



BroadWorks

## **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 Release 3.2.1 v319 validation with BroadWorks Release R20 SP1 v1.2. |
|         |  |
|         |  |

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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 a PBX Trunking Gateway that uses the Session Initiation Protocol (SIP) to communicate with BroadWorks for call control.

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 SBC 4.0 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 SIP interface. Qualitative aspects of the device or device capabilities not affecting the SIP interface, such as display features, performance, and audio qualities 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.

In the following table, *Compatible Versions* identify specific SBC 1000 / SBC 2000 versions that 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 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 testing the combination themselves, using the *BroadWorks IP-PBX/PBX Trunking Interoperability Test Plan* [7].

| Verified Versions |                    |                                      |   |
|-------------------|--------------------|--------------------------------------|---|
| Date (mm/yyyy)    | BroadWorks Release | SBC 1000 / SBC 2000 Verified Version | SBC 1000 / SBC 2000 Compatible Versions |
| 09/2014           | Release 20 SP1v1.2 | Release 3.2.1 v319                   |   |
|                   |                    |                                      |   |

### 2.2 Interface Capabilities Supported

The Sonus Networks, Inc. SBC 1000 / SBC 2000 has completed interoperability testing with BroadWorks using the *BroadWorks IP-PBX/PBX Trunking Interoperability Test Plan* [7]. 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 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 IP-PBX/PBX Trunking Interoperability Test Plan Support |   |           |                     |
|---|---|-----------|---------------------|
| 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                            | No        |                     |
|   | Early-Session                           | No        |                     |
|   | 181 Call Being Forwarded                | NT        | PBX Limitation      |
|   | Dial Plan                               | Yes       |                     |
|   | DTMF – Inband                           | Yes       |                     |
|   | DTMF – RFC 2833                         | Yes       |                     |
|   | DTMF – DTMF Relay                       | No        | Supported in R4.0.0 |
|   | Codec Negotiation                       | Yes       |                     |
|   | Codec Renegotiation                     | NT        | PBX Limitation      |
| <b>SIP Connect</b>  | GIN Registration                        | Yes       |                     |
|   | Private Branch Exchange (PBX) Redirect  | NT        | PBX Limitation      |
|   | Calling Line ID and Privacy             | Yes       |                     |
|   | Calling Line ID with Unicode Characters | No        |                     |
|   | E.164 Numbering                         | NT        |                     |
| <b>BroadWorks Services</b>  | Voice Message Deposit/Retrieval         | Yes       |                     |
|   | Message Waiting Indicator               | NT        | PBX Limitation      |

|   |   |     |                            |
|---|---|-----|----------------------------|
|   | Connected Line ID                                   | No  |                            |
|   | Connected Line ID with Unicode Characters           | No  |                            |
|   | Connected Line ID on UPDATE                         | No  |                            |
|   | Connected Line ID on Re-INVITE                      | No  |                            |
|   | Diversion Header                                    | No  | Supported in R4.0.0        |
|   | History-Info Header                                 | No  | Supported in R4.0.0        |
|   | Enterprise Trunking – Originating Trunk Group (OTG) | No  | Supported in R4.0.0        |
|   | Enterprise Trunking – Destination Trunk Group (DTG) | No  | Supported in R4.0.0        |
|   | Enterprise Trunking – Trunk Group (TGRP)            | No  | Supported in R4.0.0        |
|   | Advice of Charge                                    | No  |                            |
|   | Meet-Me Conferencing                                | Yes |                            |
|   | Meet-Me Conferencing – G722                         | No  | Supported in R4.0.0        |
|   | Meet-Me Conferencing – AMR-WB                       | No  | Supported in R4.0.0        |
| <b>DUT Services – Call Control Services</b>                             | Call Waiting  | No  | Supported in R4.1.0        |
|   | Call Hold   | Yes | PBX Limitation             |
|   | Call Transfer                                       | No  |                            |
|   | 2 B Channel Transfer                                | No  |                            |
|   | Three-Way Calling                                   | Yes |                            |
| <b>DUT Services – Registration and Authentication</b>                   | Register Authentication                             | No  |                            |
|   | Maximum Registration                                | No  |                            |
|   | Minimum Registration                                | No  |                            |
|   | Invite Authentication                               | No  |                            |
|   | Re-Invite/Update Authentication                     | No  |                            |
|   | Refer Authentication                                | No  |                            |
|   | Device Authenticating BroadWorks                    | No  |                            |
| <b>DUT Services – Fax</b>   | G711 Fax Passthrough                                | NT  | PBX Limitation (No Analog) |
|   | G711 Fax Fallback                                   | NT  | PBX Limitation (No Analog) |
|   | T38 Fax Messaging                                   | NT  | PBX Limitation (No Analog) |
| <b>Session Border Controller (SBC)/ Application Layer Gateway (ALG)</b> | Register  | No  |                            |
|   | Outgoing Invite                                     | No  |                            |
|   | Incoming Invite                                     | No  |                            |
| <b>Video – Basic Video Calls</b>  | Call Origination                                    | NA  |                            |
|   | Call Termination                                    | NA  |                            |
|   | Call Hold   | NA  |                            |

|                                      |  |     |  |
|--------------------------------------|--|-----|--|
|                                      | Call Waiting                           | NA  |  |
|                                      | Call Transfer                          | NA  |  |
| Video – BroadWorks<br>Video Services | Auto Attendant                         | NA  |  |
|                                      | Auto Attendant – HD                    | NA  |  |
|                                      | Voice Messaging                        | NA  |  |
|                                      | Voice Messaging – HD                   | NA  |  |
|                                      | Custom Ringback                        | NA  |  |
|                                      |  |     |  |
| TCP                                  | Register                               | Yes |  |
|                                      | Outgoing Invite                        | Yes |  |
|                                      | Incoming Invite                        | Yes |  |
| IPV6                                 | Call Origination                       | No  |  |
|                                      | Call Termination                       | No  |  |
|                                      | Session Audit                          | No  |  |
|                                      | Ringback                               | No  |  |
|                                      | Codec Negotiation/Renegotiation        | No  |  |
|                                      | Voice Message Deposit/Retrieval        | No  |  |
|                                      | Call Control                           | No  |  |
|                                      | Registration with Authentication       | No  |  |
|                                      | T38 Fax Messaging                      | No  |  |
|                                      | Redundancy                             | No  |  |
|                                      | SBC                                    | 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 Sonus Networks, Inc. release notes.

| Issue Number | Issue Description | Partner Version |
|--------------|-------------------|-----------------|
|--------------|-------------------|-----------------|

---

|      |      |       |  |  |  |
|------|------|-------|--|--|--|
|      |      | 3.2.1 |  |  |  |
| ---- | None |       |  |  |  |
|      |      |       |  |  |  |

### 3 Solution Configuration

The following figure shows an example of a typical deployment configuration with the SBC 1000 / SBC 2000. Typically, the SBC 1000 / SBC 2000 is placed on the customer premises to which SIP phones are registered and is on a private network, which necessitates an edge device or an SBC.

The SBC 1000 / SBC 2000 registers its main line (or pilot number) with the trunk group configured on BroadWorks via an SBC. A single registration, identifying the pilot number via GIN registration, conforms to SIP Connect standards for trunk registration. This enables all PBX users to be implicitly registered with BroadWorks via the pilot number registration. Note that the SBC deployed in the solution must support SIP Connect.

The SBC 1000 / SBC 2000 is identified as a BroadWorks PBX Classification Type A. For PBX classification descriptions, see the *BroadWorks SIP Trunking Solution Guide* [4]. To determine how to configure Oracle for this PBX classification type, see the *SIP Trunking Configuration* table in the *BroadSoft Partner Configuration Guide Oracle Net-Net 3000/4000 Series* [6].

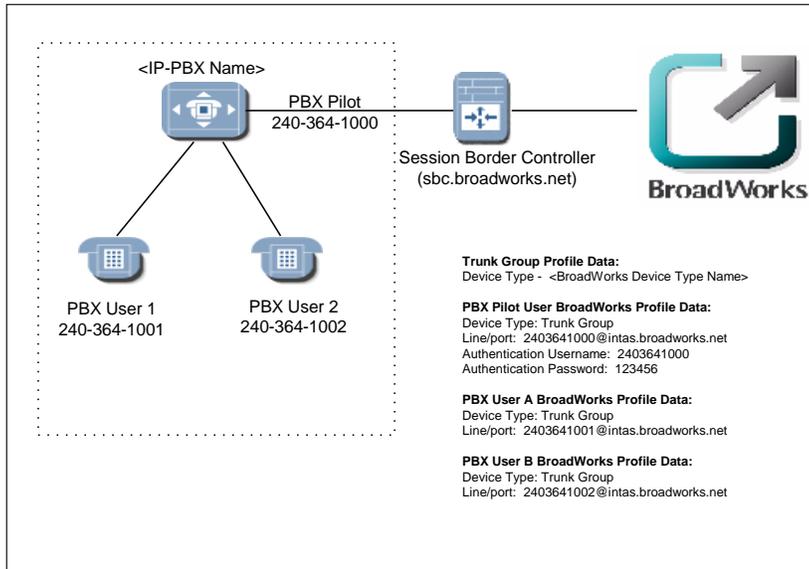


Figure 1 SBC 1000 / SBC 2000 Configuration Setup (IP-PBX)

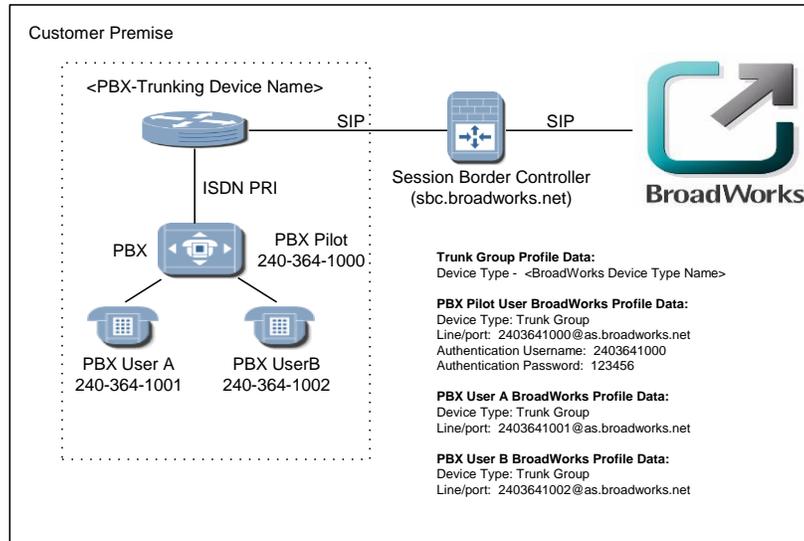


Figure 2 SBC 1000 / SBC 2000 Configuration Setup (PBX-Trunking Device)

The following configuration sections describe how to configure the SBC 1000 / SBC 2000 to support the configuration shown in the above diagram. The SBC 1000 / SBC 2000 configuration examples refer to data in the diagram.

## 4 BroadWorks Configuration

---

This section identifies the required BroadWorks device profile type settings for the Sonus Networks, Inc. SBC 1000 / SBC 2000 and any other unique BroadWorks configuration required for interoperability with the SBC 1000 / SBC 2000.

### 4.1 BroadWorks Device Profile Type Configuration

This section identifies the device profile type settings to use when deploying the Sonus Networks, Inc. SBC 1000 / SBC 2000 with BroadWorks.

Create a device profile type for the Sonus Networks, Inc. SBC 1000 / SBC 2000 as shown in the following example. The settings shown are recommended for use when deploying the Sonus Networks, Inc. SBC 1000 / SBC 2000 with BroadWorks. For an explanation of the profile parameters, see the *BroadWorks Device Management Configuration Guide* [\[1\]](#).

**Identity/Device Profile Type Modify**  
Modify an existing identity/device profile type.

OK Apply Delete Export Cancel

Identity/Device Profile Type: Sonus SBC-1000\_2000 Trunk  
Signaling Address Type: Intelligent Proxy Addressing  
 Obsolete

Standard Options

Number of Ports:  Unlimited  Limited To

Ringback Tone/Early Media Support:  RTP - Session  
 RTP - Early Session  
 Local Ringback - No Early Media

Authentication:  Enabled  
 Disabled  
 Enabled With Web Portal Credentials

Hold Normalization:  Unspecified Address  
 Inactive  
 RFC3264

Registration Capable  Authenticate REFER  
 Static Registration Capable  Video Capable  
 E164 Capable  Use History Info Header  
 Trusted

Advanced Options

Route Advance  Forwarding Override  
 Wireless Integration  Conference Device  
 PBX Integration  Mobility Manager Device  
 Add P-Called-Party-ID  Music On Hold Device  
 Auto Configuration Soft Client  Requires BroadWorks Digit Collection  
 Requires BroadWorks Call Waiting Tone  Requires MWI Subscription  
 Advice of Charge Capable  Support Call Center MIME Type  
 Support Emergency Disconnect Control  Support Identity In UPDATE and Re-INVITE  
 Enable Monitoring  Support RFC 3398  
 Static Line/Port Ordering  Support Client Session Info  
 Support Call Info Conference Subscription URI  Support Remote Party Info  
 Support Visual Device Management  Bypass Media Treatment

Reset Event:  reSync  checkSync  Not Supported  
Trunk Mode:  User  Pilot  Proxy  
Hold Announcement Method:  Inactive  Bandwidth Attributes

Unscreened Presentation Identity Policy:  Profile Presentation Identity  
 Unscreened Presentation Identity  
 Unscreened Presentation Identity With Profile Domain

Web Based Configuration URL Extension:

Device Configuration Options:  Not Supported  Device Management  Legacy

OK Apply Delete Export Cancel

Figure 3 SBC 1000 / SBC 2000 Trunk Device Profile Type

## 4.2 BroadWorks Configuration Steps

There are no additional BroadWorks configuration steps required.

## 5 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 SBC 4.0 User's Guide [1] for SBC 1000 / SBC 2000 configuration details not covered in this section.

### 5.1 Configuration Method

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

### 5.2 System Level Configuration

This section describes system-wide configuration items that are generally required for each SBC 1000 / SBC 2000 to work with BroadWorks.

### 5.2.1 Configuration Settings

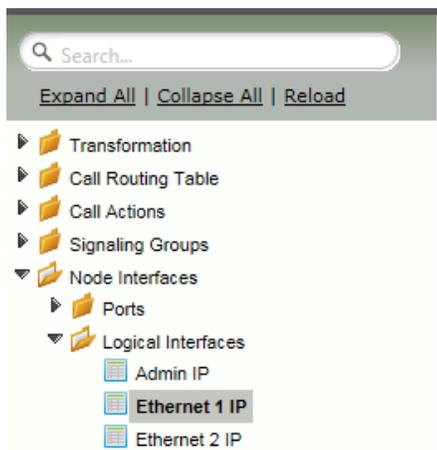
The Tabs across the top of the Sonus SBC WebUI permit the user to access various configuration subsystems. Within this document, all configurations will be performed under the SETTINGS tab.



### 5.2.2 Configure Network Settings

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.

**Ethernet 1 IP**      10.1.1.74      Disabled

**Identification/Status**

Interface Name: Ethernet 1 IP  
 I/F Index: 39  
 Alias:   
 Description:   
 Admin State: Enabled

**Networking**

MAC Address: 00:10:23:01:01:01  
 IP Assign Method: Static  
 Primary Address: 10.1.1.74  
 Primary Netmask: 255.255.255.0  
 Configure Secondary Interface: Disabled

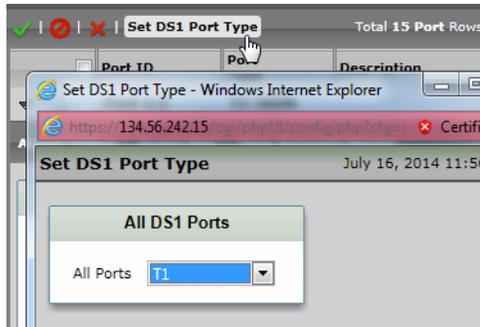
ACL In: None  
 ACL Out: None  
 ACL Forward: None

- Configure the TDM port for connectivity to the PBX. In the Navigation Tree, click *Node Interfaces | Ports | Port 1:1*

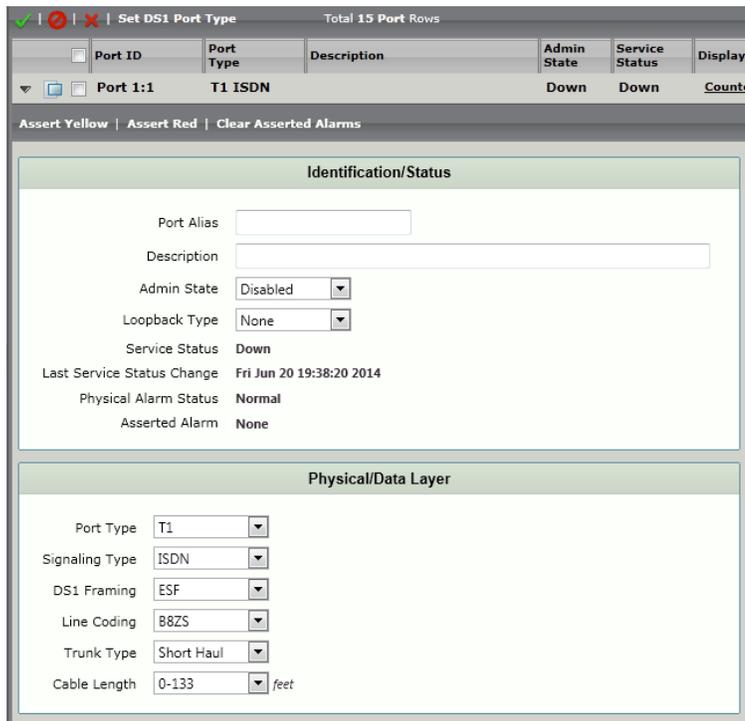


Set the TDM Port Type.

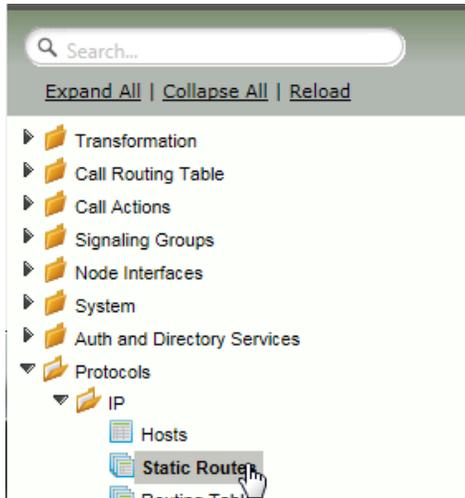
- Click *Set DS1 Port Type*
- Use the *All Ports* pulldown to select the appropriate DS1 port type for your installation.
- Click OK



- Configure the TDM port as necessary to connect to the PBX.



- 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.

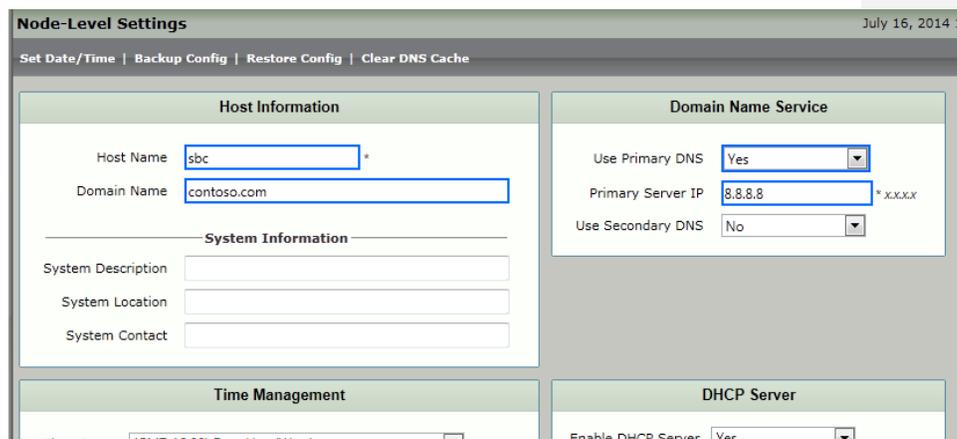
| Static IP Route Table |                |                 |              |        |
|-----------------------|----------------|-----------------|--------------|--------|
| Row ID                | Destination IP | Mask            | Gateway      | Metric |
| 1                     | 172.16.110.106 | 255.255.255.255 | 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



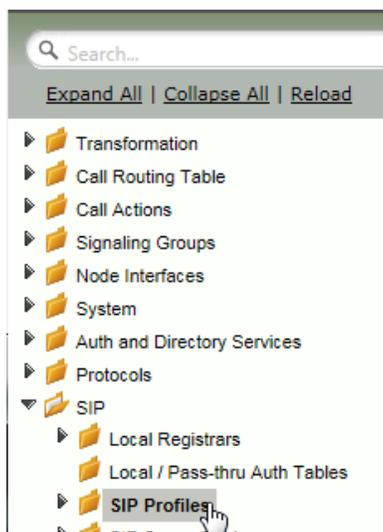
#### 5.2.2.1 Configure IPV6 Settings

*Not Supported.*

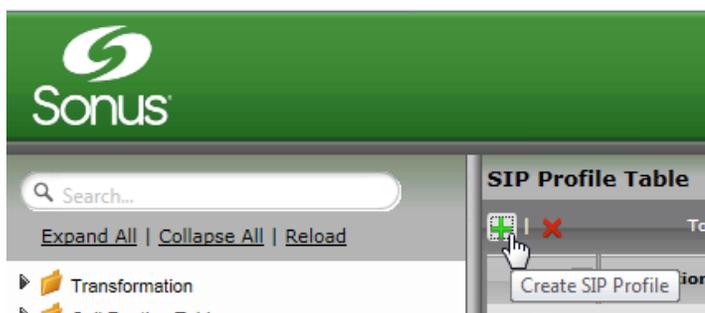
### 5.2.3 Configure SIP Interface Settings

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 +.



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

**SIP Profile Table**

Total 1 SIP Profile Row

| Description         | Primary Key |
|---------------------|-------------|
| Default SIP Profile | 1           |

Description: Default SIP Profile

**Session Timer**

Session Timer:

**MIME Payloads**

ELIN Identifier:

PIDF-LO Passthrough:

Unknown Subtype Passthrough:

**Header Customization**

UA Header:

Subscription State Passthrough:

FQDN in From Header:

Send Assert Header:

Trusted Interface:

Calling Info Source:

Diversion Header Selection:

**Options Tags**

100rel:

Update:

**Timers**

Transport Timeout Timer:  ms [5000..32000]

Maximum Retransmissions:

---

**RFC timers**

Timer T1:  ms [100..10000]

Timer T2:  ms [1000..80000](>= T1)

Timer T4:  ms [1000..100000]

Timer D:  ms [5000..640000]

Timer B: 32000 ms

Timer F: 32000 ms

Timer H: 32000 ms (64\*TimerT1)

Timer J: 32000 ms (64\*TimerT1)

**SDP Customization**

Send Number of Audio Channels:

Connection Info in Media Section:

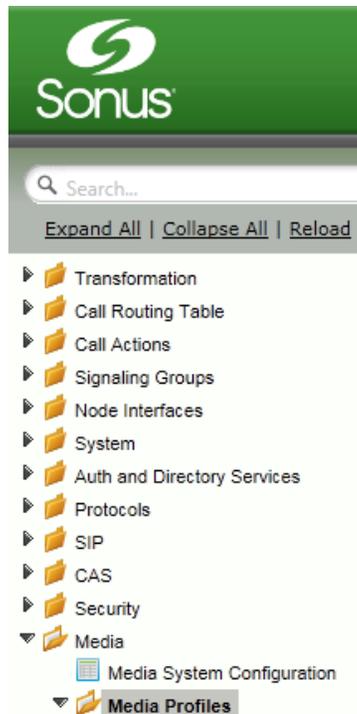
Origin Field Username:  default: SBC

Session Name:  default: VoipCall

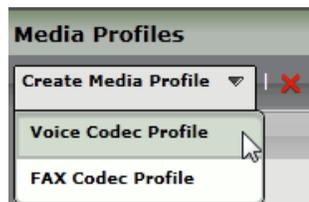
### 5.2.4 Configure Media Settings

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.

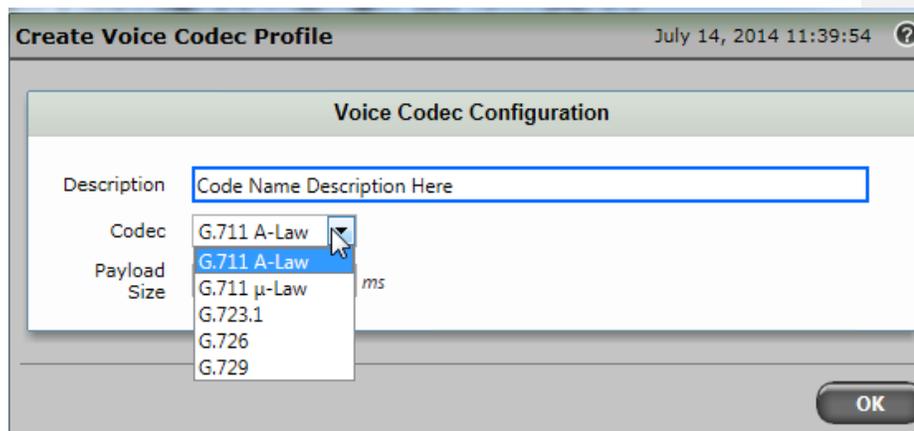
- In the Navigation Tree, click on *Media Profiles*.



- 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.

| Media Profiles |               |
|----------------|---------------|
| Codec          | Description   |
| G.711 A-Law    | Default G711A |
| G.711 μ-Law    | Default G711u |
| G.729          | G.729         |
| G.723.1        | G.723.1       |
| G.726          | G.726         |
| T.38 Fax       | T.38 Fax      |

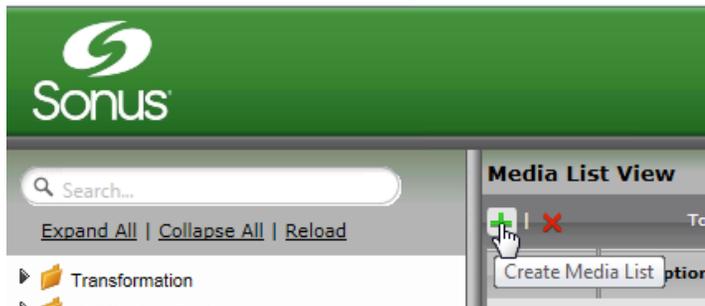
### 5.2.5 Configure Media Lists

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.

**BSFT Media List**

Description: BSFT Media List

Media Profiles List: Default G711A, Default G711u, G.729

Crypto Profile ID: None

Media DSCP: 46 \* [0..63]

RTCP Mode: RTCP

Dead Call Detection: Disabled

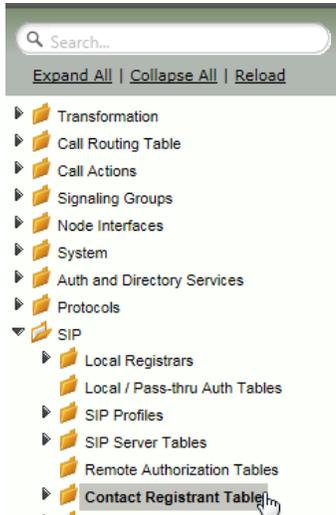
Silence Suppression: Enabled

**\*\*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.

### 5.3 Configure Broadsoft Subscriber Information

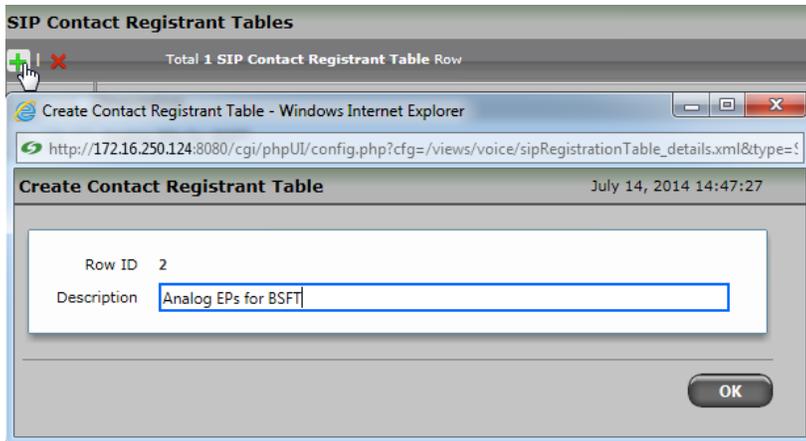
The Contact Registrant Table is used to provide user authentication to the Broadsoft server when calls are made.

- In the Navigation Tree, click on *Contact Registrant Table*

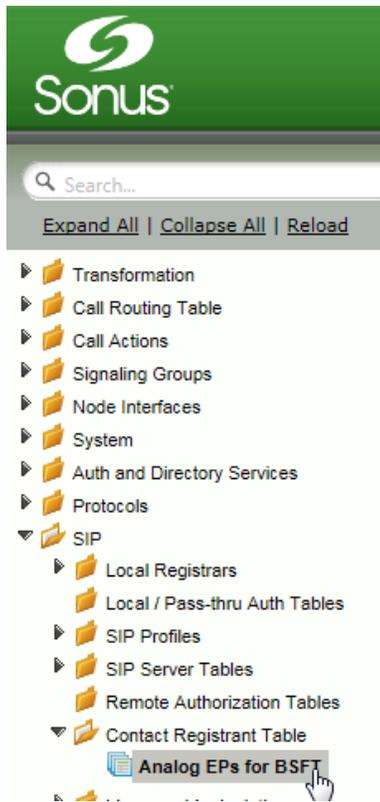


Add a Contact Registrant Table to hold the Broadsoft subscriber information.

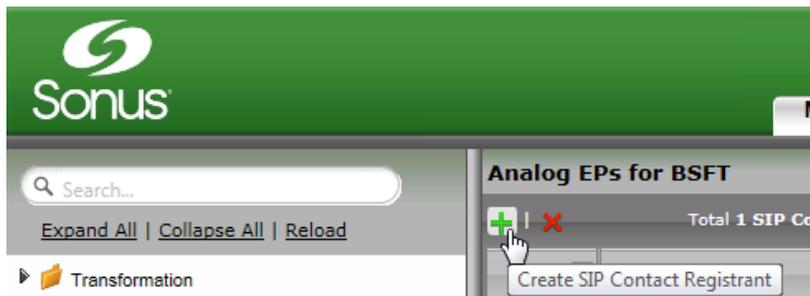
- Click the + to add a Contact Registrant Table
- Type of name of the Table
- Click OK



- Click the newly added Contact Registrant Table in the Navigation Tree.



- Click the + to add a Contact Registrant



- Add the Broadsoft subscription user in the *Address of Record URI* box. This information will be supplied by your service provider.

### Analog EPs for BSFT

+ | × Total 1 SIP Contact Registrant Entry Row

Address of Record

sip:2405556256@as.iop1.broadwo...

Type of Address of Record Static ▼

Address of Record URI sip:2405556256@as.iop1.broadworks.t \* user@host[:port]

Global Time to Live (TTL) 60 \* secs [30..86400]

Failed Registration Retry Timer 30 \* secs [30..86400]

#### SIP Contacts

+ | × Total 1 SIP User Contact Row

| Contact URI Username  | TTL (secs) | Priority (Q) |
|---|------------|--------------|
| <span style="color: yellow;">✎</span> <span style="float: right;">2405556256</span> | Inherited  | 0            |

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

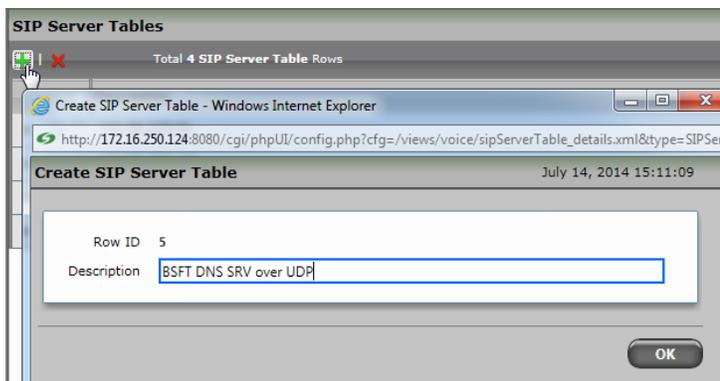
SIP Server Tables contain information about the SIP devices connected to the Sonus SBC 1000/2000. The entries in the tables provide information about the IP Addresses, ports, and protocols used to communicate with each server. The Table Entries also contain links to counters that are useful for troubleshooting.

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

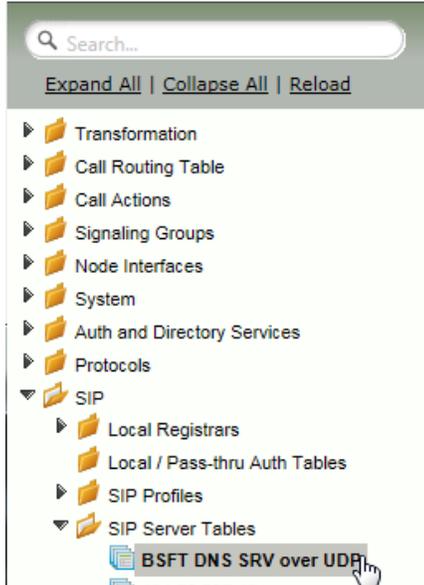


Add a Contact Registrant Table to hold the Broadsoft subscriber information Click the + to add a SIP Server Table:

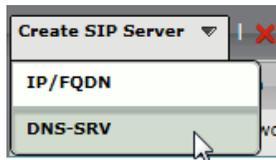
- Type of name of the Table
- Click OK



- 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
- Select the Contact Registrant Table
- Verify the Protocol

**BSFT DNS SRV over UDP**

Create SIP Server | X | Total 1 SIP Server Row

Host / Domain | Server Lookup

as.iop1.broadworks.n... | DNS SRV

**Server Host**

Server Lookup: DNS SRV

Domain Name / FQDN: as.iop1.broadworks.net \*

Service Name: sip \*

Protocol: UDP \*

**Transport**

Monitor: None

**Remote Authorization and Contacts**

Remote Authorization Table: None

Contact Registrant Table: Analog EPs for BSFT

Clear Remote Registration on Startup: False

Contact URI Randomizer: False

Stagger Registration: False

**\*\*NOTE:** You will need to repeat the steps above to create a SIP Server Table for each Enterprise-based SIP Server. Follow the template below for creating a single IP/FQDN SIP Server Entry in each SIP Server Table you create.

Enter the SIP Server information as noted below:

- Create an IP/FQDN **Enterprise** SIP Server.
- Enter the FQDN of the desired Enterprise SIP Server
- Enter the SIP Server's Port Number
- Enter the SIP Server's Protocol type
- Configure Monitor to *SIP Options*
- Click OK

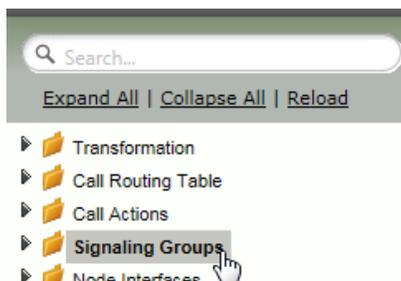
**Create SIP Server Entry** July 15, 2014 05:21:57

| Server Host  | Transport  |
|--|--|
| Row ID: 2<br>Server Lookup: IP/FQDN<br>Priority: 1<br>Host: exchange.contoso.com * FQDN or IP<br>Port: 5060 * [1024..65535]<br>Protocol: TCP * | Monitor: SIP Options<br>Keep Alive Frequency: 30 * secs [30..300]<br>Recover Frequency: 5 * secs [5..300]<br>Local Username: Anonymous * Local<br><small>Username of Sonus SBC</small><br>Peer Username: Anonymous * Peer<br><small>Username of sip server</small> |
| Remote Authorization and Contacts  | Connection Reuse   |
| Remote Authorization Table: None<br>Contact Registrant Table: None   | Reuse: True<br>Sockets: 4<br>Reuse Timeout: Forever  |

### 5.3.2 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 Media Information to *RTP Proxy Mode: Enable, RTP DSP Mode: Disable*
- Add the Broadsoft Server FQDN in the *Federated IP* with a netmask of 255.255.255.255

**SIP Signaling Group Details: BSFT Connection** Jul

Description **BSFT Connection**  
Admin State **Enabled**  
Service Status **Unknown 0**

| SIP Channels and Routing      |                                  |
|-------------------------------|----------------------------------|
| Action Set Table              | None                             |
| Call Routing Table            | From SIP                         |
| No. of Channels               | 10                               |
| SIP Profile                   | Default SIP Profile              |
| SIP Mode                      | Basic Call                       |
| SIP Server Table              | BSFT DNS SRV over UDP            |
| 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                              |

| Media Information |          |
|-------------------|----------|
| RTP Proxy Mode    | Enabled  |
| RTP DSP Mode      | 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   |

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

| Federated IP/FQDN            |                 |
|------------------------------|-----------------|
| Total 1 SIP Federated IP Row |                 |
| IP/FQDN                      | Netmask         |
| as.iop1.broadworks.n...      | 255.255.255.255 |

Message Manipulation **Disabled**

**\*\*NOTE:** You will need to repeat the steps above to create an ISDN Signaling Group for the TDM PBX. Use the diagram below to create an ISDN Signaling Group. Configure the Port and Protocol parameters to match your PBX.

Create ISDN Signaling Group
July 16, 2014 12:08:49

Description

Admin State

**Channels and Routing**

Channel Hunting

Direction

Tone Table   
Ringback \*

Action Set Table

Call Routing Table  \*

No Channel Available Override

Play Inband Message post-disconnect

Call Setup Response Timer  [180..750] secs

**Port and Protocol**

Port Name

Fractional

Switch Variant

ISDN Side

Play Ringback

Overlap Receive Mode

Overlap Send Mode

Stop Far-End T310

Indicated Channel

---

**Switch Specific Parameters**

Add Progress Indicator To Setup

Send Facility Message Passthrough

ASN.1 Protocol Identifier

ASN.1 Numbering Space

Include NFE and I-APDU

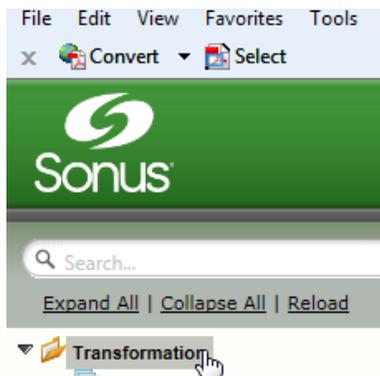
**Timeout/Timer Settings**

|      |   |               |
|------|---|---------------|
| T301 | <input style="width: 40px;" type="text" value="180"/> | [1..255] secs |
| T302 | <input style="width: 40px;" type="text" value="15"/>  | [1..255] secs |
| T303 | <input style="width: 40px;" type="text" value="4"/>   | [1..255] secs |
| T305 | <input style="width: 40px;" type="text" value="30"/>  | [1..255] secs |
| T308 | <input style="width: 40px;" type="text" value="4"/>   | [1..255] secs |
| T309 | <input style="width: 40px;" type="text" value="6"/>   | [1..255] secs |
| T310 | <input style="width: 40px;" type="text" value="10"/>  | [1..255] secs |
| T313 | <input style="width: 40px;" type="text" value="4"/>   | [1..255] secs |
| T314 | <input style="width: 40px;" type="text" value="4"/>   | [1..255] secs |
| T316 | <input style="width: 40px;" type="text" value="120"/> | [1..255] secs |
| T322 | <input style="width: 40px;" type="text" value="4"/>   | [1..255] secs |

### 5.3.3 Configure a Transformation Table to the Broadsoft Server

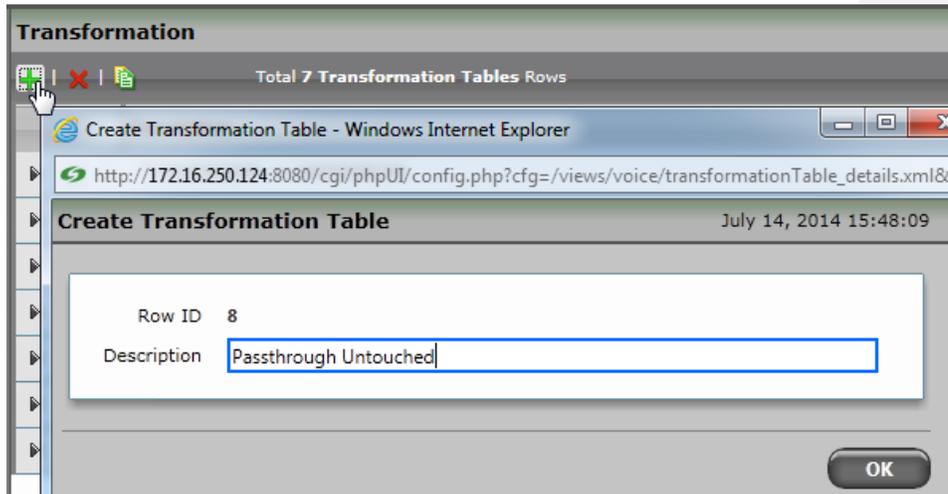
Transformation Tables facilitate the conversion of names, numbers and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number, or into a SIP address (URI). Every Call Routing Table Entry requires a Transformation Table.

- 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"/>          | Called Address/Number | (.*)              | Called Address/Number | \1                 | Mandatory  |  |
| <input 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.

**Create ISDN Signaling Group** July 16, 2014 12:08:49

Description

Admin State

---

**Channels and Routing**

Channel Hunting

Direction

Tone Table   
*Ringback \**

Action Set Table

Call Routing Table  \*

No Channel Available Override

Play Inband Message post-disconnect

Call Setup Response Timer  [180..750] secs

**Port and Protocol**

Port Name

Fractional

Switch Variant

ISDN Side

Play Ringback

Overlap Receive Mode

Overlap Send Mode

Stop Far-End T310

Indicated Channel

---

**Switch Specific Parameters**

Add Progress Indicator To Setup

Send Facility Message Passthrough

ASN.1 Protocol Identifier

ASN.1 Numbering Space

Include NFE and I-APDU

---

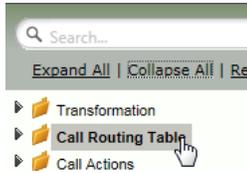
**Timeout/Timer Settings**

|      |                                  |               |
|------|----------------------------------|---------------|
| T301 | <input type="text" value="180"/> | [1..255] secs |
| T302 | <input type="text" value="15"/>  | [1..255] secs |
| T303 | <input type="text" value="4"/>   | [1..255] secs |
| T305 | <input type="text" value="30"/>  | [1..255] secs |
| T308 | <input type="text" value="4"/>   | [1..255] secs |
| T309 | <input type="text" value="6"/>   | [1..255] secs |
| T310 | <input type="text" value="10"/>  | [1..255] secs |
| T313 | <input type="text" value="4"/>   | [1..255] secs |
| T314 | <input type="text" value="4"/>   | [1..255] secs |
| T316 | <input type="text" value="120"/> | [1..255] secs |
| T322 | <input type="text" value="4"/>   | [1..255] secs |

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

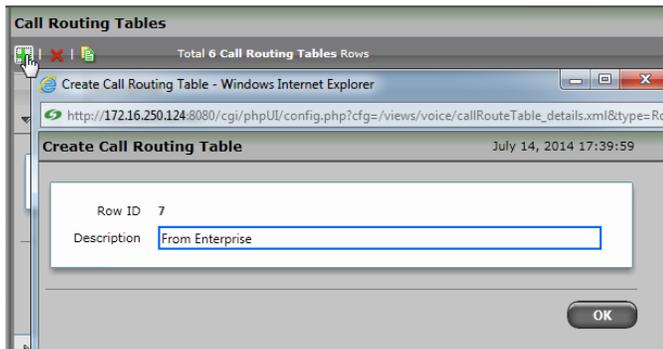
Call Routing allows calls to be carried between signalling groups, thus allowing calls to be carried between ports and between protocols (like ISDN to SIP). Call Routes are grouped into Call Routing Tables.

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

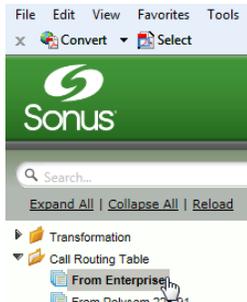


Create a new Call Routing Table. This call routing will take call from the Enterprise and route them to the Broadsoft server:

- 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 Mode* to RTP Proxy
- Click OK

**\*\*NOTE:** You will need to repeat the steps above to create a separate Call Routing Table called 'From Broadsoft' to process calls coming from Broadsoft to Enterprise-based SIP or TDM destinations. The Destination Signaling Groups in these call route entries must be configured for Enterprise-based destinations (Enterprise Signaling Groups).

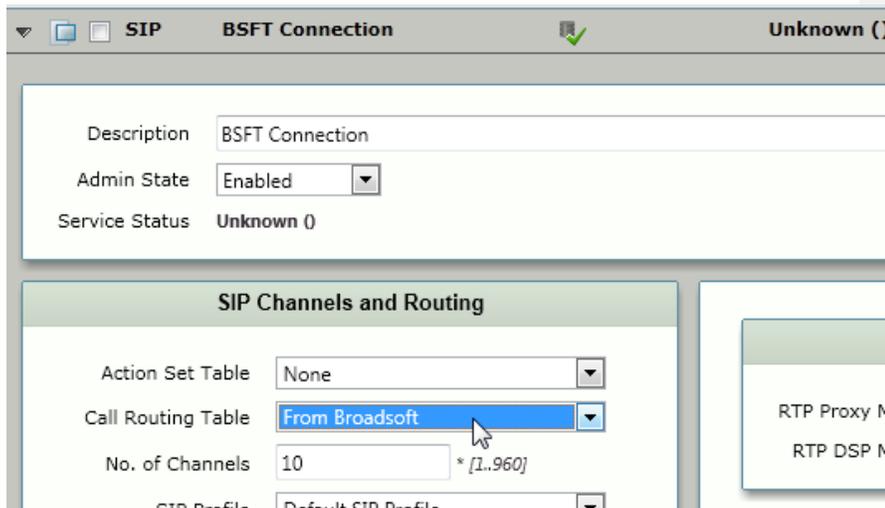
### 5.3.5 Set/Verify the Call Routing Table in the Ingress Signaling Group

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

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



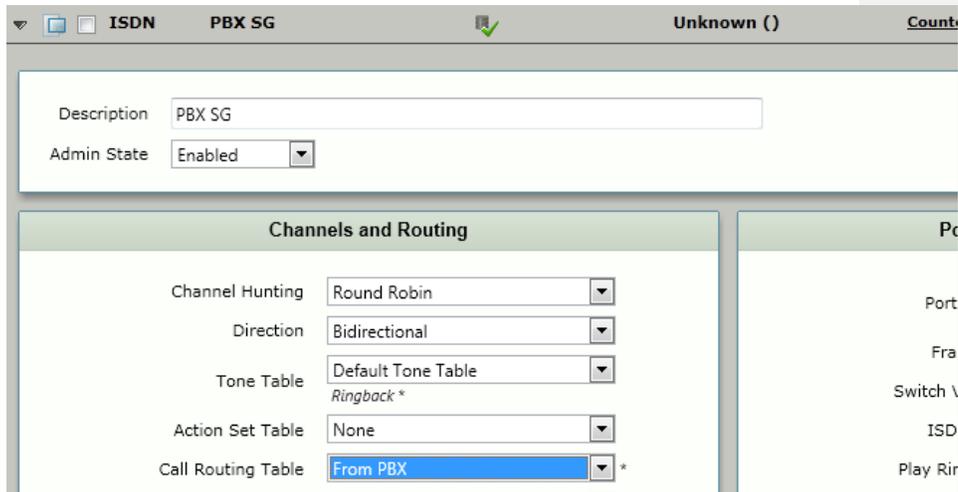
- The Broadsoft Signaling Group must be configured to use the FROM BROADSOFT Call Routing Table



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



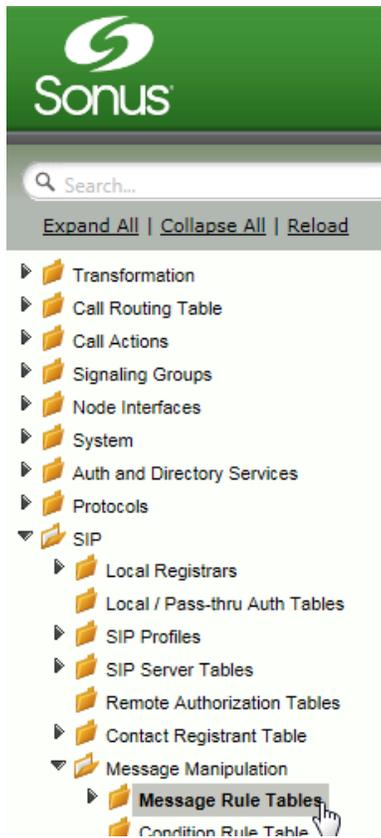
- The *Enterprise Exchange* Signaling Group must be configured to use the FROM PBX Call Routing Table



### 5.3.1 Create SIP Message Manipulation Rules

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 Transformation Table
- 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&

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

Row ID 2

Description GIN Registration

Applicable Messages Selected Messages

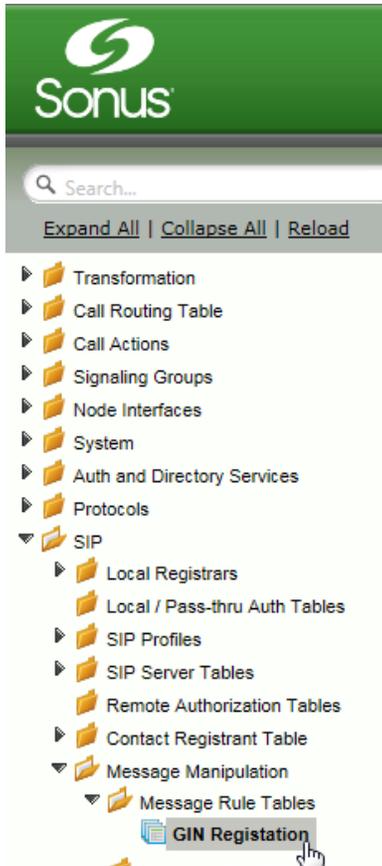
Message Selection Register

Add/Edit \*  
Remove

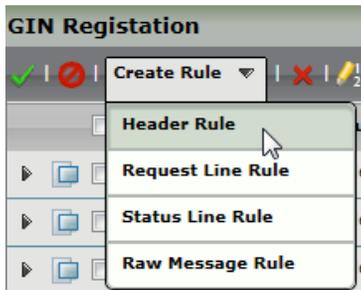
Table Result Type Mandatory

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 the following fields and controls:

- Description: Text box containing "Add Require header"
- Condition Expression: Text box containing "Add/Edit"
- Admin State: Dropdown menu set to "Enabled"
- Result Type: Dropdown menu set to "Optional"
- Header Action: Dropdown menu set to "Add"
- Header Name: Dropdown menu set to "Require"
- Header Value: Dropdown menu set to "Add" and a text box containing "Add/Edit"

Add the Header Value:

- Configure the information as shown
- Click OK

The screenshot shows a dialog box titled "Edit Message Field". It contains the following fields and controls:

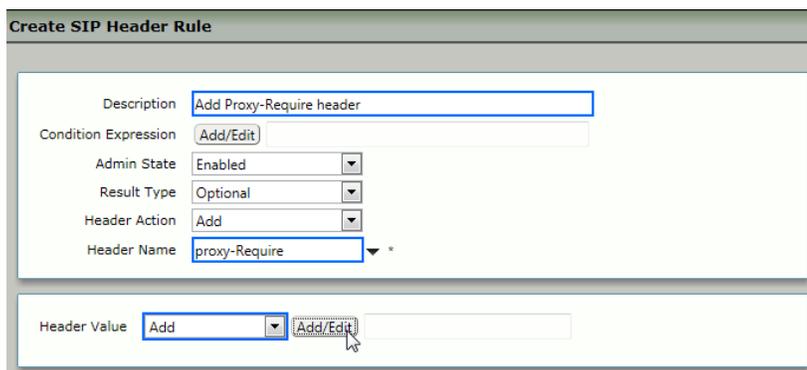
- Type of Value: Dropdown menu set to "Literal"
- Value: Text box containing "gin"
- Buttons: "OK" and "Cancel" buttons at the bottom right.

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



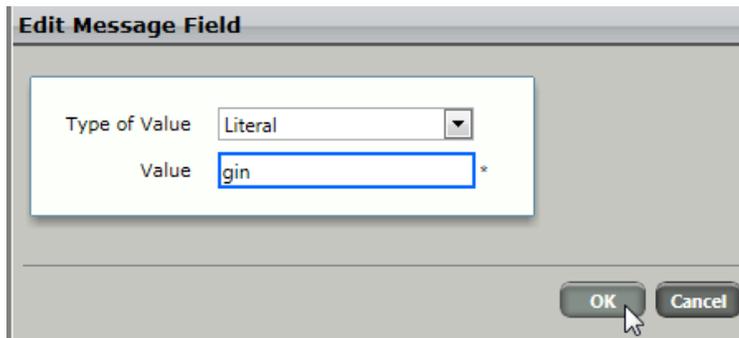
Create a new SMM Header Rule:

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

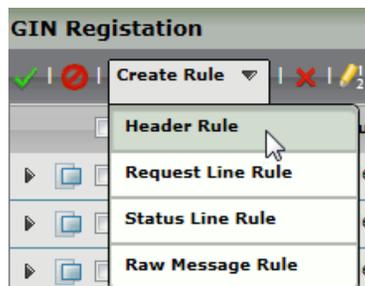


Add the Header Value:

- Configure the information as shown
- Click OK



- Create a third header rule. From the *Create Signaling Rule* 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*

**Comment [n1]:** Please move this from the From header to Contact header.

**Create SIP Header Rule**

Description: Add bnc parameter

Condition Expression: Add/Edit

Admin State: Enabled

Result Type: Optional

Header Action: Modify

Header Name: Contact

Header Ordinal Number: 1st

▼ Header Value

▼ URI

URI Scheme: Ignore

▶ URI User Info: Ignore

URI Host: Ignore

URI Port: Ignore

URI Parameters

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

Add the Header Value:

- Configure the information as shown
- Click OK

**Create Parameter**

Parameter Name  \*

Action

Type of Value

Value  \*

Prefix

Suffix

OK Cancel

Create a SMM to change calls from anonymous users to your Broadsoft Pilot Number.

- In the Navigation Tree, click the *Condition Rule Table*



Add a Condition Rule:

- Click the + to add an entry to the Condition Rule Table



Add a Condition Rule as noted below:

- Add the information as noted below
- Click Apply

The screenshot shows a configuration window with a 'Description' field containing 'Privacy:user;id;critical'. Below it is a 'Match Type' dialog box with the following fields:

|                  |                        |   |
|------------------|------------------------|---|
| Match Type       | from.uri.userinfo.user | * |
| Operation        | Equals                 |   |
| Match Value Type | Literal                |   |
| Match Value      | anonymous              | * |

- In the Navigation Tree, click the *Message Rule Tables*



Create a new SMM Rule Table:

- Click the + to add a Message Rule

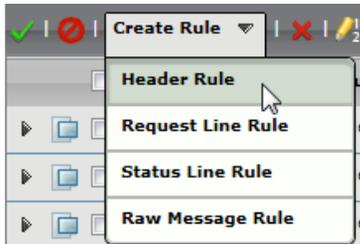


- Type the desired name of the Table and enter the information as shown
- Click OK

- In the Navigation Tree, click the newly created SMM Rule.



- Create a header rule to modify the P-Asserted-Identity header



Create a new SMM Header Rule:

- Type the desired name of the Table and set the configuration as shown

Header Rule    Optional    modify P-Asserted-identity (Sonus-TrkUser2)

Description: modify P-Asserted-identity (Sonus-TrkUser2)

Condition Expression: Add/Edit: \${2}

Admin State: Enabled

Result Type: Optional

Header Action: Modify

Header Name: P-Asserted-Identity \*

Header Ordinal Number: All

---

▼ Header Value

Display Name: Modify Add/Edit: 'Sonus-TrkUser2 Sonus-TrkUser2'

▼ URI

URI Scheme: Ignore

▶ URI User Info: Modify Add/Edit: '2404985622'

URI Host: Ignore

URI Port: Ignore

URI Parameters

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

- Click Condition Expression *Add/Edit*

Description: modify P-Asserted-identity (Sonus-TrkUser2)  
 Condition Expression: Add/Edit \${2}  
 Admin State: Enabled

- Set the condition as noted, click Apply

**Message Rule Condition**  
 Match All Conditions  
 Privacy:user;id:critical  
 Apply Cancel

- Click Display Name *Add/Edit*

Header Value  
 Display Name: Modify Add/Edit 'Sonus-TrkUser2 Sonus-TrkUser2'  
 URI  
 Click to add value for field: Display Name

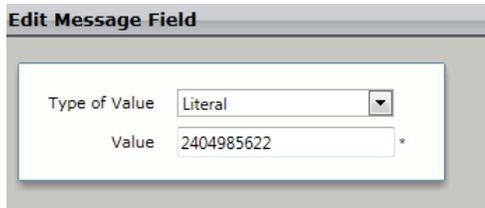
- Set the Display Name as noted, click Apply. The Trunk Identification will be supplied by the carrier.

**Edit Message Field**  
 Type of Value: Literal  
 Value: Sonus-TrkUser2+Sonus-TrkU \*

- Click URI User Info *Add/Edit*

URI Scheme: Ignore  
 URI User Info: Modify Add/Edit '2404985622'  
 URI Host: Ignore  
 Click to add value for field: URI User Info

- Set the URI User Info as noted, click Apply. Insert a valid Broadsoft number..



**Edit Message Field**

Type of Value  ▼

Value  \*

- Click Apply when finished entering the SMM Rule.



Create a SMM to change calls to add a Privacy header for calls to the Broadsoft server.

- In the Navigation Tree, click the *Message Rule Tables*



Create a new SMM Rule Table:

- Click the + to add a Message Rule



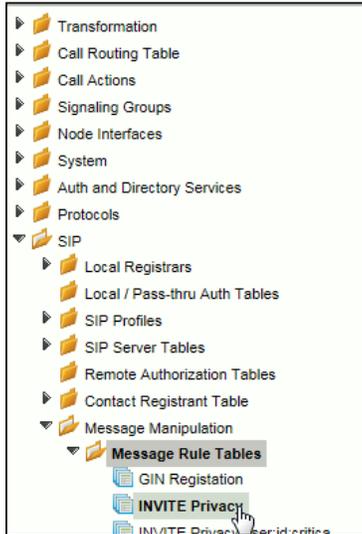
- Type the desired name of the Table and enter the information as shown and click *Apply*

The screenshot shows the 'SIP Message Rule Table' configuration form. The form has the following fields and values:

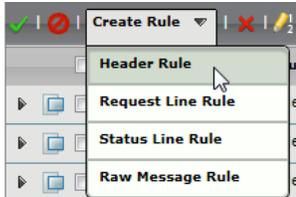
- Description:** INVITE Privacy
- Applicable Messages:** Selected Messages (dropdown menu)
- Message Selection:** Invite (list box)
- Table Result Type:** Optional (dropdown menu)

There are also 'Add/Edit' and 'Remove' buttons next to the 'Message Selection' list box.

- In the Navigation Tree, click the newly created SMM Rule.



- Create a header rule to add the Privacy header



Create a new SMM Header Rule:

- Type the desired name of the Table and set the configuration as shown

A screenshot of a configuration form for a SMM Header Rule. The form has the following fields and values:
 

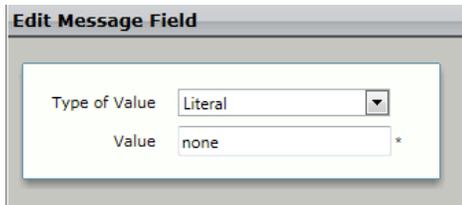
- Description: none
- Condition Expression: Add/Edit
- Admin State: Enabled
- Result Type: Optional
- Header Action: Add
- Header Name: Privacy
- Header Value: Add, Add/Edit, 'none'

- Click Header Value *Add/Edit*



A screenshot of a configuration interface. On the left, the text 'Header Value' is followed by a dropdown menu currently set to 'Add'. To the right of the dropdown is a button labeled 'Add/Edit'. Further right is a text input field containing the value ''none''. A mouse cursor is pointing at the 'Add/Edit' button, and a tooltip box below it contains the text 'Click to add value for field: Header Value'.

- Set the value to *none* and click OK.



A screenshot of a dialog box titled 'Edit Message Field'. Inside the dialog, there are two rows of controls. The first row is labeled 'Type of Value' and has a dropdown menu set to 'Literal'. The second row is labeled 'Value' and has a text input field containing the text 'none'. A small asterisk is visible to the right of the 'Value' input field.

- Click Apply when finished entering the SMM Rule.



### 5.3.2 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

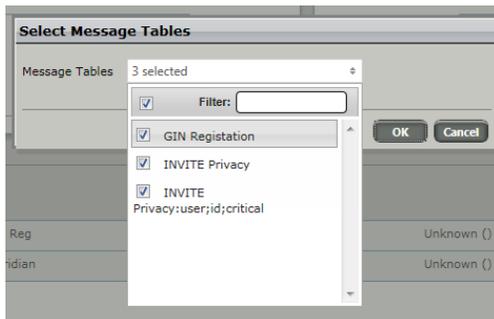


Enable the new SIP Message Manipulation (SMM) Rule:

- 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*



## 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 Device Management Configuration Guide, Release 20.0*. Available from BroadSoft at [xchange.broadsoft.com](http://xchange.broadsoft.com).
- [3] BroadSoft, Inc. 2013. *BroadWorks Redundancy Guide, Release 20.0*. Available from BroadSoft at [xchange.broadsoft.com](http://xchange.broadsoft.com).
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- [7] BroadSoft, Inc. 2014. *BroadWorks IP-PBX/PBX Trunking Interoperability Test Plan, Release 20.0*. Available from BroadSoft at [xchange.broadsoft.com](http://xchange.broadsoft.com).