



# **Dialogic® Brooktrout® SR140 Fax Software with Patton 4960 PRI and 4554 BRI Gateways**

## **Installation and Configuration Integration Note**

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## 1. Scope

This document is intended as a general guide for configuring a basic installation of the **Patton 4960 PRI and the Patton 4554 BRI Gateways** 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 should not and 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 **Patton Gateways**.

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 Patton Gateways will be denoted herein as Patton 4960 PRI and Patton 4554 BRI, or some other form thereof.

## 2. Configuration Details

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

### 2.1 Gateway

Vendor	<b>Patton</b>
Model(s)	<b>4960 PRI Gateway</b> <b>4554 BRI Gateway</b>
Software Version(s)	<b>PRI – R5.4 2009-07-20 SIP</b> <b>BRI – R5.3 2009-05-20 SIP</b>
PSTN Device	<b>Dialogic® Diva® Server BRI-2M</b>
Protocol to PSTN Device	<b>Euro ISDN PRI and BRI</b>
IP Device	<b>Dialogic® Brooktrout® SR140</b>

## 2.2 Dialogic® Brooktrout® SR140 Fax Software

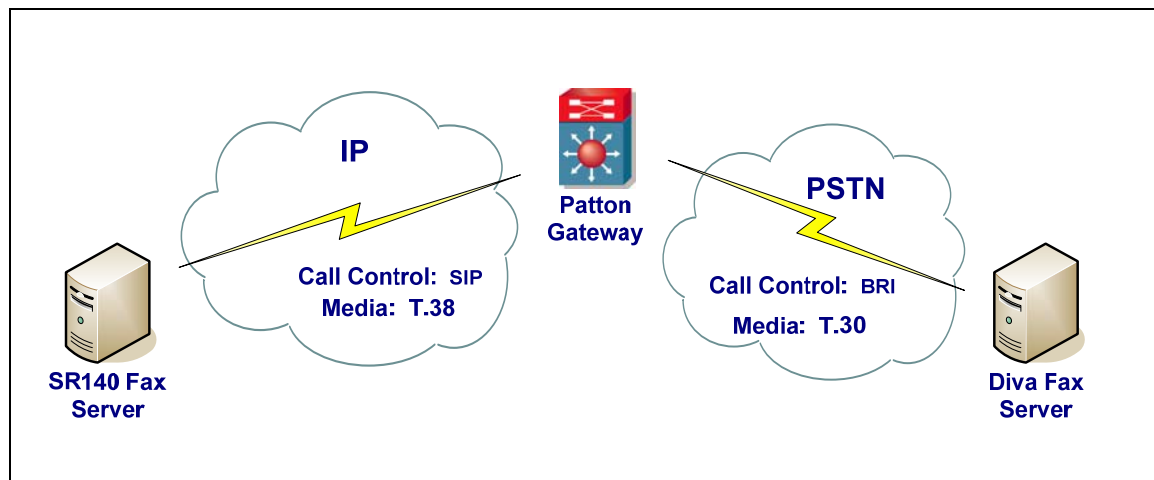
Vendor	<b>Dialogic</b>
Model	<b>Dialogic® Brooktrout® SR140 Fax Software</b>
Software Version	<b>SDK 6.1.2 build 11</b>
Protocol to Gateway or Call Manager	<b>SIP</b>
callctrl.cfg file	<b>Default callctrl.cfg file included in SDK 6.1.2</b>

## 2.3 Dialogic® Diva® BRI-2M Fax Board

Vendor	<b>Dialogic</b>
PSTN Device	<b>Dialogic® Diva® BRI-2M Fax Board</b>
Software Version	<b>SDK 5.5, Diva SU 8.5.7</b>
Protocol to PSTN Device	<b>Euro ISDN PRI and BRI</b>
callctrl.cfg file	<b>Standard setup</b>

## 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.
- Diva Fax Server = Fax Server including Dialogic® Diva® Fax Board and third party fax application.

### 3. Prerequisites

None.

### 4. Summary of Limitations

The Patton 4960 and 4554 gateways support up to V.17 and do not support V.34. SR140 was configured to initiate calls at V.17 as default during most of the tests. Some additional tests were performed with SR140 initiating calls at V.34 and the calls were negotiated to V.17 and successfully completed. All calls initiated and answered by the Diva BRI-2M fax board at V.34 and were negotiated to V.17 and successfully completed.

### 5. Deployment Details

#### 5.1 Network Addresses

Device #	Device Make, Model, and Description	Device IP Address
1	SR140 Fax Server	10.0.1.139
2	Patton 4960 PRI Gateway	10.0.1.14
3	Patton 4554 BRI Gateway	10.0.1.93

#### 5.2 Dialing Plan Overview

Dialing from SR140 was performed using a SIP URI phone\_number@Gateway\_IP\_Address. This was configured in the FoIP Interop test tool setup.

### 6. Dialogic® Brooktrout® SR140 Fax Software Setup Notes

Default setup for SR140 was used during the tests, varied by the interop test tool during the test runs.

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

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

### 7. Dialogic® Diva® BRI-2M Fax Board Setup Notes

Default setup for the Diva BRI-2M was used. Calls were terminated and generated by a Diva SDK fax sample application that had been modified to enable control of ECM and fax file to send. All calls were generated and answered at V.34 but negotiated down to V.17 by the Patton gateways.

## 8. Gateway Setup Notes

Patton 4960 PRI Configuration and Patton 4554 BRI configuration were identical apart from the Call-Routing Destination which depends on the individual call routes setup. The 4960 PRI configuration screens are shown.

### Remote User Agent Host Name/Port = SR140 Fax Server

CONFIGURATION MEN		Configuration	Incoming Call Address Translation	Outgoing Call Address Translation	Status	
Network IP/DNS NAT/NAPT ACL QoS Virtual Router DynDNS DHCP Server DHCP Relay VPN PPP Profiles Telephony Call-Router SIP VoIP Profiles Tone Profiles PSTN Profiles Ports Ethernet E1/T1 Various System AAA Time Reports Syslog Save Reload About License	SIP Gateway		<input checked="" type="checkbox"/>	GW_SIP_5060		
	Call-Routing Destination		<input checked="" type="checkbox"/>	<input type="radio"/> Interface (none) <input checked="" type="radio"/> Table SIP_to_PRI <input type="radio"/> Service (none)		
	Remote User Agent Host Name / Port		<input checked="" type="checkbox"/>	10.0.1.139	5060	
	Local User Agent Host Name / Port		<input type="checkbox"/>			
	Early Connect		<input type="checkbox"/>	Connect call when local terminal plays precall announcement		
	Early Disconnect		<input type="checkbox"/>	Release call when local terminal hangs up		
	Hold-Method			zero-ip		
	Call-Transfer		Accept:	<input checked="" type="checkbox"/>	Accepts REFER messages from the connected user agent	
			Emit:	<input checked="" type="checkbox"/>	Sends REFER messages to transfer internally looped calls	
			Pull-In:	<input checked="" type="checkbox"/>	Detects external call loops and connects intern through	
	Call-Reroute		Emit:	<input type="checkbox"/>	Sends 302 moved temporarily messages to reroute internally looped calls	
	Address-Complete Indication		Accept:	clear	Set always sets the address-complete indication; and clear never sets the address-complete indication.	
	Advice of Charge		AOC-D (Charge During The Call)	<input type="checkbox"/>	Accept (receive AOC-D from the remote SIP terminal and pass them to ISDN)	
				<input type="checkbox"/>	Emit (send AOC-D messages received from ISDN to the remote SIP terminal)	
	Privacy		<input type="checkbox"/>	Use the Identity-header for the Calling Party Number in addition to the From header. The handling of this header can be configured for incoming and outgoing direction separately.		
	Accept Address Update		<input type="checkbox"/>	wait-for-name	Proceeding Timeout [ms]	4000
					Alerting Timeout [ms]	0
	Overlap dialing		With new transaction	<input type="checkbox"/>	Accept (receive INVITE with updated called-user information from the remote SIP terminal and forward them)	
				<input type="checkbox"/>	Emit (send INVITE with updated called-user information received to the remote SIP terminal)	
	Penalty Box		<input type="checkbox"/>			
Use new session after redirect		<input type="checkbox"/>				
Session Timer		<input type="checkbox"/>	1800	seconds		
VoIP Profile			FAX_t38			
Tone Set Profile			default			
Sip Profile			SIP_Profile_TV			
					Apply	

### VoIP Profile T.38/Voice Tab

Voice		Fax	Modem	Dejitter Buffer	Status
<b>Voice Codecs</b>					
Position	Codec	Rx Length [ms]	Tx Length [ms]	Silence Suppression	
	1 g711alaw64k	<input type="text" value="20"/>	<input type="text" value="20"/>	<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no	
	2 g711ulaw64k	<input type="text" value="20"/>	<input type="text" value="20"/>	<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no	
<input type="text"/>	transparent	<input type="text"/>	<input type="text"/>	<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no	
<b>Additional Voice Parameters</b>					
Default Silence Suppression	<input type="checkbox"/> If not specified by the codec				
Highpass Filter	<input checked="" type="checkbox"/> Voice input filter for A/D conversion				
Post Filter	<input checked="" type="checkbox"/> Voice output filter for D/A conversion				
DTMF Relay	<input checked="" type="checkbox"/> <input checked="" type="radio"/> default <input type="radio"/> rtp <input type="radio"/> signaling <input type="text" value="default"/>				
Flash-hook Relay	<input type="radio"/> default <input type="radio"/> rtp <input type="radio"/> signaling <input type="text" value="default"/>				
RTP Payload Type For Tone Events (NTE)	<input type="text" value="101"/>				
RTP Payload Type For Signaling Events (NSE)	<input type="text" value="100"/>				
RTP Payload Type For Transparent Clearmode	<input type="text" value="97"/>				
RTP Payload Type For G.726-32	<input type="text" value="2"/>				
RTP Payload Type For G.726-32 Cisco Compatible	<input type="text" value="2"/>				
RTP Traffic Class	<input type="text" value="local-voice"/>				
					Apply

### VoIP Profile T.38/Fax Tab

Voice | **Fax** | Modem | Dejitter Buffer | Status

Fax Transmission Methods			
Position	Method	Protocol	
 	1	relay	t38-udp
		relay	t38-udp
		bypass	g711alaw64k


  

Additional Fax Parameters	
Fax Detection	ced-tone
Error Correction	<input checked="" type="checkbox"/>
Max Bitrate	14400  bps
HDLC Image Transfer	<input checked="" type="checkbox"/>
T.38 Redundancy	Low Speed <input type="text" value="0"/> additional packets High Speed <input type="text" value="0"/> additional packets
T.38 CED Retransmission	<input checked="" type="checkbox"/> Number of additional packets <input type="text" value="2"/>
T.38 No-Signal Retransmission	Number of packets (1 - 5) <input type="text" value="3"/>
T.38 Output Volume	-9.5
Dejitter Buffer Max Delay	<input type="text" value="200"/> milliseconds
Bypass Method	default
CED-Tone Network Side Detection	<input type="checkbox"/> Allow detection of fax/modem answer tones on the network (RTP) side

Apply



### VoIP Profile T.38/Dejitter Buffer Tab

<b>Voice</b>	<b>Fax</b>	<b>Modem</b>	<b>Dejitter Buffer</b>	<b>Status</b>
<b>General Parameters</b>				
Mode	adaptive ▾			
Max Delay	Voice	60	milliseconds	
	Fax	200	milliseconds	
				Apply ✓
<b>Adaptive Mode Parameters</b>				
 Improvident changes can cause buffer size oscillation and voice drops				
Packets Loss	4	Number of packets that must be lost before shrinking/growing the dejitter buffer		
Shrink Behaviour	Step	1	Number of packets the dejitter buffer shrinks	
Grow Behaviour	Step	1	Number of packets the dejitter buffer grows	
	Attenuation	1		
				Apply ✓

## 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.