



# **BroadSoft Partner Configuration Guide**

Sonus Networks, Inc. SBC 1000 / SBC 2000

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## BroadWorks® Guide

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## Document Revision History

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Version	Reason for Change
1.0	Introduced document for Sonus Networks, Inc. SBC 1000 / SBC 2000 version 4.0.0 validation with BroadWorks Release 19 SP1.

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## 1 Overview

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This guide describes the configuration procedures required for the Sonus Networks, Inc. SBC 1000 / SBC 2000 for interoperability with BroadWorks.

The SBC 1000 / SBC 2000 is an Enterprise SBC that has been validated with BroadWorks.

This guide describes the specific configuration items that are important for use with BroadWorks. It does not describe the purpose and use of all configuration items on the SBC 1000 / SBC 2000. For those details, see the Sonus User's Guide [\[1\]](#) supplied by Sonus Networks, Inc..

## 2 Interoperability Status

This section provides the known interoperability status of the Sonus Networks, Inc. SBC 1000 / SBC 2000 with BroadWorks. This includes the version(s) tested, the capabilities supported, and known issues.

Interoperability testing validates that the device interfaces properly with BroadWorks via the Session Initiation Protocol (SIP) interface. Qualitative aspects of the device or device capabilities not affecting the SIP interface, such as performance, are not covered by interoperability testing. Requests for information and/or issues regarding these aspects should be directed to Sonus Networks, Inc..

### 2.1 Verified Versions

The following table identifies the verified Sonus Networks, Inc. SBC 1000 / SBC 2000 and BroadWorks versions and the month/year the testing occurred. If the device has undergone more than one test cycle, versions for each test cycle are listed, with the most recent listed first.

*Compatible Versions* in the following table identify specific SBC 1000 / SBC 2000 versions which the partner has identified as compatible and should interface properly with BroadWorks. Generally, maintenance releases of the validated version are considered compatible and are not specifically listed here. For any questions concerning maintenance and compatible releases, contact Sonus Networks, Inc..

**NOTE:** Interoperability testing is usually performed with the latest generally available (GA) device firmware/software and the latest GA BroadWorks release and service pack at the time the testing occurs. If there is a need to use a non-verified mix of BroadWorks and device software versions, customers can mitigate their risk by self-testing the combination themselves using the *BroadWorks Session Controller Interoperability Test Plan* [5].

#### Verified Versions

Date (mm/yyyy)	BroadWorks Release	SBC 1000 / SBC 2000 Verified Version	SBC 1000 / SBC 2000 Compatible Versions
09/2014	Release 19 SP1	Release 4.0.0	N/A

## 2.2 Interface Capabilities Supported

The Sonus Networks, Inc. SBC 1000 / SBC 2000 has completed interoperability testing with BroadWorks using the *BroadWorks Session Controller Interoperability Test Plan* [5]. The results are summarized in the following table.

The BroadWorks test plan is composed of packages, each covering distinct interoperability areas, such as “Basic” call scenarios and “Redundancy” scenarios. Each package is composed of one or more test items, which in turn, are composed of one or more test cases. The test plan exercises the SIP interface between the device and BroadWorks with the intent to ensure interoperability sufficient to support the BroadWorks feature set.

The *Supported* column in the following table identifies the Sonus Networks, Inc. SBC 1000 / SBC 2000’s support for each of the items covered in the test plan packages, with the following designations:

- Yes Test item is supported.
- No Test item is not supported.
- NA Test item is not applicable to the device type.
- NT Test item was not tested.

Caveats and clarifications are identified in the *Comments* column.

**NOTE:** *DUT* in the following table refers to the *Device Under Test*, which in this case is the Sonus Networks, Inc. SBC 1000 / SBC 2000.

BroadWorks Session Controller Interoperability Test Plan Support Table			
Test Plan Package	Test Plan Package Items	Supported	Comments
<b>Basic</b>	Call Origination	Yes	
	Call Termination	Yes	
	Session Audit	Yes	
	Session Timer	Yes	
	Ringback	Yes	
	Forked Dialog	Yes	
	Early UPDATE	Yes	
	Early-Session	No	
	181 Call Being Forwarded	Yes	
	DTMF – Inband	Yes	
	DTMF – RFC 2833	Yes	
	DTMF – DTMF Relay	Yes	
<b>BroadWorks Services</b>	Third-Party Call Control – Basic	Yes	
	Third-Party Call Control – Advanced	Yes	
	Message Waiting Indicator –	Yes	

	Unsolicited		
	Message Waiting Indicator – Solicited	Yes	
	Voice Portal Outcall	Yes	
	Advanced Alerting	Yes	
	Calling Line ID – Non-Trusted Endpoint	Yes	
	Calling Line ID with Unicode Characters – Non-Trusted Endpoint	Yes	
	Calling Line ID – Trusted Endpoint	Yes	
	Calling Line ID with Unicode Characters – Trusted Endpoint	Yes	
	Diversion Header	Yes	
	History-Info Header	Yes	
	Deny Calls from Unregistered Users	Yes	
	Enterprise Trunking – Originating Trunk Group (OTG)	Yes	
	Enterprise Trunking – Destination Trunk Group (DTG)	Yes	
	Enterprise Trunking – Trunk Group (TGRP)	Yes	
<b>Access Device Services – Call Control Services</b>	Call Waiting	Yes	
	Call Hold	Yes	
	Call Transfer	Yes	
	Local Conference	Yes	
	Network Conference	Yes	
	Call Forwarding	Yes	
<b>Access Device Services – Registration and Authentication</b>	Registration – Register Authentication	Yes	
	Registration – Maximum Registration	Yes	
	Registration – Minimum Registration	Yes	
	Authentication – Invite Authentication	Yes	
	Authentication – Re-Invite or Update Authentication	Yes	
	Authentication – Refer Authentication	Yes	
	Authentication – Access Device Authenticating BroadWorks	Yes	
	SIP Connect GIN Registration – GIN Register	Yes	
	SIP Connect GIN Registration – Call to PBX User	Yes	
	SIP Connect GIN Registration – Call from PBX User	Yes	

<b>Access Device Services – Fax</b>	G711 Fax Passthrough	Yes	
	G711 Fax Fallback	Yes	
	T38 Fax Messaging	Yes	
<b>Advanced Phone Services – Busy Lamp Field</b>	Busy Lamp Field	Yes	
	Maximum Monitored Users	Yes	
<b>Advanced Phone Services – Feature Key Synchronization</b>	Do Not Disturb	Yes	
	Call Forwarding	Yes	
	Call Center Agent Logon/Logoff	Yes	
	Call Recording	No	
	Security Classification	No	
<b>Advanced Phone Services – Shared Call Appearance</b>	Line-Seize Events	Yes	
	Call-Info Events	Yes	
	Multiple Call Arrangement	Yes	
	Bridging	Yes	
<b>Advanced Phone Services – Call Recording</b>	Call Recording Controls	No	
	Call Recording Video	No	
<b>Advanced Phone Services – Security Classification</b>	Security Classification	No	
<b>Redundancy</b>	DNS SRV Lookup	Yes	
	Register Failover/Failback	Yes	
	Invite Failover/Failback	Yes	
	Bye Failover	Yes	
<b>Video – Basic Video Calls</b>	Call Origination	Yes	
	Call Termination	Yes	
	Call Hold	Yes	
	Call Transfer	Yes	
<b>Video – BroadWorks Video Services</b>	Auto Attendant	Yes	
	Auto Attendant – HD	Yes	
	Voice Messaging	Yes	
<b>Remote Survivability</b>	Register	Yes	
	Local Calls – Without Subscriber Data	Yes	
	PSTN Calls – Without Subscriber Data	Yes	
	SCA Call – Without Subscriber Data	No	
	Register for Subscriber Data	Yes	
	Local Calls – With Subscriber Data	Yes	

	PSTN Calls – With Subscriber Data	Yes	
	SCA Call – With Subscriber Data	No	
IPV6	Call Origination	No	
	Call Termination	No	
	Ringback	No	
	Call Control	No	
	Registration with Authentication	No	
	T38 Fax Messaging	No	
	Busy Lamp Field	No	
	Redundancy	No	
	Video	No	
	Dual Stack with Alternate Connectivity	No	

### 2.3 Known Issues

This section lists the known interoperability issues between BroadWorks and specific partner release(s). Issues identified during interoperability testing and known issues identified in the field are listed.

The following table provides a description of each issue and, where possible, identifies a workaround. The verified partner device versions are listed with an “X” indicating that the issue occurs in the specific release. The issues identified are device deficiencies or bugs, so typically not BroadWorks release dependent.

If the testing was performed by BroadSoft, then the *Issue Number* is a BroadSoft ExtraView partner issue number. If the testing was performed by the partner or a third party, then the partner may or may not supply a tracking number.

For more information on any issues related to the particular partner device release, see the partner release notes.

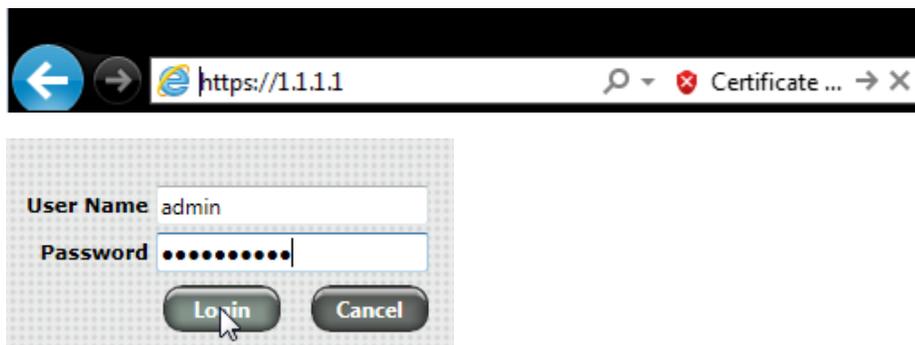
Issue Number	Issue Description	Partner Version			
		4.0.0v340			
	None				

### 3 SBC 1000 / SBC 2000 Configuration

This section describes the configuration settings required for the SBC 1000 / SBC 2000 integration with BroadWorks, primarily focusing on the SIP interface configuration. The SBC 1000 / SBC 2000 configuration settings identified in this section have been derived and verified through interoperability testing with BroadWorks. Refer to the SBC 4.0 User's Guide [1] for SBC 1000 / SBC 2000 configuration details not covered in this section.

#### 3.1 Configuration Method

Out of the box, the Sonus SBC 1000 / SBC 2000/2000 is configured primarily using a web browser via a web interface hosted on the Sonus SBC 1000 / SBC 2000/2000 system.



The WebUI provides a full range of configuration options to end-users. To list a few, the ability to configure IP interfaces, setting the telephony ports, configuring routes and digit manipulation, and managing Users and Groups.

## 3.2 System Configuration

This section describes system configuration items required for the SBC 1000 / SBC 2000.

### 3.2.1 Configuration Settings

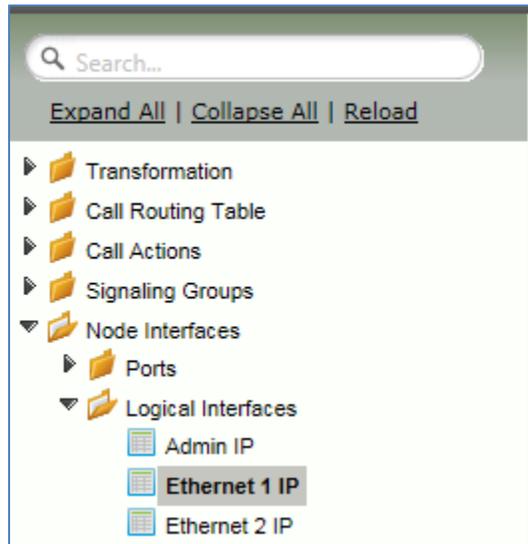
The Tabs across the top of the Sonus SBC WebUI permit the user to access various configuration subsystems. Within this document, all configurations will be performed under the SETTINGS tab.



### 3.2.2 Configure Network Interfaces

Configure the SBC's basic network connectivity items to permit the SBC to interoperate with the Broadsoft Server as well as Enterprise network.

- In the Navigation tree, click on *Ethernet 1 IP*



- Configure the Ethernet IP 1 port as necessary to connect to the Broadsoft server.

The screenshot displays the configuration page for the 'Ethernet 1 IP' interface. At the top, the interface name is 'Ethernet 1 IP', the IP address is '10.1.1.74', and the status is 'Disabled'. The page is divided into two main sections: 'Identification/Status' and 'Networking'.

**Identification/Status Section:**

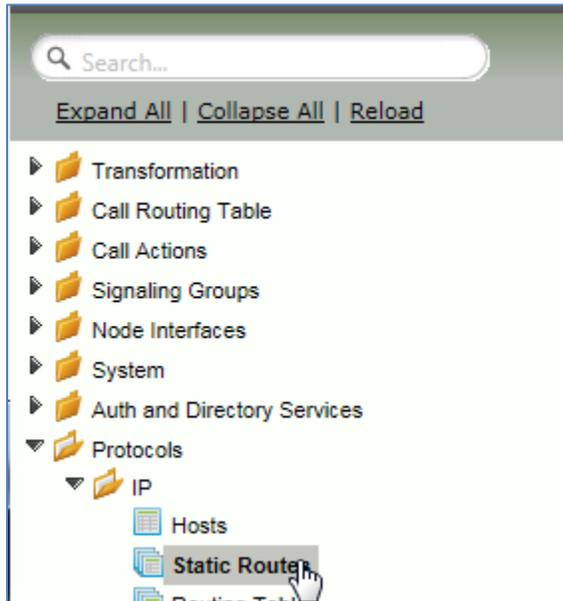
- Interface Name: Ethernet 1 IP
- I/F Index: 39
- Alias: [Empty text box]
- Description: [Empty text box]
- Admin State: Enabled (dropdown menu)

**Networking Section:**

- MAC Address: 00:10:23:01:01:01
- IP Assign Method: Static (dropdown menu)
- Primary Address: 10.1.1.74 (with x.x.x.x placeholder)
- Primary Netmask: 255.255.255.0 (with x.x.x.x placeholder)
- Configure Secondary Interface: Disabled (dropdown menu)
- ACL In: None (dropdown menu)
- ACL Out: None (dropdown menu)
- ACL Forward: None (dropdown menu)

**Note:** Your installation may require additional IP/Ethernet interfaces be configured.

- In the Navigation Tree, click on *Static Routes*



- Configure any IP routes required to provide connectivity between the SBC and the Broadsoft server, as well as any IP routes required to provide connectivity to the Enterprise LAN.

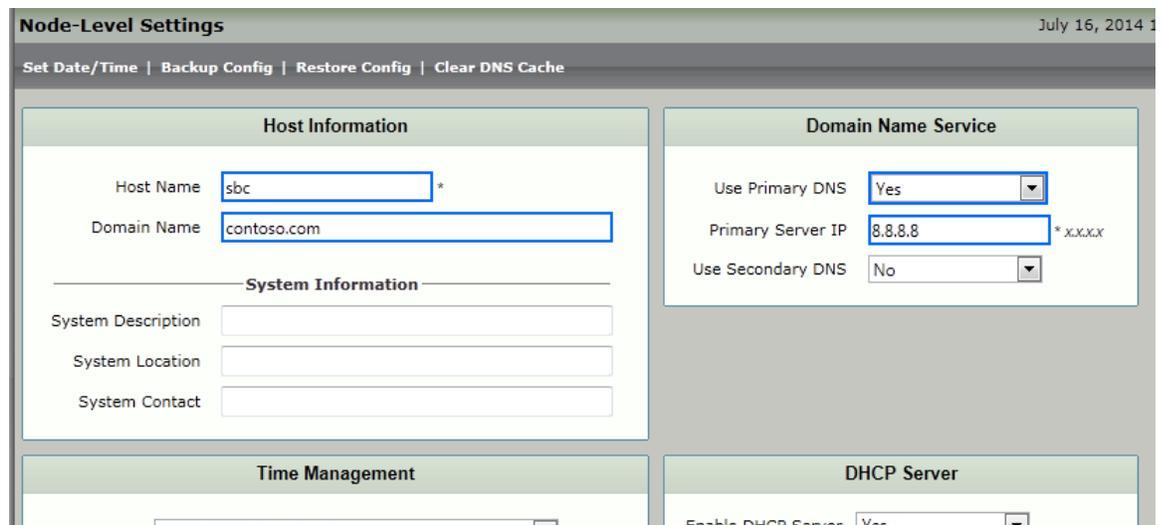
Static IP Route Table				
Total 4 IP Route Rows				
Row ID	Destination IP	Mask	Gateway	Metric
1	172.16.110.106	255.255.255.255	134.56.227.5	1
2	199.19.193.0	255.255.255.0	134.56.242.1	1

- In the Navigation Tree, click on *System | Node-Level Settings*



Verify or add the following information to the Node-Level Settings:

- Ensure the SBC has a configured Host Name
- Ensure the SBC has a configured Domain Name
- Ensure Primary DNS Server IP is set to an appropriate DNS server
- Click Apply



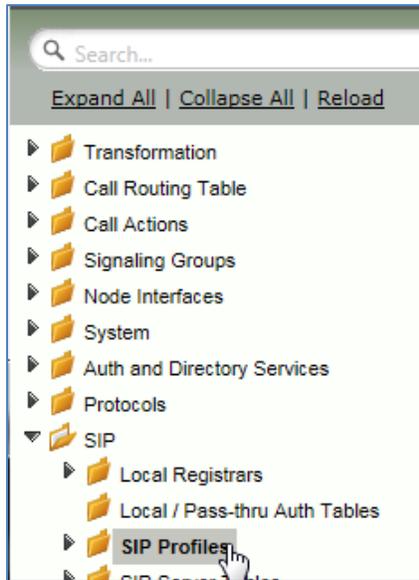
### 3.2.2.1 Configure IPV6 Settings

*Not Supported.*

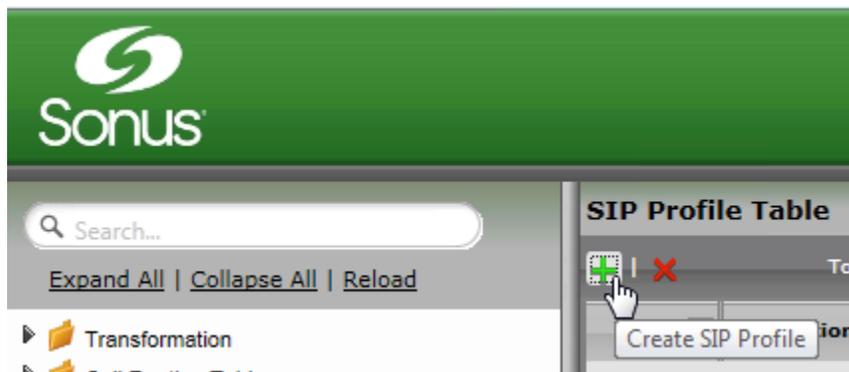
### 3.2.3 Configure SIP Interface Settings

Create the *Default SIP Profile* as noted below. If it already exists, correct as noted in the *Default SIP Profile* configuration picture below.

- In the Navigation Tree, click on *SIP Profiles*



- Create a SIP Profile by clicking +.



**Create SIP Profile Entry** August 28, 2014 13:04:20

Row ID	3
Description	<input type="text" value="Broadworks Profile"/>

- Configure the SIP Profile as noted below to permit proper connectivity to the Broadsoft Server.

<p><b>Session Timer</b></p> <p>Session Timer <input type="text" value="Disable"/></p>	<p><b>MIME Payloads</b></p> <p>ELIN Identifier <input type="text" value="LOC"/></p> <p>PIDF-LO Passthrough <input type="text" value="Enable"/></p> <p>Unknown Subtype Passthrough <input type="text" value="Disable"/></p>
<p><b>Header Customization</b></p> <p>FQDN in From Header <input type="text" value="Static"/></p> <p>Static Host <input type="text" value="as.iop2.broadwork"/> <small>FQDN or IP [port]</small></p> <p>Send Assert Header <input type="text" value="Always"/></p> <p>Trusted Interface <input type="text" value="Enable"/></p> <p>UA Header <input type="text"/></p> <p>Calling Info Source <input type="text" value="RFC Standard"/></p> <p>Diversion Header Selection <input type="text" value="First"/></p>	<p><b>Options Tags</b></p> <p>100rel <input type="text" value="Not Present"/></p> <p>Update <input type="text" value="Supported"/></p>
<p><b>Timers</b></p> <p>Transport Timeout Timer <input type="text" value="5000"/> <small>ms [5000..32000]</small></p> <p>Maximum Retransmissions <input type="text" value="3"/></p> <hr/> <p><b>RFC timers</b></p> <p>Timer T1 <input type="text" value="500"/> <small>ms [100..10000]</small></p> <p>Timer T2 <input type="text" value="4000"/> <small>ms [1000..80000](&gt;= T1)</small></p> <p>Timer T4 <input type="text" value="5000"/> <small>ms [1000..100000]</small></p> <p>Timer D <input type="text" value="32000"/> <small>ms [5000..640000]</small></p> <p>Timer B <b>7500 ms</b></p> <p>Timer F <b>7500 ms</b></p> <p>Timer H <b>32000 ms (64*TimerT1)</b></p> <p>Timer J <b>32000 ms (64*TimerT1)</b></p>	<p><b>SDP Customization</b></p> <p>Send Number of Audio Channels <input type="text" value="True"/></p> <p>Connection Info in Media Section <input type="text" value="True"/></p> <p>Origin Field Username <input type="text" value="SBC"/> <small>default: SBC</small></p> <p>Session Name <input type="text" value="VoipCall"/> <small>default: VoipCall</small></p> <p>Digit Transmission Preference <input type="text" value="SIP INFO"/></p>

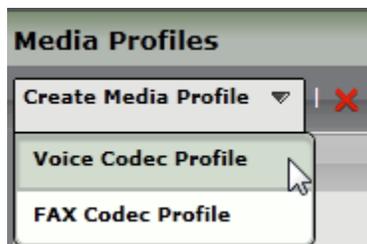
### 3.2.4 Configure Media Settings

Media Profiles allow you to specify the individual voice and fax compression codecs and their associated settings, for inclusion in a [Media List](#). Different codecs provide varying levels of compression, allowing one to reduce bandwidth requirements at the expense of voice quality.

- In the Navigation Tree, click on *Media Profiles*.



- Create a *Voice Codec Profile*.



- Add any codecs required for your configuration Broadworks or Enterprise applications. Repeat these steps until all the desired codecs are added.

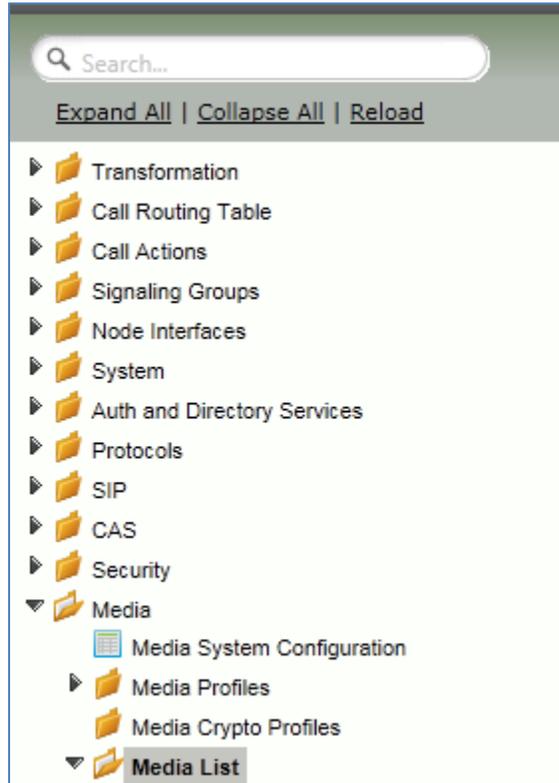
- When completed, your codec configuration will list all the codecs you've created.

Codec	Description
<input type="checkbox"/> G.711 A-Law	Default G711A
<input type="checkbox"/> G.711 μ-Law	Default G711u
<input type="checkbox"/> G.729	G.729
<input type="checkbox"/> G.723.1	G.723.1
<input type="checkbox"/> G.726	G.726
<input type="checkbox"/> T.38 Fax	T.38 Fax

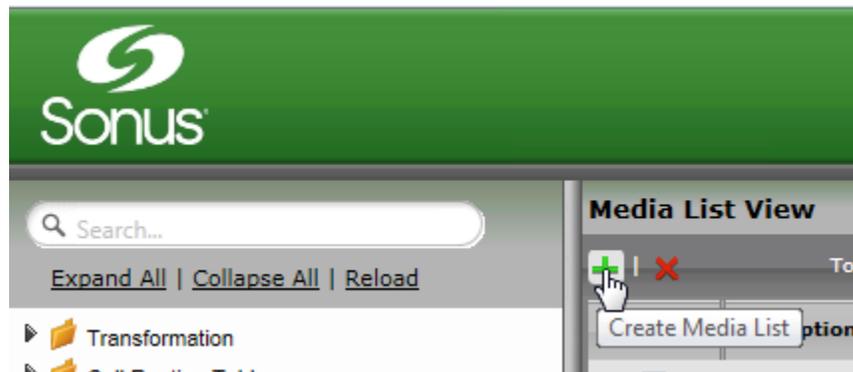
### 3.2.5 Configure Media Lists

Media Lists allow you to specify a set of codecs and fax profiles that are allowed on a given SIP Signaling Group. They contain one or more Media Profiles, which must first be defined in [Media Profiles](#). These lists allow you to accommodate specific transmission requirements, and SIP devices that only implement a subset of the available voice codecs.

- In the Navigation Tree, click on *Media List*



- Create a Media List for the Broadsoft application



- Add any codecs to be available from the Broadsoft application.

**Media List Details: BSFT Media List**

Description	BSFT Media List	
Media Profiles List	Default G711u Default G711A G.729 fax	*
Crypto Profile ID	None	
Media DSCP	46	
RTCP Mode	RTCP	
Dead Call Detection	Enabled	
Silence Suppression	Enabled	

**Gain Control**

Receive Gain	0
Transmit Gain	0

**Digit Relay**

Digit (DTMF) Relay Type	RFC 2833
Digit Relay Payload Type	101

**Passthrough/Tone Detection**

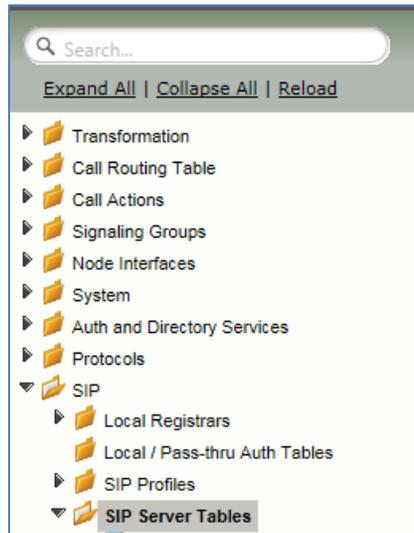
Modem Passthrough	Enabled
Fax Passthrough	Enabled
CNG Tone Detection	Disabled

**\*\*NOTE:** You will need to repeat the steps above to create another Media List for the Enterprise network if the codec list for Enterprise devices is different than those you added to the Media List above.

### 3.2.6 Configure a SIP Server Table and Entry for the Broadsoft Server

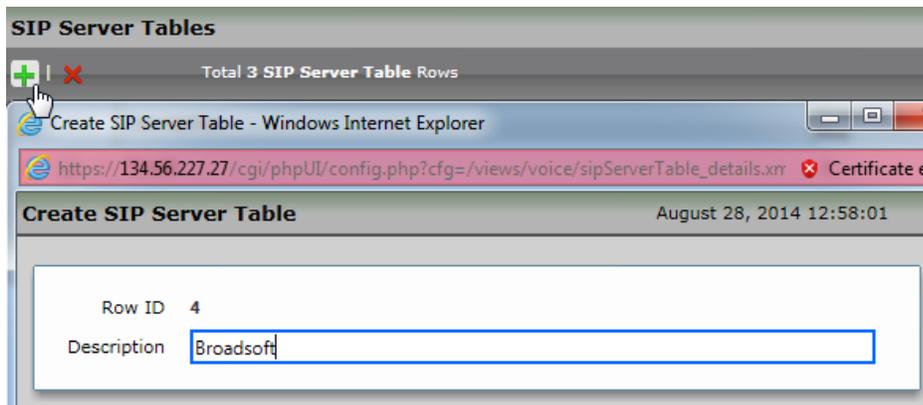
SIP Server Tables contain information about the SIP devices connected to the Sonus SBC 1000 / SBC 2000/2000. The entries in the tables provide information about the IP Addresses, ports, and protocols used to communicate with each server. The Table Entries also contain links to counters that are useful for troubleshooting.

- In the Navigation tree, click on SIP Server Table



Add a SIP Server Table:

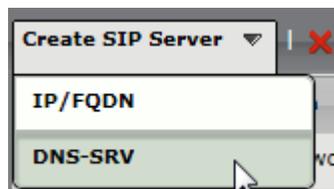
- Click the + to add a SIP Server Table
- Type of name of the Table
- Click OK



- In the Navigation tree, click on the name of the new *SIP Server Table* that you just added.



- From the *Create SIP Server* pulldown, select *DNS-SRV*. This will place a SIP Server Entry in the newly created SIP Server Table.



Enter the SIP Server information as noted below:

- Enter the FQDN of the **Broadworks** Server
- Verify the Protocol

Host / Domain	Server Lookup	Port	Protocol
as.iop2.broadworks.n...	DNS SRV	N/A	UDP

### Server Host

Server Lookup: DNS SRV

Domain Name / FQDN:  \*

Service Name:  \*

Protocol:  \*

### Transport

Monitor:  ▼

### Remote Authorization and Contacts

Remote Authorization Table:  ▼

Contact Registrant Table:  ▼

### SRV Servers

Total 2 SipSrvServer Rows

Server ID	FQDN/Domain Name	Protocol	Port	Time to Live	Priority	Weight
101	as1.iop2.broadworks....	UDP	5060	29108	1	50
100	as2.iop2.broadworks....	UDP	5060	29108	2	50

- Create an additional DNS SRV entry as below

Create SIP Server Total 1 SIP Server Row

Host / Domain	Server Lookup	Port	Protocol	Display Counters
revas.iop2.broadwork...	DNS SRV	N/A	UDP	<a href="#">Counters</a>

**Server Host**

Server Lookup: **DNS SRV**

Domain Name / FQDN:

Service Name:

Protocol:

**Transport**

Monitor:

**Remote Authorization and Contacts**

Remote Authorization Table:

Contact Registrant Table:

**SRV Servers**

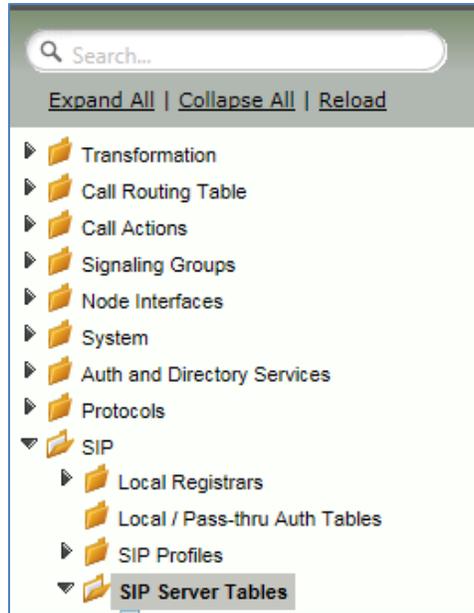
Total 2 SipSrvServer Rows

Server ID	FQDN/Domain Name	Protocol	Port	Time to Live	Priority	Weight
101	as2.iop2.broadworks....	UDP	5060	3600	0	0
100	as1.iop2.broadworks....	UDP	5060	3600	1	0

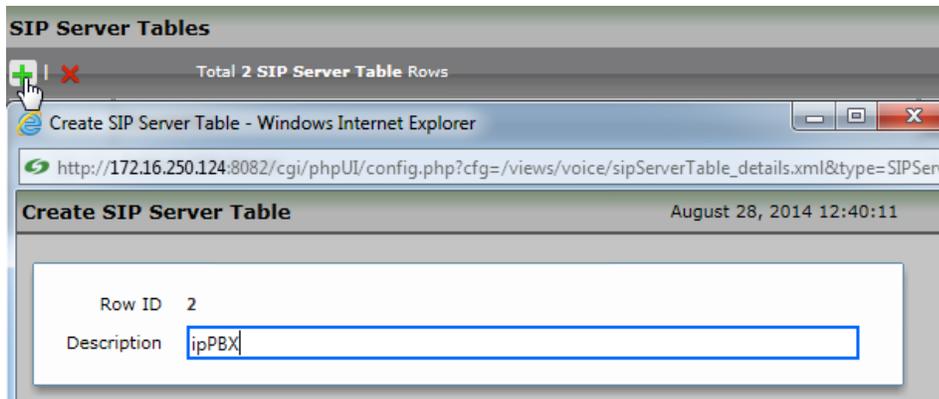
### 3.2.8 Configure a SIP Server Table and Entry for the IP PBXr

Create a SIP Server Table for the Enterprise IP PBX

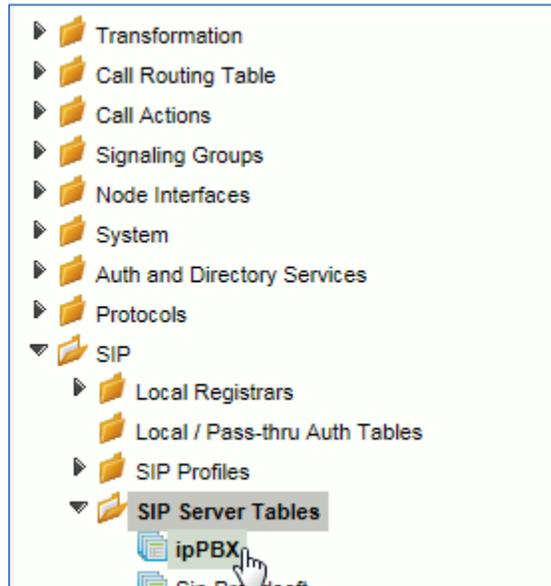
- In the Navigation tree, click on *SIP Server Table*.



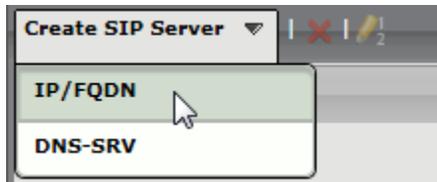
- Enter the desired name of the SIP Server Table, click OK



- In the Navigation Tree, select the newly created SIP Server Table



- Create an IP/FQDN **Enterprise** SIP Server.



- Enter the FQDN of the IP PBX
- Enter the SIP Server's Port Number
- Enter the SIP Server's Protocol type
- Configure Monitor to *SIP Options*
- Click OK

August 28, 2014 12:47:09

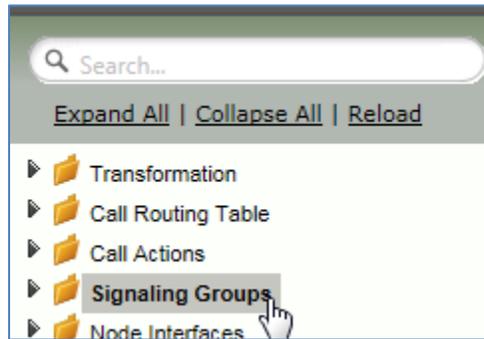
### Create SIP Server Entry

Server Host		Transport	
Row ID	1	Monitor	SIP Options ▼
Server Lookup	IP/FQDN	Keep Alive Frequency	30 * secs [30..300]
Priority	1 ▼	Recover Frequency	5 * secs [5..300]
Host	ippbx.contoso.net * FQDN <small>or IP</small>	Local Username	Anonymous * Local Username <small>of Sonus SBC</small>
Port	5060 * [1024..65535]	Peer Username	Anonymous * Peer Username <small>of sip server</small>
Protocol	TCP ▼ *		
Remote Authorization and Contacts		Connection Reuse	
Remote Authorization Table	None ▼	Reuse	True ▼
Contact Registrant Table	None ▼	Sockets	4 ▼
		Reuse Timeout	Forever ▼

### 3.2.9 Configure a Signaling Group for the Broadsoft Server

Signaling groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which [Call Routes](#) are selected. In the case of SIP, they specify protocol settings and link to server, media and mapping tables

- In the Navigation Tree, click *Signaling Groups*



- From the *Create Signaling Group* pulldown, select *SIP Signaling Group*



Enter the Broadsoft Signaling Group information as noted below:

- Select the *SIP Profile* you created earlier
- Select the Broadsoft *SIP Server Table*
- Verify/Delete/Create *Listening Ports* that the SBC will use to receive SIP from the Broadsoft Server
- Set all Media Modes to *Enable*
- Add the Broadsoft Server FQDNs in the *Federated IP* with a netmask of 255.255.255.255. **The list of servers to add to as Federated IPs will be provided by your ISP provider.**

Signaling Group Table August 26, 2014 16:04:47

✓ | ✗ | Create Signaling Group | Total 4 Signaling Group Rows

Type	Description	Admin State	Service Status	Display	Primary Key
SIP	Broadworks SG	Enabled	Up	Counters   Channels   Subscriber Data	2

Description: Broadworks SG  
Admin State: Enabled  
Service Status: Up

**SIP Channels and Routing**

Action Set Table: None

Call Routing Table: Default Route Table

No. of Channels: 60 \* [1..960]

SIP Profile: Broadworks Profile

SIP Mode: Basic Call

Agent Type: Access Mode

Interop Mode: BroadSoft Extension

Registrant TTL: 3600 \* secs [0..60400]

SIP Server Table: BroadSoft

Channel Hunting: Most Idle

Notify Lync CAC Profile: Disable

Challenge Request: Disable

Outbound Proxy: IP/FQDN

Outbound Proxy Port: 5060 [1024..65535]

No Channel Available Override: 34: No Circuit/Channel Available

Call Setup Response Timer: 255 [180..750] secs

**Media Information**

Audio/Fax Stream Proxy Mode: Enabled

Audio/Fax Stream DSP Mode: Enabled

Video/Application Stream Proxy Mode: Enabled

Media List ID: Broadsoft Media List

Play Ringback: Auto

Tone Table: Default Tone Table

Early 183: Disabled

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**

NAT Traversal: None

Signaling/Media Source IP: Auto

Signaling DSCP: 40 \* [0..63]

**Listen Ports**

Total 2 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	UDP	N/A
5060	TCP	N/A

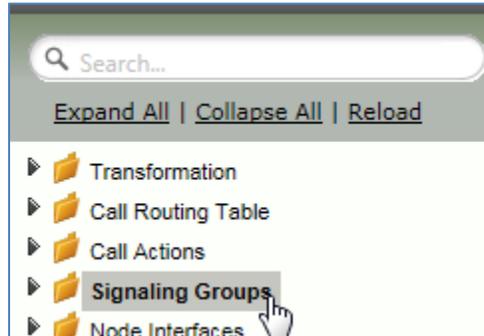
**Federated IP/FQDN**

Total 2 SIP Federated IP Rows

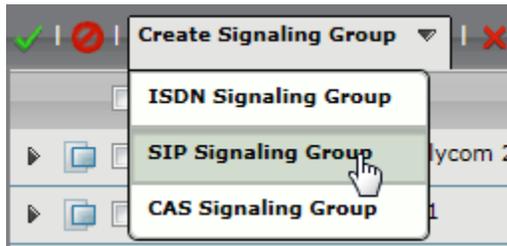
IP/FQDN	Netmask
as.iop2.broadworks.n...	255.255.255.255
as.iop1.broadworks.n...	255.255.255.255

### 3.2.10 Configure a Signaling Group for the Enterprise IP PBX

- In the Navigation Tree, click *Signaling Groups*



- From the *Create Signaling Group* pulldown, select *SIP Signaling Group*



Enter the IP PBX Signaling Group information as noted below:

- Select the *SIP Profile* you created earlier
- Select the IP PBX *SIP Server Table*
- Verify/Delete/Create *Listening Ports* that the SBC will use to receive SIP from the IP PBX Server
- Set all Media Modes to *Enable*
- Add the IP PBX Server FQDNs in the *Federated IP* with a netmask of 255.255.255.255

**Create SIP Signaling Group** August 28, 2014 11:41:08

Description:

Admin State:

---

**SIP Channels and Routing**

Action Set Table:

Call Routing Table:

No. of Channels:  \* [1..960]

SIP Profile:

SIP Mode:

Agent Type:

SIP Server Table:

Load Balancing:

Channel Hunting:

Notify Lync CAC Profile:

Challenge Request:

Outbound Proxy:

Outbound Proxy Port:  [1024..65535]

No Channel Available Override:

Call Setup Response Timer:  [180..750] secs

QoE Reporting:

**Media Information**

Audio/Fax Stream Proxy Mode:

Audio/Fax Stream DSP Mode:

Video/Application Stream Proxy Mode:

Media List ID:

Play Ringback:

Tone Table:

Early 183:

Music on Hold:

---

**Listen Ports**

Total 2 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
<input type="checkbox"/> 5060	UDP	N/A
<input type="checkbox"/> 5060	TCP	N/A

**Federated IP/FQDN**

Total 0 SIP Federated IP Rows

IP/FQDN	Netmask
<input type="checkbox"/> ippbx.contoso.net	255.255.255.255

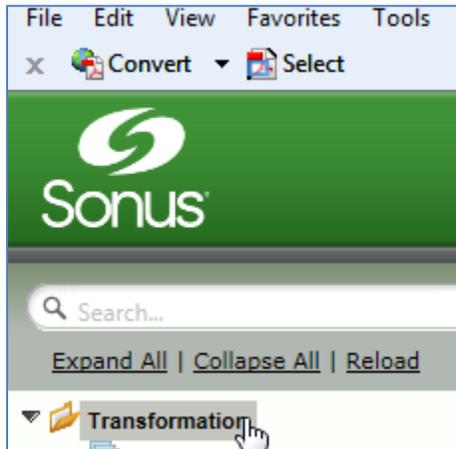
---

Message Manipulation:

### 3.2.11 Configure a Transformation Table

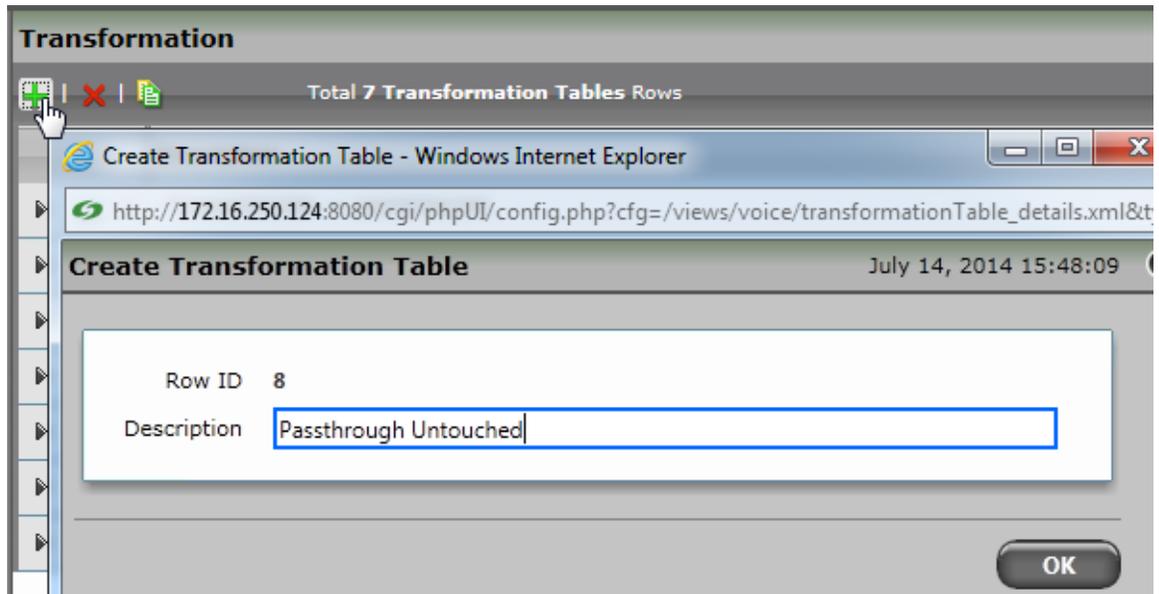
Transformation Tables facilitate the conversion of names, numbers and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number, or into a SIP address (URI). Every Call Routing Table Entry requires a Transformation Table.

- In the Navigation tree, click on Transformations

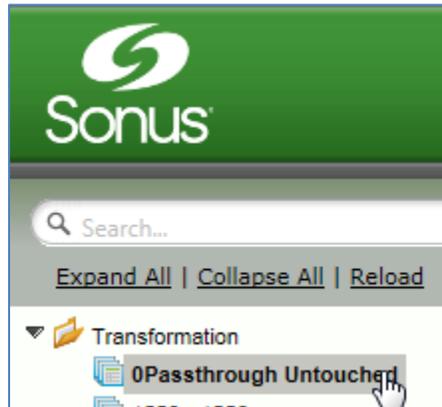


Create a new Transformation Table:

- Click the + to add a Transformation Table
- Type the desired name of the Table
- Click OK



- In the Navigation tree, click on the name of the new Transformation Table that you just added.



- Use the + to create the Transformation Entries as desired for your installation.

Passthrough Untouched						
Total 2 Transformation Entry Rows						
	Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
▶	<input type="checkbox"/> <input type="checkbox"/> <input checked="" type="checkbox"/>	Called Address/Number	(.*)	Called Address/Number	\1	Mandatory
▶	<input type="checkbox"/> <input type="checkbox"/> <input checked="" type="checkbox"/>	Calling Name	(.*)	Calling Name	\1	Optional

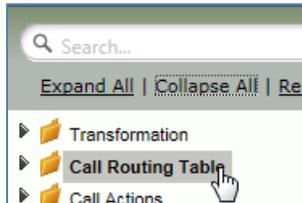
**\*\*NOTE:** You will likely need to create a separate Transformation Table for each Enterprise-based SIP Server or TDM destination.

The sample transformation above simply passes the calling and called number unchanged through the SBC. Modify the (number) transformations to properly manipulate the called and calling number for your installation.

### 3.2.12 Configure a Call Routing Table to the Broadsoft Server

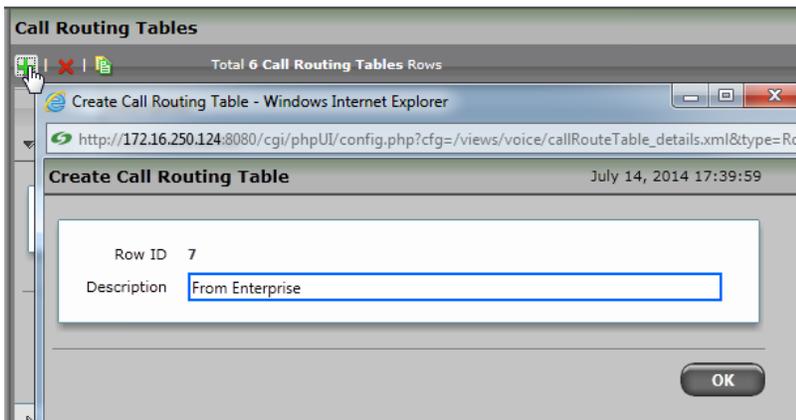
Call Routing allows calls to be carried between signaling groups, thus allowing calls to be carried between ports and between protocols (like ISDN to SIP). Call Routes are grouped into Call Routing Tables.

- In the Navigation tree, click on *Call Routing Table*



Create a new Call Routing Table. This call routing will take call from the Enterprise and route them to the Broadsoft server:

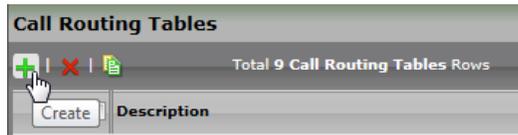
- Click the + to add a Call Routing Table
- Type the desired name of the Table
- Click OK



- In the Navigation tree, click on the name of the new Call Routing Table that you just added.



- Use the + to create the Call Routing Entries as desired for your installation



- Select the *Transformation Table* created in the previous step
- Set the *Destination Signaling Group* to the Broadsoft Signaling Group
- Set the *Media Modes* as noted below
- Click OK

**Route Details**

Description

Admin State

Route Priority

Call Priority

Number/Name Transformation Table

---

**Destination Information**

Destination Type

Message Translation Table

Cause Code Reroutes

Cancel Others upon Forwarding

Fork Call

Destination Signaling Groups

\*

---

**Media**

Audio/Fax Stream Mode

Video/Application Stream Mode

**Quality of Service**

Quality Metrics Number of Calls  [1..100]

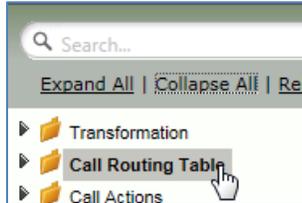
Quality Metrics Time Before Retry  [1-60] min.

Min. ASR Threshold  % [0..100]

### 3.2.13 Configure a Call Routing Table to the Enterprise IP PBX

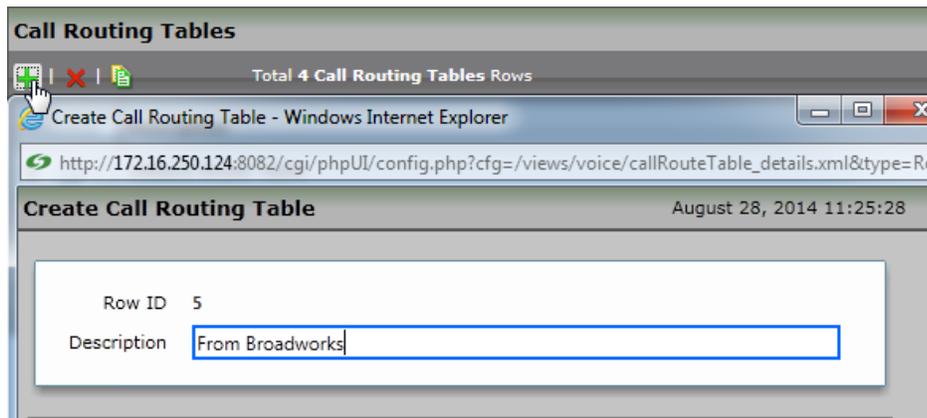
Call Routing allows calls to be carried between signaling groups, thus allowing calls to be carried between ports and between protocols (like ISDN to SIP). Call Routes are grouped into Call Routing Tables.

- In the Navigation tree, click on *Call Routing Table*

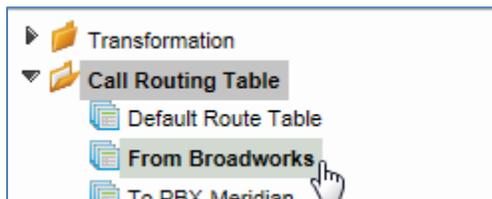


Create a new Call Routing Table. This call routing will take call from the Broadsoft and route them to the Enterprise:

- Click the + to add a Call Routing Table
- Type the desired name of the Table
- Click OK



- In the Navigation tree, click on the name of the new Call Routing Table that you just added.



- Use the + to create the Call Routing Entries as desired for your installation



- Select the *Transformation Table* created in the previous step
- Set the *Destination Signaling Group* to the ipPBX Signaling Group
- Set the *Media Modes* as noted below
- Click OK

Route Details

Description:

Admin State:

Route Priority:

Call Priority:

Number/Name Transformation Table:

---

Destination Information

Destination Type:

Message Translation Table:

Cause Code Reroutes:

Cancel Others upon Forwarding:

Fork Call:

Destination Signaling Groups:

---

Media

Quality of Service

Audio/Fax Stream Mode:

Video/Application Stream Mode:

Media Transcoding:

Media List:

Quality Metrics Number of Calls:  [1..100]

Quality Metrics Time Before Retry:  [1-60] min.

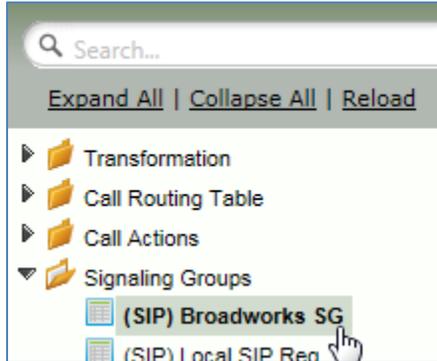
Min. ASR Threshold:  % [0..100]

Enable Max. R/T Delay:

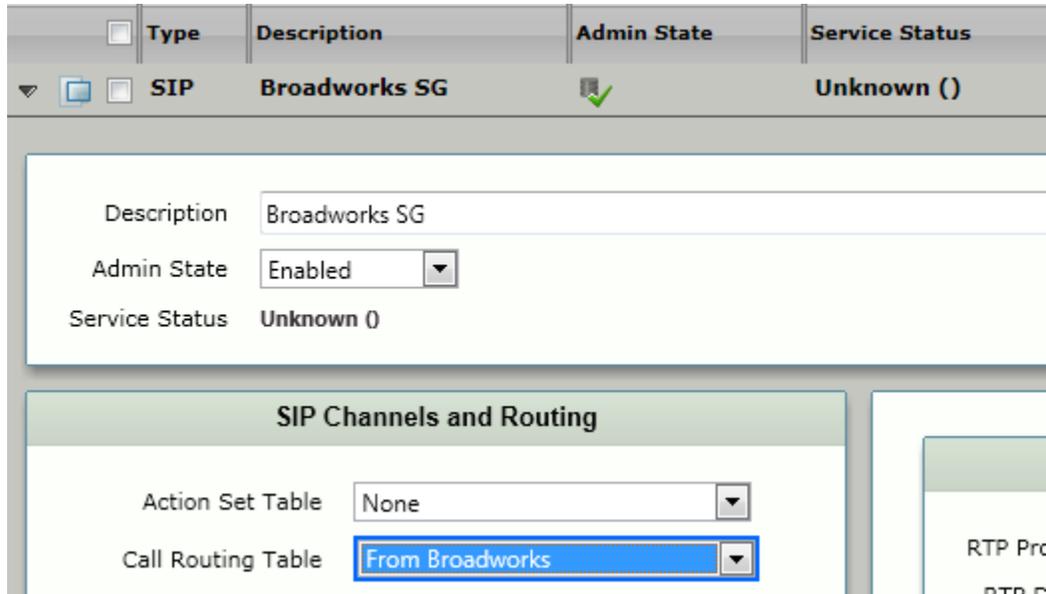
### 3.2.14 Set/Verify the Call Routing Table in the Broadsoft Signaling Group

Ensure that each Signaling Group is configured using an appropriate Call Route Table.

- In the Navigation Tree, click the *Broadworks SG* Signaling Group

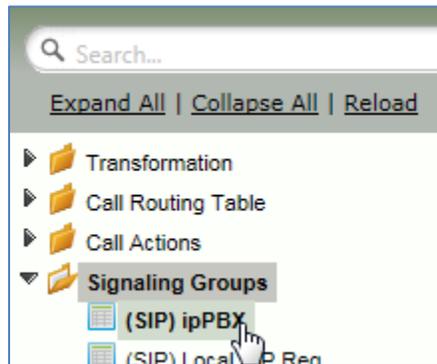


- The Broadsoft Signaling Group must be configured to use the *From Broadworks* Call Routing Table

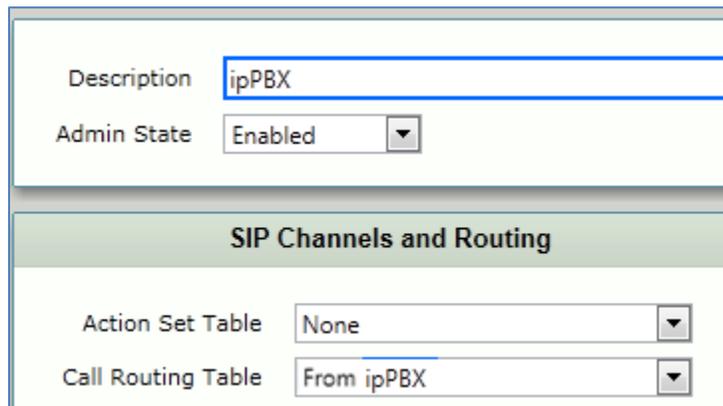


### 3.2.15 Set/Verify the Call Routing Table in the IP PBX Signaling Group

- In the Navigation Tree, click the *ipPBX SG* Signaling Group



- The *Enterprise Exchange* Signaling Group must be configured to use the FROM ipPBX Call Routing Table

A screenshot of a configuration page for a signaling group. The page has a light green header and a white body. The 'Description' field contains 'ipPBX'. The 'Admin State' is set to 'Enabled' with a dropdown arrow. Below this is a section titled 'SIP Channels and Routing' with a green header. Under this section, there are two dropdown menus: 'Action Set Table' is set to 'None' and 'Call Routing Table' is set to 'From ipPBX'.

### 3.2.16 Configure Trunking Identification

- In the Navigation Tree, click the *ipPBX SG* Signaling Group



Create a new Trunk Group entry.

- Click the + to add a Trunk Group Table



- Supply the information per your installation requirements
- Click OK

A screenshot of a configuration form for a new Trunk Group. The form contains the following fields and controls:

- Description:
- Trunk Group ID:  \*
- Trunk Group Type:  ▼
- Include ID in Outbound Calls:  ▼
- Use ID for Routing Inbound Calls:  ▼
- Associated Signaling Groups: 
  - \*
  -

### 3.2.16.1 Configure SIP Parameters

This section describes how to configure SIP parameters such as timers and headers.

Step	Command	Description
Step 1		
Step 2		
Step 3		

### 3.2.16.2 Configure BroadWorks SIP Peers

This section describes how to configure the SBC 1000 / SBC 2000 with BroadWorks Application Server and Network Server Peer(s).

Step	Command	Description
Step 1		
Step 2		
Step 3		

### 3.2.16.3 Configure Registration

This section describes how to configure the SBC 1000 / SBC 2000 core-side registration settings.

Step	Command	Description
Step 1		
Step 2		
Step 3		

### 3.2.16.4 Configure Failover/Failback for BroadWorks Redundancy

This section describes how to configure the SBC 1000 / SBC 2000 failover and failback mechanism to support BroadWorks redundant Application Servers and Network Servers.

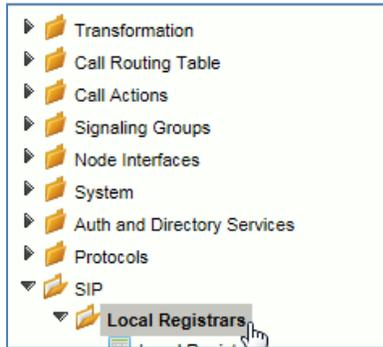
Step	Command	Description
Step 1		
Step 2		
Step 3		

### 3.3.1 Configure SIP Access Side

#### 3.3.1.1 Configure Registering Peer (Registering Access Device)

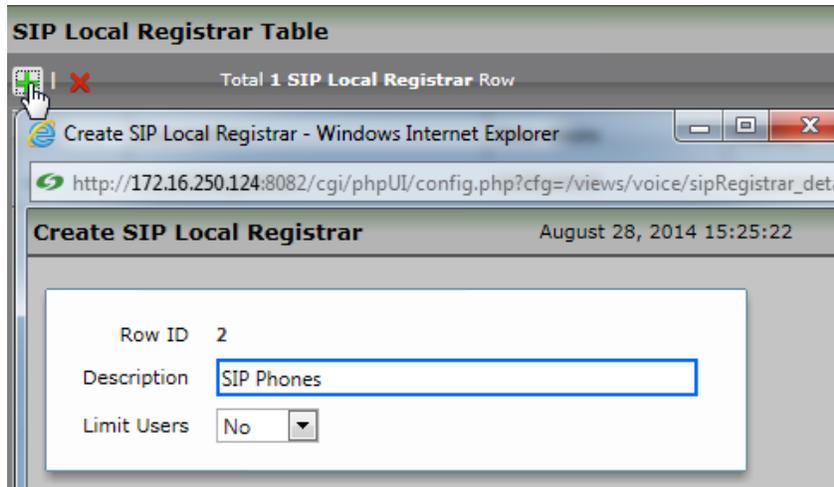
This section describes how to configure the SBC 1000 / SBC 2000 for registering access device peers.

- In the Navigation Tree, click *SIP | Local Registrars*



Create a new Trunk Group entry.

- Click the + to add a SIP Registrar
- Supply the Description Name
- Click OK



**See Section 3.3, Remove Survivability, to configure Signaling Groups and Call Routes to use SIP Phone Registration.**

#### 3.3.1.2 Configure Static Peer (Non-registering Access Device)

This section describes how to configure the SBC 1000 / SBC 2000 for static peers, that is, access devices that do not register.

*{If this function is not supported by the session controller, indicate "Not Supported."}*

Step	Command	Description
Step 1		
Step 2		
Step 3		

### 3.3.1.3 Configure Static Peer with NAT Traversal (Non-registering Access Device with NAT Traversal)

This section describes how to configure the SBC 1000 / SBC 2000 for static peers requiring Network Address Translation (NAT) traversal, that is, access devices that do not register and require NAT traversal.

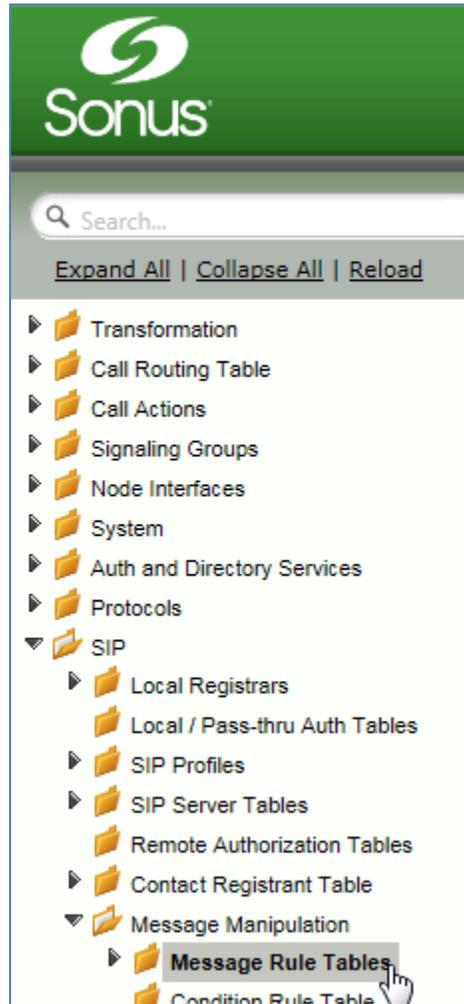
*{If this function is not supported by the session controller, indicate "Not Supported."}*

Step	Command	Description
Step 1		
Step 2		
Step 3		

### 3.3.1.4 Configure SIP Connect Peer (GIN Registering Access Device)

Create a SMM to add the GIN Registration for call from the SBC to the Broadsoft Server.

- In the Navigation Tree, click the *Message Manipulation | Message Rules Table*



Create a new SMM Rule Table:

- Click the + to add a Message Rule
- Type the desired name of the Table and enter the information as shown
- Click OK

**SIP Message Rule Table**  
Total 1 SIP Message Manipulation Table Row

Create Message Rule Table - Windows Internet Explorer  
[http://172.16.250.124:8080/cgi/phpUI/config.php?cfg=/views/voice/sipMessageRuleTable\\_details.xml&](http://172.16.250.124:8080/cgi/phpUI/config.php?cfg=/views/voice/sipMessageRuleTable_details.xml&)

**Create Message Rule Table** July 30, 2014 11:30:35

Row ID 2

Description GIN Registration

Applicable Messages Selected Messages

Message Selection Register

Add/Edit \*  
Remove

Table Result Type Mandatory

OK

- In the Navigation Tree, click the *GIN Registration* SMM Table



- From the *Create Signaling Rule* pulldown, select *Header Rule*



Create a new SMM Header Rule:

- Type the desired name of the Table and set the configuration as shown
- Click *Add/Edit*

The screenshot shows a dialog box titled "Create SIP Header Rule". It contains several fields and dropdown menus:

- Description: Add Require header
- Condition Expression: Add/Edit
- Admin State: Enabled
- Result Type: Optional
- Header Action: Add
- Header Name: Require
- Header Value: Add

The "Add/Edit" button next to the Header Value field is highlighted with a mouse cursor.

Add the Header Value:

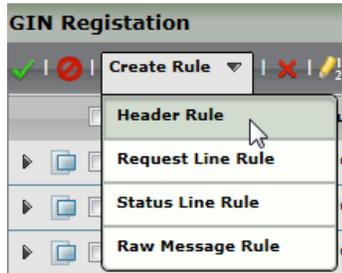
- Configure the information as shown
- Click OK

The screenshot shows a dialog box titled "Edit Message Field". It contains two fields and two buttons:

- Type of Value: Literal
- Value: gin
- Buttons: OK, Cancel

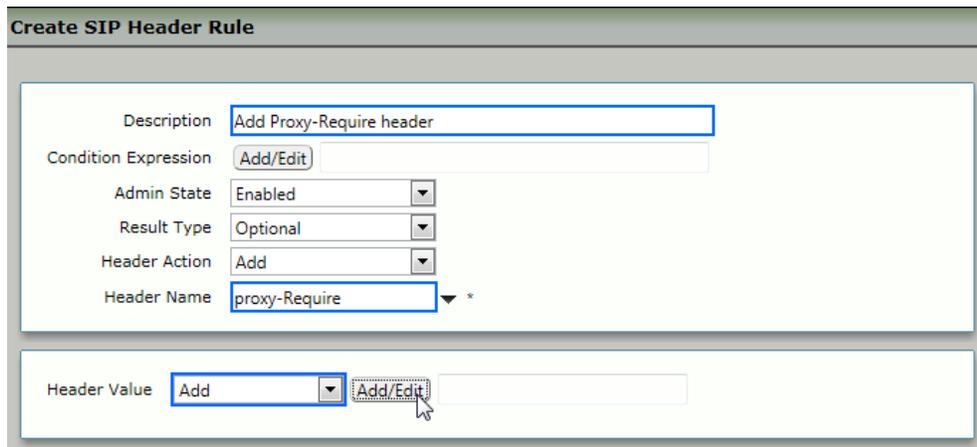
The "OK" button is highlighted with a mouse cursor.

- Add a second Header Rule. From the *Create Signaling Rule* pulldown, select *Header Rule*



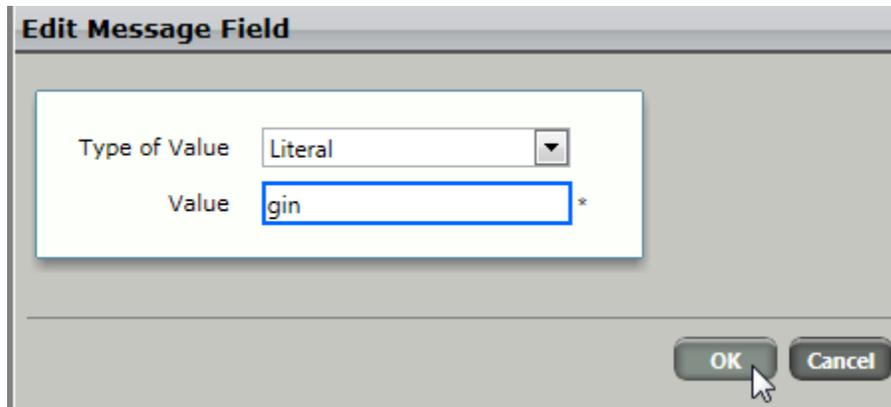
Create a new SMM Header Rule:

- Type the desired name of the Table and set the configuration as shown
- Click *Add/Edit*



Add the Header Value:

- Configure the information as shown
- Click OK



- Create a third header rule. From the *Create Signaling Rule* pull-down, select *Header Rule*



Create a new SMM Header Rule:

- Type the desired name of the Table and set the configuration as shown
- Click *Add/Edit*

The screenshot shows the 'Create SIP Header Rule' configuration window. The fields are as follows:

- Description: Add bnc parameter
- Condition Expression: Add/Edit
- Admin State: Enabled
- Result Type: Optional
- Header Action: Modify
- Header Name: Contact
- Header Ordinal Number: 1st

Below the main configuration, there is a section for 'Header Value' with a sub-section for 'URI' parameters:

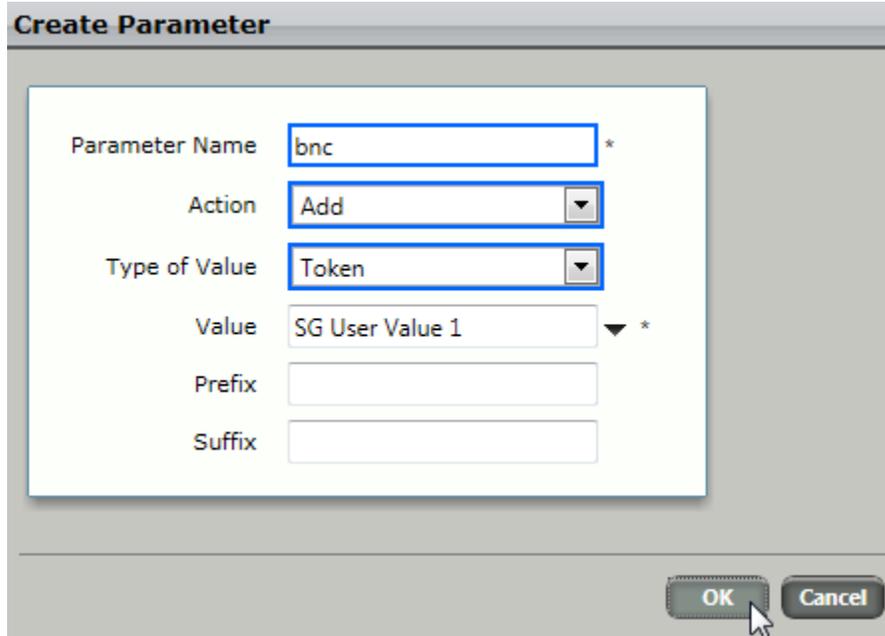
- URI Scheme: Ignore
- URI User Info: Ignore
- URI Host: Ignore
- URI Port: Ignore

At the bottom, there is a table for 'URI Parameters' with the following structure:

Total 0 SPRUriParam Rows			
	Name	Value	Action
-- Table is empty --			

Add the Header Value:

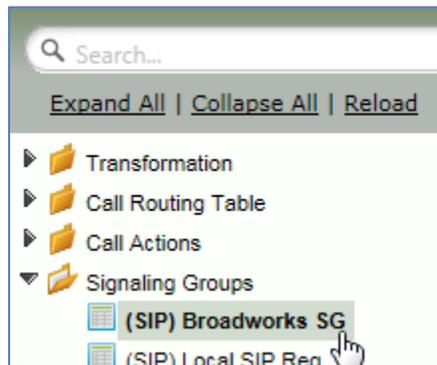
- Configure the information as shown
- Click OK



Configure the SMM Rule in the Broadsoft Signaling Group

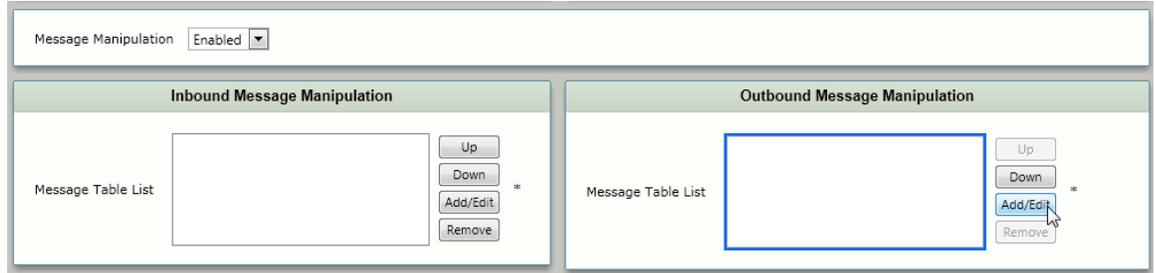
Configure the Broadsoft Signaling Group with the newly created SMM Rule.

- In the Navigation Tree, click the *BSFT Connection* Signaling Group

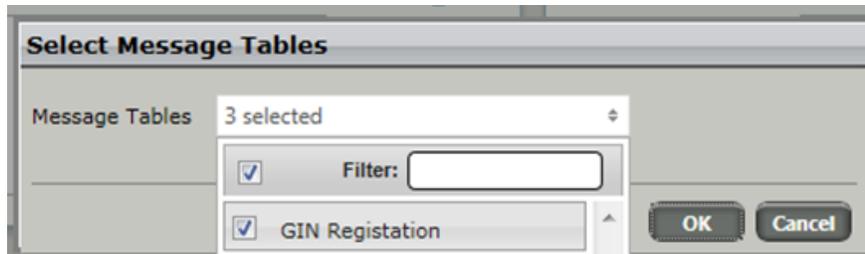


Enable the new SIP Message Manipulation (SMM) Rule:

- Set the Message Manipulation to *Enable*
- In the Outgoing Message Manipulation pane, click *Add/Edit*



- In the pop-up window, select the newly created SMM Rule, then click *OK*



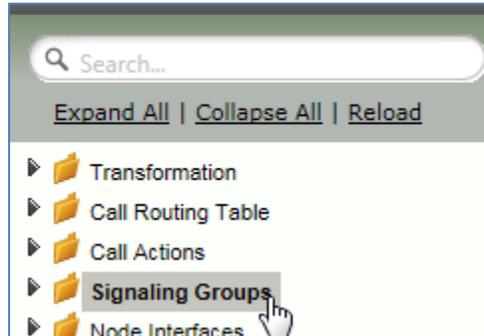
- Click *Apply*



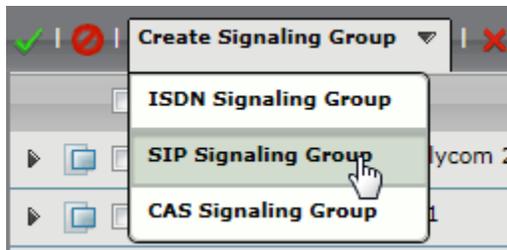
### 3.4 Remote Survivability Configuration

This section describes the settings necessary for remote survivability when the BroadWorks server(s) cannot be reached for call control. In this case, the session controller facilitates calls between users behind the SBC as well as off-net calls.

- In the Navigation Tree, click *Signaling Groups*



- From the *Create Signaling Group* pulldown, select *SIP Signaling Group*



Enter the SIP Phone Signaling Group information as noted below:

- Select the *Default SIP Profile*
- Select the Broadsoft *SIP Server Table*
- Set the SIP Mode to *Local Registrar*
- Set the Registrar to *SIP Phones*
- Verify/Delete/Create *Listening Ports* that the SBC will use to receive SIP from the Broadsoft Server
- Set all Media Modes as per the capabilities of your IP Phones
- Add the Federated IP with the IP address and netmask to permit the IP Phones within your network

**Signaling Group Table** August 26, 2014 16:11:24

✓ | ⏪ | ⏩ | ✖ | Create Signaling Group ▼ | ✖ Total 4 Signaling Group Rows

Type	Description	Admin State	Service Status	Display	Prim Key
SIP	Broadworks SG	✓	Up	<a href="#">Counters</a>   <a href="#">Channels</a>   <a href="#">Subscriber Data</a>	2
SIP	SIP In from Phones	✓	Up	<a href="#">Counters</a>   <a href="#">Channels</a>	3

---

Description:

Admin State:

Service Status: Up

**SIP Channels and Routing**

Action Set Table:

Call Routing Table:

No. of Channels:  \* [1..960]

SIP Profile:

SIP Mode:

Registrar:

Agent Type:

Interop Mode:

Channel Hunting:

Notify Lync CAC Profile:

Outbound Proxy:

Outbound Proxy Port:  [1024..65535]

No Channel Available Override:

Call Setup Response Timer:  [180..750] secs

**Media Information**

Audio/Fax Stream Proxy Mode:

Audio/Fax Stream DSP Mode:

Video/Application Stream Proxy Mode:

Media List ID:

Play Ringback:

Tone Table:

Early 183:

Music on Hold:

**Mapping Tables**

SIP To Q.850 Override Table:

Q.850 To SIP Override Table:

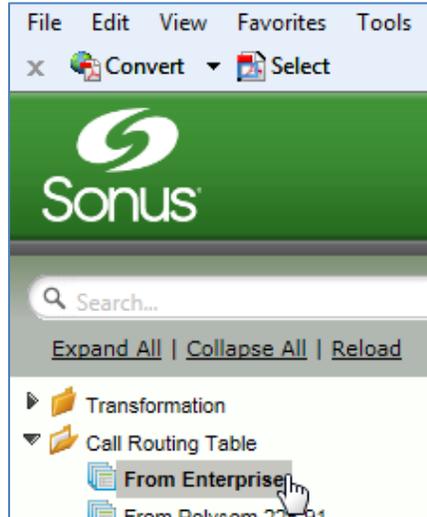
Pass-thru Peer SIP Response Code:

Listen Ports			Federated IP/FQDN	
+   X Total 1 SIP Listen Port Row			+   X Total 1 SIP Federated IP Row	
<input type="checkbox"/>	Port	Protocol	IP/FQDN	Netmask
<input type="checkbox"/>	5060	UDP	172.10.8.0	255.255.252.0
N/A				

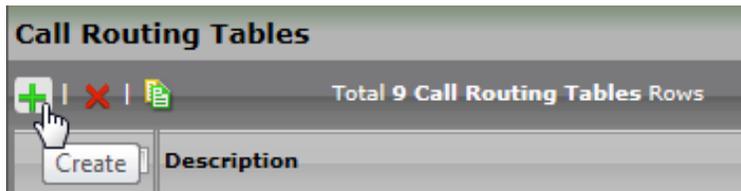
Message Manipulation  Disabled ▾

Create an additional Call Route entry in the Enterprise Call Route table. The additional entry will be used for routing calls if the Broadsoft service becomes unavailable.

- In the Navigation tree, click on the name of the new Call Routing Table that you just added.



- Use the + to create the Call Routing Entries as desired for your installation



- Select the *Passthrough Untouched* Transformation Table
- Set the Destination Signaling Group to the *SIP In from Phones*
- Set the *Media Modes* as noted below
- Click OK

**From SIP Phones** August 26, 2014 17:01:58

✓ | ✗ | + | - | 🗨 | 📊 | Display Counters Total 4 Call Route Entry Rows

Admin State	Priority	Transformation Table	First Signaling Group	Description	Fork Call	Prima Key
▶	1	Passthrough Untouched	(SIP) Broadworks SG	Route to Broadworks	No	1
▼	1	<b>Passthrough Untouched</b>	<b>(SIP) SIP In from Phones</b>	<b>Try Registrar</b>	No	2

**Route Details**

Description:

Admin State:

Route Priority:

Call Priority:

Number/Name Transformation Table:

**Destination Information**

Destination Type:

Message Translation Table:

Cause Code Reroutes:

Cancel Others upon Forwarding:

Fork Call:

Destination Signaling Groups:

**Media**

Audio/Fax Stream Mode:

Video/Application Stream Mode:

Media Transcoding:

Media List:

**Quality of Service**

Quality Metrics Number of Calls:  [1..100]

Quality Metrics Time Before Retry:  min. [1..60]

Min. ASR Threshold:  % [0..100]

Enable Max. R/T Delay:

Max. R/T Delay:  ms [1..65535]

Enable Max. Jitter:

Max. Jitter:  ms [1..3000]

## **Appendix A: Reference SBC 1000 / SBC 2000 Configuration**

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The following is a reference configuration for the SBC 1000 / SBC 2000 configured for use with BroadWorks.

## References

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- [1] Sonus Networks 2014 SBC 4.0 User's Guide , available at <https://support.sonus.net/display/ALLDOC/SBC+1000-2000+Documentation>
- [2] BroadSoft, Inc. 2013. *BroadWorks Redundancy Guide, Release 20.0*. Available from BroadSoft at [xchange.broadsoft.com](http://xchange.broadsoft.com).
- [3] BroadSoft, Inc. 2013. *BroadWorks SIP Access Interface Interworking Guide, Release 20.0*. Available from BroadSoft at [xchange.broadsoft.com](http://xchange.broadsoft.com).
- [4] BroadSoft, Inc. 2013. *BroadWorks SIP Trunking Solution Guide, Release 20.0*. Available from BroadSoft at [xchange.broadsoft.com](http://xchange.broadsoft.com)
- [5] BroadSoft, Inc. 2014. *BroadWorks Session Controller Interoperability Test Plan, Release 20.0*. Available from BroadSoft at [xchange.broadsoft.com](http://xchange.broadsoft.com).