SIP Trunking Configuration Guide
for
Vertical Wave IP500 Certification
ISM 5.0.0 (4370)

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1 Audience

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities.

2 Introduction

This Configuration Guide describes configuration steps for IntelePeer SIP Trunking to a Vertical Wave IP500 PBX.

2.1 tekVizion Labs

tekVizion Labs™ is an independent testing and Verification facility offered by tekVizion PVS, Inc. ("tekVizion"). tekVizion Labs offers several types of testing services including:

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing
- Verification Testing – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration
- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

tekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services helps service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our practice areas which include: SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

The tekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. TekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. TekVizion's headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in Plano, Texas.

For more information on tekVizion and its practice areas, please visit tekVizion Labs website at www.tekvizion.com

3 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of a Vertical Wave IP500 PBX configuration.

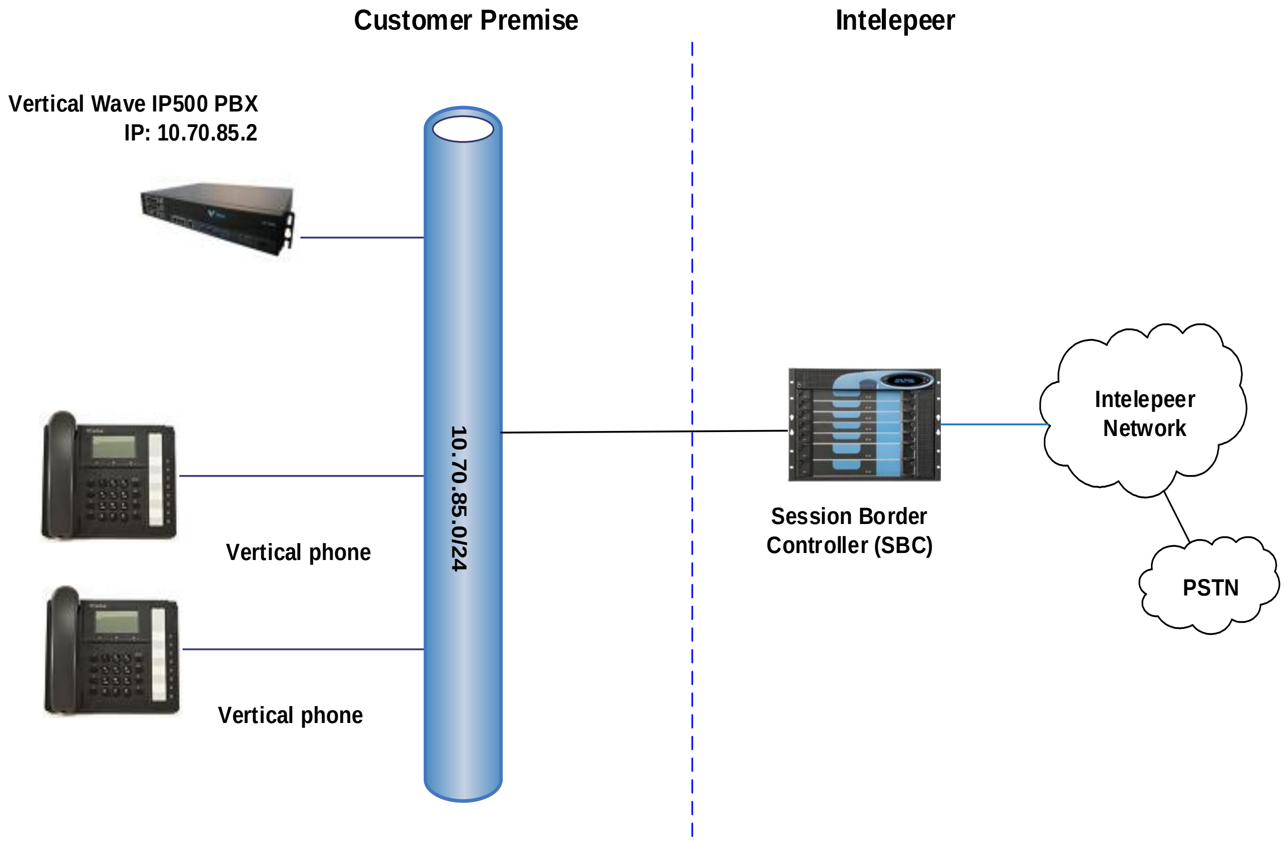


Figure 1: SIP Trunk Lab Reference Network

The lab network consists of the following components:

- Vertical Wave IP500 PBX for voice features, SIP proxy and SIP trunk termination
- Vertical IP and digital phones connected to the local LAN

3.1 Hardware Components

- Vertical Wave IP500
- Analog fax machine

3.2 Software Requirements

- Wave ISM 5.0.0 (4370)

4 Features

4.1 Features Supported

- Basic calls using G.711ulaw
- Anonymous call
- International Call
- Call Transfer
- Call Forwarding
- Call Hold and Resume
- Call Pickup
- Call Waiting
- DND
- Call Park
- Do Not Disturb
- Hunt groups (Simultaneous and Sequential Ring)
- Three-Way Calling
- PBX Auto Attendant to Off-net Numbers
- PBX Authorization Codes
- Casual dialing
- Fax Send
- Fax Receive
- Dial-Up Modem
- E911 Call
- 900 Call blocking
- 911 Call
- RFC2833 transcoding
- PBX-Defined Caller ID (spoofing)

4.2 Features Not Supported by PBX

- PBX Account Codes

4.3 Caveats and Limitations

- Intermittent no way speech path observed for an inbound calls. This issue is not indicative of Vertical Wave and Taqua standard behavior but is a Layer 3 issue. The intermittent audio issue is due to the nature of the lab used for testing by IntelePeer. IntelePeer has customers running this deployment load on Taqua in production without audio issues.
- Vertical Wave IP500 sends a re-INVITE with T.38 negotiation. As a result the fax transmission stops. Vertical support states that the ability to disable T.38 re-INVITE will be available in the next software release.
- Refer to Section 6 Summary of Tests and Results

5 Configuration

5.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure the Vertical Wave IP500 PBX for SIP Trunking as well as all features that were tested.

Table 1 – PBX Configuration Steps

Step	Description	Reference
Step 1	IP Network Settings	Section 6.3.1
Step 2	SIP carrier Options	Section 6.3.2
Step 3	Signaling Control Point Configuration	Section 6.3.3

5.2 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Table 2 – IP Addresses

Component	Lab Value	Customer Value
IntelePeer SIP Trunk		
• Signaling IP Address	68.68.123.XXX	
• Media IP Address	68.68.123.XXX	
• Protocol/Port	UDP/5060	
Vertical Wave IP500 PBX		
• System LAN IP Address The Internet Connection will typically be on the same subnet as the LAN IP Address of the E-SBC. If this is not the case, then Layer 3 routing must be in place.	10.70.85.2	
• LAN Default Gateway The Default Gateway must be the LAN Network default Gateway. This will allow the administrator to log in via his\her workstation if the workstation is on a different network	10.70.85.1	

5.3 Vertical Wave IP 500 Detailed Configuration Steps

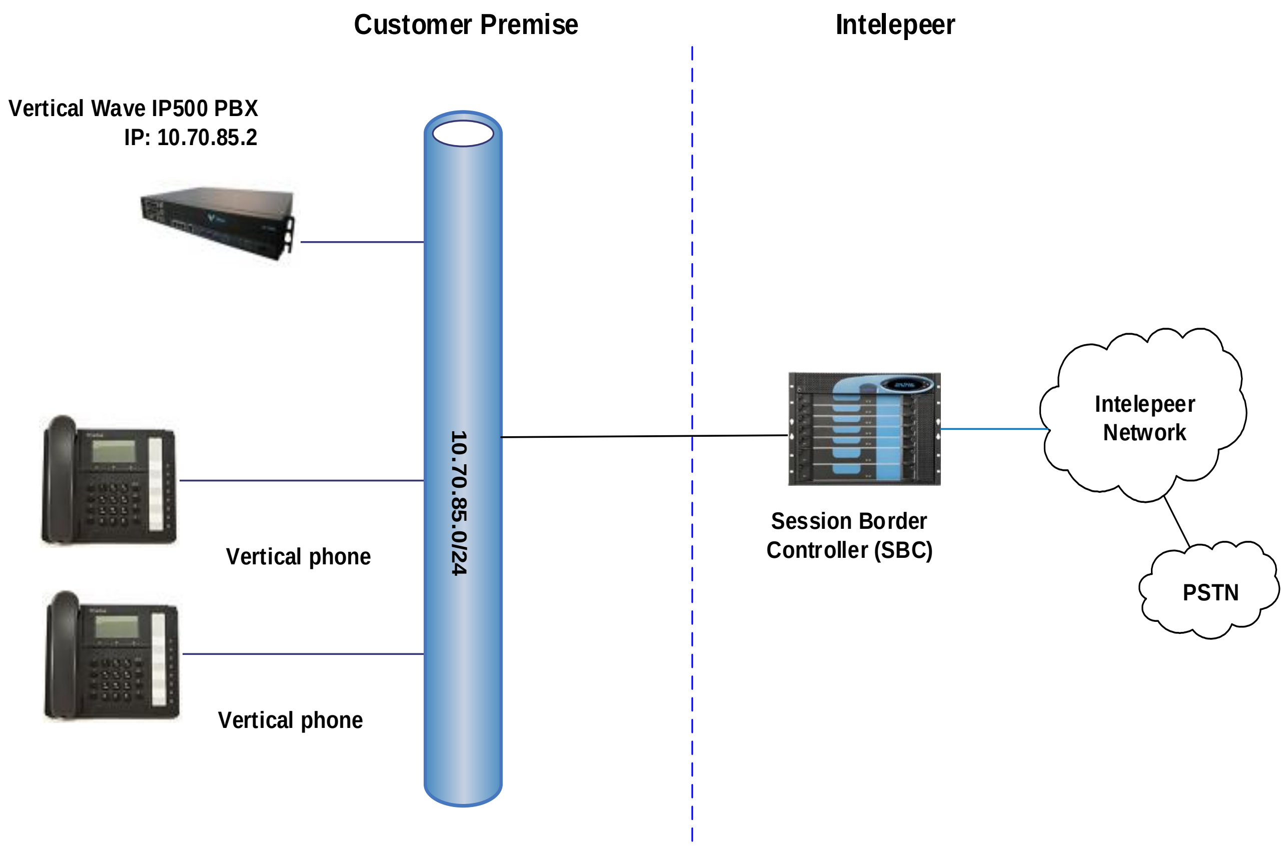
Equipment used for configuration setup:

- Vertical Wave IP 500
- Wave ISM 5.0.0 (4370)

![InelePeer logo]

5.3.1 IP Network Settings

1. Open a web browser and enter the URL of http://10.70.85.2/
2. When the dialog window ‘‘V Wave Global Administrator Log on“ page appears, log in using User name and Password
3. Navigate to **Data Administration > IP Network System**
4. Set **Host Name**: For this example io-default is used
5. Set **DNS Domain Name**: Wave.Local is used for this example
6. Set **Network interface**: Select *Vertical Wave Application Module* from drop down menu
7. Set **IP Address**: Enter the PBX IP address. Please use the actual PBX IP for your network. Please see



8. **Figure** 1: SIP Trunk Lab Reference Network
9. and **Table 2** for the IP address scheme used for this example.
10. Set the **Subnet mask** to (FFFFFFFF00) 255.255.255.0 Class C
11. Set **Default Gateway**: 10.70.85.1 as shown in **Figure 2**
12. Click **Apply**

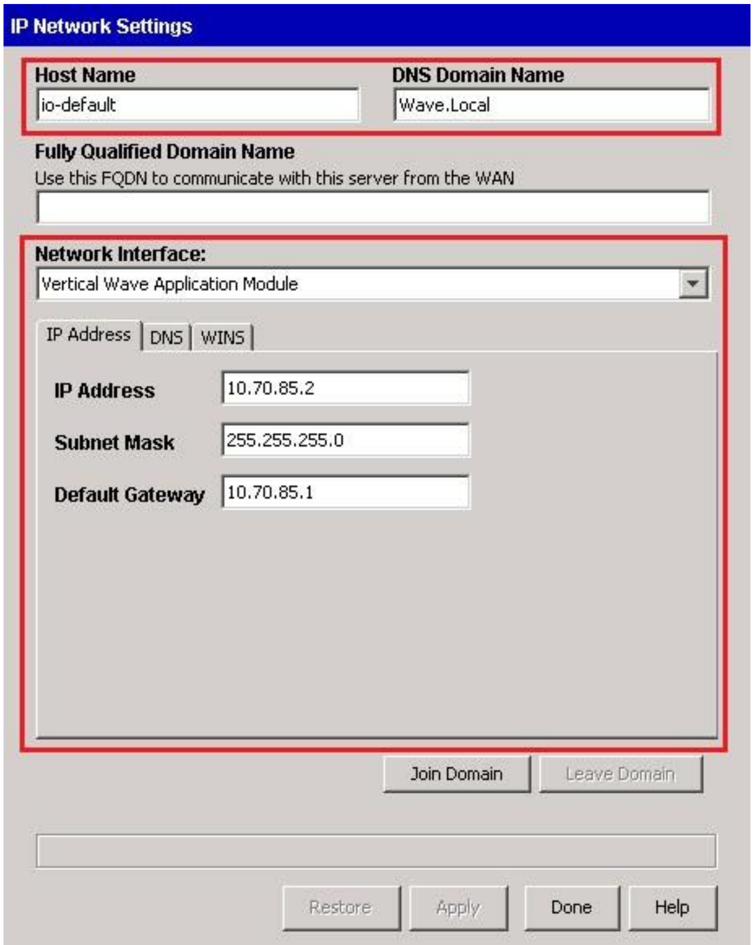


Figure 2: IP Network Settings

To configure the **(Slot 1) Integrated Services Card** which allows communication internally:

13. Set **Host Name**: For this example io-default is used for this example
14. Set **DNS Domain Name**: Wave.Local is used for this example
15. Set **Network interface**: Select *(Slot 1) Integrated Services Card* from drop down menu

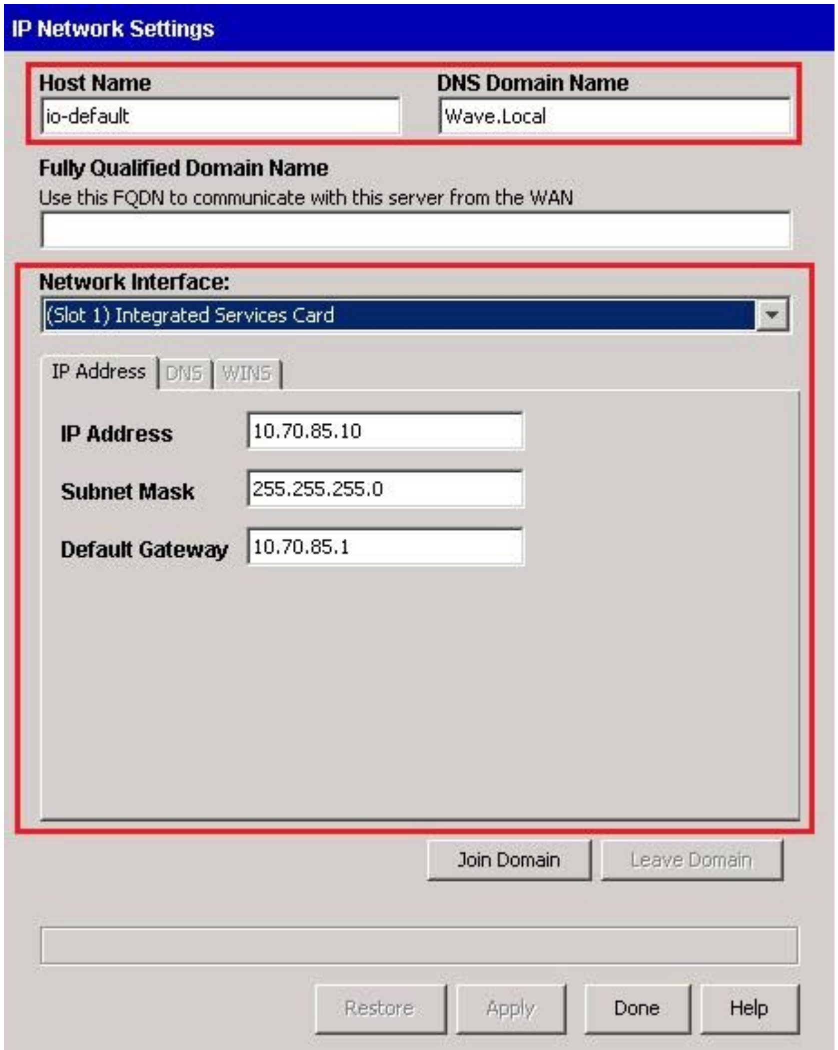


Figure 3: IP Network Settings (Cont.)

16. Select the **DNS** tab
17. To add a DNS server, click **Add** to add the new entry. The DNS IP Address shown is used as an example.
18. Click **Apply** to save changes
19. Click **Done** to exit

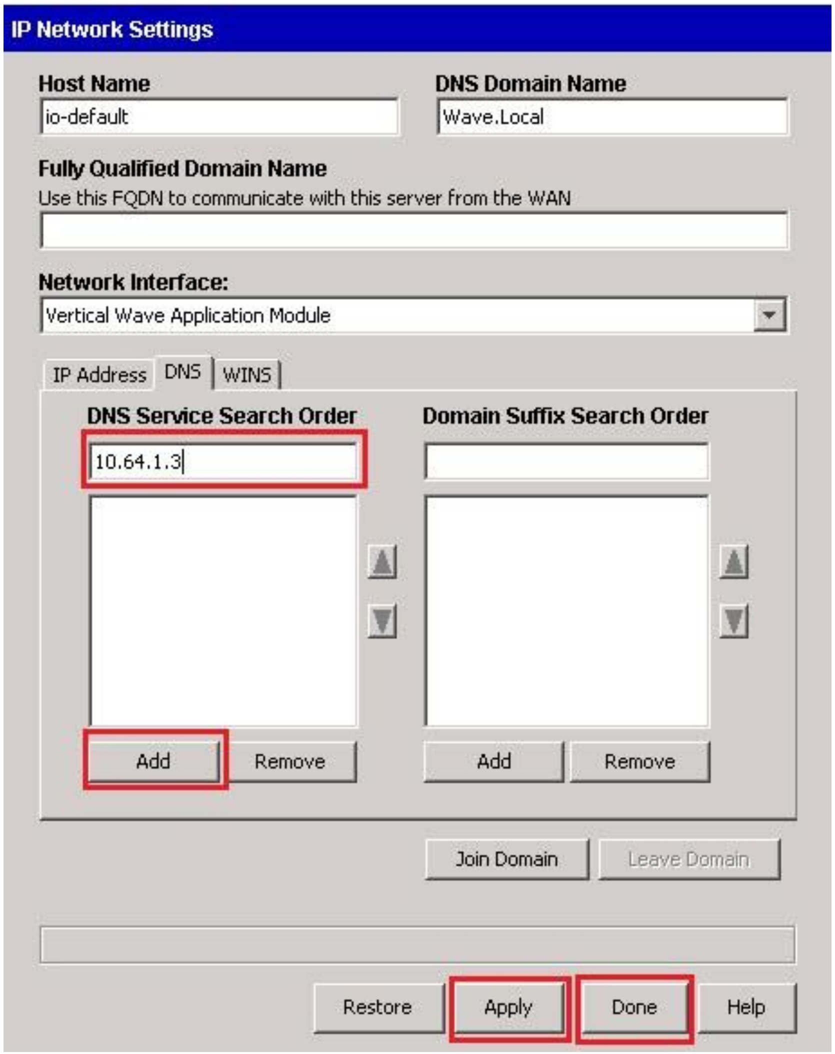


Figure 4: IP Network Settings (Cont.)

5.3.2 SIP Carrier Options

1. Navigate to **IP Telephony**
2. Check **SIP Enabled**. By default SIP Enabled is turned off.
3. Confirm that **SIP Local IP Address** is set to the local IP address of the Vertical Wave IP500. This comes by default from SIP configuration.

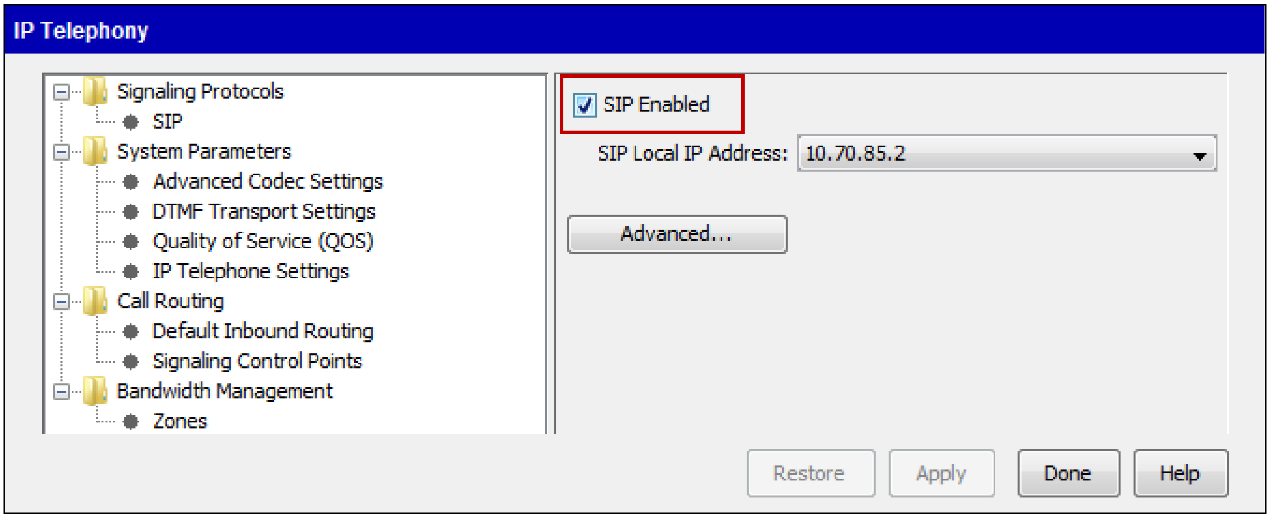


Figure 5: SIP Settings

5.3.3 Signaling Control Point Configuration

1. Navigate to **IP Telephony > Call Routing > Signaling Control Points**
2. Click **New…** to add new Signaling Control Point

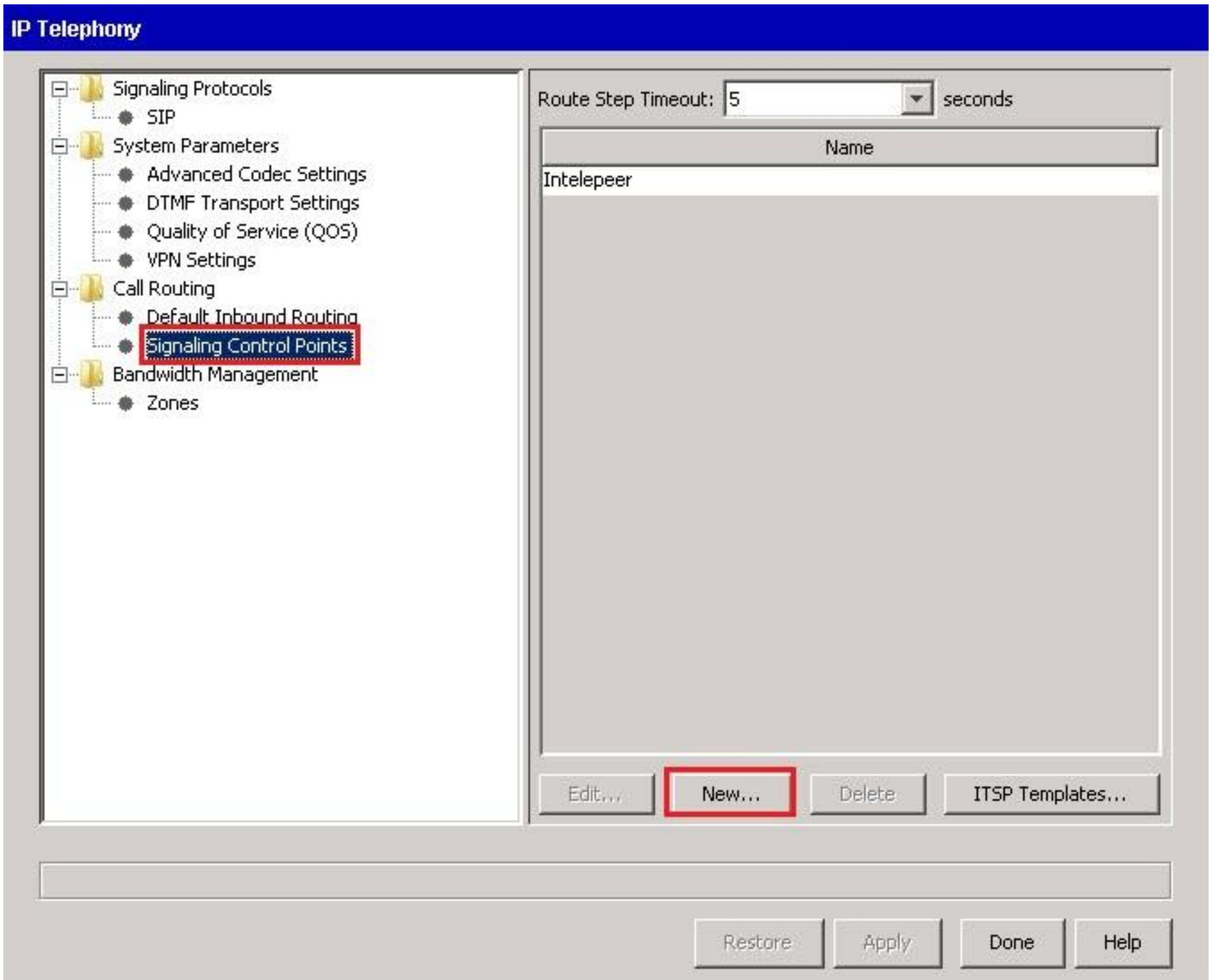


Figure 6: Signaling Control Point

3. Select the **SIP Settings** tab
4. Set **Name:** Enter a name for the new SCP
5. Set **User Name:** Enter the SIP trunk BTN or ITSP Name assigned by the ITSP
6. Set **Proxy Server**: Enter the static LAN IP address of the IntelePeer E-SBC. Please use the actual SBC LAN IP for your network. Please see **Table 2**.
7. Check **SCP is located outside of Wave's network**
8. Ensure **Register with a Proxy/Registrar** is enabled
9. Set **Preferred DTMF Transport**: Select *RFC 2833* from the down menu

The actual SIP Registration Password and Username will be provided by your IntelePeer Account Representative and must be kept confidential! The Trunk Group Pilot Number (username) is used here for illustration purposes only!

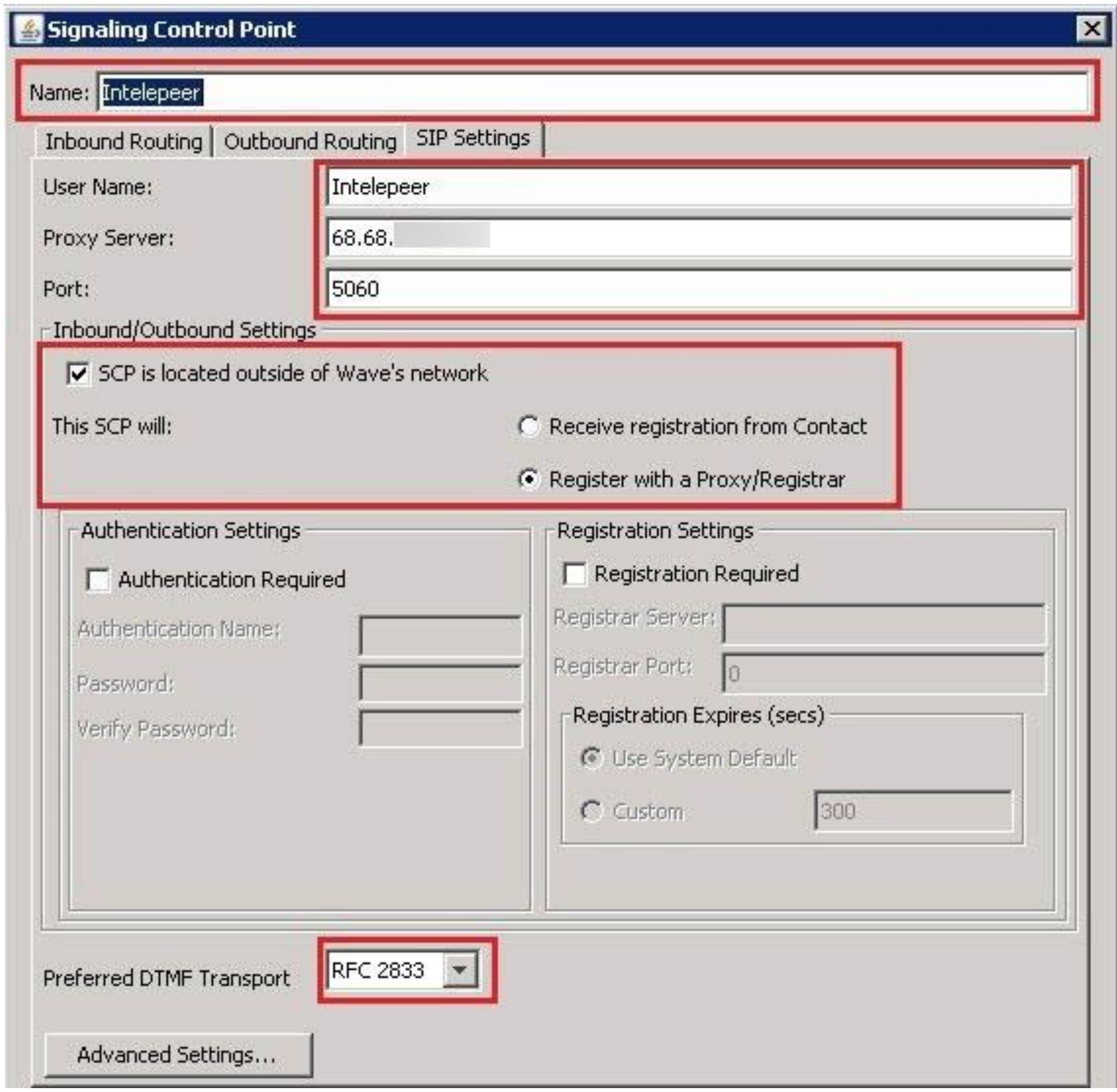


Figure 7: SIP Settings – Cont.

10. Select **Advanced Settings**
11. Uncheck **Attempt Hairpin Elimination on Supervised Transfer**
12. Uncheck **Attempt Hairpin Elimination on Blind Transfer**
13. Set Diversion Source: Select **Send when redirect information is in setup**
14. Click **OK**
15. Click **OK** in the next screen to apply changes

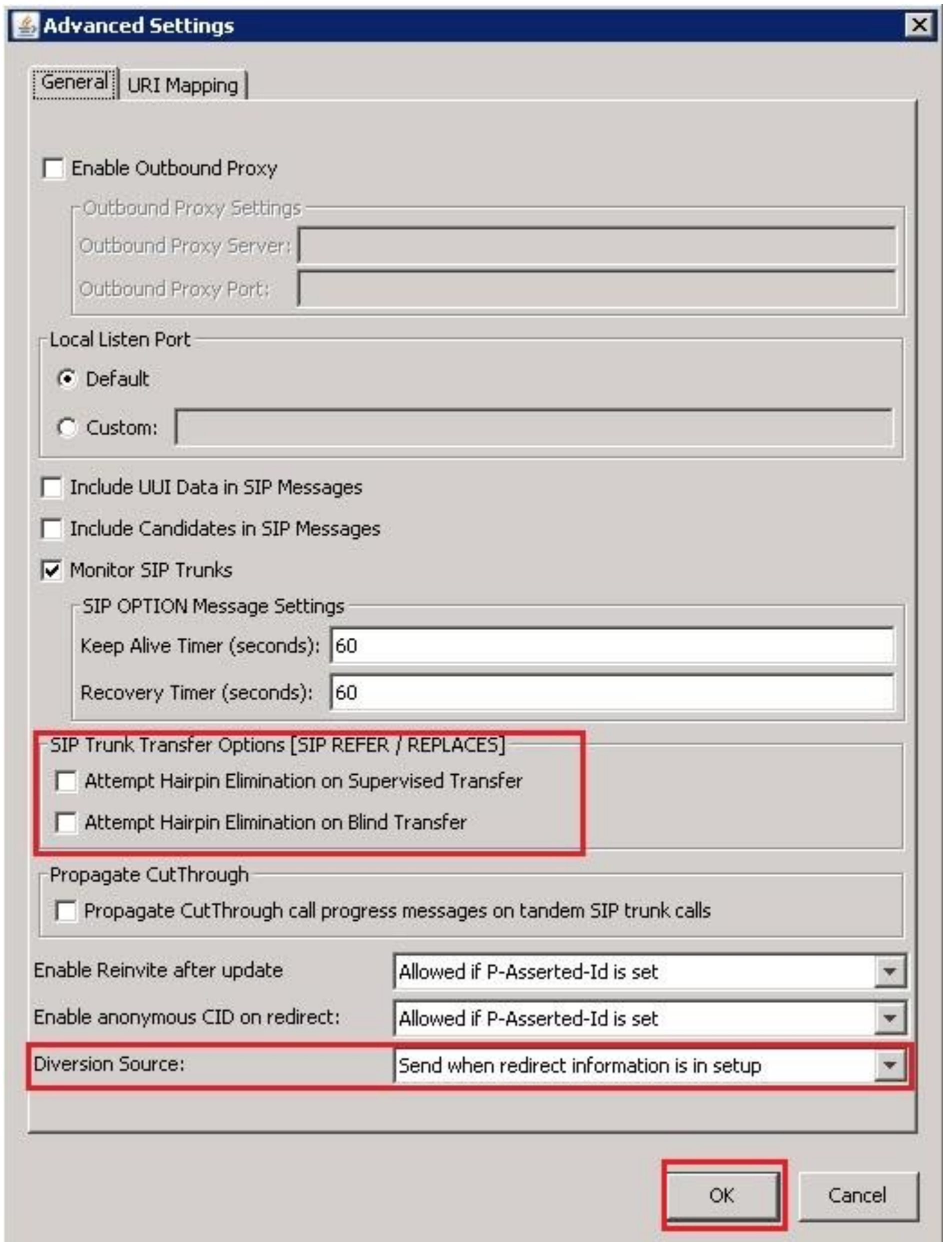


Figure 8: SIP Advanced Settings

6 Summary of Tests and Results

N/S = Not Supported   N/T= Not Tested   N/A= Not Applicable

Test Case #	Test Case Description	Results	Notes
0	Verify Setup	PASS	
1.01	Authenticate on REGISTER Wave resolves SCP/ITSP name	N/A	IntelePeer doesn’t require trunk authorization from the PBX to SIP trunk
1.02	Registration Period	N/A	IntelePeer doesn't require trunk authorization from the PBX to SIP trunk
1.03	No registration-authentication required	PASS	
2.01	Outbound call from SIP phone	PASS	
2.02	Ringback – Outbound from SIP phone	PASS	
2.03	Hold – Outbound from SIP phone	PASS	
2.04	IP MOH on hold call – Outbound from SIP phone	PASS	
2.05	Repeat multiple hold /unhold – Outbound from SIP phone	PASS	
2.06	Remote hold – Outbound from SIP phone	PASS	
2.07	Both end hold – Outbound from SIP phone	PASS	
2.08	Park – Outbound from SIP phone	PASS	
2.09	Multiple outbound calls from SIP Phone	PASS	
2.10	Outbound call from digital phone	PASS	
2.11	Ringback – Outbound from digital phone	PASS	
2.12	Hold – Outbound from digital phone	PASS	
2.13	IP MOH on hold call – Outbound from digital phone	PASS	
2.14	Repeat multiple hold /unhold – Outbound from digital phone	PASS	
2.15	Remote hold – Outbound from digital phone	PASS	
2.16	Both end hold – Outbound from digital phone	PASS	
2.17	Park – Outbound to digital phone	PASS	
3.01	Inbound call to SIP phone	PASS	
3.02	Ringback – Inbound to SIP phone	PASS	
3.03	Hold – Inbound to SIP phone	PASS	
3.04	IP MOH on hold call – Inbound to SIP phone	PASS	
3.05	Repeat multiple hold /unhold –	PASS	

	Inbound to SIP phone		
3.06	Remote hold – Inbound to SIP phone	PASS	
3.07	Both end hold – Inbound to SIP phone	PASS	
3.08	Park – Inbound to SIP phone	PASS	
3.09	Inbound call to digital phone	PASS	
3.10	Ringback – Inbound to digital phone	PASS	
3.11	Hold – Inbound to digital phone	PASS	
3.12	IP MOH on hold call - Inbound to digital phone	PASS	
3.13	Repeat multiple hold/unhold - Inbound to digital phone	PASS	
3.14	Remote hold - Inbound to digital phone	PASS	
3.15	Both end hold - Inbound to digital phone	PASS	
3.16	Park - Inbound to digital phone	PASS	
4.01	Outbound call termination from SIP phone - Wave disconnects after answer	PASS	
4.02	Outbound call termination from SIP phone - Wave disconnects before answer	PASS	
4.03	Outbound call termination from SIP phone - ITSP disconnects after answer	PASS	
4.04	Inbound call termination from SIP phone - Wave disconnects after answer	PASS	
4.05	Inbound call termination from SIP phone - ITSP disconnects after answer	PASS	
4.06	Inbound call termination from SIP phone - ITSP disconnects before answer	PASS	
4.07	Outbound call termination from digital phone - Wave disconnects after answer	PASS	
4.08	Outbound call termination from digital phone - Wave disconnects before answer	PASS	
4.09	Outbound call termination from digital phone - ITSP disconnects after answer	PASS	
4.10	Inbound call termination from digital phone - Wave disconnects after answer	PASS	

4.11	Inbound call termination from digital phone - ITSP disconnects after answer	PASS	
4.12	Inbound call termination from digital phone - ITSP disconnects before answer	PASS	
5.01	Inbound DTMF recognition	N/S	IntelePeer doesn't support SIP INFO messaging
5.02	Outbound DTMF transmission from SIP phone	N/S	IntelePeer doesn't support SIP INFO messaging
5.03	Outbound DTMF transmission from digital phone	N/S	IntelePeer doesn't support SIP INFO messaging
5.04	DTMF- Inband - Inbound DTMF recognition	PASS	
5.05	Outbound DTMF transmission from digital phone	PASS	
5.06	Outbound DTMF transmission from SIP phone	PASS	
5.07	Inbound DTMF recognition - RFC2833	PASS	
5.08	Outbound DTMF transmission from SIP phone	PASS	
5.09	Outbound DTMF transmission from digital phone	PASS	
6.01	Caller-ID on inbound call - SIP phone; Calling party number	PASS	
6.02	Caller-ID on inbound call - SIP phone; Calling party name	PASS	IntelePeer doesn't enable CNAM for inbound calls on the trunk by design
6.03	Inbound call on digital phone - Calling party number	PASS	
6.04	Inbound call on digital phone - Calling party name	PASS	IntelePeer doesn't enable CNAM for inbound calls on the trunk by design
6.05	Caller-ID on inbound call into AA >> SIP phone - Calling party number	PASS	
6.06	Caller-ID on inbound call into AA >> SIP phone - Calling party name	PASS	IntelePeer doesn't enable CNAM for inbound calls on the trunk by design
6.07	AA >> digital phone - Calling party number	PASS	
6.08	AA >> digital phone - Calling party name	PASS	IntelePeer doesn't enable CNAM for inbound calls on the trunk by design
6.09	Caller-ID on outbound call - Calling party number	PASS	
6.10	Caller-ID on outbound call - Calling party name	PASS	
6.11	Use External Caller ID from User Configuration - Calling party number	PASS	IntelePeer doesn't enable CNAM for inbound calls on the trunk by design

6.12	Use External Caller ID from User Configuration - Calling party name	PASS	
6.13	Send Company Name and Main Number - Calling party number	PASS	
6.14	Send Company Name and Main Number - Calling party name	PASS	
6.15	Send station Name and Internal Extension Number - Calling party number	PASS	
6.16	Send station Name and Internal Extension Number - Calling party name	PASS	
6.17	Send station Name and this Number - Calling party number	PASS	
6.18	Send station Name and this Number - Calling party name	PASS	
6.19	Do Not Send Caller ID	PASS	
6.20	Inbound Call forward caller ID - Calling party number	PASS	
6.21	Inbound Call forward caller ID - Calling party name	PASS	
6.22	Inbound - Blind Call transfer Caller ID -Calling party number	PASS	
6.23	Inbound - Blind Call transfer Caller ID -Calling party name	PASS	
6.24	Inbound Supervised Call transfer Caller ID - Calling party number	PASS	
6.25	Inbound Supervised Call transfer Caller ID - Calling party name	PASS	
6.26	Voicemail call Forward Call ID - Calling party number	PASS	
6.27	Voicemail call Forward Call ID - Calling party name	PASS	
7.01	Inbound call to a busy extension	PASS	
7.02	Inbound call to a non-existent extension	PASS	
7.03	Inbound call to a busy extension with VM	PASS	
7.04	Outbound call to a busy extension	PASS	
7.05	Outbound call to a non-existent extension	PASS	
7.06	Outbound call to invalid number	PASS	
8.01	9-1-xxxxxxxxxx	PASS	
8.02	9- xxxxxxx	PASS	7-digits are PASS from the Vertical IP500, but IntelePeer doesn't accept 7-digit dialing

8.03	9-011-xxxxxxxxxx	PASS	
9.01	Call into Auto Attendant	PASS	
9.02	Navigate AA	PASS	
9.03	Inbound call to AA, route to SIP ext	PASS	
9.04	Inbound call to AA, route to digital ext	PASS	
10.01	Call forward to a station digital Ext	PASS	
10.02	Call forward to a station SIP Ext	PASS	
10.03	Call forward to busy station. Configure digital x3772 with no VM	PASS	
10.04	Call forward to busy station (with VM)	PASS	
10.05	Call forward to RNA	PASS	
11.01	Call forward to Off-Net location	PASS	
11.02	Voicemail call notification to OffNet location	PASS	
12.01	Inbound call Blind Transfer station	PASS	
12.02	Inbound call Supervised Transfer station	PASS	
12.03	Outbound call Blind Transfer station	PASS	
12.04	Outbound call Supervised Transfer station	PASS	
12.05	Inbound call Blind Transfer station	PASS	
12.06	Inbound call Supervised Transfer station	PASS	
12.07	Outbound call Blind Transfer station	PASS	
12.08	Outbound call Supervised Transfer station	PASS	
12.09	Inbound call Blind Transfer station	PASS	
12.1	Inbound call Supervised Transfer station		
12.11	Outbound call Blind Transfer station	PASS	
12.12	Outbound call Supervised Transfer station	PASS	
12.13	Inbound call Blind Transfer station	PASS	
12.14	Inbound call Supervised Transfer station	PASS	
12.15	Outbound call Blind Transfer station	PASS	
12.16	Outbound call Supervised Transfer to station	PASS	
13.01	Inbound call, Blind Trunk to Trunk Transfer	PASS	
13.02	Inbound call, Supervised Trunk to Trunk Transfer	PASS	
13.03	Outbound call, Blind Trunk to Trunk Transfer	PASS	
13.04	Outbound call, Trunk to Trunk Supervised Transfer	PASS	

13.05	Inbound Trunk to Trunk Blind Transfer	PASS	
13.06	Inbound Trunk to Trunk Supervised Transfer	PASS	
13.07	Outbound Trunk to Trunk blind Transfer	PASS	
13.08	Outbound Supervised Trunk to Trunk Transfer	PASS	
14.01	Transfer inbound Call to a Forwarded station - Blind Transfer	PASS	
14.02	Transfer Inbound Call to a Forwarded station - Supervised Transfer	PASS	
14.03	Transfer Outbound Call to a Forwarded station - Blind Transfer	PASS	
14.04	Transfer Outbound Call to a Forwarded station-Supervised Transfer	PASS	
14.05	Inbound call blind hairpin transfer	PASS	
14.06	Inbound call Supervised hairpin transfer	PASS	
14.07	Outbound call blind hairpin transfer	PASS	
14.08	Outbound call Supervised hairpin transfer	PASS	
14.09	Inbound Blind Trunk to Trunk Transfer	PASS	
14.10	Inbound Supervised Trunk to Trunk Transfer		
14.11	Outbound Blind Trunk to Trunk Transfer	PASS	
14.12	Outbound Supervised Trunk to Trunk Transfer	PASS	
14.13	Transfer Inbound Call to a Forwarded Off-Net location - Blind Transfer	PASS	
14.14	Transfer Inbound Call to a Forwarded Off-Net location - Supervised Transfer	PASS	
14.15	Transfer Outbound Call to a Forwarded Off-Net location - Blind Transfer	PASS	
14.16	Transfer Outbound Call to a Forwarded OffNet location - Supervised Transfer	PASS	
15.01	Inbound SIP trunk call DID routing	PASS	
16.01	Conference inbound SIP trunk call	PASS	
16.02	Conference outbound SIP trunk call	PASS	

17.01	Inbound call with G711 codec only OR Inbound call with G729 codec only	PASS	
18.01	Successful Session Audit on inbound call	PASS	
18.02	Successful Session Audit on outbound call	PASS	
18.03	Successful Session Audit on Hold call	PASS	
19.01	ITSP QoS settings for SIP	PASS	
19.02	ITSP QoS settings for RTP	PASS	
19.03	Wave QoS settings for SIP	PASS	
19.04	Wave QoS settings for RTP	PASS	
20.01	Emergency Service dialing	N/S	IntelePeer doesn't support 911 testing
21.01	Trunk Monitoring	PASS	
22.01	Fax over IP (Not T.38)	N/A	Vertical is going remove this test case from Vertical test plan
22.02	Inbound Fax	FAIL	Vertical Wave IP500 sends a re-INVITE with T.38 negotiation. As a result the fax transmission stops. Vertical support states that the ability to disable T.38 re-INVITE will be available in the nest software release.
22.03	Outbound Fax	PASS	
22.04	Fax Redirect	N/A	Vertical states that this test case doesn't need to be executed
22.05	Fax Manager	N/A	Vertical states that this test case doesn't need to be executed
23.01	NAT traversal	PASS	