Dialogic® Brooktrout® SR140 Fax Software with Linksys SPA8000 8-Port Telephony Gateway

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 **Linksys SPA8000** 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 **Linksys SPA8000**.

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 Linksys SPA8000 will be denoted herein as Linksys SPA8000 and SPA8000, 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	***Linksys***
Model	***SPA8000***
Software Version	***5.1.10***
Hardware Version	***1.0.0***
IP Device	***Dialogic® Brooktrout® SR140 Fax Software***
Protocol to Dialogic® Brooktrout® SR140 Fax Software	***SIP***
PSTN Device	***Dialogic® Brooktrout® TR1034 Analog Fax Board***
Protocol to PSTN Device	***Analog***
Additional Notes	***N/A***

2.2 Dialogic® Brooktrout® SR140 Fax Software

Vendor	***Dialogic***
Model	***Dialogic® Brooktrout® SR140 Fax Software***
Software Version	***SDK 6.2.0***
Protocol to Gateway or Call Manager	***SIP***
callctrl.cfg file	***All defaults***

2.3 Dialogic® Brooktrout® TR1034 Fax Board

Vendor	***Dialogic***
PSTN Device	***Dialogic® Brooktrout® TR1034 Fax Board***
Software Version	***SDK 6.2.0***
Protocol to PSTN Device	***Analog***
callctrl.cfg file	***All defaults***

2.4 Network System Configuration

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

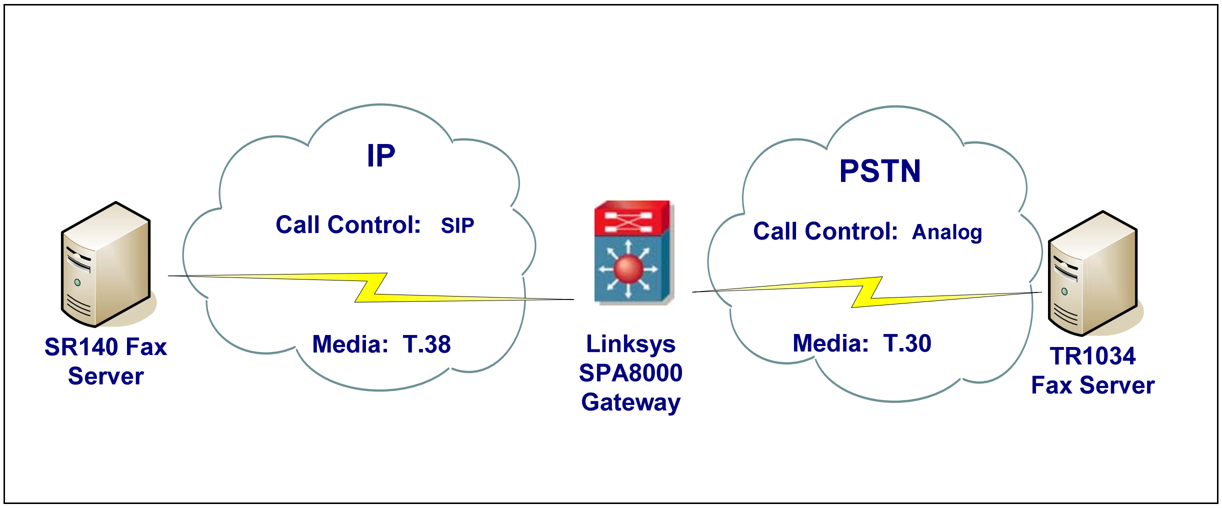


Diagram Notes:

- SR140 Fax Server = Fax Server including Dialogic® Brooktrout® SR140 Fax Software and third party fax application.
- TR1034 Fax Server = Fax Server including Dialogic® Brooktrout® TR1034 Fax Board and third party fax application.

3. Prerequisites

None.

4. Summary of Limitations

None.

5. Gateway Setup Notes

5.1 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	Linksys ASP8000 Gateway	10.128.30.230 (WAN IP Address)
2	Linksys ASP8000 Gateway	192.168.0.1 (LAN IP Address)
3	SR140	10.128.30.100

5.2 Router Configuration

Follow the quick installation guide below which ships with the SPA8000 gateway to configure the IP address that will be used for voice and fax calls, which is known as the WAN IP address.

http://www.cisco.com/en/US/docs/voice_ip_comm/csbpvga/spa8000/quick_start/guide/spa8000_quick.pdf

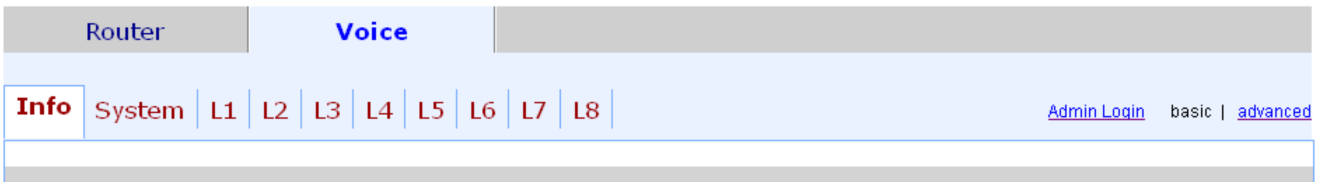
By default, the gateway’s web interface can be accessed by directing a web browser to URL http://192.168.0.1. 192.168.0.1 is the default LAN IP address. The WAN IP address can be configured in the ‘Wan Setup’ tab of the ‘Router’ menu. Click on the ‘Submit All Changes’ button once changes are made to the gateway’s configuration.

For a detailed explanation of all Router settings, see the SPA8000 Administration Guide below:

http://www.cisco.com/en/US/docs/voice_ip_comm/csbpvga/ata/administration/guide/ATA_AG_v3_NC-WEB.pdf

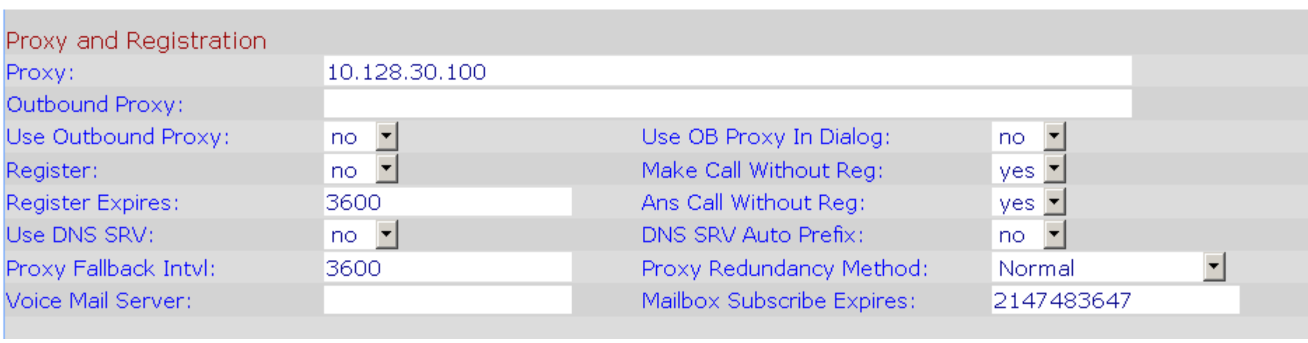
5.3 SIP Proxy and Registration Configuration

Once the WAN IP address is configured, click on the ‘Voice’ tab and obtain administrative privileges by clicking on the ‘Admin Login’ hyperlink on the right side of the web page. Click on the ‘advanced’ hyperlink next to expose all telephony configuration options.



Click of the ‘L1’ tab and scroll down to the ‘Proxy and Registration’ section to configure the first analog line. Each ‘L#’ tab configures a corresponding analog line.

If the SPA8000 is required to register with a Proxy Server, the ‘Proxy’ field needs to be filled in with the IP address or FQDN of the Proxy Server and the ‘Register’ drop-down list needs to be set to ‘yes’. If a Proxy Server is not required, the ‘Proxy’ field can be filled in with the IP address or FQDN of the server that will receive all outgoing calls originating from the first analog line and the ‘Register’ drop-down list should be set to ‘no’. In the following screenshot, the ‘Proxy’ field contains the IP address of the SR140 Fax Server that was used to perform interoperability testing. The ‘Make Call Without Reg:’ and ‘Ans Call Without Reg’ drop down lists are set to ‘yes’ since a Proxy Server was not used.



If the SPA8000 is required to provide a user id and password to the Proxy Server, the ‘User ID’ and ‘Password’ fields of the ‘Subscriber Information’ section fields need to be filled in. The ‘User ID’ field also represents the user part of the SIP URI, which is used by the gateway to route incoming VoIP calls. For example, if the ‘User ID’ field of the first analog line is filled in with the text ‘joe’ and an IP call destined for sip:joe@gatewayaddr.com is received by the SPA8000 gateway, the first analog line will ring. If a Proxy Server is not used, only the ‘User ID’ field is needed to route SIP calls to the appropriate analog line.

5.4 Dialing Plan Overview

5.4.1 VoIP to POTS

The ‘User ID’ field of the ‘L#’ tab represents the user part of the SIP URI, which is used by the gateway to route incoming VoIP calls. For example, if the ‘User ID’ field of the first analog line is filled in with the text ‘joe’ and an IP call destined for sip:joe@gatewayaddr.com is received by the SPA8000 gateway, the first analog line will ring.

5.4.2 POTS to VoIP

The ‘Dial Plan’ field of the of the ‘L#’ tab has to be filled in to make outbound VoIP calls. The ‘Enable IP Dialing’ drop down list also has to be set to ‘yes’ to allow outbound VoIP calls. For example, setting this field to the text ‘11’ of the ‘L1’ tab will make the SPA8000 gateway generate a SIP call using SIP URI sip:11@proxyaddr.com after the first analog line dials the number 11. For all allowed dial plan patterns, see the SPA8000 Administration Guide.

5.5 Call Routing Configuration

5.5.1 VoIP to POTS

The following screenshot uses the text ‘1’ as the ‘User ID’ of ‘L1’ tab; thus, a SIP call destined for sip:1@10.128.30.230 rings the first analog line.

Subscriber Information			
Display Name:	1st line of Linksys SPA8	User ID:	1
Password:		Use Auth ID:	no
Auth ID:			
Mini Certificate:			
SRTP Private Key:			

5.5.2 POTS to VoIP

The following screenshot uses the text ‘1’ as the ‘Dial Plan’ of the ‘L1’ tab; thus, when the first analog line dials the number 1, an outbound SIP call is generated using SIP URI sip:1@10.128.30.100.

Dial Plan			
Dial Plan:	1		
Enable IP Dialing:	yes	Emergency Number:	

6. Dialogic® Brooktrout® SR140 Fax Software Setup Notes

The Installation and Configuration Guides used to setup the SR140 is available from the site:

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

The default setup for SR140 and the 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_0004_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=14400
  t38_fax_version=0
  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_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 pcma
  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=0.0.0.0:0
  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=Anonymous <sip:no_from_info@anonymous.invalid>
  sip_Contact=0.0.0.0:0
  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_interface=
  sip_ip_interface_port=5060
  sip_redirect_as_calling_party=0
  sip_redirect_as_called_party=0
  sip_user_agent=Brktsip/6.2.0B5 (Dialogic)
[module.41]
  model=SR140
  virtual=1
  exists=1
  vb_firm=C:\fdtool-6.2.0\bin\bostvb.dll
  channels=120
[module.41/ethernet.1]
  ip_interface={22EE3B90-BC80-4F06-9076-E64284265B67}:0
  media_port_min=56000
  media_port_max=57000
[module.41/host_cc.1]
  host_module=1
  number_of_channels=120
```

7. Dialogic® Brooktrout® TR1034 Fax PSTN 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.

8. 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.