Dialogic®

Dialogic® Brooktrout® SR140 Fax Software with Broadvox GO! SIP Trunking Service

Installation and Configuration Integration Note

December 2010 64-0600-23

www.dialogic.com

Copyright and Legal Notice

1. Scope

This document is intended as a general guide for configuring a basic installation of the ***Broadvox GO! SIP Trunking Service*** for use with Dialogic® Brooktrout® SR140 Fax over IP (FoIP) software platform. The interoperability includes ***SIP*** call control and ***T.38/T.30*** media.

This document is not intended to be comprehensive and thus does not replace the manufacturer's detailed configuration documentation. Users of this document should already have a general knowledge of how to install and configure the ***Broadvox GO! SIP Trunking Service.***

The sample configuration shown and/or referred in the subsequent sections was used for lab validation testing by Dialogic. Therefore, it is possible and even likely that the example configuration will not match the exact configuration and versions that would be present in a deployed environment. However, the sample configuration does provide a possible starting point to work with the equipment vendor for configuring your device. Please consult the appropriate manufacturer's documentation for details on setting up your specific end user configuration.

For ease of reference, the Dialogic® Brooktrout® SR140 Fax Software and Dialogic® Brooktrout® TR1034 Fax Boards will sometimes be denoted herein, respectively, as SR140 and TR1034. All references to the SDK herein refer to the Dialogic® Brooktrout® Fax Products SDK. The Broadvox GO! SIP Trunking Service will sometimes be denoted herein as Broadvox SIP Trunk, or some other form thereof.

2. Configuration Details

The following systems were used for the sample configuration described in the document.

2.1 Broadvox GO! SIP Trunking Service

Vendor	***Broadvox***
Model	***GO! SIP Trunking Service***
Software Version	***N/A***
Protocol to SIP Trunking Service	***SIP***
IP Device	***Dialogic® Brooktrout® SR140***
Protocol to IP Device	***SIP***
Additional Notes	***Broadvox SIP Trunking Service only supports PCMU as an RTP codec.***

2.2 Dialogic® Brooktrout® SR140 Fax Software

Vendor	***Dialogic***
Model	***Dialogic® Brooktrout® SR140 Fax Software***
Software Version	***SDK 6.2.8***
Protocol to Gateway or Call Manager	***SIP***
callctrl.cfg file	***The Broadvox SIP trunk only supports PCMU as an RTP codec.***

2.3 Dialogic® Brooktrout® TR1034 Fax Board

Vendor	***Dialogic***
PSTN Device	***Dialogic® Brooktrout® TR1034 Fax Board***
Software Version	***SDK 6.2.8***
Protocol to PSTN Device	***Analog Loop Start***
callctrl.cfg file	***All defaults***

2.4 Network System Configuration

The diagram below details the sample configuration used in connection with this document. On the IP side, the SR140 was configured to send and receive T.38 faxes. On the PSTN side, the TR1034 board was configured to send and receive T.30 faxes over an analog loopstart connection. Carrying traffic between the two was the Broadvox GO! SIP trunk. Testing consisted of the full suite of interop calls between the two endpoints: first the SR140 sending and the TR1034 receiving and then the TR1034 sending with the SR140 receiving.

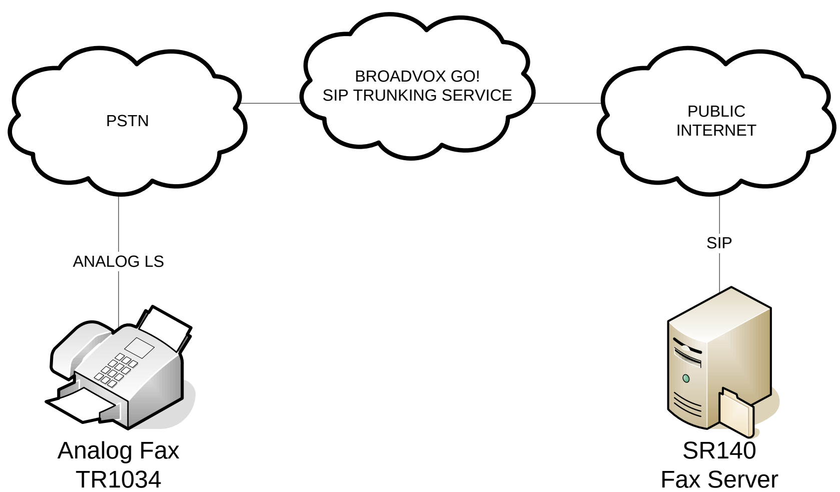


Diagram Notes: SR140 Fax Server = Fax Server including Dialogic® Brooktrout® SR140 Fax Software and third party fax application

3. Prerequisites

The SR140 based fax server must be assigned an IP address able to be reached by the SIP Trunking service. A fax server that has been assigned a private IP address will have communication issues talking to the Broadvox SIP Trunking service.

4. Summary of Limitations

Broadvox SIP Trunking Service does not support T.38 with V.34 (version 3) support. The SR140 default setting for T.38 version will work without issues. Invites that include T.38 Version 3 will be rejected with a '488 Not Supported Here' response from the SIP Trunk.

The Broadvox SIP trunk only supports PCMU as an RTP codec. Only PCMU should be listed in the RTP codec list under the "RTP Parameters" tab in the advanced view of the SR140 Configuration Tool.

5. Broadvox SIP Trunk Setup Notes

For the sample test configuration, the Broadvox SIP Trunk was configured as described below.

5.1 Network Addresses

Device #	Device Make, Model, and Description	Device IP Address
1	BROADVOX GO! SIP TRUNKING SERVICE	64.115.X.X*

* Broadvox will provide an unique IP address to use for the Broadvox SIP Trunking Service.

5.2 IP Trunk Configuration

There is no need to configure the IP trunk itself. The provided IP address should be used as the "gateway" IP address in setting up the SR140 software.

In order for your PBX, IAD, or Gateway to receive signaling and media from Broadvox, you must configure your firewall or NAT to allow the following IP addresses and port ranges:

Traffic Type	IP Addresses	Protocol	Port Range
SIP	208.93.224.224/28 208.93.226.208/28 208.93.227.208/28	UDP and TCP	5060
SIPS (SIP over TLS)	208.93.224.224/28 208.93.226.208/28 208.93.227.208/28	TCP	5061
Media	208.93.224.224/28 208.93.226.208/28 208.93.227.208/28 209.249.3.164 64.158.162.71 64.158.162.100 64.152.60.71 64.152.60.164 209.249.3.71 209.249.3.81 64.156.174.71 208.93.227.5 208.93.226.5	UDP	1024-65535

In order to send traffic to Broadvox, you should create peer / gateway / trunk definitions inside your PBX, IAD, or Gateway device for all three of these locations:

City	DNS A Record	DNS SRV Record	IP Address
New York City, NY	nyc01-01.fs.broadvox.net	nyc01-01.fs.broadvox.net	208.93.226.212
Dallas, TX	dfw01-01.fs.broadvox.net	dfw01-01.fs.broadvox.net	208.93.224.228
Los Angeles, CA	lax01-01.fs.broadvox.net	lax01-01.fs.broadvox.net	208.93.227.212
All Cities	fs.broadvox.net	fs.broadvox.net	N/A

6. Dialogic® Brooktrout® SR140 Fax Software Setup Notes

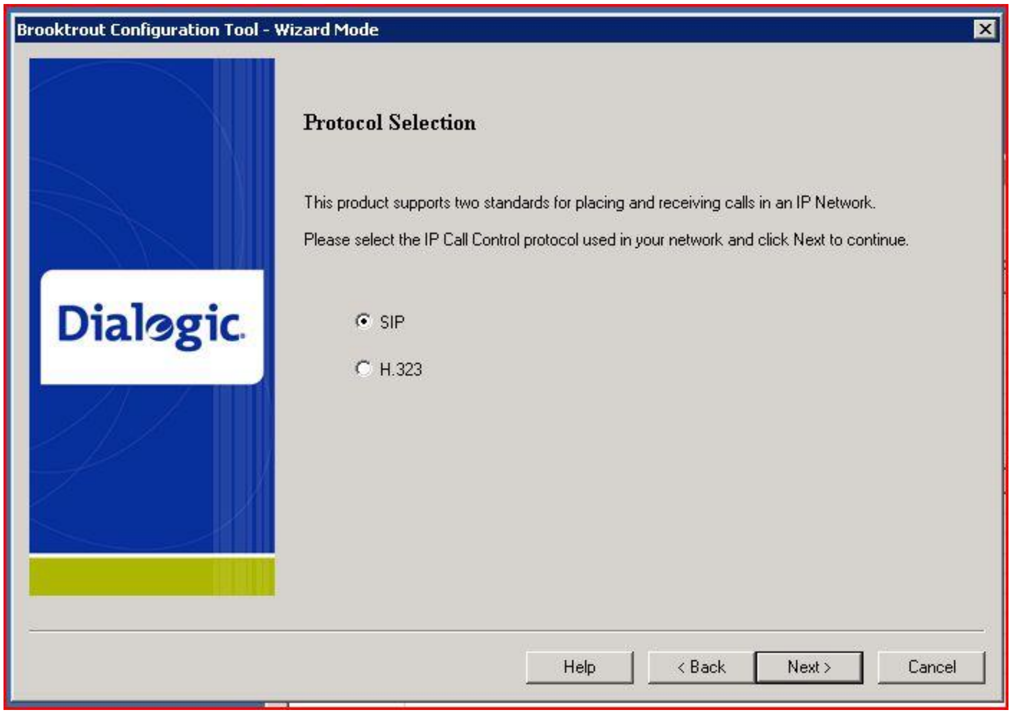
The *Installation and Configuration Guides* for the SR140 are available from the site:

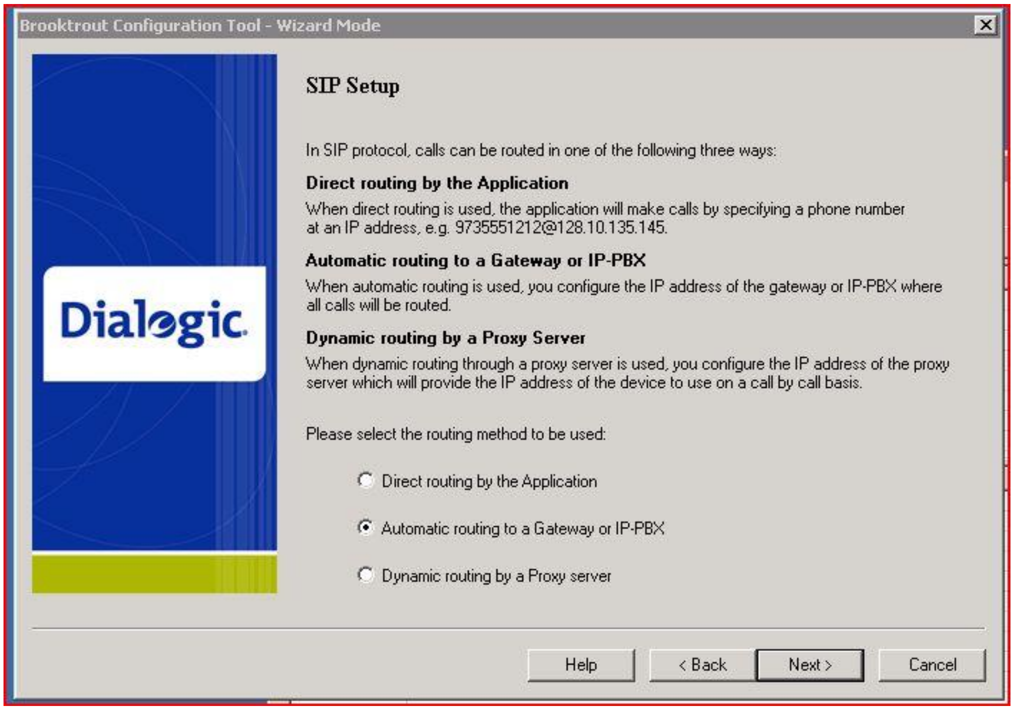
http://www.dialogic.com/manuals/brooktrout/default.htm

Please note that if you plan to place your fax server behind a firewall, you must keep all necessary ports open so as not to impede fax traffic.

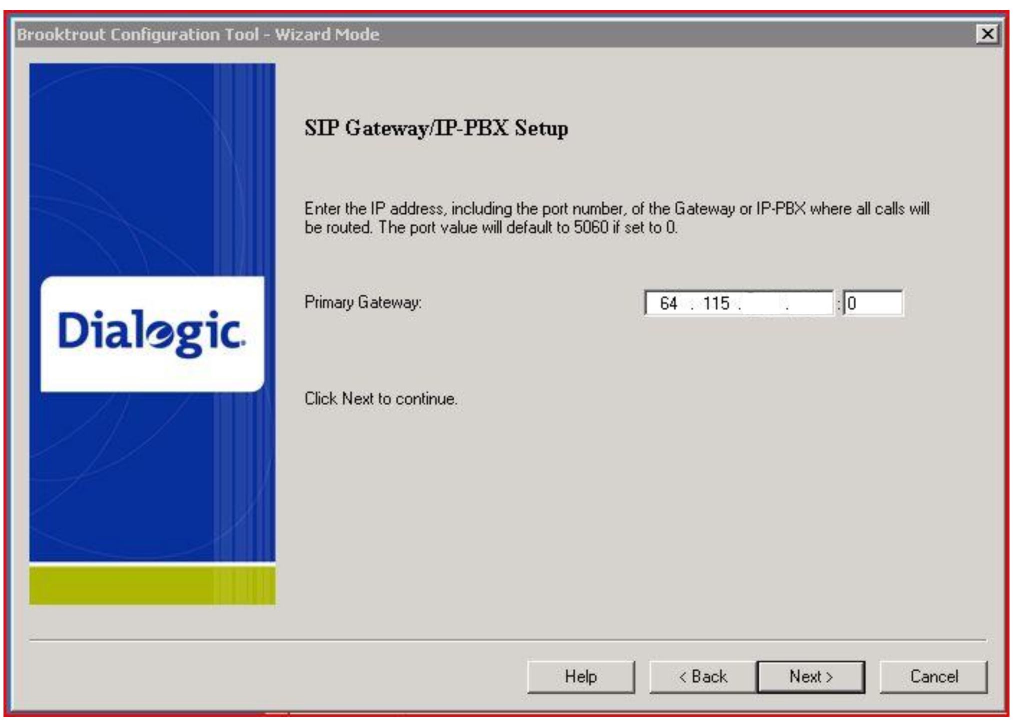
Dialogic SR140 Ports:
- Port 5060 – SIP signaling port
- Ports 56000 to 57000 – UDP ports for FoIP traffic (configurable)

The following SR140 Setup Wizard screenshots illustrate how to test configuration was setup to interop with the Broadvox SIP Trunking Service.

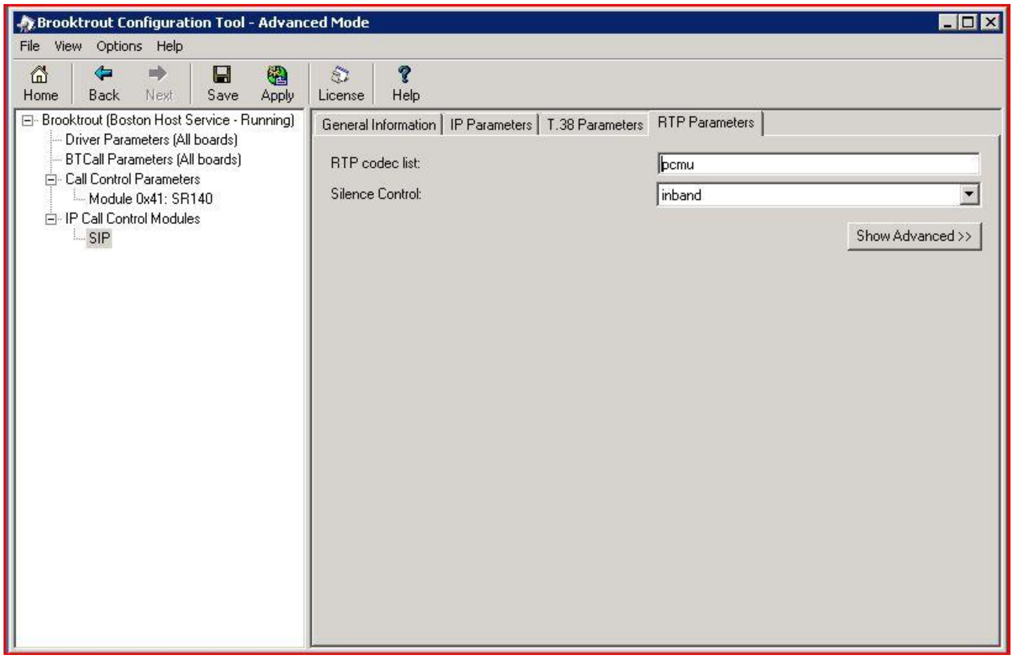




Enter the IP Address provided by Broadvox.



All settings were default except for RTP codec which was set to PCMU as the Broadvox SIP trunk only supports PCMU as an RTP codec. This setting can be adjusted through the "advance configuration" option. Select the SIP heading under the IP Call Control Modules section. Selecting the “RTP Parameters” tab and set the RTP codec list to read “pcmu”, as pictured below.



For the sample test configuration, the SR140 callctrl.cfg is shown below for reference.

```
l3l4_trace=verbose
 l4l3_trace=verbose
 api_trace=verbose
 internal_trace=verbose
 host_module_trace=verbose
 ip_stack_trace=warning
# Most of the time a path should be used for this file name.
 trace_file=test_0004_ecc.log
 max_trace_files=1
 max_trace_file_size=10
[host_module.1]
 module_library=brktsip.dll
 enabled=true
[host_module.1/t38parameters]
 t38_fax_rate_management=transferredTCF
 fax_transport_protocol=t38_only
 t38_fax_udp_ec=t38UDPRedundancy
 rtp_ced_enable=true
 t38_max_bit_rate=14400
 t38_fax_version=0
 media_renegotiate_delay_inbound=1000
 media_renegotiate_delay_outbound=-1
 t38_fax_fill_bit_removal=false
 t38_fax_transcoding_jbig=false
 t38_fax_transcoding_mmr=false
 t38_t30_fastnotify=false
 t38_type_of_service=0
 t38_UDPTL_redundancy_depth_control=5
 t38_UDPTL_redundancy_depth_image=2
[host_module.1/rtp]
 rtp_frame_duration=20
 rtp_jitter_buffer_depth=100
```

```
  rtp_codec=pcmu
  rtp_silence_control=inband
  rtp_type_of_service=0
  rtp_voice_frame_replacement=0
[host_module.1/parameters]
  sip_max_sessions=256
  sip_default_gateway=64.115.XXX.XXX:5060
  sip_proxy_server1=
  sip_proxy_server2=
  sip_proxy_server3=
  sip_proxy_server4=
  sip_registration_server1=
  sip_registration_server1_aor=
  sip_registration_server1_username=
  sip_registration_server1_password=
  sip_registration_server1_expires=3600
  sip_registration_server2=
  sip_registration_server2_aor=
  sip_registration_server2_username=
  sip_registration_server2_password=
  sip_registration_server2_expires=3600
  sip_registration_server3=
  sip_registration_server3_aor=
  sip_registration_server3_username=
  sip_registration_server3_password=
  sip_registration_server3_expires=3600
  sip_registration_server4=
  sip_registration_server4_aor=
  sip_registration_server4_username=
  sip_registration_server4_password=
  sip_registration_server4_expires=3600
  sip_registration_interval=60
  sip_Max-Forwards=70
  sip_From=Test <sip:test@dialogic.com>
  sip_Contact=0.0.0.0:0
  sip_username=-
  sip_session_name=Broadvox Interop
  sip_session_description=
  sip_description_URI=
  sip_email=
  sip_phone=
  sip_Route=
  sip_session_timer_session_expires=0
  sip_session_timer_minse=-1
  sip_session_timer_refresh_method=0
  sip_ip_interface=
  sip_ip_interface_port=5060
  sip_redirect_as_calling_party=0
  sip_redirect_as_called_party=0
[module.41]
  model=SR140
  virtual=1
  exists=1
  vb_firm=C:\Interop kit SDK611 v1.2\fdtool-6.1.1\bin\bostvb.dll
  channels=26
[module.41/ethernet.1]
  ip_interface={B583A089-A6B5-431A-8443-8F9956C3C1CB}:0
  media_port_min=56000
  media_port_max=57000
[module.41/host_cc.1]
  host_module=1
  number_of_channels=26
```

7. Frequently Asked Questions

- ”*I’m configured as near as possible to this the sample configuration described in this document, but calls are still not successful; what is my next step?”*
  - ➔ Provide this document to your gateway support.
  - ➔ Ensure T.38 is enabled on the gateway.
  - ➔ Confirm that basic network access is possible by pinging the gateway.

- ”*How do I obtain Wireshark traces?*”
  - ➔ The traces can be viewed using the Wireshark network analyzer program, which can be freely downloaded from *http://www.wireshark.org*.
  - ➔ To view the call flow in Wireshark, open the desired network trace file and select “Statistics->VoIP Calls” from the drop down menu. Then highlight the call and click on the “Graph” button.

- ”*I try to call the SR140 port, but I get a network busy - why?*”
  - ➔ Most likely you do not have the proper ports open on your firewall. Check settings against the above recommendations and be sure your efforts match up.

- ”*I’ve followed this guide to the letter, but I can’t connect to Broadvox, why?*”
  - ➔ Make sure your fax server is assigned a non-private IP reachable from the Internet. If you’re assigning a private IP to the FoIP server, that will be communicated in the “connection information” of the SDP message. Broadvox needs a public IP to communicate with the server, as this is how Broadvox authenticates your connection and routes traffic to and from your fax server.
  - ➔ Make certain that “PCMU” is the only codec listed under the “RTP Parameters” tab in the advanced setup mode when setting up SR140. Failing to do so can cause setup to fail.

8. References

http://www.broadvox.com/SIPTrunking.aspx