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](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.

![Diagram of OpenText RightFax with Avaya SIP-based Network](image)

**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:

<table>
<thead>
<tr>
<th>Equipment/Software</th>
<th>Release/Version</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya Aura® Communication Manager</td>
<td>8.1.2.0.0-FP2</td>
</tr>
<tr>
<td>Avaya G450 Media Gateway</td>
<td>FW 41.24.0</td>
</tr>
<tr>
<td>Avaya Aura® System Manager</td>
<td>8.1.2.0</td>
</tr>
<tr>
<td></td>
<td>Build No. – 8.1.0.0.733078</td>
</tr>
<tr>
<td></td>
<td>Software Update Revision No: 8.1.2.0.06111167</td>
</tr>
<tr>
<td></td>
<td>Feature Pack 2</td>
</tr>
<tr>
<td>Avaya Aura® Session Manager</td>
<td>8.1.2.0.812039</td>
</tr>
<tr>
<td>OpenText RightFax with</td>
<td>20.2.0.846</td>
</tr>
<tr>
<td>Dialogic® Brooktrout SR140 Fax Software</td>
<td>6.5.0.14</td>
</tr>
</tbody>
</table>

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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

IP NODE NAMES

<table>
<thead>
<tr>
<th>Name</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>default</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>devcon-aes</td>
<td>10.64.102.119</td>
</tr>
<tr>
<td>devcon-ams</td>
<td>10.64.102.118</td>
</tr>
<tr>
<td>devcon-sm</td>
<td>10.64.102.117</td>
</tr>
<tr>
<td>procr</td>
<td>10.64.102.115</td>
</tr>
<tr>
<td>procr6</td>
<td>::</td>
</tr>
</tbody>
</table>

( 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.

<table>
<thead>
<tr>
<th>Codec Set: 1</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Audio Codec</strong></td>
</tr>
<tr>
<td>G.711MU</td>
</tr>
</tbody>
</table>

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.

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

<table>
<thead>
<tr>
<th>FAX Mode</th>
<th>Redundancy</th>
<th>Packet Size(ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>t.38-standard</td>
<td>0</td>
<td>ECM: y</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Modem</th>
<th>off</th>
</tr>
</thead>
<tbody>
<tr>
<td>TDD/TTY</td>
<td>US</td>
</tr>
<tr>
<td>H.323 Clear-channel</td>
<td>n</td>
</tr>
<tr>
<td>SIP 64K Data</td>
<td>n</td>
</tr>
</tbody>
</table>

Media Connection IP Address Type Preferences

1: IPv4
2: IPv6
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
MEDUSA PARAMETERS
Codec Set: 1   Intra-region IP-IP Direct Audio: yes
              Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048
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
H.323 IP ENDPOINTS
  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.
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

TRUNK GROUP

Group Number: 10  
Group Name: To devcon-sm  
Direction: two-way  
Dial Access? n  
Queue Length: 0

Group Type: sip  
CDR Reports: y  
COR: 1  
TN: 1  
TAC: 1010  
Outgoing Display? n  
Night Service:

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

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.

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

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.

<table>
<thead>
<tr>
<th>Change private-numbering 0</th>
<th>Page 1 of 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>NUMBERING – PRIVATE FORMAT</td>
<td></td>
</tr>
<tr>
<td>Ext Ext Len Code Trk Grp(s) Private Prefix Total Len Total Administered: 1 Maximum Entries: 540</td>
<td></td>
</tr>
<tr>
<td>5 7</td>
<td></td>
</tr>
</tbody>
</table>

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).

<table>
<thead>
<tr>
<th>Change uniform-dialplan 7</th>
<th>Page 1 of 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>UNIFORM DIAL PLAN TABLE</td>
<td></td>
</tr>
<tr>
<td>Matching Pattern Len Del Insert Digits Net Conv Num Node</td>
<td></td>
</tr>
<tr>
<td>78 5 0</td>
<td>aar n</td>
</tr>
</tbody>
</table>
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.

<table>
<thead>
<tr>
<th>Dialed String</th>
<th>Total Min</th>
<th>Max</th>
<th>Route Pattern</th>
<th>Call Type</th>
<th>Node Num</th>
<th>ANI Reqd</th>
</tr>
</thead>
<tbody>
<tr>
<td>7</td>
<td>7</td>
<td>7</td>
<td>254</td>
<td>aar</td>
<td>n</td>
<td>n</td>
</tr>
<tr>
<td>78</td>
<td>5</td>
<td>5</td>
<td>10</td>
<td>lev0</td>
<td>n</td>
<td>n</td>
</tr>
<tr>
<td>8</td>
<td>7</td>
<td>7</td>
<td>254</td>
<td>aar</td>
<td>n</td>
<td>n</td>
</tr>
<tr>
<td>9</td>
<td>7</td>
<td>7</td>
<td>254</td>
<td>aar</td>
<td>n</td>
<td>n</td>
</tr>
</tbody>
</table>

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

Dial patterns must be defined to direct calls to the appropriate SIP Entity. In the sample configuration, a 5-digit number beginning with ‘7877’ will be routed to RightFax.

To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under **General**:
- **Pattern**: Dialed number or prefix.
- **Min**: Minimum length of dialed number.
- **Max**: Maximum length of dialed number.
- **SIP Domain**: SIP domain of dial pattern.
- **Notes**: Comment on purpose of dial pattern (optional).

Under **Originating Locations and Routing Policies**:
Click **Add** and then select the appropriate location and routing policy from the list. In this case, the RightFax routing policy is selected.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definitions for RightFax extensions.
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 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.
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.

![DocTransport Configuration Window](image-url)
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](image)

- 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:** Rightfax

**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.
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*.

![Brooktrout Configuration Tool - Advanced Mode](image)

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.*

![Brooktrout Configuration Tool - Advanced Mode](image)
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](image)

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

![RightFax User Management Screen](image-url)
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.
In the **Routing** tab, **Fax Number/Routing Code** is set to the user’s fax number (e.g., 78771).
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.
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.

![SIP Entity, Entity Link Connection Status](image)

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.

![RightFax Login](image)
3. 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.
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).
