SIP Trunk Qualification and Config Guide

SBC1000/SBC2000

BT SIP Trunk Service (NOAS)

Sonus

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.

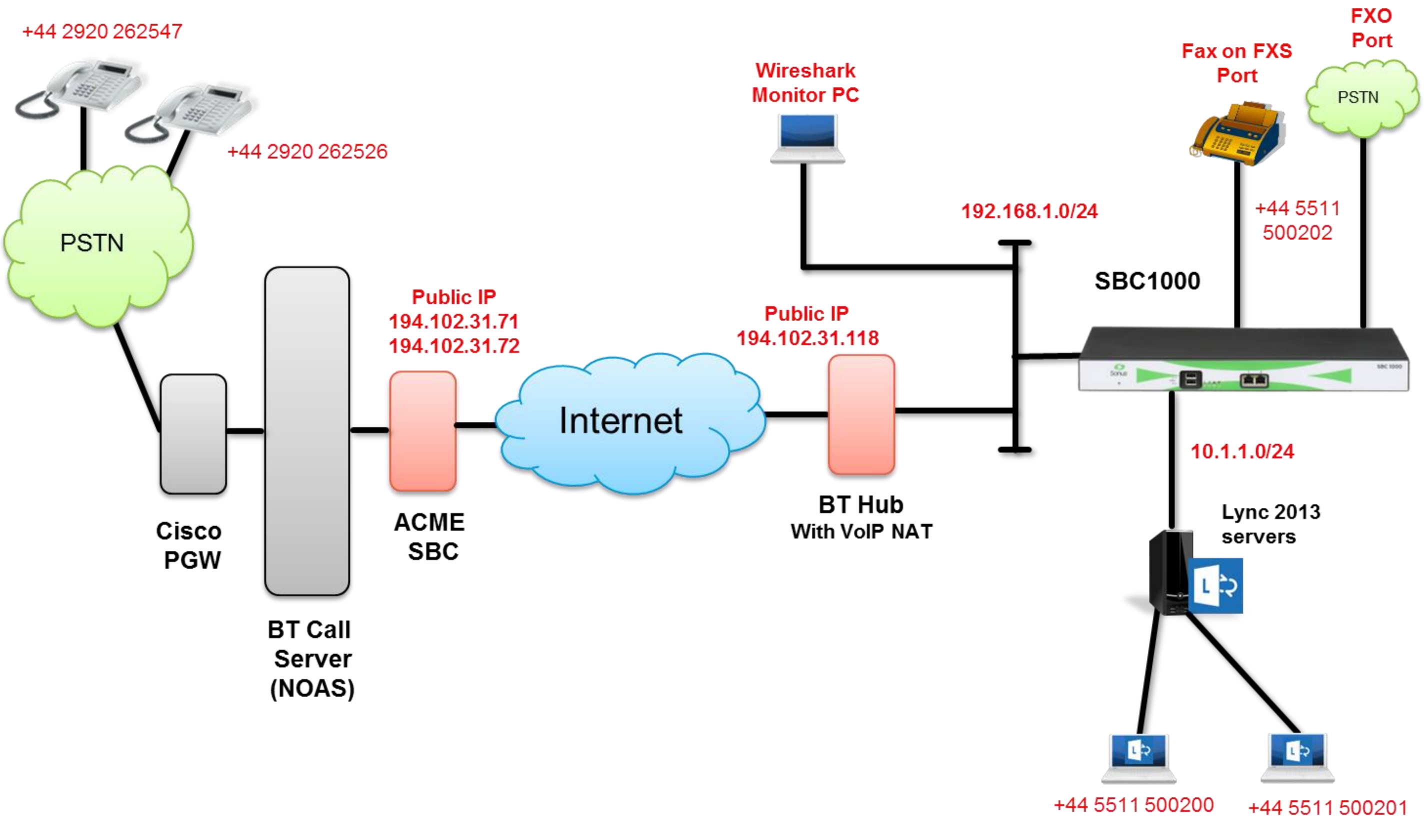
2. Testing Architecture

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)**

BT October 28, 2036 20:59:45

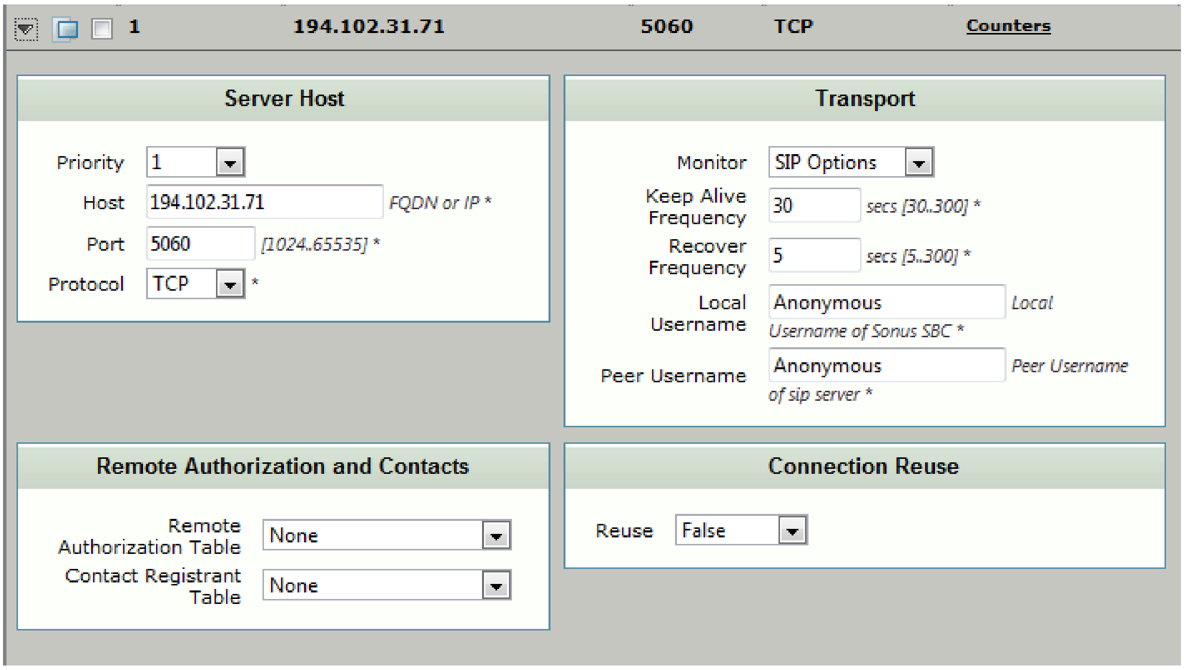
Total 2 SIP Server Rows

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.



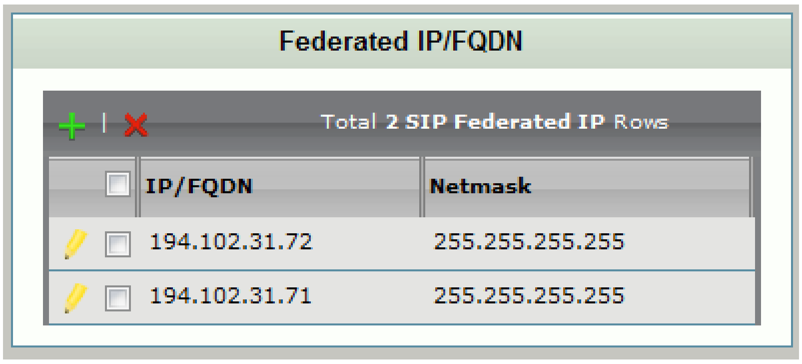
Server Priority

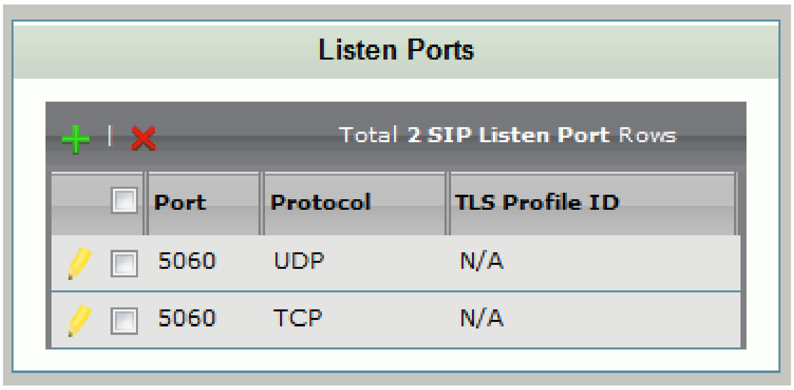
It is possible to assign a priority to each SIP server in the table, enabling server prioritisation from within the ***Signalling Group***.

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.

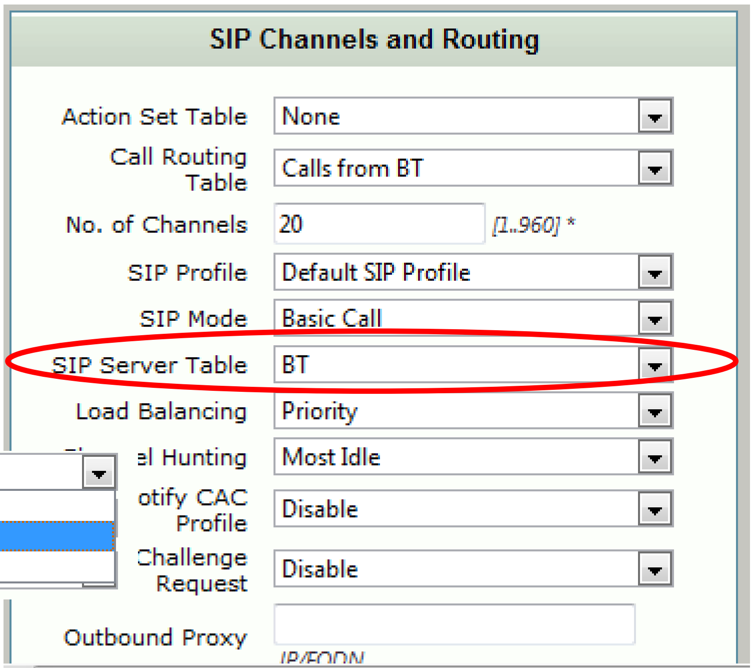


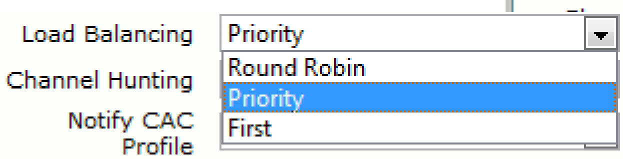


Next, configure the ports and protocols that the SBC will listen for traffic from the above authorised servers.
Note that you can support both UDP/TCP/TLS on the same Signalling Group, and ports are fully customisable.

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:

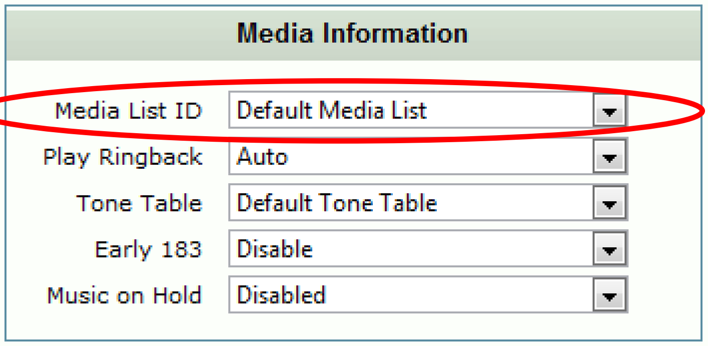


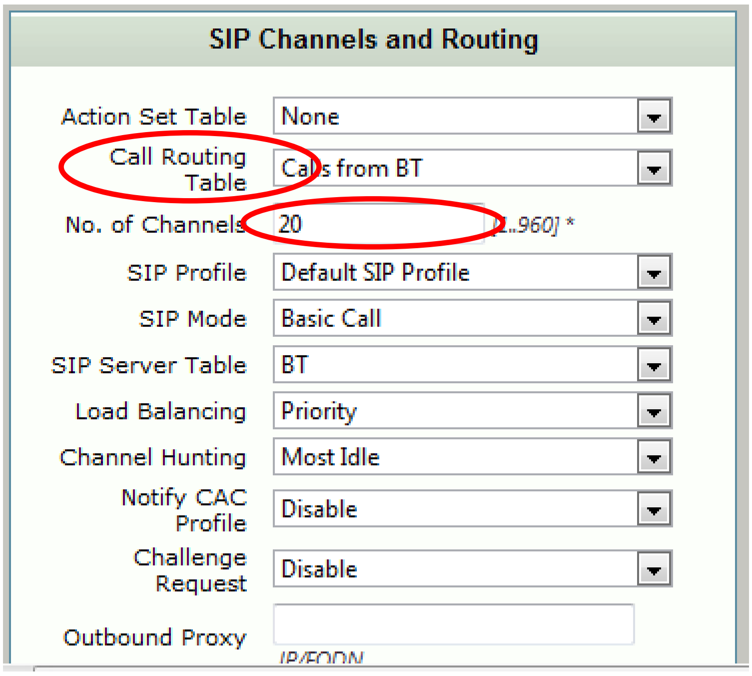


Also within the Signalling group are the Media related parameters that the traffic inbound and outbound will use.

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.





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 <u>most specific</u> matches at the top of the list.

Calls from BT October 28, 2036 23:15:52

Display Counters Total **2 Call Route Entry** Rows

Admin State	Priority	Transformation Table	First Signaling Group	Description	Fork Call
	1	Match FAX Extension	(CAS) FAX	To FAX	No
	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.

BT Calls to Lync October 28, 2036 23:26:41

Total **1 Transformation Entry** Row

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description
	Called Address/Number	(.*)	Called Address/Number	\1	Mandatory	Match All

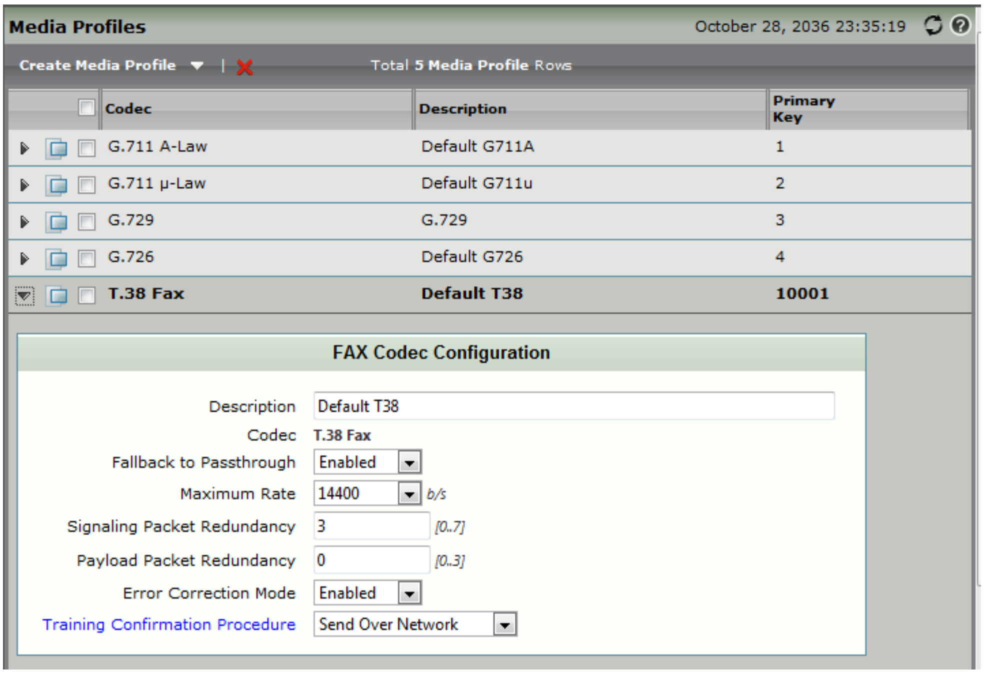
Note:

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

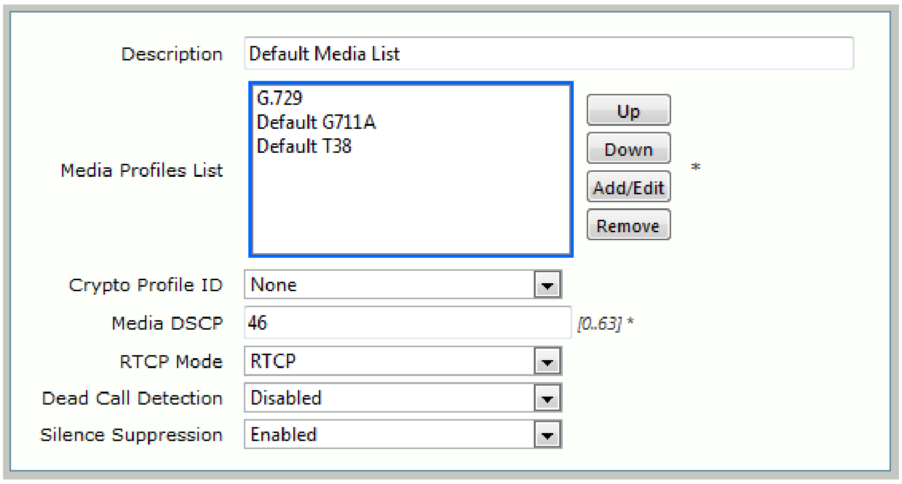
6. Media Parameters

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.



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



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

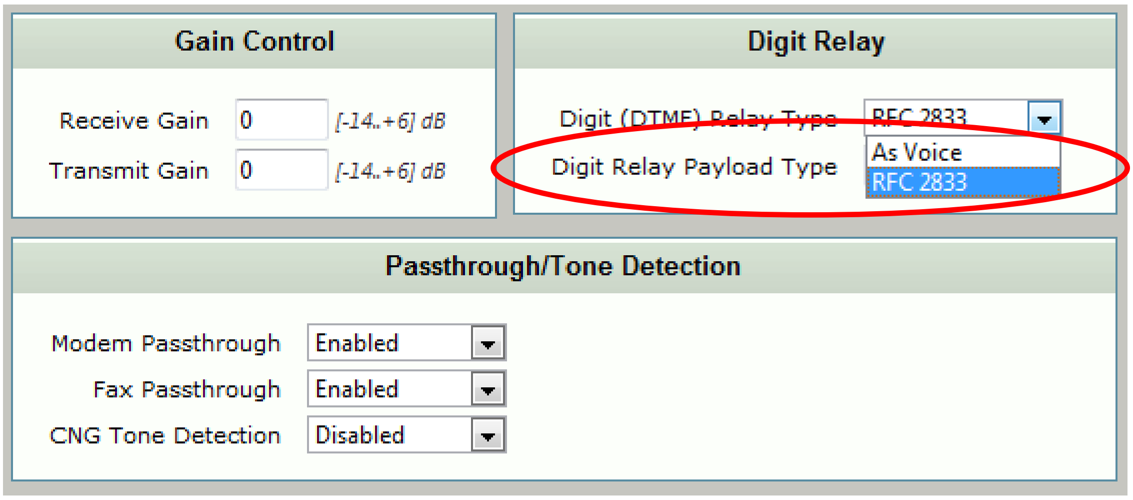
The BT SIP service supports G711, G729 and T38 for Fax transmission. Ensure these codecs are configured with the default payload size of 20ms.

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.



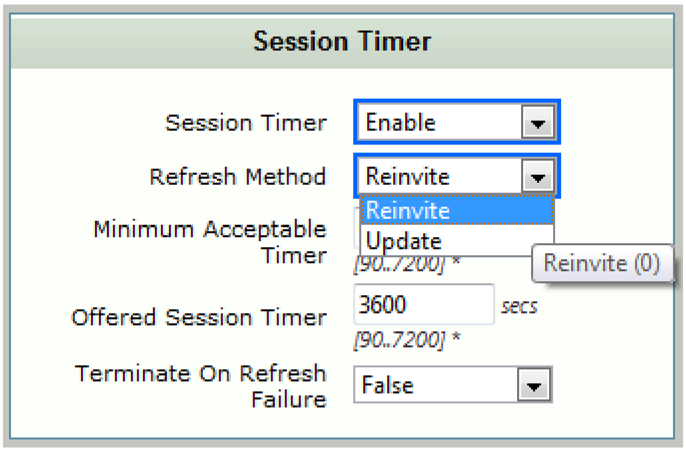
These **Media Lists** are applied to the Signalling Groups as already mentioned, and also to call routing lines if required, enabling transcoding between different codecs on each side of the SBC.

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.



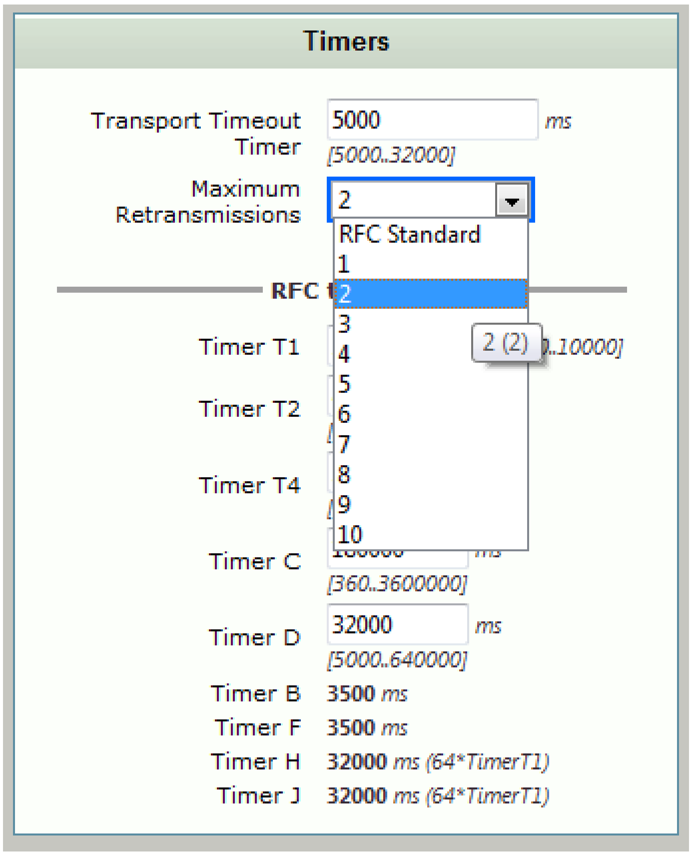
Also within the **SIP Profile** are various timers that can be adjusted, changing the response times during SIP signalling.

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.



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.

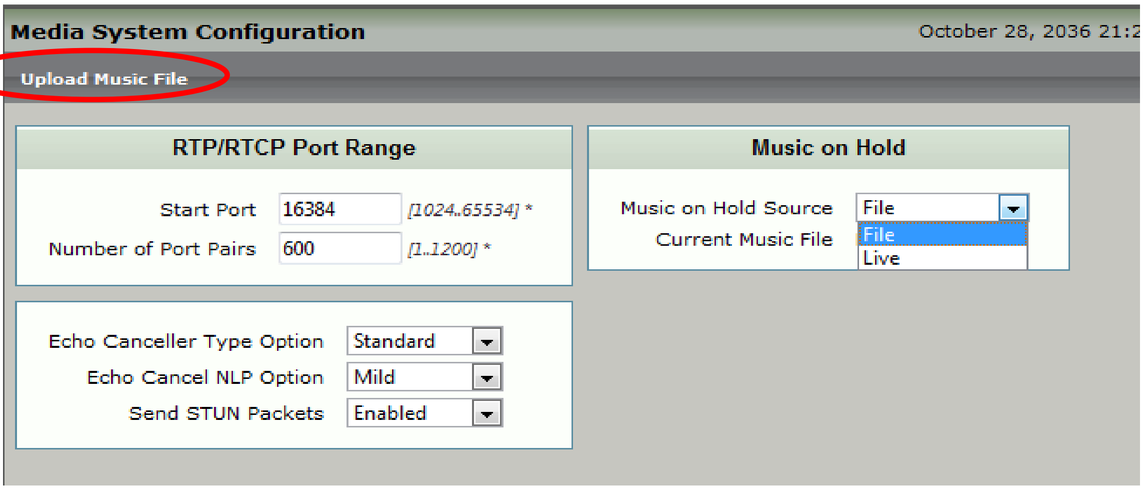
8. Music on Hold

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)*

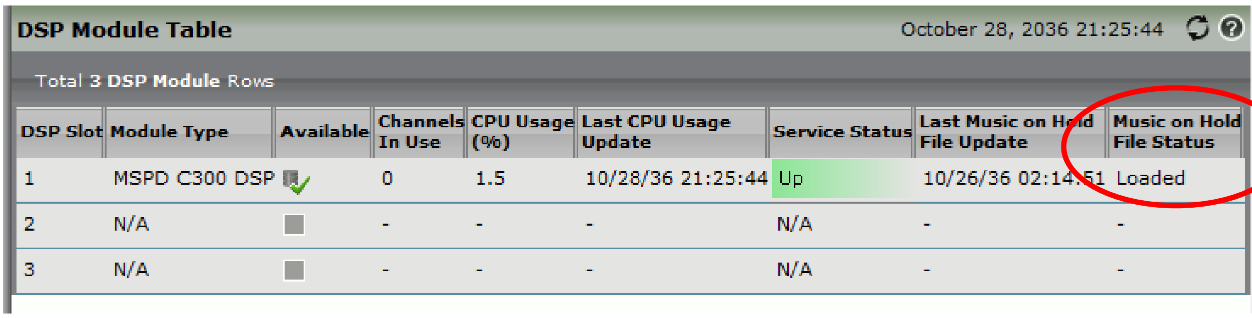
In order to prepare the SBC with a Music Source, follow the configuration below.

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:

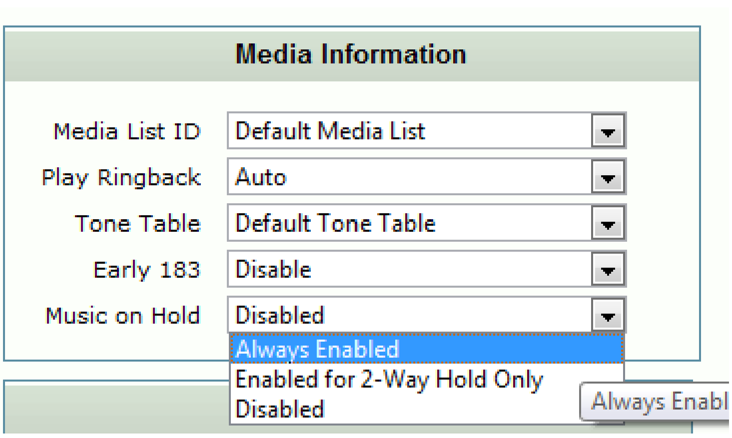


Once uploaded, check the DSP cards under **System Settings** – you can verify the MoH file status:



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		0	1.5	10/28/36 21:25:44	Up	10/26/36 02:14:51	Loaded
2	N/A		-	-	-	N/A	-	-
3	N/A		-	-	-	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.



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.

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