Path | Not Present |
| Timer | Supported |
| Update | Supported |

| Timers | |
|-------------------------|-----------------------|
| Transport Timeout Timer | 5000 |
| Maximum Retransmissions | RFC Standard |
| RFC Timers | |
| Timer T1 | 500 |
| Timer T2 | 4000 |
| Timer T4 | 5000 |
| Timer D | 32000 |
| Timer B | 32000 ms |
| Timer F | 32000 ms |
| Timer H | 32000 ms (64*TimerT1) |
| Timer J | 4000 |

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

5. Remote Authorization Table

Select **Settings > SIP > Remote Authorization Tables**.

Remote Authorization Tables entries contain information for responses to request message challenges by an upstream server.

Figure 9: Remote Authorization Table



6. Contact Registrant Table

Select **Settings > SIP > Contact Registrant Table**.

Contact Registrant Tables manage contacts that are registered to a SIP server. The SIP Server Configuration can specify a Contact Registrant Table. The username portion of the table is used for outbound calls.

Figure 10: Contact Registrant Table



7. Server Table

Select **Settings > SIP > SIP Server Tables**

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

Figure 11: Airtel SIP server

SIP Server Entry: ims.airtel.in:5060 UDP November 30, 2018 11:57:47

| Server Host | |
|-----------------|---------------|
| Server Lookup | IP/FQDN |
| Priority | 1 |
| Host | ims.airtel.in |
| Host IP Version | IPv4 |
| Port | 5060 |
| Protocol | UDP |

| Transport | |
|----------------------|--------------|
| Monitor | SIP Options |
| Keep Alive Frequency | 30 |
| Recover Frequency | 5 |
| Local Username | +91124123456 |
| Peer Username | +91124123456 |

| Remote Authorization and Contacts | |
|--------------------------------------|--------------------|
| Remote Authorization Table | Airtel Auth |
| Contact Registrant Table | Airtel Contact Reg |
| Clear Remote Registration on Startup | False |
| Contact URI Randomizer | False |
| Stagger Registration | False |
| Retry Non-Stale Nonce | True |
| Authorization on Refresh | True |
| Session URI Validation | Strict |

Figure 12: SFB Mediation Server

SIP Server Entry: sfb.example.com:5060 TCP November 30, 2018 12:14:12

| Server Host | |
|-----------------|-----------------|
| Server Lookup | IP/FQDN |
| Priority | 1 |
| Host | sfb.example.com |
| Host IP Version | IPv4 |
| Port | 5060 |
| Protocol | TCP |

| Transport | |
|----------------------|-------------|
| Monitor | SIP Options |
| Keep Alive Frequency | 30 |
| Recover Frequency | 5 |
| Local Username | Anonymous |
| Peer Username | Anonymous |

| Remote Authorization and Contacts | |
|-----------------------------------|---------|
| Remote Authorization Table | None |
| Contact Registrant Table | None |
| Session URI Validation | Liberal |

| Connection Reuse | |
|------------------|---------|
| Reuse | True |
| Sockets | 4 |
| Reuse Timeout | Forever |

8. Signaling Group

Select **Settings > Signaling Groups**

Signaling groups allow telephony channels to be grouped together for the purposes of routing and sharing configuration data. Calls are routed to signaling groups along with the location data used in Call Route selection. A signaling group also specifies the location from which Tone Tables and Action Sets are selected. For SIP, signaling groups specify protocol settings and link to server, media, and mapping tables.

Figure 13: SFB Signaling Group

Description **SFB Signaling Group**
Admin State **Enabled**
Service Status **Down**

SIP Channels and Routing

| | |
|-------------------------------|----------------------------------|
| Action Set Table | None |
| Call Routing Table | calls from SFB to Airtel |
| No. of Channels | 60 |
| SIP Profile | SFB Profile |
| SIP Mode | Basic Call |
| Agent Type | Back-to-Back User Agent |
| Interop Mode | Standard |
| SIP Server Table | SFB Mediation Server |
| Load Balancing | Round Robin |
| Channel Hunting | Most Idle |
| Notify Lync CAC Profile | Disable |
| Challenge Request | Disable |
| Outbound Proxy | |
| Outbound Proxy Port | 5060 |
| No Channel Available Override | 34: No Circuit/Channel Available |
| Call Setup Response Timer | 255 |
| Call Proceeding Timer | 180 |
| QoE Reporting | Disabled |
| Use Register as Keep Alive | Enable |
| Forked Call Answered Too Soon | Disable |

Media Information

| | |
|-------------------------------|----------------------------------|
| Audio/Fax Stream Mode | <input type="text" value="DSP"/> |
| Video/Application Stream Mode | Disabled |
| Media List ID | SFB Media List |
| Play Ringback | Auto on 180/183 |
| Tone Table | Default Tone Table |
| Play Congestion Tone | Disable |
| Early 183 | Disable |
| Allow Refresh SDP | Enable |
| Music on Hold | Disabled |

Mapping Tables

| | |
|----------------------------------|-------------------|
| SIP To Q.850 Override Table | Default (RFC4497) |
| Q.850 To SIP Override Table | Default (RFC4497) |
| Pass-thru Peer SIP Response Code | Enable |

SIP IP Details

| | |
|---------------------------------|------------------------------|
| Signaling/Media Source IP | Ethernet 1 IP (10.158.11.34) |
| Signaling DSCP | 40 |
| -----Static NAT - Outbound----- | |
| Outbound NAT Traversal | None |
| -----Static NAT - Inbound----- | |
| Detection | Disabled |

Listen Ports

| Total 1 SIP Listen Port Row | | |
|-----------------------------|----------|----------------|
| Port | Protocol | TLS Profile ID |
| 5060 | TCP | N/A |

Federated IP/FQDN

| Total 1 SIP Federated IP Row | |
|------------------------------|-----------------|
| IP/FQDN | Netmask/Prefix |
| 10.0.1.0 | 255.255.255.255 |

Message Manipulation **Disabled**

Figure 14: Airtel Signaling Group

Description **Airtel Signaling Group**
Admin State **Enabled**
Service Status **Down**

SIP Channels and Routing

Action Set Table **None**
Call Routing Table **Calls from Airtel to SFB**
No. of Channels **60**
SIP Profile **Airtel Profile**
SIP Mode **Basic Call**
Agent Type **Back-to-Back User Agent**
Interop Mode **Standard**
SIP Server Table **Airtel Sip Server**
Load Balancing **Round Robin**
Channel Hunting **Most Idle**
Notify Lync CAC Profile **Disable**
Challenge Request **Disable**
Outbound Proxy
Outbound Proxy Port **5060**
No Channel Available Override **34: No Circuit/Channel Available**
Call Setup Response Timer **255**
Call Proceeding Timer **180**
QoE Reporting **Disabled**
Use Register as Keep Alive **Enable**
Forked Call Answered Too Soon **Disable**

Media Information

Audio/Fax Stream Mode **DSP**
Video/Application Stream Mode **Disabled**
Media List ID **Airtel Media List**
Play Ringback **Auto on 180**
Tone Table **Default Tone Table**
Play Congestion Tone **Disable**
Early 183 **Enable**
Allow Refresh SDP **Enable**
Music on Hold **Disabled**

Mapping Tables

SIP To Q.850 Override Table **Default (RFC4497)**
Q.850 To SIP Override Table **Default (RFC4497)**
Pass-thru Peer SIP Response Code **Enable**

SIP IP Details

Signaling/Media Source IP **Ethernet 2 IP (192.168.129.2)**
Signaling DSCP **40**
Static NAT - Outbound
Outbound NAT Traversal **None**
Static NAT - Inbound
Detection **Disabled**

Listen Ports

| Total 1 SIP Listen Port Row | | |
|-----------------------------|----------|----------------|
| Port | Protocol | TLS Profile ID |
| 5060 | UDP | N/A |

Federated IP/FQDN

| Total 5 SIP Federated IP Rows | |
|-------------------------------|-----------------|
| IP/FQDN | Netmask/Prefix |
| ims.airtel.in | 255.255.255.255 |
| 10.232.130.171 | 255.255.255.255 |
| 10.232.130.172 | 255.255.255.255 |
| 10.232.130.178 | 255.255.255.255 |
| 10.232.130.179 | 255.255.255.255 |

Message Manipulation **Disabled**

Note: Be sure to add all of the SIP Federated IP addresses.

Table 1: Federated IP addresses

| Delhi SBC | Mumbai SBC |
|----------------|----------------|
| 10.232.130.171 | 10.232.146.150 |
| 10.232.130.172 | 10.232.146.151 |
| 10.232.130.178 | 10.232.146.138 |
| 10.232.130.179 | 10.232.146.139 |
| 10.232.130.180 | |

| | |
|----------------|--|
| 10.232.130.186 | |
| 10.232.130.187 | |
| 10.232.130.178 | |
| 10.232.131.98 | |
| 10.232.131.99 | |
| 10.232.131.100 | |
| 10.232.131.106 | |
| 10.232.131.107 | |
| 10.232.131.114 | |
| 10.232.131.115 | |
| 10.232.131.116 | |
| 10.232.131.122 | |
| 10.232.131.123 | |
| 10.232.131.124 | |
| 10.232.131.130 | |
| 10.232.131.131 | |
| 10.232.131.132 | |

Test Results

| S.No | Procedure | Observation | Result | Comment |
|------|-----------|-------------|--------|---------|
| | | | | |
| | | | | |
| | | | | |

Conclusion

These Application Notes describe the configuration steps required for **Ribbon SBC 1000 / 2000** to successfully interoperate with **Skype for Business 2015**. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in [Test Results](#).

Appendix A
