Dialogic® Brooktrout® SR140 Fax Software with Cisco Unified Border Element

Installation and Configuration Integration Note

IMPORTANT NOTE

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1. Scope

This document is intended as a general guide for configuring a basic installation of the **Cisco Unified Border Element (CUBE)** 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 **Cisco Unified Border Element**.

The sample configuration shown and/or referred in the subsequent sections was used for lab validation testing by Dialogic. Therefore, it is quite possible that the sample configuration will not match an exact configuration or 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 Cisco Unified Border Element will sometimes be denoted herein as Cisco CUBE, or some other form thereof.

2. Configuration Details

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

2.1 Cisco Unified Border Element

Vendor	***Cisco***
Model	***Cisco Unified Border Element w/ Cisco 2911***
Software Version	***Cube Version 9.0*** ***IOS Version 15.2-3.T1***
IP Device	***Dialogic® Brooktrout® SR140 Fax Software***
Protocol to SR140 Fax Software	***SIP***
PSTN Device	***Dialogic® Brooktrout® TR1034***
Protocol to PSTN Device	***E1 ISDN***
Additional Notes	

2.2 Dialogic® Brooktrout® SR140 Fax Software

Vendor	***Dialogic***
Model	***Dialogic® Brooktrout® SR140 Fax Software***
Software Version	***Tested with SDK 6.5.0***
Protocol to Gateway or Call Manager	***SIP***
callctrl.cfg file	***All defaults except "rtp_codec = pcma"***

2.3 Dialogic® Brooktrout® TR1034 Fax Server

Vendor	***Dialogic***
Model	***Dialogic® Brooktrout® TR1034+P30V30FH-E1-1N***
Software Version	***Tested with SDK 6.5.0***
Protocol to Gateway or Call Manager	***E1 ISDN***
callctrl.cfg file	***All defaults***

2.4 Network System Configuration

The diagram below details the sample configuration used in connection with this document.

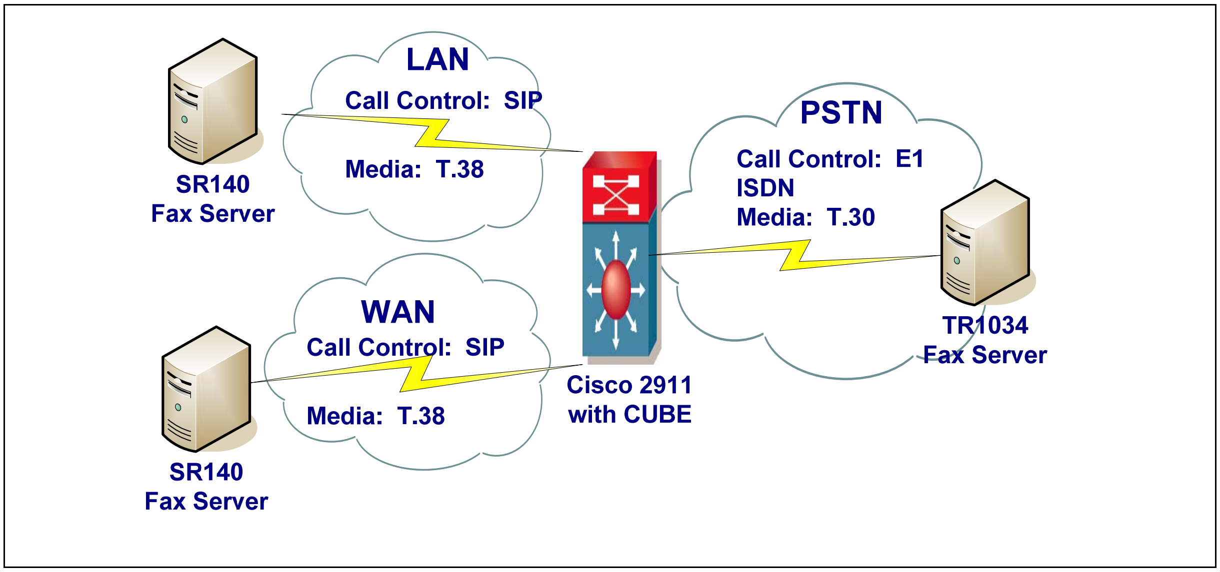


Diagram Notes:

- SR140 Fax Server = Fax Server including Dialogic® Brooktrout® SR140 Fax Software and third party fax application.

2.5 Network Addresses

The following table lists the IP addresses and their descriptions used in subsequent sections.

Device #	Device Make, Model, and Description	Device IP Address
1	Cisco Unified Border Element	10.128.28.43
2	SR140 Fax Server WAN	10.128.16.131
3	SR140 Fax Server LAN	10.128.28.201

3. Prerequisites

- None.

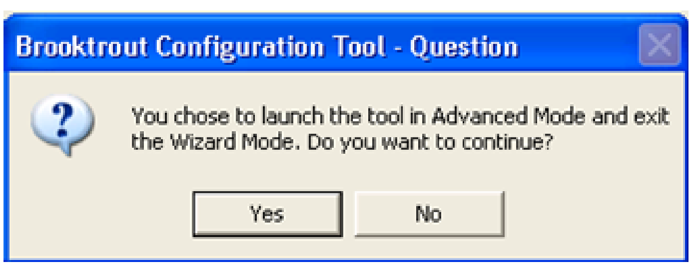
4. Summary of Limitations

By default, the SR140 configures G.711 codec support for both alaw and ulaw. The Cisco CUBE does not support an M= line for T.38 as well as an M= line for g711ulaw and g711alaw. Cisco has an enhancement to add support for this behaviour (CSCsi10343).

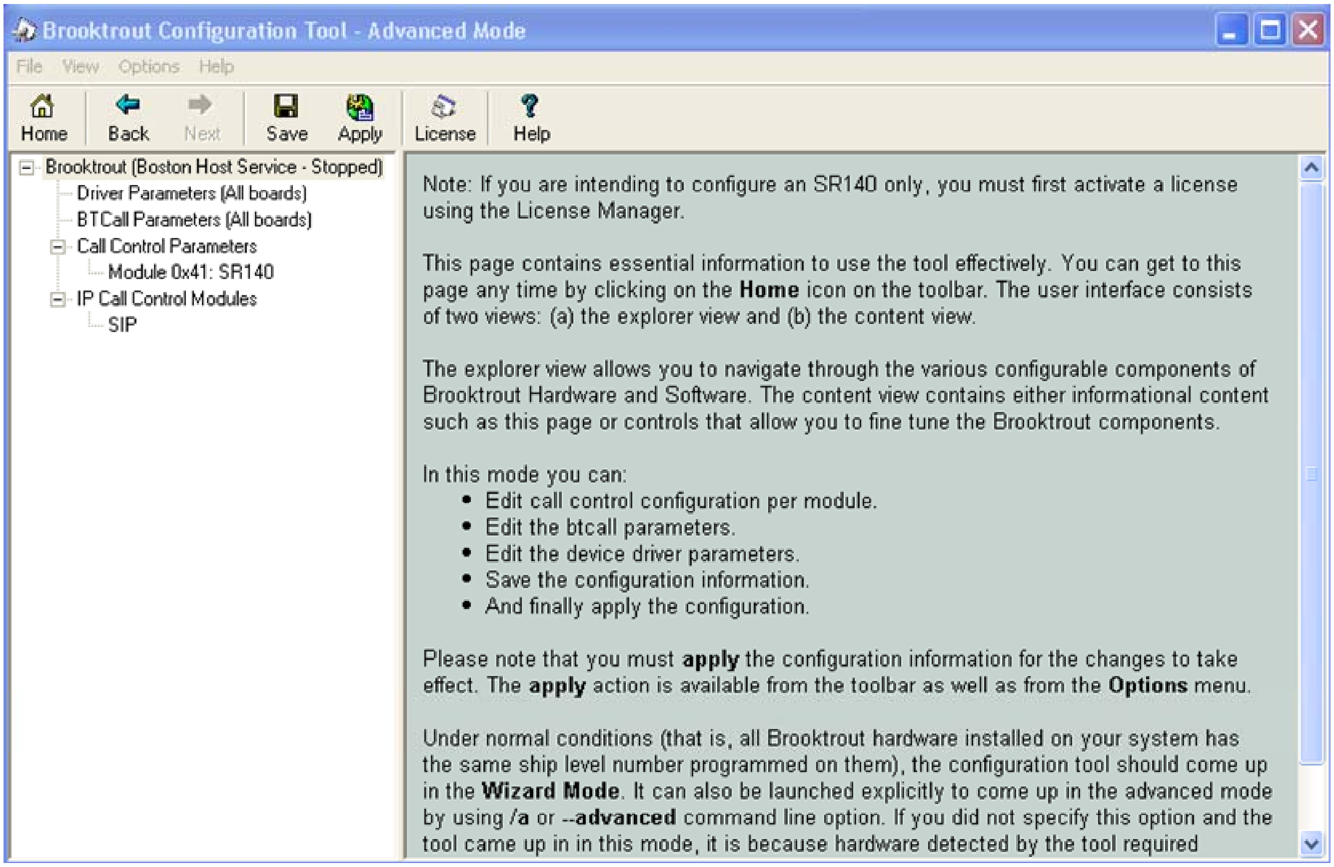
As a workaround, the SR140 must be configured to present only one audio codec. The following are the configuration steps to perform this change when using the configuration tool.

Launch the Config Tool (Start->Programs->Brooktrout->Brooktrout Configuration Tool

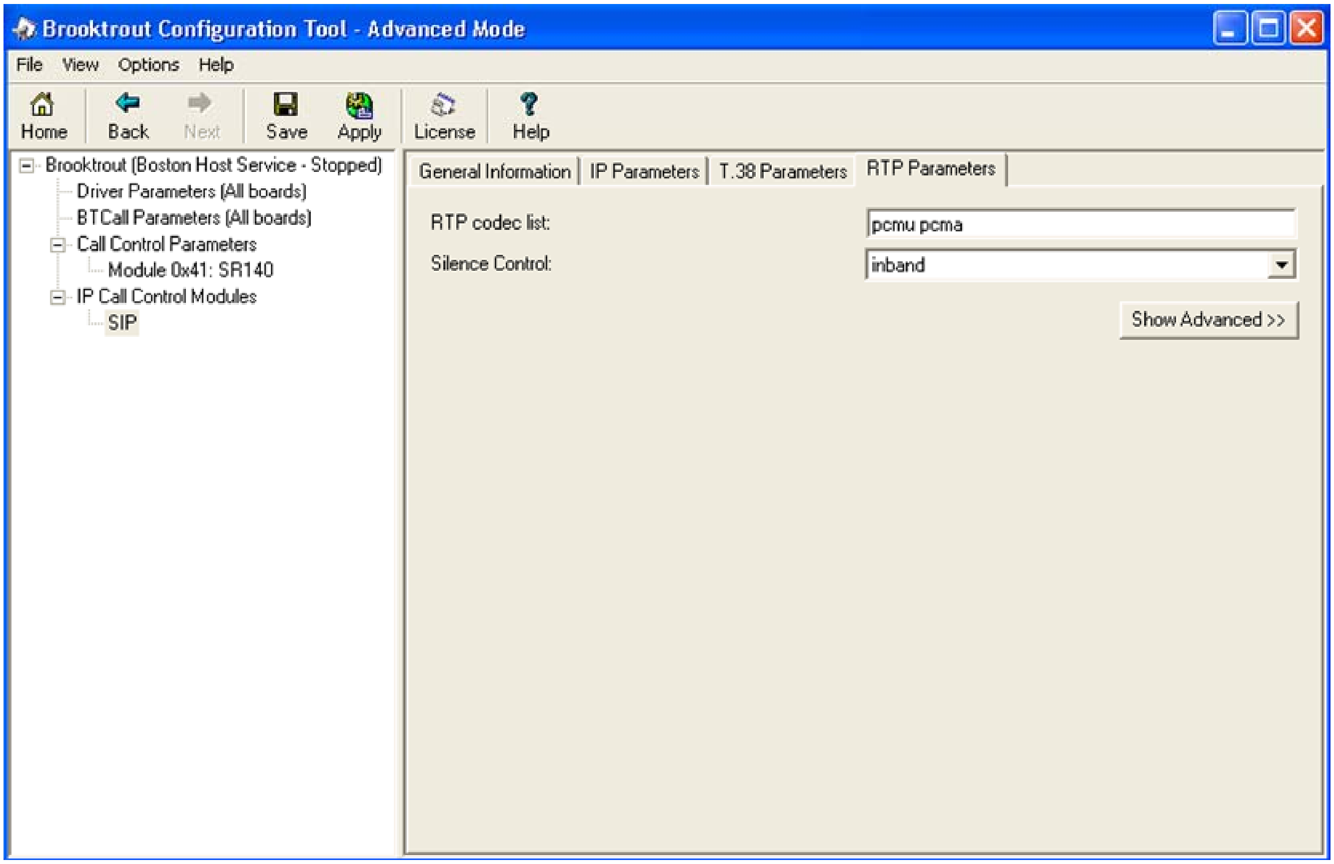
Select **Advanced Mode**.



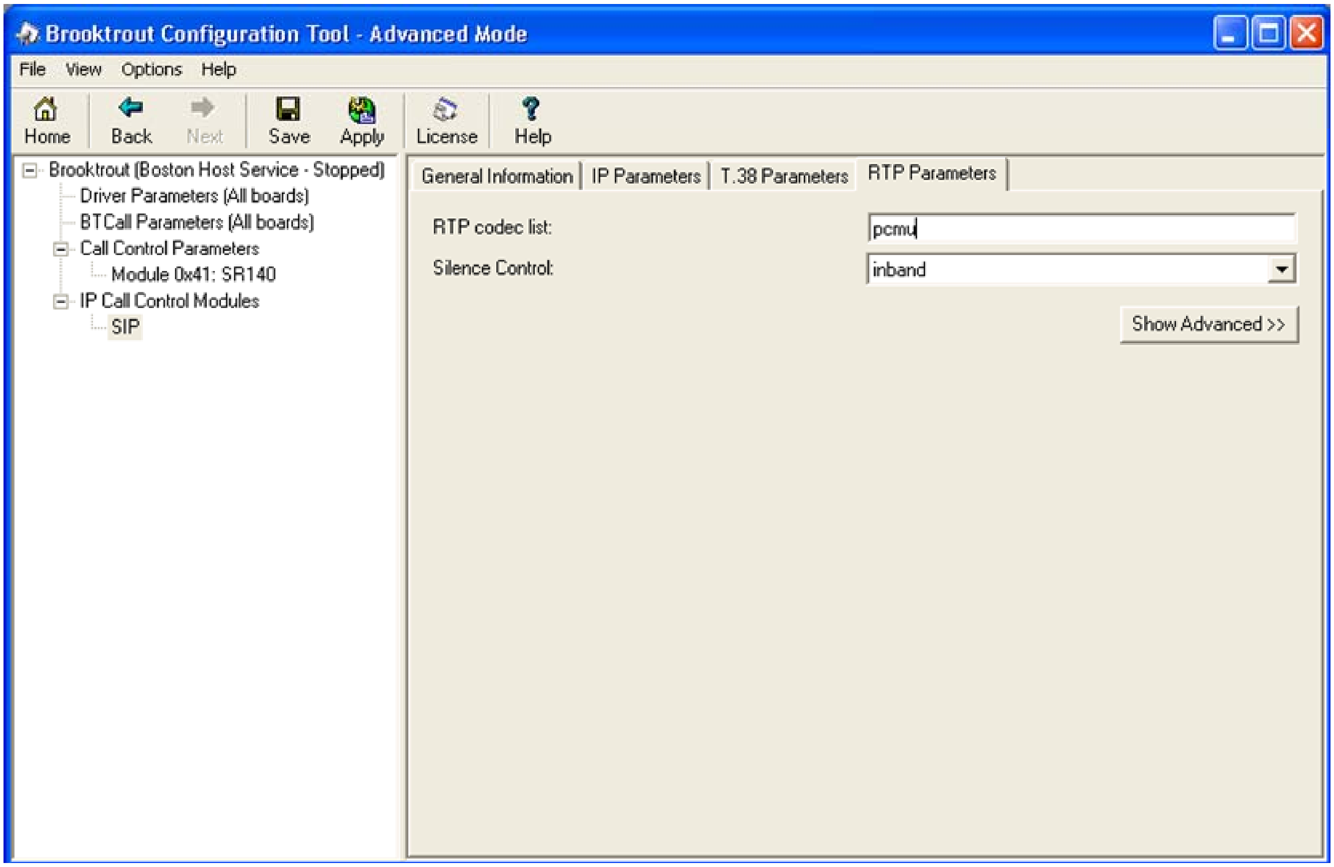
Select **Yes** to enter Advanced Mode.



Select **SIP** under **IP Call Control Modules.**



Select **RTP Parameters** TAB.
Change **RTP codec list** to only include one codec from the default of pcmu pcma.



Click **Save** and then close the Configuration Tool.

5. Cisco Unified Border Element Setup Notes

The Cisco Unified Border Element software was pre-installed on the unit that was tested. There were no changes to the default CUBE configuration. Please refer to the *Cisco Unified Border Element Configuration Guide* for details.

> **Activation Cisco Product Authorization Key (PAK)—**A Cisco Product Authorization Key (PAK) is required to configure some of the Cisco features described in this guide. Before you start the configuration process, please make sure that you have registered your products and activate your PAK at the following URL http://www.cisco.com/go/license.

To display the Cisco Unified Border Element (Cisco UBE) status, the software version, the license capacity, the image version, and the platform name of the device, use the show cube status command in user EXEC or privileged EXEC mode.

Device> show cube status

CUBE-Version : 9.0
SW-Version : 15.2.3.T1, Platform CISCO2911/K9
HA-Type : none
Licensed-Capacity : 50

6. Cisco Gateway Configuration

The Cisco Media Gateway was configured using the CLI. For this sample test configuration, Cisco IOS 15.x with support for Super G3 Fax (V.34 T.38) was used. Cisco configuration instructions for configuring dial-peers are located the following site: http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/dial_peer/dpeer_c.html

Important Note: The CUBE requires both the incoming and outgoing dial-peer to have support for fax relay. If no voip dial-peer can be matched for the present string of digits being sent, the dial-peer 0 (the default dial-peer) will be used. When dial-peer 0 is used, the CUBE limits the fax capabilities of a router to audio only (non-T.38 fax). In the example configuration below, dial-peer 6 was created to match incoming fax called-numbers and add fax capabilities to the inbound direction.

In the outbound direction, a destination pattern was used to direct calls to the correct SIP endpoint or to the PSTN trunk card.

```
voice service voip
 no ip address trusted authenticate
 mode border-element license capacity 50
 allow-connections sip to sip
 fax protocol t38 version 3 ls-redundancy 2 hs-redundancy 2 fallback none
 sip
  sip-profiles 100

voice class sip-profiles 100
!

voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 clear-channel

controller E1 0/0/0
```

```
 clock source internal
 pri-group timeslots 1-31

dial-peer voice 441 pots
 destination-pattern 777
 no digit-strip
 direct-inward-dial
 port 0/0/0:15

dial-peer voice 3 voip
 destination-pattern 7810016131
 session protocol sipv2
 session target ipv4:10.128.16.131
 session transport udp
 voice-class codec 1

dial-peer voice 6 voip
 session protocol sipv2
 incoming called-number 781.......
 voice-class codec 1
 no fax-relay sg3-to-g3
 fax nsf 000000
 fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none

dial-peer voice 5 voip
 destination-pattern 7819999999
 session protocol sipv2
 session target ipv4:10.128.28.201
 session transport udp
 voice-class codec 1
 no fax-relay sg3-to-g3
 fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
```

7. Dialogic® Brooktrout® SR140 Fax Software Setup Notes

The Installation and Configuration Guide used to set up the SR140 is available from the site below:

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

The SR140 callctrl.cfg file used in the sample test configuration is shown below for reference.

```
 api_trace=verbose
 internal_trace=verbose
 l3l4_trace=verbose
 l4l3_trace=verbose
 host_module_trace=verbose
 ip_stack_trace=warning
 vtty_trace=true
 max_trace_files=1
 max_trace_file_size=100
 trace_file=test_0017_ecc.log
[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=33600
 t38_fax_version=3
 media_passthrough_timeout_inbound=1000
 media_passthrough_timeout_outbound=4000
 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_stream_renegotiation=single
 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=pcma
 rtp_silence_control=inband
 t38_offer_as_ced=true
 rtp_type_of_service=0
 rtp_voice_frame_replacement=0
[host_module.1/parameters]
 sip_max_sessions=256
 sip_default_gateway=
 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_registration_interval_delta=5
  sip_Max-Forwards=70
  sip_From=Anonymous <sip:no_from_info@anonymous.invalid>
  sip_Contact=0.0.0.0:0
  sip_ContactV6=
  sip_username=-
  sip_session_name=no_session_name
  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_preference=ipv4_only
  sip_ip_interface=
  sip_ip_interfaceV6=
  sip_ip_interface_port=5060
  sip_ip_interface_portV6=5060
  sip_redirect_as_calling_party=0
  sip_T1_timeout=500
  sip_max_invite_retransmissions=7
  sip_redirect_as_called_party=0
  sip_user_agent=Brktsip/6.5.0B4 (Dialogic)
  sip_RFC3325_Identity=0
[module.41]
  model=SR140
  virtual=1
  exists=1
  vb_firm=C:\fdtool-6.5.0\bin\bostvb.dll
  channels=120
[module.41/ethernet.1]
  ip_preference=ipv4_only
  ip_interface={A95D8EEC-EE58-4B5B-A3FF-657D851AC2E0}:0
  ip_interfaceV6=
  ip_address=0.0.0.0
  ip_addressV6=
  media_port_min=56000
  media_port_max=56999
[module.41/host_cc.1]
  host_module=1
  number_of_channels=120
```

8. Dialogic® Brooktrout® TR1034 Setup Notes

For the sample test configuration, the TR1034 was configured using the default values, consult the *Dialogic® Brooktrout® Fax Products Installation and Configuration Guide* for details.

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

9. 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.