



Avaya Solution & Interoperability Test Lab

Application Notes for OpenText RightFax with Avaya Aura[®] Communication Manager and Avaya Aura[®] Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate OpenText RightFax with Avaya Aura[®] Communication Manager and Avaya Aura[®] Session Manager. OpenText RightFax is a fax server solution that interfaces with Avaya Aura[®] Session Manager via a SIP trunk and supports T.38 and G.711 pass-through for fax calls.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to integrate OpenText RightFax with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. OpenText RightFax is a fax server solution that interfaces with Avaya Aura® Session Manager via a SIP trunk and supports T.38 and G.711 pass-through for fax calls. OpenText RightFax uses the Dialogic® Brooktrout SR140 Fax Software as its Fax over IP (FoIP) engine.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on sending and receiving fax calls between RightFax and fax software connected to the PSTN and local to Communication Manager using T.38 and G.711 pass-through.

The serviceability testing focused on verifying that RightFax came back into service after re-connecting the Ethernet cable (i.e., restoring network connectivity and rebooting the RightFax server. In addition, it as verifies that any interrupted faxes were successfully re-sent.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and OpenText RightFax did not include use of any specific encryption features as requested by OpenText.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing focused on verifying the following:

- Establishing a SIP trunk between Session Manager and RightFax and verifying the proper response by RightFax to SIP OPTIONS.
- Proper handling of faxes via the SIP trunk including sending and receiving internal fax (i.e., between two RightFax users), local fax (i.e., fax software connected to Communication Manager via analog line), external fax to the PSTN over ISDN-PRI, simultaneous bi-directional faxes, and various failure scenarios, such as fax call to invalid number and busy endpoint.
- Verifying that any interrupted faxes were retried and successfully sent by RightFax based on its configured retry strategy.
- Proper handling of faxes with different number of pages, complexity, format and data rates.

The serviceability testing focused on verifying the ability of RightFax to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to the RightFax server and restarting RightFax.

2.2. Test Results

The compliance test passes with following observation:

- The maximum bit rate supported by the G450 Media Gateway for T.38 fax is 9600 bps (V.29). Refer to [3] for additional information.

2.3. Support

For technical support on OpenText RightFax, contact OpenText Customer Support at:

- Phone: + 1 (800) 540-7292
- Website: <https://www.opentext.com/support>
- Email: support@opentext.com

3. Reference Configuration

Figure 1 illustrates a sample configuration consisting of OpenText RightFax with an Avaya SIP-based network, including Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and Avaya G450 Media Gateway. RightFax connected to Session Manager via a SIP trunk and calls were routed through Communication Manager, except for fax calls between two RightFax users. The G450 Media Gateway provided media resources for the fax calls and supported T.38 and G.711 relay/pass-through. For T.38 fax calls, the G450 Media Gateway supported a maximum bit rate of 9600 bps (V.29). Fax calls were made from fax software connected to the PSTN and connected directly to a G450 Media Gateway via an analog line. Fax calls were routed to RightFax using fax extensions assigned to each user account in RightFax.

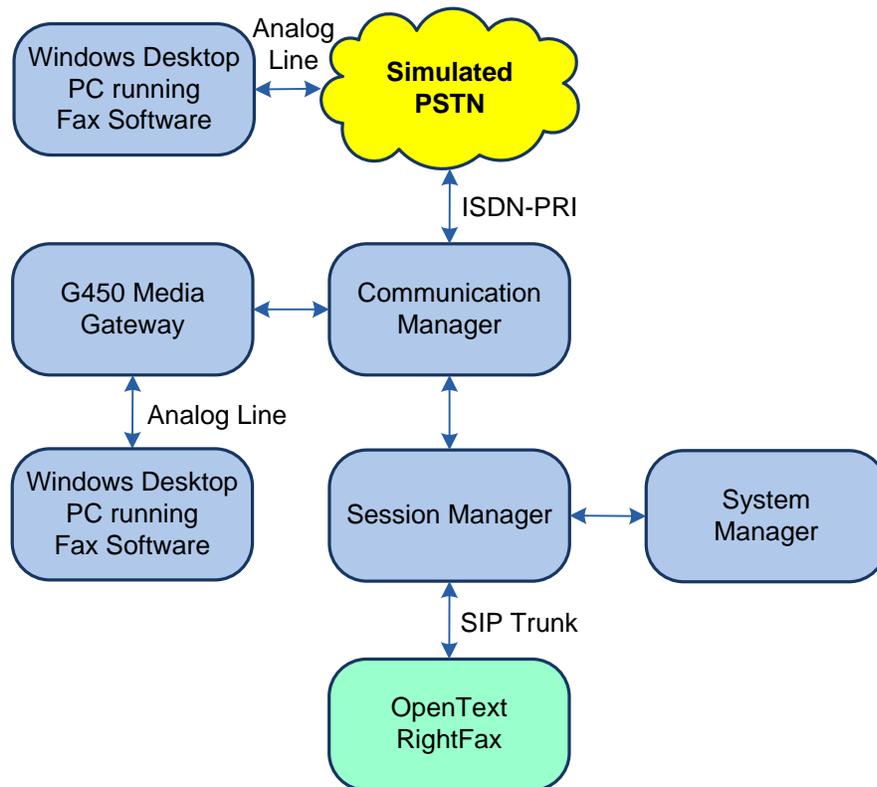


Figure 1: OpenText RightFax with Avaya SIP-based Network

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	8.1.2.0.0-FP2
Avaya G450 Media Gateway	FW 41.24.0
Avaya Aura® System Manager	8.1.2.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.2.0.0611167 Feature Pack 2
Avaya Aura® Session Manager	8.1.2.0.812039
OpenText RightFax with Dialogic® Brooktrout SR140 Fax Software	20.2.0.846 6.5.0.14

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring a SIP trunk to Session Manager and routing calls to RightFax. Administration of Communication Manager was performed using the System Access Terminal (SAT).

This section covers the following configuration:

- **IP Node Names** to associate names with IP addresses.
- **IP Codec Set** to specify the codec type and fax mode used for fax calls to RightFax.
- **IP Network Region** to specify the domain name and the IP codec set.
- **SIP trunk** for calls towards Session Manager and RightFax.
- **Private Numbering** to allow the caller's extension to be sent to RightFax.
- **Call Routing** to route fax calls to RightFax using AAR.

5.1. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                     Page 1 of 2
                                                    IP NODE NAMES
      Name                IP Address
default                 0.0.0.0
devcon-aes              10.64.102.119
devcon-ams              10.64.102.118
devcon-sm              10.64.102.117
procr                  10.64.102.115
procr6                  ::
( 6 of 6 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.2. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to RightFax. The form is accessed via the **change ip-codec-set 1** command. Note the codec set number since it will be used in the IP Network Region covered in the next section. For the compliance test, G.711MU codec was used.

```
change ip-codec-set 1                                     Page 1 of 2

                                IP MEDIA PARAMETERS

Codec Set: 1

Audio          Silence      Frames      Packet
Codec          Suppression  Per Pkt    Size (ms)
1: G.711MU      n                2          20
2:
3:
4:
5:
6:
7:

Media Encryption                                Encrypted SRTCP: best-effort
1: 1-srtp-aescm128-hmac80
2: none
3:
4:
5:
```

On **Page 2**, set the **FAX Mode** to **t.38-standard**. Default values may be used for other fields.

```
change ip-codec-set 1                                     Page 2 of 2

                                IP MEDIA PARAMETERS

                                Allow Direct-IP Multimedia? y
                                Maximum Call Rate for Direct-IP Multimedia: 4096:Kbits
                                Maximum Call Rate for Priority Direct-IP Multimedia: 4096:Kbits

FAX          Mode          Redun-          Packet
Modem        t.38-standard  dancy          Size (ms)
TDD/TTY      off            0              ECM: y
H.323 Clear-channel  US            3
SIP 64K Data  n              0              20

Media Connection IP Address Type Preferences
1: IPv4
2:
```

5.3. Administer IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between audio IP endpoints without using media resources in the Avaya G450 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

```
change ip-network-region 1                                     Page 1 of 20
                                                           IP NETWORK REGION
  Region: 1
Location: 1          Authoritative Domain: avaya.com
  Name:              Stub Network Region: n
MEDIA PARAMETERS    Intra-region IP-IP Direct Audio: yes
  Codec Set: 1      Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 2048      IP Audio Hairpinning? n
  UDP Port Max: 50999
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 46
  Audio PHB Value: 46
  Video PHB Value: 26
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 6
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
```

5.4. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Specify the Ethernet processor (*procr*) of Communication Manager and Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form in **Section 5.1**.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                     Page 1 of 2
                                                         SIGNALING GROUP
Group Number: 10                                         Group Type: sip
IMS Enabled? n                                           Transport Method: tls
  Q-SIP? n
  IP Video? n                                           Enforce SIPS URI for SRTP? n
Peer Detection Enabled? y Peer Server: SM                 Clustered? n
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
Near-end Node Name: procr                               Far-end Node Name: devcon-sm
Near-end Listen Port: 5061                              Far-end Listen Port: 5061
                                                         Far-end Network Region: 1
Far-end Domain: avaya.com
                                                         Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                    RFC 3389 Comfort Noise? n
  DTMF over IP: rtp-payload                             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
  Enable Layer 3 Test? y                               Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n                 Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for fax calls to RightFax. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

```
add trunk-group 10                                     Page 1 of 5
                                     TRUNK GROUP
Group Number: 10                                     Group Type: sip                                     CDR Reports: y
  Group Name: To devcon-sm                             COR: 1                                     TN: 1                                     TAC: 1010
  Direction: two-way                                   Outgoing Display? n
Dial Access? n                                         Night Service:
Queue Length: 0
Service Type: tie                                   Auth Code? n
                                                    Member Assignment Method: auto
                                                    Signaling Group: 10
                                                    Number of Members: 10
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 10                                     Page 3 of 5
TRUNK FEATURES
  ACA Assignment? n                                     Measured: none
                                                    Maintenance Tests? y
  Suppress # Outpulsing? n Numbering Format: private
                                                    UUI Treatment: service-provider
Maximum Size of UUI Contents: 128
  Replace Restricted Numbers? n
  Replace Unavailable Numbers? n
                                                    Hold/Unhold Notifications? y
Modify Tandem Calling Number: no
Show ANSWERED BY on Display? y
```

On **Page 5** of the trunk group form, the default settings were used as shown below.

```

add trunk-group 10
                                Page 5 of 5
                                PROTOCOL VARIATIONS
                                Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                                Send Transferring Party Information? n
                                Network Call Redirection? n
                                Send Diversion Header? n
                                Support Request History? y
                                Telephone Event Payload Type:
                                Convert 180 to 183 for Early Media? n
                                Always Use re-INVITE for Display Updates? n
                                Identity for Calling Party Display: P-Asserted-Identity
Block Sending Calling Party Location in INVITE? n
                                Accept Redirect to Blank User Destination? n
                                Enable Q-SIP? N
                                Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
  
```

5.5. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with ‘7’ whose calls are routed over any trunk group, including SIP trunk group 10, have the extension sent to RightFax.

```

change private-numbering 0
                                Page 1 of 2
                                NUMBERING - PRIVATE FORMAT
Ext  Ext      Trk      Private      Total
Len  Code      Grp(s)     Prefix      Len
5   7
                                Total Administered: 1
                                Maximum Entries: 540
  
```

5.6. AAR Call Routing

Configure the uniform dial plan table to route calls using AAR for dialed digits that are 5-digits long and begin with ‘78’. This would cover call routing to RightFax extensions (i.e., 78771 – 78722).

```

change uniform-dialplan 7
                                Page 1 of 2
                                UNIFORM DIAL PLAN TABLE
                                Percent Full: 0
Matching      Insert      Node
Pattern      Digits     Net Conv Num
78           5 0       aar  n
  
```

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and add an entry that routes digits beginning with “78” to route pattern 10 as shown below. Note that the **Call Type** was set to *lev0*. This routes calls to RightFax.

```
change aar analysis 7
```

Page 1 of 2

AAR DIGIT ANALYSIS TABLE
Location: all Percent Full: 2

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
7	7	7	254	aar		n
78	5	5	10	lev0		n
8	7	7	254	aar		n
9	7	7	254	aar		n
						n
						n

Configure a preference in **Route Pattern 10** to route calls over SIP trunk group 10 as shown below.

```
change route-pattern 10
```

Page 1 of 3

Pattern Number: 10 **Pattern Name: To devcon-sm**

SCCAN? n Secure SIP? n Used for SIP stations? n

Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Digits	DCS/ IXC QSIG Intw
1:	10	0						n user
2:								n user
3:								n user
4:								n user
5:								n user
6:								n user

	BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	Sub	Numbering	LAR
	0	1	2	M	4	W	Request		Dgts	Format	
1:	y	y	y	y	y	n	n			unk-unk	none
2:	y	y	y	y	y	n	n				none
3:	y	y	y	y	y	n	n				none
4:	y	y	y	y	y	n	n				none
5:	y	y	y	y	y	n	n				none
6:	y	y	y	y	y	n	n				none

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP Entity for RightFax
- Entity Link, which defines the SIP trunk parameters used by Session Manager when routing calls to/from RightFax
- Routing Policies and Dial Patterns
- Session Manager, corresponding to the Avaya Aura® Session Manager Server to be managed by Avaya Aura® System Manager

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager using the URL “https://<ip-address>/SMGR”, where <ip-address> is the IP address of Avaya Aura® System Manager. Log in with the appropriate credentials.

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of the SIP entity, entity link, and call routing for RightFax.

6.1. Add SIP Entity for RightFax

In the sample configuration, a SIP trunk was configured for RightFax.

A SIP Entity must be added for RightFax. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of RightFax.
- **Type:** Select *SIP trunk*.
- **Location:** Select the location defined previously (not shown).
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The main content area is titled 'SIP Entity Details' and is divided into sections: 'General', 'Loop Detection', and 'Monitoring'. The 'General' section contains the following fields:

- Name:** RightFax
- FQDN or IP Address:** 10.64.102.102
- Type:** SIP Trunk
- Notes:** (empty)
- Adaptation:** (empty)
- Location:** (empty)
- Time Zone:** America/New_York
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)
- Securable:**
- Call Detail Recording:** egress

The 'Loop Detection' section contains:

- Loop Detection Mode:** On
- Loop Count Threshold:** 5
- Loop Detection Interval (in msec):** 200

The 'Monitoring' section contains:

- SIP Link Monitoring:** Use Session Manager Configuration
- CRLF Keep Alive Monitoring:** Use Session Manager Configuration

6.2. Add Entity Link for RightFax

This section covers the configuration of an Entity Link for RightFax. This entity link will specify the SIP entity configured in **Section 6.1**.

The SIP trunk from Session Manager to RightFax is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *RightFax Link*).
- **SIP Entity 1:** Select Session Manager.
- **Protocol:** Select UDP transport protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the RightFax entity configured in **Section 6.1**.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Select *Trusted*. *Note: If Trusted is not selected, calls from the associated SIP Entity specified in Section 6.1 will be denied.*

Click **Commit** to save the Entity Link definition.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and various menu items like 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are also visible. The main content area is titled 'Entity Links' and features a table with one entry:

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port
<input type="checkbox"/>	* RightFax Link	* devcon-sm	UDP	* 5060	* RightFax	* 5060

Below the table, there is a 'Select : All, None' option. The interface also includes 'Commit' and 'Cancel' buttons at the top right and bottom right of the main content area.

6.3. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the RightFax SIP Entity specified in **Section 6.1**. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select** and then select the appropriate SIP entity to which this routing policy applies. In this case, the RightFax SIP entity is selected.

Defaults can be used for the remaining fields. Click **Commit** to save the Routing Policy definition. The following screen shows the Routing Policy for RightFax.

The screenshot displays the Avaya Aura System Manager 8.1 interface for configuring a Routing Policy. The left sidebar shows the navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes 'Commit' and 'Cancel' buttons. The 'General' section contains the following fields:

- Name:** RightFax Policy
- Disabled:**
- Retries:** 0
- Notes:** RightFax Users

The 'SIP Entity as Destination' section features a 'Select' button and a table with the following data:

Name	FQDN or IP Address	Type	Notes
RightFax	10.64.102.102	SIP Trunk	

The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons, along with a 'Filter: Enable' option. It displays a table with one item:

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7						

At the bottom, there is a 'Select : All, None' option.

6.5. Add Session Manager

Adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

Under *General*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager

Use default values for the remaining fields. Click **Commit** to add this Session Manager.

The screenshot displays the 'Edit Session Manager' configuration interface in Avaya Aura System Manager 8.1. The interface is divided into two main sections: 'General' and 'Security Module'. The 'General' section contains the following fields: 'SIP Entity Name' (devcon-sm), 'Description' (empty), '*Management Access Point Host Name/IP' (10.64.102.116), '*Direct Routing to Endpoints' (Enable), 'Data Center' (None), 'Avaya Aura Device Services Server Pairing' (None), and 'Maintenance Mode' (unchecked). The 'Security Module' section contains: 'SIP Entity IP Address' (10.64.102.117), '*Network Mask' (255.255.255.0), '*Default Gateway' (10.64.102.1), '*Call Control PHB' (46), and '*SIP Firewall Configuration' (SM 6.3.8.0). The page includes a dark header with the Avaya logo, user information (admin), and navigation menus. A left sidebar shows the configuration tree with 'Session Manager Administration' selected. Buttons for 'Commit' and 'Cancel' are visible at the top right of the configuration area.

The following screen shows the **Monitoring** section, which determines how frequently Session Manager sends SIP Options messages to RightFax. Use default values for the remaining fields. Click **Commit** to add this Session Manager. In the following configuration, Session Manager sends a SIP Options message every 900 secs. If there is no response, Session Manager will send a SIP Options message every 120 secs.

Monitoring ▾

Enable SIP Monitoring

*Proactive cycle time (secs)

*Reactive cycle time (secs)

*Number of Tries

*Number of Successes

Enable CRLF Keep Alive Monitoring

*CRLF Ping Interval (secs)

7. Configure OpenText RightFax

This section covers the configuration of OpenText RightFax, including the Dialogic® Brooktrout SR140 Fax Software, using the RightFax Enterprise Fax Manager (EFM) and Brooktrout Configuration Tool. For additional information on configuring RightFax, refer to [7]. This section covers the following procedures:

- Launch RightFax Enterprise Fax Manager and Brooktrout Configuration Tool
- Configure SIP IP Parameters
- Configure T.38 Parameters
- Configure RTP Parameters
- Configure Call Control Parameters
- Configure Other RightFax Options
- Add RightFax Users
- Restart RightFax Services

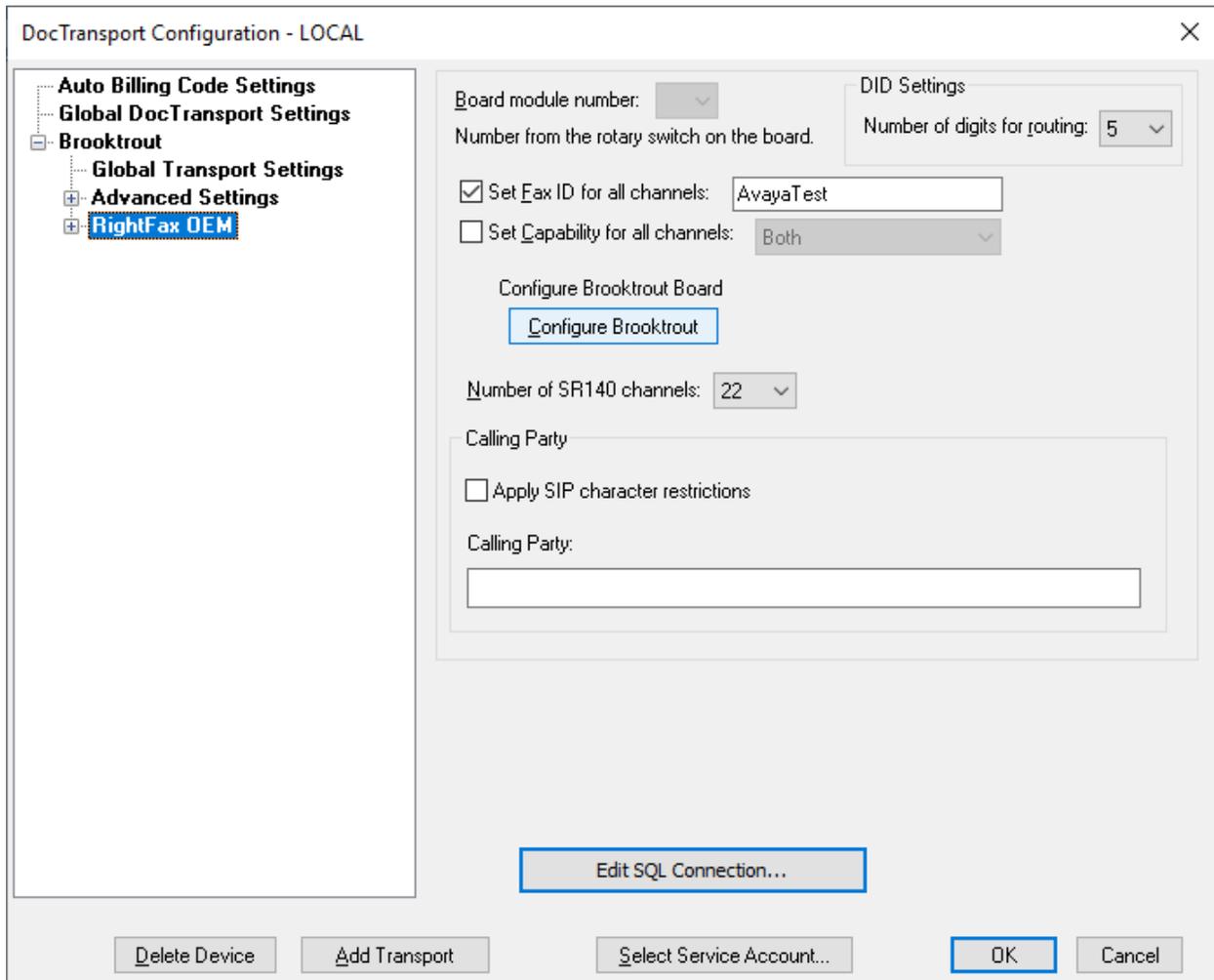
7.1. Launch RightFax Enterprise Fax Manager and Brooktrout Configuration Tool

Launch RightFax Enterprise Fax Manager (EFM) to configure RightFax. In the EFM, expand the server, which is **R1-1** in this case, and select **Services** in the left pane. The list of RightFax services is displayed in the right pane as shown below. Note that the services have already been started. Double-click on **RightFax DocTransport Module** to open the **DocTransport Configuration** window.

Service Name	Status	Running Time (ddd:hh:mm:ss)	Startup
RightFax DocTransport Module	Running	0008:20:36:52	Automatic
RightFax Server Module	Running	0008:22:30:51	Automatic
RightFax Database Module	Running	0008:20:33:22	Manual
RightFax RPC Server Module	Running	0008:17:31:01	Automatic
RightFax Queue Handler	Running	0008:17:57:17	Automatic
RightFax WorkServer1 Module	Running	0008:22:30:56	Manual
RightFax WorkServer2 Module	Running	0008:22:30:55	Manual
RightFax WorkServer3 Module	Running	0008:22:30:52	Manual
RightFax Email Gateway Module	Not configured	N/A	
RightFax SAP Gateway1 Module	Not configured	N/A	
RightFax Integration Module	Not configured	N/A	
RightFax eTransport Module	Running	0008:22:31:00	Manual
RightFax AutoReply Module	Not configured	N/A	
RightFax Remoting Service	Running	0008:22:31:00	Automatic
RightFax Sync Module	Running	0008:22:30:59	Automatic
RightFax Conversion Engine	Running	0008:22:30:59	Automatic
RightFax Licensing Service	Running	0009:20:47:06	Automatic
RightFax EWS Connector	Not configured	N/A	
RightFax SharePoint Gateway Module	Not configured	N/A	
RightFax Worker Host	Running	0008:22:30:59	Automatic

20 services last updated at 6/26/2020 8:57:3

The **DocTransport Configuration** window is shown below. Expand **Brooktrout** in the left pane and select **RightFax as OEM**. Next, click on **Configure Brooktrout**.



Log into the Brooktrout Configuration Tool using the appropriate access credentials as shown below.

Account access information

The RightFax OEM service account must have administrative user rights on the computer that runs the service.

Enter the Username and Password of the account with which the service will log on.

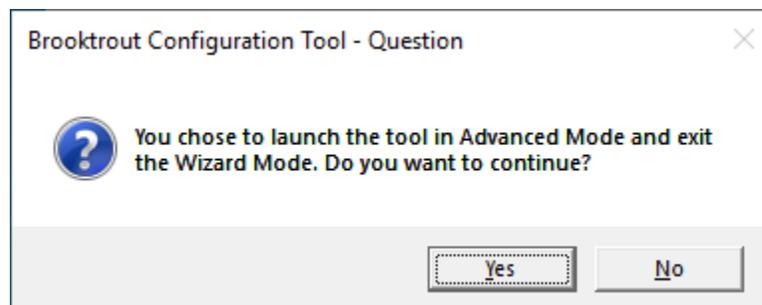
Username:

Password:

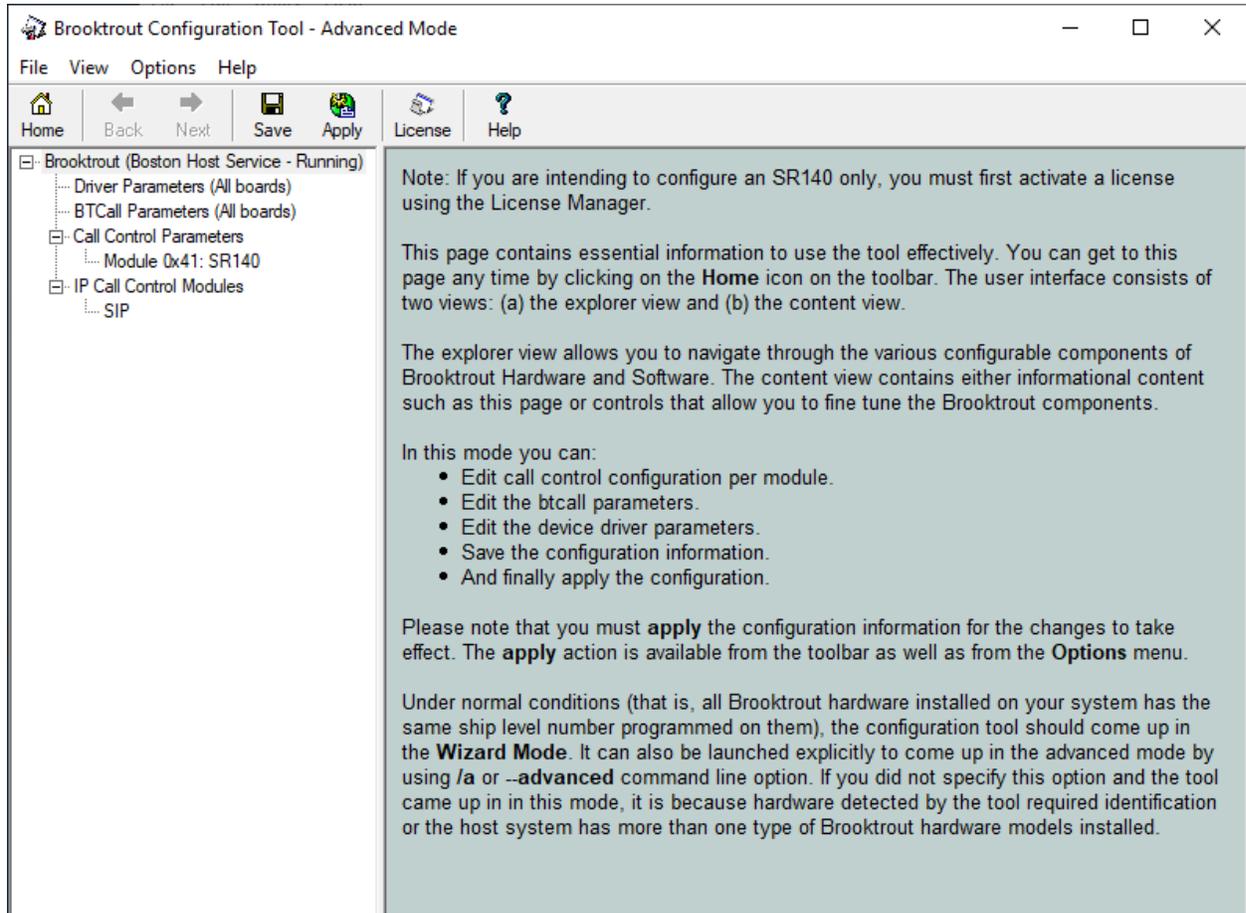
The **Brooktrout Configuration Tool – Wizard Mode** window is displayed. Click **Advanced Mode**.



Click **Yes** when prompted to launch the Configuration Tool in Advanced mode.



Brooktrout Configuration Tool – Advanced Mode window is displayed as shown below.



7.2. Configure SIP IP Parameters

In the **Brooktrout Configuration Tool – Advanced Mode** window, navigate to **Brooktrout → IP Call Control Modules → SIP** in the left pane and select the **IP Parameters** tab in the right pane. For the **Primary Gateway**, specify the Session Manager signaling IP address and port (e.g., *10.64.102.117* and *5060*) for the SIP trunk and set **From Value** to the desired value. The **From Value** will be specified in the From header of SIP messages. The default values for the other fields may be used.

The screenshot shows the 'Brooktrout Configuration Tool - Advanced Mode' window. The left pane displays a tree view with the following structure:

- Brooktrout (Boston Host Service - Running)
 - Driver Parameters (All boards)
 - BTCall Parameters (All boards)
 - Call Control Parameters
 - Module 0x41: SR140
 - IP Call Control Modules
 - SIP**

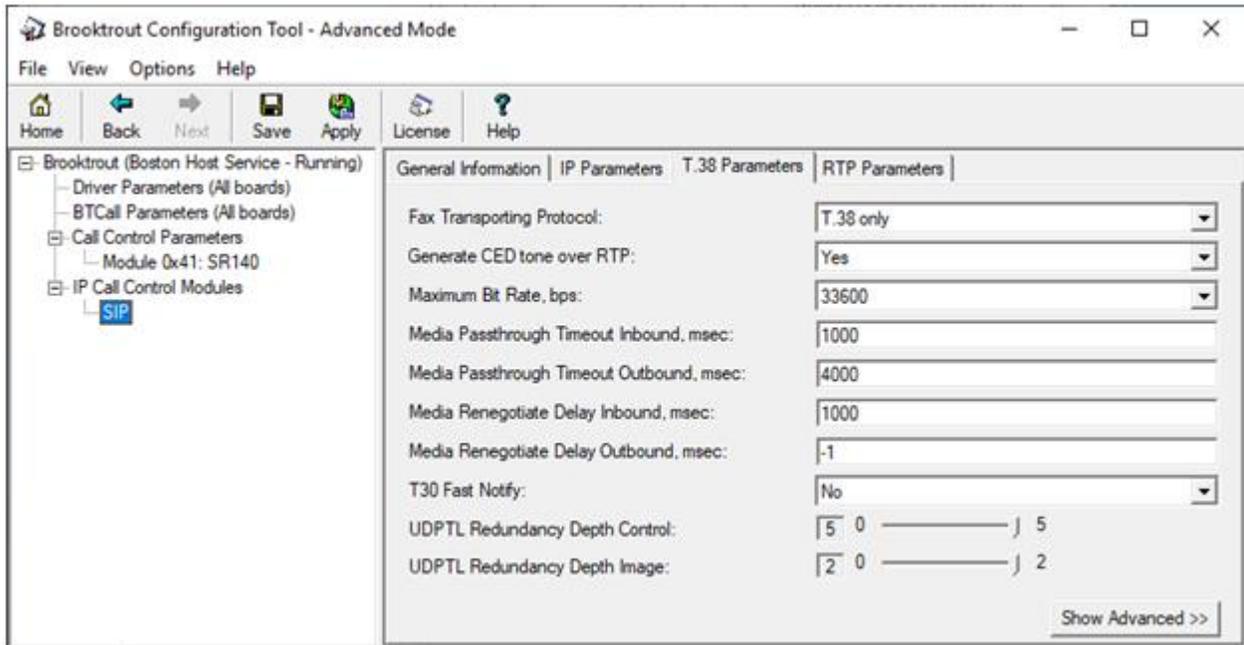
The right pane is titled 'IP Parameters' and contains the following configuration fields:

Field	Value
Maximum SIP Sessions:	256
Primary Gateway:	10.64.102.117 :5060
Additional SIP Gateway #2:	:0
Additional SIP Gateway #3:	:0
Additional SIP Gateway #4:	:0
Primary Proxy Server:	:0
Additional Proxy Server #2:	:0
Additional Proxy Server #3:	:0
Additional Proxy Server #4:	:0
Primary Registrar Server URL:	:0
Additional Registrar Server #2:	:0
Additional Registrar Server #3:	:0
Additional Registrar Server #4:	:0
From Value:	RightFax <sip:no_from_info@10.64.102.102>
Contact IPv4 Address:	
Username:	-
Session Name:	no_session_name
Session Description:	
Description URI:	
Email Address:	
Phone Number:	

A 'Show Advanced >>' button is located at the bottom right of the configuration area.

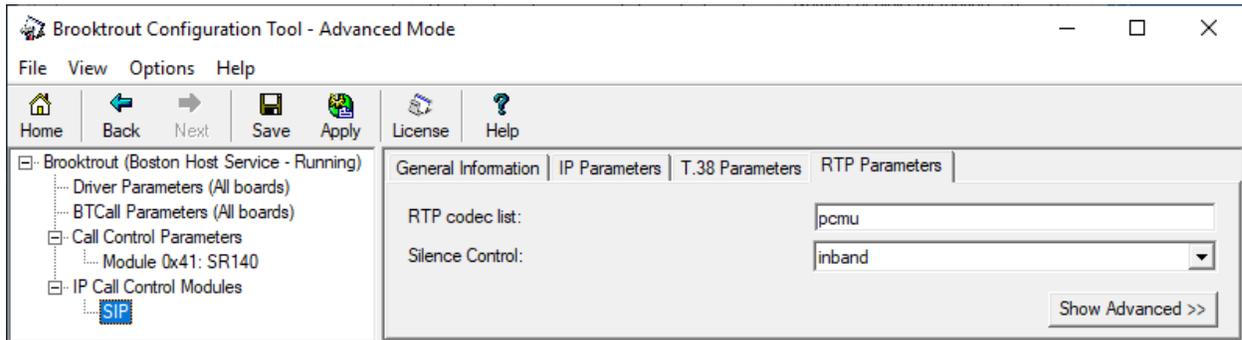
7.3. Configure T.38 Parameters

In the **T.38 Parameters** tab, set the Fax Transporting Protocol to T.38 only and the Maximum Bit Rate, bps to the appropriate value. Note that the G450 Media Gateway supports a maximum bit rate of 9600 bps for T.38 fax.



7.4. Configure RTP Parameters

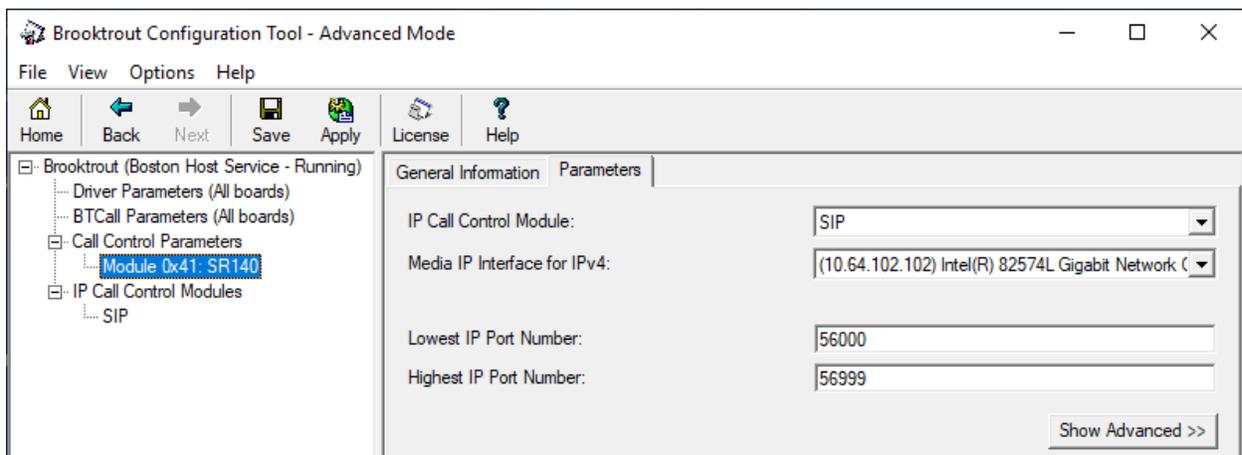
In the **RTP Parameters** tab, **RTP codec list** was set to *pcmu*.



7.5. Configure Call Control Parameters

Navigate to **Brooktrout** → **Call Control Parameters** → **Module 0x41: SR140** in the left pane. Ensure the following settings in the **Parameters** tab are correct for your environment.

- **IP Call Control Module:** SIP
- **Media IP Interface for IPv4:** Ensure this field is set to the network interface used to communication with Session Manager.
- **Lowest/Highest IP Port Numbers:** Ensure the RTP range is within the port range supported by the Avaya SIP infrastructure. *By default, the port range for SR140 is 56000 to 56999. A maximum range of 1000 ports may be specified.*



7.6. Configure Other RightFax Options

In EFM, double-click on **RightFax Server Module** to display the **Server Configuration** window displayed below. Select the **General** tab. **High Resolution** may be enabled and the **Retry Settings** may be modified. The **Retry Settings** specify the retry strategy for interrupted fax calls.

Server Configuration - LOCAL

User Messages **Admin Messages** **Data Sharing** **eTransport**

General **Advanced** **SQL Connections** **Feature Activation** **Custom Messages**

Enable CSID Routing

Enable Quick Headers

Create new user when printing to the fax queue

Disable routing of faxes with errors

Event LogLevel: Terse

Delay all Faxes Until: None Set

Auto delete failed gateway faxes

Enable Shared Services client failover

Enable Notifications

Require strong password

Enable Secure TCP/IP communication

Enable High Resolution

FaxUtil Timeout

0 minutes (0 = no timeout)

Reminder Notifications

For faxes not viewed or printed:

Send reminders every: 6 minutes

Reminders will be sent in this sequence:

User: Duration 60 minutes

Group Monitor: Duration 60 minutes

Alternate Group Monitor: Duration 60 minutes

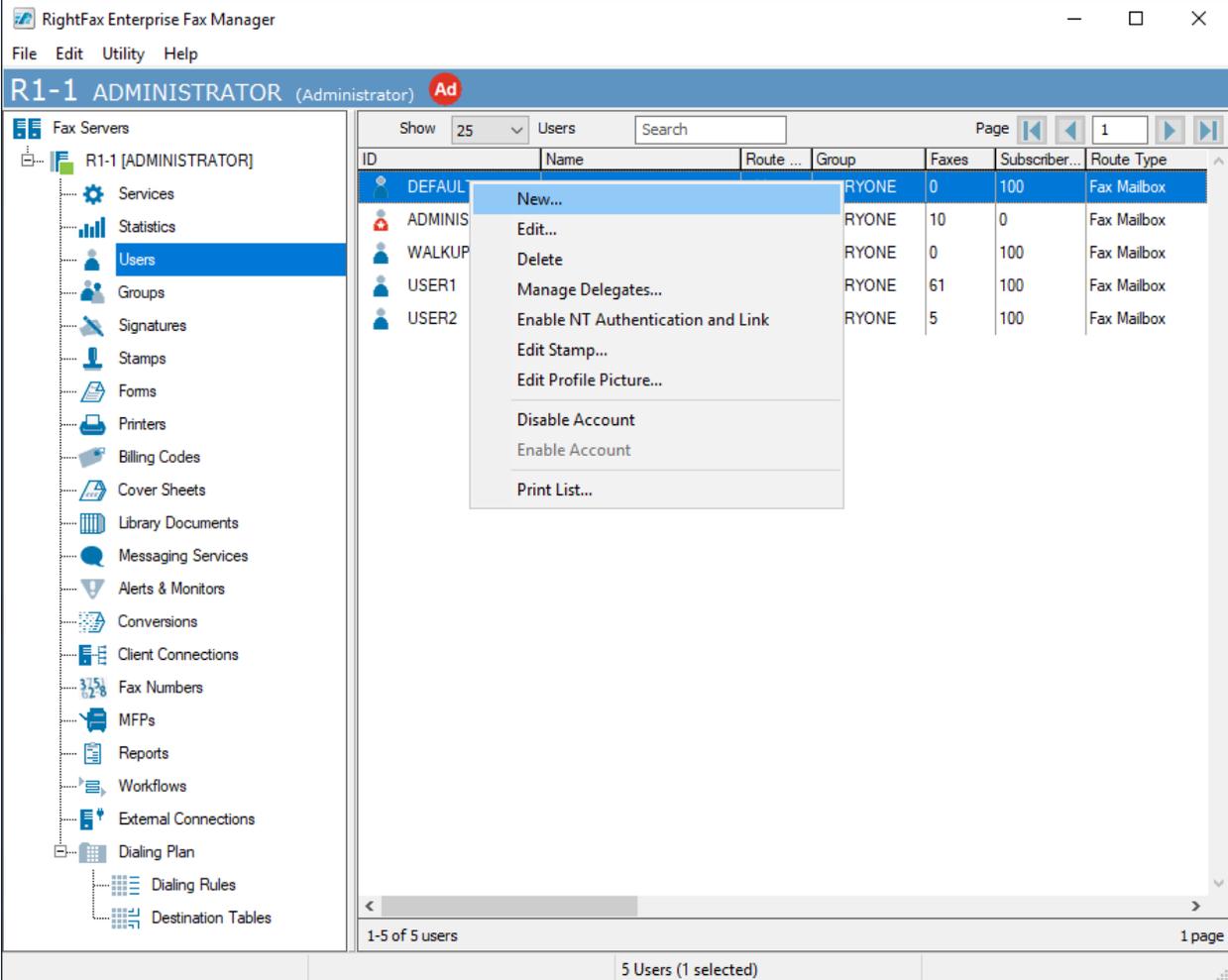
Retry Settings

	Count:	Interval (in minutes):
Busy:	5	5
Human Answered:	5	5
Special Information Tone:	5	5
No Loop/Errors:	5	5
Other:	5	5
SMS:	5	5

OK Cancel

7.7. Add RightFax Users

A user is created on the RightFax server and associated with a fax number. From a user's account, a fax may be received or sent. To view the list of users, select **Users** in the left pane of EFM. To add a new user, right-click on one of the users in the right pane and select **New** from the pop-up menu as shown below.



The screenshot displays the RightFax Enterprise Fax Manager interface. The left-hand navigation pane shows a tree view with 'Users' selected. The main window title is 'R1-1 ADMINISTRATOR (Administrator) Ad'. The 'Users' section is active, showing a table of users. A context menu is open over the 'ADMINIS' user, with 'New...' selected. The table below shows the user data:

ID	Name	Route ...	Group	Faxes	Subscriber...	Route Type
DEFAUL			RYONE	0	100	Fax Mailbox
ADMINIS			RYONE	10	0	Fax Mailbox
WALKUP			RYONE	0	100	Fax Mailbox
USER1			RYONE	61	100	Fax Mailbox
USER2			RYONE	5	100	Fax Mailbox

The status bar at the bottom indicates '1-5 of 5 users' and '5 Users (1 selected)'. The page number is '1 page'.

The user details are specified in the subsequent window. In the **Identification** tab, the **User ID** (e.g., *USER1*), **User Name** (e.g., *user1*) and **Password** are provided.

The screenshot shows a 'User Edit' dialog box with the 'Identification' tab selected. The dialog has a title bar with a close button (X) and a tabbed interface. The tabs are: Default Inbound Settings, Notification, Other, Messaging, Administrative Alerts, Identification (selected), Permissions, Routing, Default Outbound Settings, and Automatic Printing. The 'Identification' tab contains the following fields and controls:

- User ID:** A text box containing 'USER1'.
- Use Integrated Windows NT Security**
- Select NT Account:** A button.
- User Name:** A text box containing 'user1'.
- Password:** A text box with 10 black dots.
- Confirm password:** A text box with 10 black dots.
- Distinguished Name:** An empty text box.
- Group ID:** A dropdown menu showing 'EVERYONE'.
- Voice Mail Subscriber ID:** A text box containing '100'.
- Email address:** An empty text box.
- SMS/Mobile Address:** An empty text box.
- Compute Disk Usage:** A button.
- May take several seconds on a server with many faxes:** A note.

At the bottom right, there are 'OK' and 'Cancel' buttons.

In the **Routing** tab, **Fax Number/Routing Code** is set to the user's fax number (e.g., 78771).

User Edit [X]

Default Inbound Settings | Notification | Other | Messaging | Administrative Alerts
Identification | Permissions | **Routing** | Default Outbound Settings | Automatic Printing

Fax Number/Routing Code:

Routing Type:

File Format:

Routing Info:

When routing to a Fax Mailbox, no additional information is necessary. If notifications occur through email, the email address should be specified in the Routing Info field.

Routing Filename Format:

Received Fax Routing Form:

Include Web Delivery URL

Delete after routing

7.8. Restart RightFax Services

After the configuration is complete, click **Save** and exit the Brooktrout Configuration Tool. In the **DocTransport Configuration** window, click the **OK** button (shown in **Section 7.1**). Restart all RightFax service modules by right-clicking the **RightFax DocTransport Module** in EFM and select **Restart All Services** as shown below.

The screenshot shows the RightFax Enterprise Fax Manager interface. On the left is a navigation tree with 'Services' selected. The main area displays a table of services. A context menu is open over the 'RightFax DocTransport Module' row.

Service Name	Status	Running Time (ddd:hh:mm:ss)	Startup
RightFax DocTransport Module	Running	0008:20:48:20	Automatic
RightFax Server Module			Automatic
RightFax Database Module			Manual
RightFax RPC Server Module			Automatic
RightFax Queue Handler			Automatic
RightFax WorkServer1 Module			Manual
RightFax WorkServer2 Module			Manual
RightFax WorkServer3 Module			Manual
RightFax Email Gateway Module			
RightFax SAP Gateway1 Module			
RightFax Integration Module			
RightFax eTransport Module			
RightFax AutoReply Module			
RightFax Remoting Service			Automatic
RightFax Sync Module	Running	0008:22:42:34	Automatic
RightFax Conversion Engine	Running	0008:22:42:33	Automatic
RightFax Licensing Service	Running	0009:20:58:40	Automatic
RightFax EWS Connector	Not configured	N/A	
RightFax SharePoint Gateway Module	Not configured	N/A	
RightFax Worker Host	Running	0008:22:42:33	Automatic

The context menu for the 'RightFax DocTransport Module' includes the following options:

- Start Service
- Stop Service
- Restart Service
- Configure Service**
- Copy Settings to Other Nodes...
- Start All Services
- Stop All Services
- Restart All Services**
- Status
- Debug
- Debug with Options...
- Reconnect
- Cancel

At the bottom of the window, a status bar indicates: 20 services last updated at 6/26/2020 9:09:00.

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of OpenText RightFax with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The following steps can be used to verify deployments in the field.

1. Verify that the SIP trunk between Session Manager and the RightFax server is up by navigating to **Home→Elements→Session Manager→System Status→SIP Entity Monitoring** on System Manager. The status of the SIP trunk to the RightFax server is shown below.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The main content area is titled 'SIP Entity, Entity Link Connection Status' and displays a table of entity links to the SIP entity 'RightFax'. The table has columns for Session Manager Name, Session Manager IP Address, SIP Entity Resolved IP, Port, Proto., Deny, Conn. Status, Reason Code, and Link Status. One item is listed: 'devcon-sm' with IP address 'IPv4', resolved IP '10.64.102.102', port '5060', protocol 'UDP', deny 'FALSE', connection status 'UP', reason code '200 OK', and link status 'UP'.

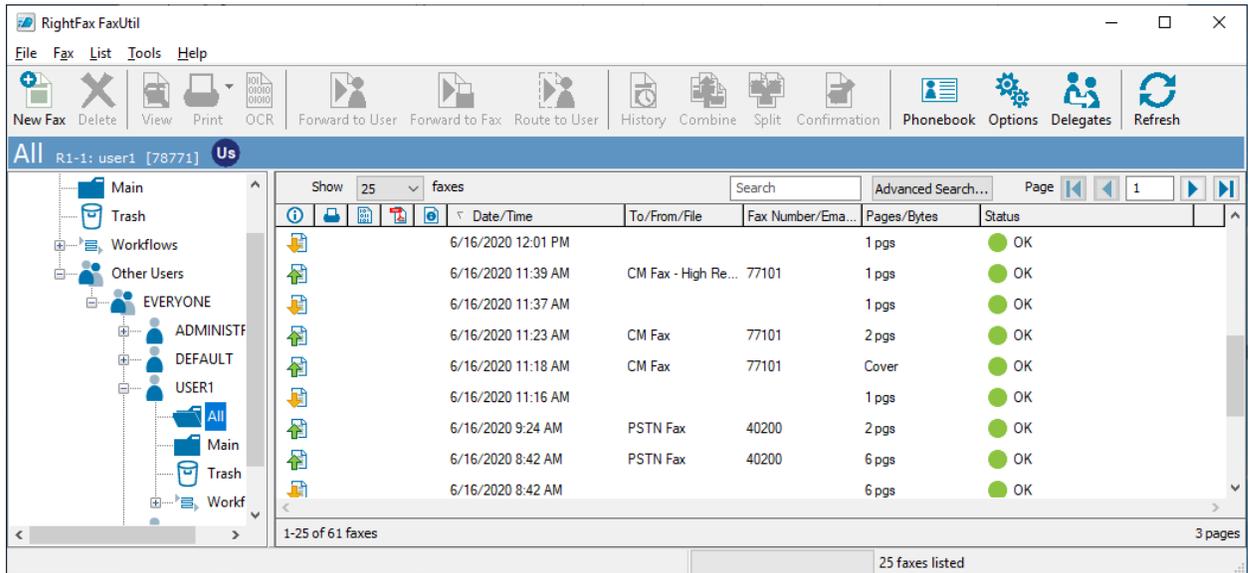
Session Manager Name	Session Manager IP Address	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
devcon-sm	IPv4	10.64.102.102	5060	UDP	FALSE	UP	200 OK	UP

2. Verify that fax messages can be sent and received by RightFax. Fax messages can be sent or received via **RightFax FaxUtil** or accessing FaxUtil from a web browser. The following steps uses the FaxUtil application. Launch **RightFax FaxUtil** and log in as shown below.

The RightFax Login dialog box contains the following fields and options:

- User ID: ADMINISTRATOR
- Password: (empty)
- Remember password
- Server: R1-1
- Protocol: TCP/IP
- Buttons: OK, Cancel

- In **RightFax FaxUtil**, navigate to the appropriate user in the left pane. The fax messages that were sent or received are shown in the left pane as shown below. Click on **New Fax** to send a fax or double-click on a fax message in the right pane to open the fax.



4. In EFM, the status of each RightFax channel is displayed. In the following example, a fax message was sent on channel 0 and received on channel 21.

The screenshot shows the 'RightFax Enterprise Fax Manager' interface. The main window displays a list of channels under the 'R1-1 ADMINISTRATOR' session. The left sidebar contains a navigation menu with various system components. The main table lists 22 channels, each with a status indicator (green circle with a dot or a green circle with a dot and a plus sign), a name, an operation, a routing code, a phone number, a user ID, and a state description.

Channel	Operati...	Routing Code	Phone Number	User ID	State
0 (B) Brooktrout	Send		78772	USER1	Sending 321a5072-0d81-45...
1 (B) Brooktrout	Wait				Checking for inbound docum.
2 (B) Brooktrout	Wait				Checking for inbound docum.
3 (B) Brooktrout	Wait				Checking for inbound docum.
4 (B) Brooktrout	Wait				Checking for inbound docum.
5 (B) Brooktrout	Wait				Checking for inbound docum.
6 (B) Brooktrout	Wait				Checking for inbound docum.
7 (B) Brooktrout	Wait				Checking for inbound docum.
8 (B) Brooktrout	Wait				Checking for inbound docum.
9 (B) Brooktrout	Wait				Checking for inbound docum.
10 (B) Brooktrout	Wait				Checking for inbound docum.
11 (B) Brooktrout	Wait				Checking for inbound docum.
12 (B) Brooktrout	Wait				Checking for inbound docum.
13 (B) Brooktrout	Wait				Checking for inbound docum.
14 (B) Brooktrout	Wait				Checking for inbound docum.
15 (B) Brooktrout	Wait				Checking for inbound docum.
16 (B) Brooktrout	Wait				Checking for inbound docum.
17 (B) Brooktrout	Wait				Checking for inbound docum.
18 (B) Brooktrout	Wait				Checking for inbound docum.
19 (B) Brooktrout	Wait				Checking for inbound docum.
20 (B) Brooktrout	Wait				Checking for inbound docum.
21 (B) Brooktrout	Rcv	78772			Receiving C:\PROGRA~2\...

At the bottom of the window, a status bar indicates '22 channels (1 selected)'.

9. Conclusion

These Application Notes describe the configuration steps required to integrate OpenText RightFax with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using a SIP trunk. Fax messages were successfully sent and received between RightFax and fax software on the PSTN and local to the enterprise. The compliance test passed with observations noted in **Section 2.2**.

10. Additional References

This section references the documentation relevant to these Application Notes. The following and additional Avaya product documentation is available at <http://support.avaya.com> and <http://www.opentext.com> (login required).

- [1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 6, March 2020.
- [2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 8.1.x, issue 9, June 2020.
- [3] *Administering Network Connectivity on Avaya Aura® Communication Manager*, Release 8.1.x, Issue 3, February 2020.
- [4] *Administering Avaya Aura® System Manager for Release 8.1.x*, Release 8.1.x, Issue 6, April 2020.
- [5] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 4, May 2020.
- [6] *OpenText RightFax 20.2 Installation Guide*, April 22, 2020.
- [7] *OpenText RightFax 20.2 Administrator Guide*, April 22, 2020.

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