

Wave SIP Trunk Configuration Guide

FOR BROADVOX

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Overview

This guide describes how to configure SIP trunks on the Wave Server when you are using SIP trunking from Broadvox LLC.

Throughout this guide, "your ITSP" refers specifically and only to Broadvox.

SIP trunk configuration consists of the following tasks:

- Enable SIP trunking on the Wave Server.
- Create a new signaling control point (SCP) for your ITSP.
- Configure bandwidth management zones.
- Configure outbound routing for SIP calls.
- Allocate VoIP resources.

Optional configuration tasks include:

- Set up emergency 911 service.
- Configure a backup proxy server to maintain SIP trunk service if the primary SIP proxy server fails, your ITSP has provided you with backup proxy server configuration information.

Special Notes

Wave Software Version 4.0

- Does not support RTCP.
- Codec support: The system is configured for G.729 and G.711 codecs, with G.729 being the top Priority codec. However reINVITEs from Wave contain G711 as the top priority codec. Issue resolve when hotfixes No 18, 22 & 23 are applied.
- Diversion header is not supported (in Call forwarding scenarios).
- Off net blind transfer No Ringback tone to external caller (Ref JIRA TM-10/12)
- Fax (G711 and T.38): This configuration does NOT support fax, for workarounds use Analog/T1/ISDN trunks or see Wave Manuals for more information.
- Backup proxy server configuration not supported (page 12/20) (Ref JIRA TM-125), for workaround setup backup SCP & backup secondary outbound routing table.

Before you begin

Required SIP trunk provisioning and configuration information

Before you begin, make sure that you have obtained the following information from your ITSP:

• Proxy server information:

Registrar Server:	nd01-03.fs.broadvox.net
Proxy Server:	nd01-03.fs.broadvox.net

Outbound proxy1 _____

Outbound proxy2 _____

• Pilot Identity Bulk Registration Credentials:

Username _____

Password _____

- SIP Domain Name:
- SIP codec to use:
- DID phone numbers:
- Technical support contact information

Wave Server requirements

- Wave ISM version: Verify that the Wave Server is running one of following versions :
 - Wave 4.0 base version
 - Wave 3.0 + Feature Pack 1 or higher
 - Wave ISM 2.5 + HotFix 16 or higher
 - Wave ISM 2.0 Service Pack 1 + HotFix 24 or higher
- **Wave licensing:** Add the following Wave licenses, in addition to any other Wave licenses required to support your configuration:
 - Wave IP User license. Depending on the IP phones that you use, you will need one or more of the following types of licenses:
 - Wave IP User Edge IP and ViewPoint Phone license: For Edge 5000-series IP phones or the ViewPoint Softphone.
 - Wave IP User Certified Third Party IP Phone license: For supported Aastra or Edge 1500-series IP phones
 - Wave IP User Generic Third Party IP Phone: For third-party IP phones.
 - Wave SIP Trunk license

• Wave IP Gateway license

 Registry: Add setting the ReinviteAfterUpdate Registry setting to 1 HKEY_LOCAL_MACHINE\SOFTWARE\Vertical Networks\InstantOffice\IpTelephony\Sip\SCP\scpname



Router requirements

You can use any NAT router with a DMZ option as a default gateway for Wave and all IP phones. Wave registers with your ITSP and handles all inbound and outbound calls over the SIP trunks.



The following information assumes that you are using a Linksys RV042 router. If you are using a different router, equivalent configuration settings should be available.

•	Network settings:	
	LAN IP:	IP address ON of router (LAN)
	WAN IP:	IP address ON of router WAN I/f
	DMZ IP:	0.0.0.0
	Mode:	Gateway
	DNS:	Provided by your ISP or ITSP
	DDNS:	OFF
	DMZ Host:	Wave IP address
•	Firewall settings:	
	SPI (Stateful Packet Inspection):	ON
	DoS (Denial of Service):	ON
	Block WAN Request:	ON
•	VPN settings:	
	Tunnel(s) Used:	0
	Tunnel(s) Available:	50
	No VPN Group is defined.	

Configuring SIP trunks for Broadvox

The steps and screenshots in this guide reflect the configuration process on a Wave 4.0 Server. Since the Wave user interface has been modified over time, if you are configuring SIP trunking on an earlier version of Wave, if necessary see Chapter 6 in the *Wave Global Administrator Guide* for that version for specific details.

Enabling SIP trunking on the Wave Server

- 1. Start the Wave Global Administrator Management Console. Click **IP Telephony**, located in the PBX Administration section.
- 2. Expand **Signaling Protocols** in the left pane and then click **SIP**. Select the **SIP Enabled** checkbox. **SIP Local IP Address** is selected automatically.

IP Telephony	
Signaling Protocols SIP Advanced Codec Settings Advanced Codec Settings Quality of Service (QOS) Call Routing Call Routing Signaling Control Points Canes Some	SIP Enabled SIP Local IP Address: Advanced
	Restore Apply Done Help

Creating a new signaling control point (SCP)

This section describes how to create a new SCP and then:

- Set up inbound routing.
- Add a rule to the inbound routing table.
- Set up Caller ID.
- Configure SIP settings.

Creating a new SCP and setting up inbound routing

1. Expand Call Routing in the left pane and choose Signaling Control Points.

Generating Protocols	Route Step Timeout: 20 seconds
SIP	Name
Advanced Codec Settings OTMF Transport Settings Quality of Service (QOS) IP Telephone Settings Call Routing Offault Inbound Routing Signaling Control Points Sones	AccessLine2WG Bandwidth.com Voice4Net IVSG CenturyLink Verizon
	Restore Apply Done Help

2. Click New.

3. In the Signaling Control Point dialog, enter a **Name** for the new SCP, for example "Broadvox". (When you configure outbound call routing, this name will appear in the Routing Table as "IP", a vertical bar (|), and the name you enter here, for example "IP|Broadvox".)

You can enter alphanumeric characters as well as the following special characters:

Signaling Control Point			
ame: Broadvox			
Inbound Routing Outbound	Routing SIP Settings		
Edit Inbound Routing Tabl	e		
ntercept Destination:	None		-
Access Profile for Tandem Ca	alls: None		-
	•		
		ОК	Cancel

~!#\$%&*	()-=+ {	[/ < > ?
----------	---------	---	---------

Setting up inbound routing for the new SCP

- 1. On the Inbound Routing tab, for **Intercept Destination**, select the extension from the drop-down list to which any incoming call from this SCP that is not matched in the Inbound Routing Table will be directed.
- 2. For Access Profile for Tandem Calls, select the access profile to apply to calls received from this SCP that will be connecting to another trunk. (Access profiles identify the different calling privileges that can be associated with SCPs, and can also be associated with extensions, trunk groups, and digital connections. For more information, see "Configuring specific access profiles" in Chapter 9 in the *Wave Global Administrator Guide*.)

If you are not sure which access profile to choose, you can select the "Unrestricted" profile.

Warning! If you have not modified the default "Unrestricted" access profile, selecting "Unrestricted" here could leave your system vulnerable to hackers who are able to identify your Tandem Access Profile number.

🍝 Signaling Control Point		×
Name: Broadvox		
Inbound Routing Outbound Rout	ing SIP Settings	
Edit Inbound Routing Table		
Intercept Destination:	5000 - Vertical Communications	•
Access Profile for Tandem Calls:	Local and Toll Free Calls	•
		OK Cancel

Adding a rule to the inbound routing table for the new SCP

- 1. Still on the Inbound Routing tab, click Edit Inbound Routing Table.
- In the Inbound Routing Table dialog, verify that Route By Source or Dialed Number is selected. (This setting lets you decide how calls from this SCP get routed based on the DID digits the caller dialed.)

	i <u>Kodi</u> c	s by Source or Dialec		ieddied Roading 👈	bour	
Call Source	Dialed Number	Destination	DNIS Name	Night Answer Mode	Night Answer Destination	Up
						Dov
			1			

- 3. Click Add to add a new rule to the table for the DID numbers provided to you by your ITSP.
- 4. For each DID number, double-click in the following columns in the new rule:
 - For **Dialed Number**, enter one of the DID numbers provided by your ITSP. You can enter the entire 10-digit number, or use "x" characters as wildcards.
 - For **Destination**, enter the extension or external phone number to which calls with this DID number will be routed. This number is interpreted as if dialed from an internal station, so for an external number, be sure to enter the external access digit as defined in the First Digit Table.

Call	Dialed	Destination	DNIS	Night Answer	Night Answer	Up
Default Default	xxxxxx0550 xxxxx0551	560 03771	Harris	Not Used Not Used	Destination	Dowr
		[] [

5. Click **OK** to save your changes to the Inbound Routing table.

Setting up Caller ID for the new SCP

- 1. Still in the Signaling Control Point dialog, select the Outbound Routing tab.
- Choose one of the Caller ID formats to send on outbound calls to this SCP. (Note that your ITSP may not support all of the listed formats.)

🕌 Signaling Control Point	×
Name: Broadvox	
Inbound Routing Outbound Routing SIP Settings	
C Use External Caller ID from User Configuration	
Send Company Name and Main Number	
C Send Station Name and Internal Extension Number	
© Send Station Name and this Number:	
🗖 plus last 3 🗾 digits of calling extension number	
C Do Not Send Caller ID	
ОК	Cancel

Configuring SIP settings for the new SCP

1. Still in the Signaling Control Point dialog, select the SIP Settings tab.

差 Signaling Control Point	×
Name: Broadvox	
Inbound Routing Outbound Routing	SIP Settings
User Name:	
Proxy Server:	
Port: 5060	
Inbound/Outbound Settings	
SCP is located outside of Wave's i	network
This SCP will:	C Receive registration from Contact
	Register with a Proxy/Registrar
Authentication Settings	Registration Settings
C Authentication Required	Registration Required
Authentication Name:	Registrar Server:
Password:	Registrar Port;
Verify Password:	Registration Expires (secs)
	C Use System Default
	C Custom 300
Preferred DTMF Transport SIP INFO	v
Advanced Settings	
	OK Cancel

- 2. Enter the following information:
 - For **User Name**, enter the main telephone number (also known as "Pilot Identity") provided by your ITSP, for example "4084180145".
 - For **Proxy Server**, enter the DNS name or IP address of the proxy server provided by your ITSP.
 - For **Port**, leave the default port number, "5060".

- 3. In the Inbound/Outbound Settings section:
 - Select the SCP is located outside of Wave's network checkbox.
 - Select Register with a Proxy/Registrar.
- 4. In the Authentication Settings section:
 - Select the Authentication Required checkbox.
 - Enter the Authentication Name and Password provided by your ITSP.
- 5. In the Registration Settings section:
 - Select the Registration Required checkbox.
 - Enter the **Registrar Server** DNS name or IP address provided by your ITSP. (Typically, this is the same as the **Proxy Server** address that you entered above.)
 - For **Registrar Port** number, enter "5060".

🛓 Signaling Control Poir	nt		×
Name: Broadvox			-
, Inbound Routing Outboun	d Routing SIP Settings		Í.
User Name:	4084180145		
Proxy Server:	nd01-03.fs.broadvox.net		
Port:	5060		
Inbound/Outbound Setting	ļs —		
SCP is located outside	of Wave's network		
This SCP will:	C Rece	eive registration from Contact	
Register with a Proxy/Registrar			
-Authentication Settings		Registration Settings	
Authentication Requ	ired	Registration Required	
Authentication Name:	4084180145	Registrar Server: nd01-03.fs.broadvox.net	
Password:	*******	Registrar Port: 5060	
Verify Password:	****	Registration Expires (secs)	
	,	• Use System Default	
		C Custom 300	
Preferred DTMF Transport	Inband 💌		
Advanced Settings			
		OK Cancel	

- 6. Click Advanced Settings.
- 7. If your ITSP has provided you with backup proxy server configuration information, do the following:
 - Select the Enable Outbound Proxy checkbox, and then enter the Outbound Proxy Server and Outbound Proxy Server Port (not supported Ref. JIRA TM-125).
 - Select the **Monitor SIP Trunks** checkbox, and then specify a **Keep Alive Timer** and **Recovery Timer** value in seconds.

For more about using a backup proxy server, including an additional required configuration step, see "Configuring a backup proxy server" on page 20.

- 8. In the SIP Trunk Transfer Options section, deselect both of the following:
 - Attempt Hairpin Elimination on Supervised Transfer
 - Attempt Hairpin Elimination on Blind Transfer

🕹 Advanced Settings					X
Enable Outbound Pr	oxyl				
Outbound Proxy S	ettings-				
Outbound Proxy Se	erver:				
Outbound Proxy Po	ort; 0				
Local Listen Port					
Oefault					
C Custom:					
Include UUI Data in	SIP Messages				
Monitor SIP Trunks					
SIP OPTION Messa	age Settings				
Keep Alive Timer (s	econds): 0				
Recovery Timer (se	econds): 180				
SIP URI To Wave Map	ping				
Called Party Source:	Request URI				-
Calling Party Source:	From URI				-
SCP User Name Source	Contact URI				-
Wave To SIP URI Map	ping				
To Source:	Called Party				-
From Source:	SCP User Name		💌 @ Proxy		-
Contact Source:	SCP User Name				T
P-Asserted-ID Source:	Calling Party		💌 @ Wave IP		v
SIP Trunk Transfer Op	tions [SIP REFER / REPLACES]				
🥅 Attempt Hairpin Eli	mination on Supervised Transfer				
🔲 Attempt Hairpin Eli	mination on Blind Transfer				
Propagate CutThrough	1				
Propagate CutThro	ough call progress messages on tar	ndem SIP trunk calls			
					1
				ОК	Cancel

- 9. Leave all other advanced settings at their default values, unless instructed otherwise by your Wave provider. Click **OK** to save your changes, and exit the Signaling Control Point dialog.
- 10. Back on the main IP Telephony screen, select the new SCP and change the Route Step Timeout (at the top of the screen) to 20 seconds. This setting adjusts the amount of time that the system waits on this SCP before trying the next step in the outbound routing table, to allow for network or other delays.

IP Telephony	
P Telephony	Route Step Timeout: 20 seconds Name AccessLine2WG AccessLine2WG Bandwidth.com Broadvox Voice4Net IVSG CenturyLink Verizon
	Edit New Delete Restore Apply Done Help

11. Click **Apply** to save your changes. Do not exit IP Telephony yet.

Configuring bandwidth management zones

Configuring the Home zone

- 1. Expand Bandwidth Management in the left pane and click Zones.
- 2. In the Zone Name list, select "Home" and then click Edit.
- 3. In the Bandwidth Management dialog, select the IP Address Ranges tab.
- 4. Review the IP Address Range for the Home zone and make corrections if necessary. Leave all other values on this tab unchanged---these are expert settings that should not be modified unless you are instructed to do so by your Wave technical support representative.

Bandwidth Management	<
Name: Home	
IP Address Ranges Bandwidth Settings Inter-Zone Codecs Intra-Zone Codecs	
IP Address Range	
172.55.0.0 - 172.155.255.255	
Edit New Delete	
OK	

- 5. On the Inter-Zone Codecs tab, do the following:
 - Make sure that the VoIP codec to use with your ITSP is in the Step 1 position. To change a codec's position in the list, select it and then click **Up** or **Down**.
 - Optionally, for each codec select the **Silence Suppression** checkbox.

Ba	andwidth Man Jame: Home	agement			×
	, IP Address Ran	ges Bandwidth Settings	Inter-Zone Codecs	Intra-Zone Codecs	
	Step	Audio Codec	Packet Time (ms)	Silence Suppression	Up
	1 2	G.729AB G.711 Mu-Law 64 kbps	20 20	2	Down
		Add	Remove		
				ОК	Cancel

- 6. On the Intra-Zone Codecs tab, make the same changes that you did in the previous step.
- 7. Click **OK** to save your changes, but don't exit IP Telephony yet.

Configuring the Remote Default zone

All IP addresses outside of the Home zone are automatically in the Remote Default zone.

- 1. In the Zone Name list, select "Remote Default" and then click Edit.
- 2. On the Inter-Zone Codecs tab, do the following:
 - Make sure that the VoIP codec to use with your ITSP is in the Step 1 position. To change a codec's position in the list, select it and then click **Up** or **Down**.
 - Optionally, for each codec select the **Silence Suppression** checkbox.
- 3. On the Intra-Zone Codecs tab, make the same changes that you did in the previous step.
- 4. Click **OK** to exit the Bandwidth Management dialog.
- 5. Click **Apply** to apply all your changes, and then click **Done** to exit IP Telephony.

Configuring outbound routing for SIP calls

- 1. In the Global Administrator Management Console, click **Outbound Routing** in the Trunk Administration section.
- 2. In the Outbound Routing dialog, select the "Unrestricted" access profile and then click Edit.

Outbound Routing	
Edit Global Access Profile	Edit Private Network
Access Profiles	
Status	Name
	A Fax Ports Internal Calls Only Local and Domestic Calls Local and Toll Free Calls Local, Domestic, and Intl. Calls Modems System Ports Unrestricted
Edit Copy	New Delete
F	kestore Apply Done Help

3. The Access Profile dialog opens. On the Area Code Table tab, click Add.

Access Profile	d ivileges Destination Acce	ss Codes
Sort By Rout	ing Edit	t Routing Table
Area Code	Office Code Range	Routing Table
408 Default	Default Default	AccessLine-Local AccessLine-SIP
	Add Remove	
		OK Cancel

A new entry is added at the bottom of the Area Code list:

Access Profile		
ame: Unrestricted	1	
Area Code Table Pri	ivileges Destination Acces	ss Codes
Sort By Rout	ing Edit	Routing Table
Area Code	Office Code Range	Routing Table
408 Default	Default Default	AccessLine-Local AccessLine-SIP
	Default	*Blocked*
	Add Remove	
		OK Cancel

- 4. Double-click in the following columns:
 - For Area Code, enter "Default".
 - Leave Office Code Range set to "Default".
 - For Routing Table, select "(New Routing Table)" from the drop-down list.
- 5. The Routing Table dialog opens. Enter a name for the new Routing Table entry, for example "Broadvox".

n Digits	Digits	Digits	Settings	
				Dow

- 6. Click **Add** to add a new route.
- 7. Click in the **Destination** column and select "IP|Broadvox" from the drop-down list.

Step	Strip First n Digits	Keep Last n Digits	Prepend Digits	Postpend Digits	Destination	ISDN Settings	Up
	0				IP Broadvox	N/A	Down

- 8. Leave all of the other default settings unchanged, and click **OK** to save the new route.
- 9. Click Apply to save your changes, and then click Done to exit Outbound Routing.

Allocating VoIP resources

- 1. In the Global Administrator Management Console, click **Resource Management**, located in the PBX Administration section.
- 2. Expand IP Telephony Resources / Voice Over IP Group in the left pane.
- Select the appropriate Low Bit Rate G.729A/G.711 or Standard Bit Rate G.711 resource, and then allocate the number of VoIP resources to use by selecting a value from the drop-down list. (You only need to allocate VoIP resources to *one* codec.)

source Management	
Conference Resources Application Resources Fax Resources Conference Resource	Standard Bit Rate (G.711) with QOS 20 -
Available Resources	·
Ports: 206 Power (MCPS): 2.0	Resource Management Advisor
	Restare Apply Done Help

4. Click **Apply** to apply your changes, and then click **Done** to exit Resource Management.

This completes the SIP trunk configuration process. Contact your Wave provider if you have any further questions.

Making a test call

To verify that your SIP trunks are configured correctly:

- Make an outbound call to an external number from a Wave phone, answer the call and verify that there is a two-way voice path.
- Make an inbound call to a DID number from outside the Wave network, for example from a cell phone. Answer the call and verify that there is a two-way voice path.

Setting up emergency 911 service

Consult with your ITSP's technical support representative for detailed guidance on how to configure emergency 911 service. See "Setting up emergency dialing" in Chapter 9 in the *Wave Global Administrator Guide* for more information.

Important!

- Ensure that all employees, visitors, and any other people who may attempt to make an emergency call using a SIP trunk are aware of an alternate method to use to access emergency services in the event that VoIP service fails.
- If your specific configuration includes centralized trunking or multiple Wave Servers at different locations, it is imperative that emergency 911 calls are routed to the local public safety answering point (PSAP) that serves a specific caller's location.

Configuring a backup proxy server (not supported Ref. JIRA TM-125)

This section applies to you if your ITSP has provided you with backup proxy server configuration information.

When you configure and enable a backup proxy server, if the primary SIP proxy server fails, Wave will automatically switch to the backup proxy server to maintain SIP trunk service. When the primary SIP proxy server becomes available again, Wave will switch back automatically.

Backup proxy server configuration consists of the following tasks:

- 1. Configure and enable required settings when you create the SCP for your ITSP, as described in step 7 on page 12.
- 2. Edit the registry and add the following STRING registry value for the SCP you created for your ITSP:

SOFTWARE\Vertical Networks\InstantOffice\IpTelephony\Sip\Scp\[SCP_NAME] \OutboundProxy2

For example:

SOFTWARE\Vertical Networks\InstantOffice\IpTelephony\Sip\Scp\Broadvox\OutboundProxy2

Set OutboundProxy2 to the fully-qualified domain name of the backup proxy server as provided by your ITSP.

Troubleshooting

Please see Vertical University course:



This advanced course will be available after you pass Wave advanced instructor lead training.

This course has 3 modules;

- Introduction to SIP
- SIP Operation
- Troubleshooting SIP

For assistance please contact Vertical Technical Support at 1-877-Vertical. Prior to contacting Tech Support, please have available Wave Trace files, a Wireshark packet capture of the failure and a network diagram.