

SIP Trunk Qualification and Config Guide

SBC1000/SBC2000

BT SIP Trunk Service
(NOAS)



Line	Author	Date	Revision	Changes Made
1	Matt Hurst	22/11/2013	v.1.0	Initial Draft
2				

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1. Introduction

This document has been written as part of the Interop testing and certification of the Sonus SBC1000 and SBC2000 products against the BT SIP trunk service, known as NOAS.

Full testing and interoperability certification has been completed with the Sonus SBC devices, and this document details the relevant configuration elements within the Sonus SBC platforms that enable smooth and seamless integration to the BT SIP trunk service.

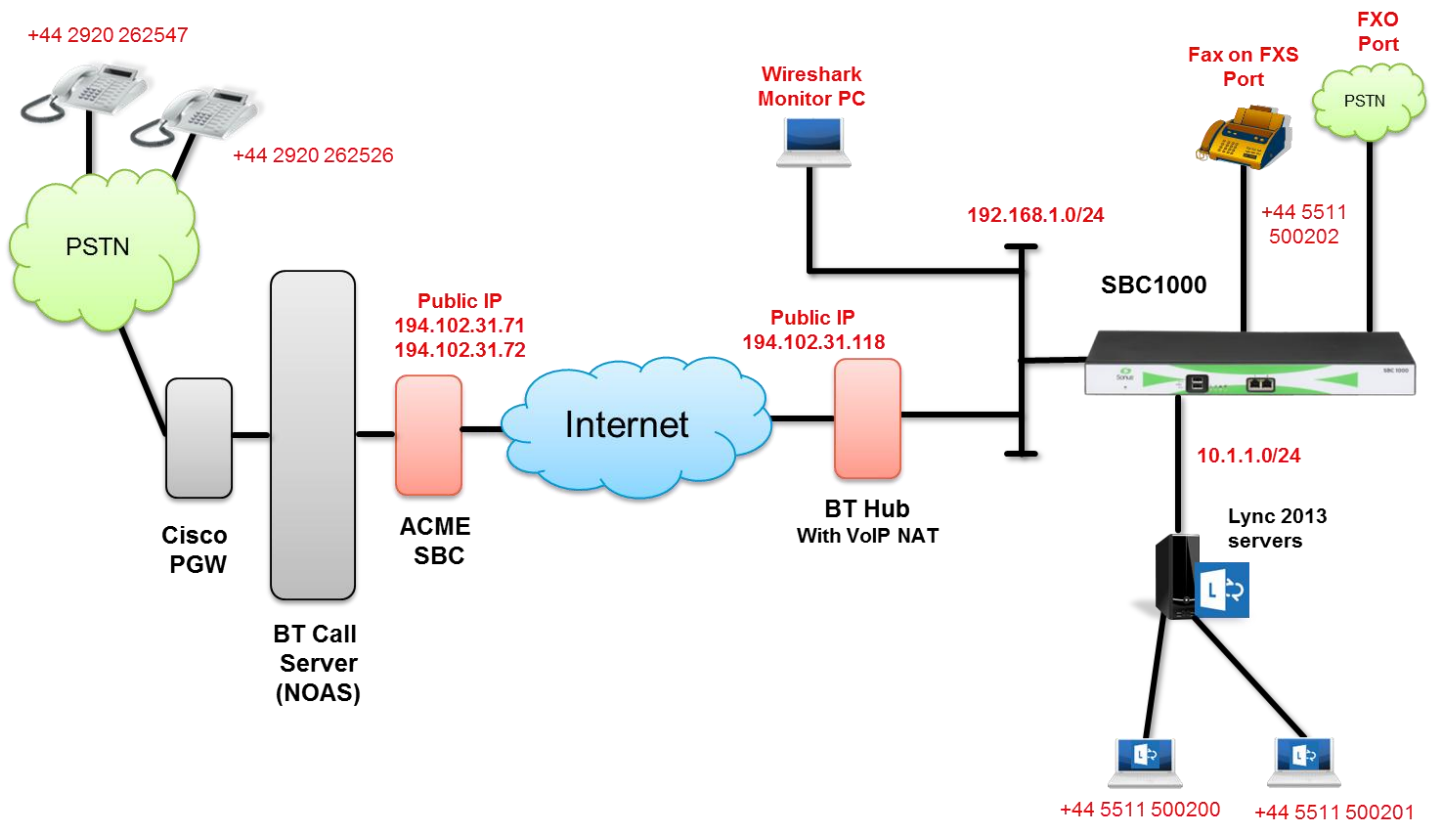
2. Testing Architecture

The following architecture was used for the certification and interop testing of the Sonus SBC against the BT NOAS SIP Trunk service.

The SBC used for testing was a Sonus SBC1000

S/W Version: R3.0.2 v262

System Name	MattDemo
Chassis Model	Sonus SBC 1000
Software Version	3.0.2
Build Number	262
Node Serial Number	A4022511270029
Hardware ID	c5a00ee77ecd89fbb15832
Up Time	17 hrs, 52 mins, 22 secs



All tests were performed from the Lync 2013 environment.

3. SIP Server Table Configuration

The SBC supports one or more outbound SIP servers in order to provide redundancy when configured to the BT SIP trunk service.

The servers are configured under the **SIP** configuration area on the SBC – as you can see it is possible to mix both TCP and UDP based servers within the same group. **(Default protocol for BT SIP service is UDP)**

Row ID	Host	Port	Protocol	Display Counters
1	194.102.31.71	5060	TCP	Counters
2	194.102.31.72	5060	UDP	Counters

Each server can be uniquely configured with either IP or FQDN, and the remote listening port. *Note that if FQDN is used here within the SIP Server table, the SIP messages will contain the FQDN in all SIP messages sent to the server – ensure that DNS is set accordingly on the SBC to enable resolution of the FQDN.*

Remote SIP Server Monitoring

It is recommended to enable the Transport Monitor to the remote SIP server – this will enable active monitoring of the SIP service, and allow dynamic fail-over and re-routing to an alternate SIP server should any failure occur. The SBC will pro-actively monitor the SIP server, and generate alarms should the remote server fail to respond, failing calls over to the next priority SIP server.

1
194.102.31.71
5060
TCP
[Counters](#)

Server Host

Priority:

Host: FQDN or IP *

Port: [1024..65535] *

Protocol: *

Transport

Monitor:

Keep Alive Frequency: secs [30..300] *

Recover Frequency: secs [5..300] *

Local Username: Local Username of Sonus SBC *

Peer Username: Peer Username of sip server *

Remote Authorization and Contacts

Remote Authorization Table:

Contact Registrant Table:

Connection Reuse

Reuse:

Server Priority

It is possible to assign a priority to each SIP server in the table, enabling server prioritisation from within the **Signalling Group**.

4. Signalling Group

The Signalling Group configuration towards the BT SIP trunk service is the main element that links together both inbound and outbound traffic flows.

Firstly, configure the remote SIP servers that are permitted to send SIP traffic inbound to the SBC.

Add in the IP or FQDN of all BT SIP servers that are configured to provide your SIP service.

ALL other SIP traffic from any other source will **not** be permitted into this Signalling Group on the SBC.

Federated IP/FQDN		
Total 2 SIP Federated IP Rows		
<input type="checkbox"/>	IP/FQDN	Netmask
<input type="checkbox"/>	194.102.31.72	255.255.255.255
<input type="checkbox"/>	194.102.31.71	255.255.255.255

Listen Ports			
Total 2 SIP Listen Port Rows			
<input type="checkbox"/>	Port	Protocol	TLS Profile ID
<input type="checkbox"/>	5060	UDP	N/A
<input type="checkbox"/>	5060	TCP	N/A

Next, configure the ports and protocols that the SBC will listen for traffic from the above authorised servers.

Note that you can support both UDP/TCP/TLS on the same Signalling Group, and ports are fully customisable.

For the outbound traffic, select the **SIP Server Table** created previously – these will be the servers used for all outbound traffic.

You can also set here the Load Balancing that is required between the SIP servers – either ‘round robin’ equally across servers, or prioritized:

Load Balancing	Priority
Channel Hunting	Round Robin
Notify CAC Profile	Priority
	First

SIP Channels and Routing	
Action Set Table	None
Call Routing Table	Calls from BT
No. of Channels	20 [1..960] *
SIP Profile	Default SIP Profile
SIP Mode	Basic Call
SIP Server Table	BT
Load Balancing	Priority
Channel Hunting	Most Idle
Notify CAC Profile	Disable
Challenge Request	Disable
Outbound Proxy	IP/FQDN

Also within the Signalling group are the Media related parameters that the traffic inbound and outbound will use.

Select a **Media List** to be used for all traffic inbound and outbound.

Also, **Music on Hold** behaviour is set here within the Signalling Group – refer to the section on MoH for full details.

Media Information		
Media List ID	Default Media List	▼
Play Ringback	Auto	▼
Tone Table	Default Tone Table	▼
Early 183	Disable	▼
Music on Hold	Disabled	▼

SIP Channels and Routing	
Action Set Table	None
Call Routing Table	Calls from BT
No. of Channels	20 [1..960] *
SIP Profile	Default SIP Profile
SIP Mode	Basic Call
SIP Server Table	BT
Load Balancing	Priority
Channel Hunting	Most Idle
Notify CAC Profile	Disable
Challenge Request	Disable
Outbound Proxy	<input type="text"/>

For all calls inbound from the SIP trunk, the next logical step is to link the traffic from the Signalling Group into a **Call Routing Table** for onward routing decisions.

Select the **Call Routing Table** you wish to have associated with the traffic from the SIP trunk. *(Leave at default if you have not yet created a specific routing table –you can always come back and change later once a table is created)*

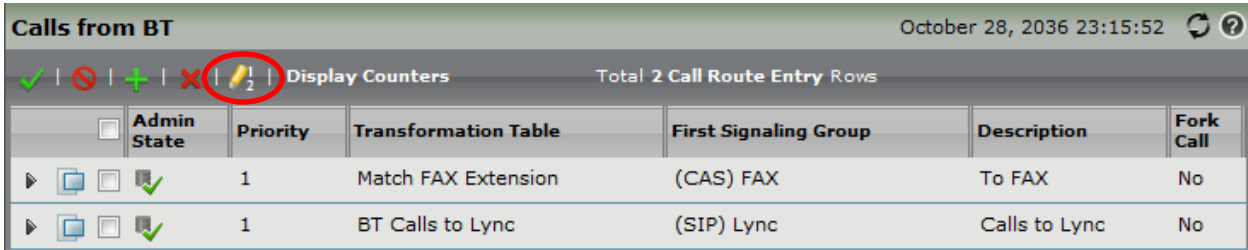
It is also possible to set the max number of Sessions (*calls*) that the SIP trunk will allow for all inbound and outbound calls combined. This can be dynamically increased or decreased as required in the future.

5. Call Routing

As the inbound calls from the SIP trunk are processed by the SBC, they will hit the relevant **Call Routing Table**, as specified in the Signalling Group.

It is possible to have multiple Call Routing tables within the SBC, each one associated to a different Signalling Group – this enables great flexibility and ease of handling differing dial plans from different sources.

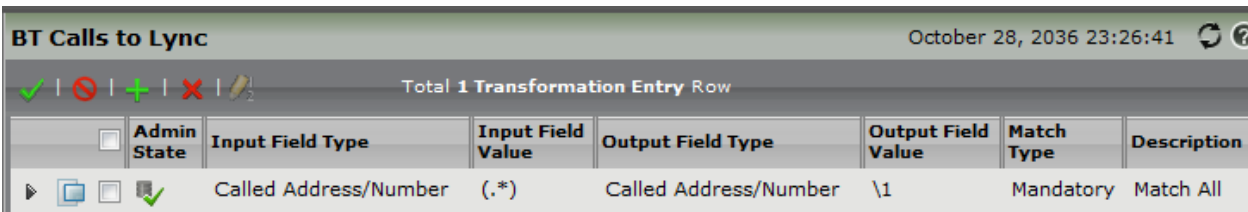
A call routing table will contain one or more *routing lines* that are processed by the SBC in **table order**. Remember to re-sequence your routing entries to have the most specific matches at the top of the list.



<input type="checkbox"/>	Admin State	Priority	Transformation Table	First Signaling Group	Description	Fork Call
<input type="checkbox"/>	<input checked="" type="checkbox"/>	1	Match FAX Extension	(CAS) FAX	To FAX	No
<input type="checkbox"/>	<input checked="" type="checkbox"/>	1	BT Calls to Lync	(SIP) Lync	Calls to Lync	No

The routing lines will refer to a **Transformation Table** – this is where the call number matching and manipulation is performed.

A Transformational Table can have one or more statements within it to determine a call route match – these matches utilise Regular Expressions in order to match and manipulate the various call header fields such as Called and Calling Numbers, as well as things like Diversion headers.



<input type="checkbox"/>	Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description
<input type="checkbox"/>	<input checked="" type="checkbox"/>	Called Address/Number	(.*)	Called Address/Number	\\1	Mandatory	Match All

Note:

Refer to the Sonus SBC documentation for full details about configuring call routing, transformation table matching and number manipulation.

6. Media Parameters

The Media List associated with the BT Signalling Group must be configured with the appropriate Codec types, as well as specifying the DTMF digit relay mechanism.

Firstly, it is required to configure all of the **Media Profiles** that are to be used within the SBC – these profiles define the individual Voice or Fax codecs that can be used, and allows customisation of things like the Packetisation time, or Fax rates and redundancy.

Codec	Description	Primary Key
G.711 A-Law	Default G711A	1
G.711 μ-Law	Default G711u	2
G.729	G.729	3
G.726	Default G726	4
T.38 Fax	Default T38	10001

FAX Codec Configuration

Description: Default T38

Codec: T.38 Fax

Fallback to Passthrough: Enabled

Maximum Rate: 14400 b/s

Signaling Packet Redundancy: 3 [0..7]

Payload Packet Redundancy: 0 [0..3]

Error Correction Mode: Enabled

Training Confirmation Procedure: Send Over Network

These **Media Profiles** can then be used and grouped together within a **Media List**.

Description: Default Media List

Media Profiles List:

- G.729
- Default G711A
- Default T38

Crypto Profile ID: None

Media DSCP: 46 [0..63] *

RTCP Mode: RTCP

Dead Call Detection: Disabled

Silence Suppression: Enabled

Note that the **Order** in which the Codecs are listed is the order in which they are offered within any SDP exchange.

The BT SIP service supports G711, G729 and T38 for Fax transmission. Ensure these codecs are configured with the default payload size of 20ms.

Also to be configured within the **Media List** is the DTMF Digit Relay transport method – the SBC can support either In-Band audio, or out of band RFC2833 method.

For the BT SIP trunk, it is required to utilise **RFC2833**.

- **Set the Media List that is used against the BT SIP Trunk to have RFC2833 configured.**

Note that the SBC will *transcode* DTMF from In Band to Out Band if required, sending the out of band RFC2833 towards the BT SIP service.

Gain Control		Digit Relay	
Receive Gain	<input type="text" value="0"/> [-14..+6] dB	Digit (DTMF) Relay Type	<input type="text" value="RFC 2833"/>
Transmit Gain	<input type="text" value="0"/> [-14..+6] dB	Digit Relay Payload Type	<input type="text" value="As Voice"/> <input type="text" value="RFC 2833"/>

Passthrough/Tone Detection	
Modem Passthrough	<input type="text" value="Enabled"/>
Fax Passthrough	<input type="text" value="Enabled"/>
CNG Tone Detection	<input type="text" value="Disabled"/>

These **Media Lists** are applied to the Signalling Groups as already mentioned, and also to call routing lines if required, enabling transcoding between different codecs on each side of the SBC.

The BT SIP trunk service supports and has been tested with G711 A and Mu law, G729 and T38 Fax codec types.

7. SIP Profile

Within the SIP profile there is access to a number of parameters that change the way the SBC communicates to the SIP trunk service.

One element is the 'Keep alive' Session timer that by default is disabled – the BT SIP trunk service supports the Reinvite method as a means of keep alive for active calls – select the Reinvite method from the drop down list. The associated timer parameters can be left at default.

The screenshot shows the 'Session Timer' configuration window. It contains the following fields:

- Session Timer: Enable (dropdown)
- Refresh Method: Reinvite (dropdown)
- Minimum Acceptable Timer: Update (dropdown), with a tooltip 'Reinvite (0)'
- Offered Session Timer: 3600 secs (input field)
- Terminate On Refresh Failure: False (dropdown)

Also within the **SIP Profile** are various timers that can be adjusted, changing the response times during SIP signalling.

One element to configure for the BT SIP service is the **Maximum Retransmissions** – this affects how many re-transmissions of SIP messages are allowed before a fail-over mechanism is invoked.

Change this from the default 'RFC Standard' to **2** – this changes the B and F timers to 3500ms.

This will result in a *fast fail-over* to the redundant BT SIP server in the event that an INVITE message is not responded to correctly. (The RFC default will result in a fail-over after 32 seconds of no response.)

Note that this fail-over mechanism is **In addition** to the pro-active SIP Options monitoring that is configured against the SIP servers in section 3.

The screenshot shows the 'Timers' configuration window. It contains the following fields:

- Transport Timeout Timer: 5000 ms (input field)
- Maximum Retransmissions: 2 (dropdown), with a tooltip '2 (2) [0..10000]'
- RFC Standard: 2 (dropdown)
- Timer T1: 3 (input field)
- Timer T2: 4 (input field)
- Timer T4: 8 (input field)
- Timer C: 1000000 ms (input field)
- Timer D: 32000 ms (input field)
- Timer B: 3500 ms (input field)
- Timer F: 3500 ms (input field)
- Timer H: 32000 ms (64*TimerT1) (input field)
- Timer J: 32000 ms (64*TimerT1) (input field)

Should a SIP server fail to respond to the Options packets being sent, the system will automatically fail-over to the redundant SIP server before any re-transmission timers are utilised.

8. Music on Hold

The SBC can insert Music on Hold to a media stream if required, upon receipt of a hold indication within the call Re-Invite SDP information. (*a=inactive or a=sendonly*)

In order to prepare the SBC with a Music Source, follow the configuration below.

The SBC can take an audio source from either a live feed (*inserted via an FXS port*) or from an uploaded audio file to the DSP cards directly.

In order to set the audio source, and upload a file to the DSP cards, check the **Media System Configuration** – here you can choose either Live feed or File, and also you can upload the audio file to the system directly:

Media System Configuration October 28, 2036 21:2

Upload Music File

RTP/RTCP Port Range

Start Port: [1024..65534] *

Number of Port Pairs: [1..1200] *

Music on Hold

Music on Hold Source:

Current Music File:

Echo Canceller Type Option:

Echo Cancel NLP Option:

Send STUN Packets:

Once uploaded, check the DSP cards under **System Settings** – you can verify the MoH file status:

DSP Module Table October 28, 2036 21:25:44

Total 3 DSP Module Rows

DSP Slot	Module Type	Available	Channels In Use	CPU Usage (%)	Last CPU Usage Update	Service Status	Last Music on Hold File Update	Music on Hold File Status
1	MSPD C300 DSP	<input checked="" type="checkbox"/>	0	1.5	10/28/36 21:25:44	Up	10/26/36 02:14:51	Loaded
2	N/A	<input type="checkbox"/>	-	-	-	N/A	-	-
3	N/A	<input type="checkbox"/>	-	-	-	N/A	-	-

Now the system is prepared with the Music on Hold source, you can set within the **Signalling Group** whether you are to offer the MoH media as a hold state is initiated from the remote party.

Media Information

Media List ID:

Play Ringback:

Tone Table:

Early 183:

Music on Hold:

Always Enabled

About Sonus Networks

Sonus Networks is a global leader in SIP communications, session management and voice security solutions for service providers and carriers.

Sonus solutions enable service providers and enterprises to reduce their recurring telecom costs, gracefully manage the migration from legacy voice to SIP multimedia communications, and mitigate business continuity and security issues for critical enterprise voice and contact centre infrastructure.

To learn more about our solutions for hosted Unified Communications services, visit www.sonusnet.com or contact a Sonus representative today at sales@sonusnet.com.

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