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.

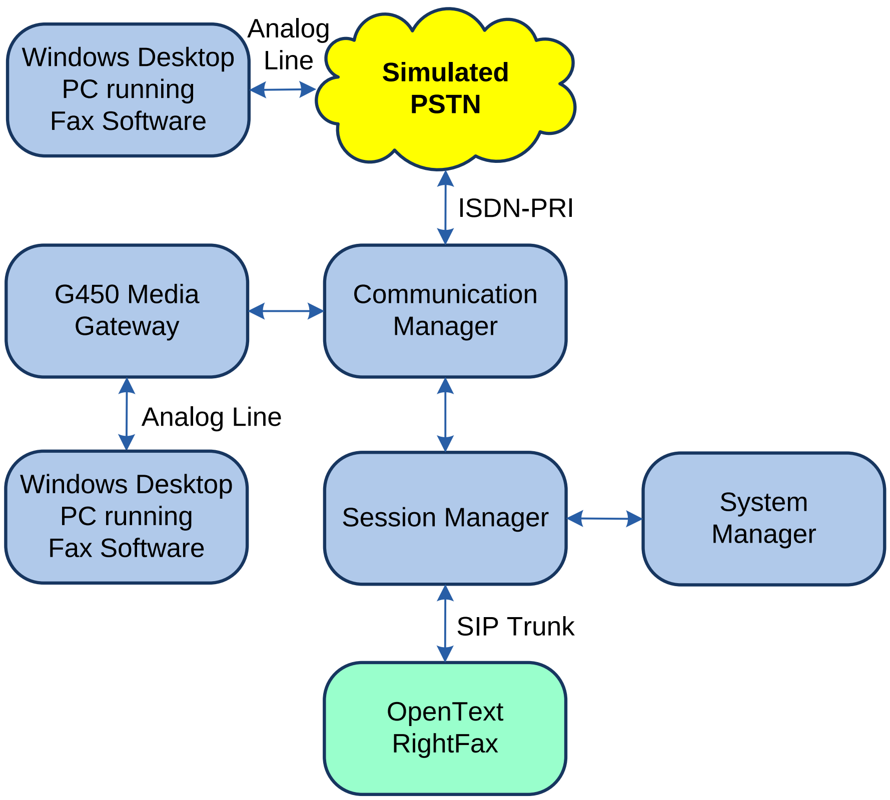


Figure 1: OpenText RightFax with Avaya SIP-based Network

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

                                             Redun-                     Packet
                          Mode               dancy                      Size(ms)
    FAX                   t.38-standard      0     ECM: y
    Modem                 off                0
    TDD/TTY               US                 3
    H.323 Clear-channel   n                  0
    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                                            5     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       Len Del      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            Total      Route     Call    Node   ANI
          String           Min  Max  Pattern    Type    Num    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 FRL NPA Pfx Hop Toll No.  Inserted                               DCS/ IXC
    No          Mrk Lmt List Del  Digits                                 QSIG
                             Dgts                                        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            rest                               unk-unk   none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         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

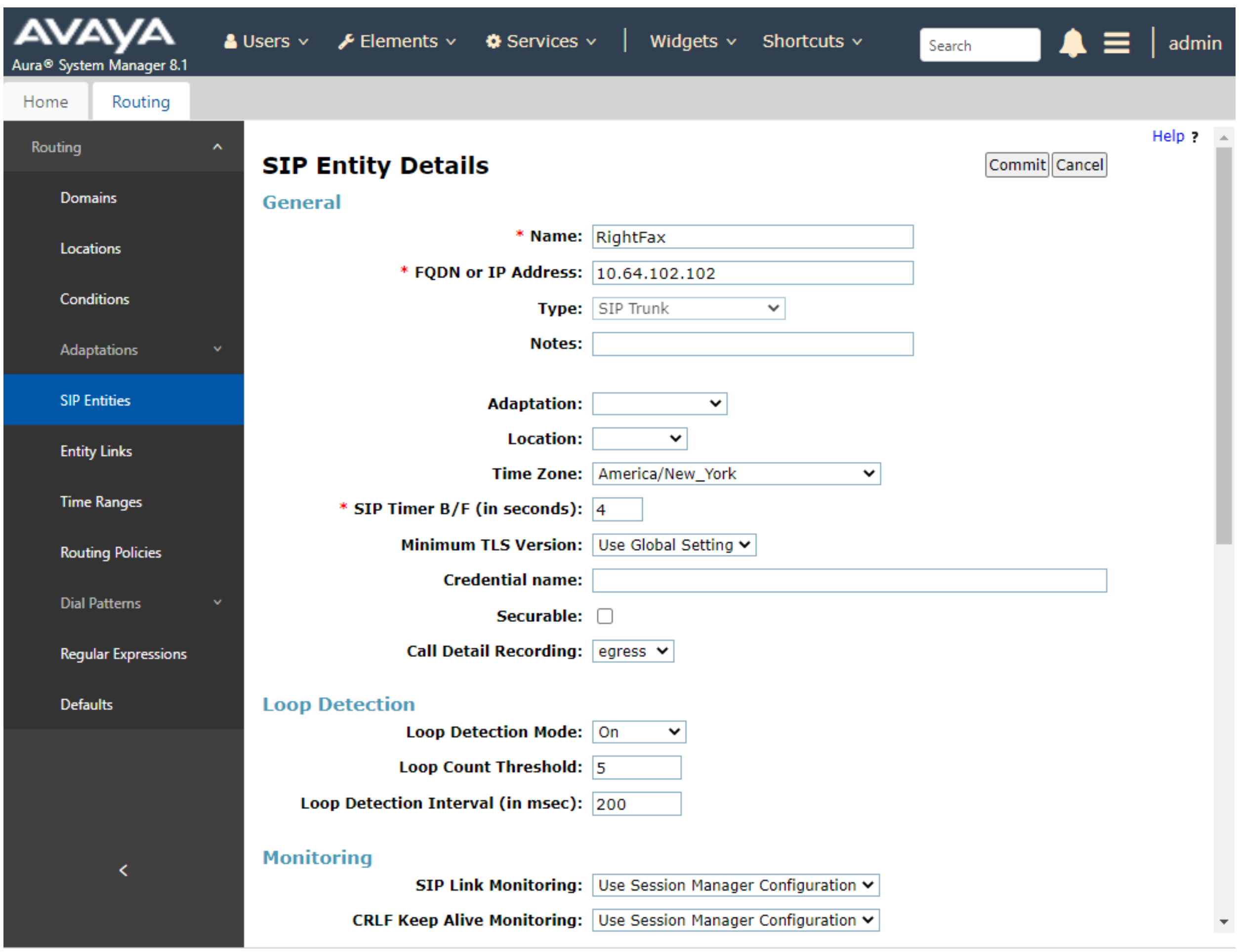
In the sample configuration, a SIP trunk was configured for RightFax.

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.



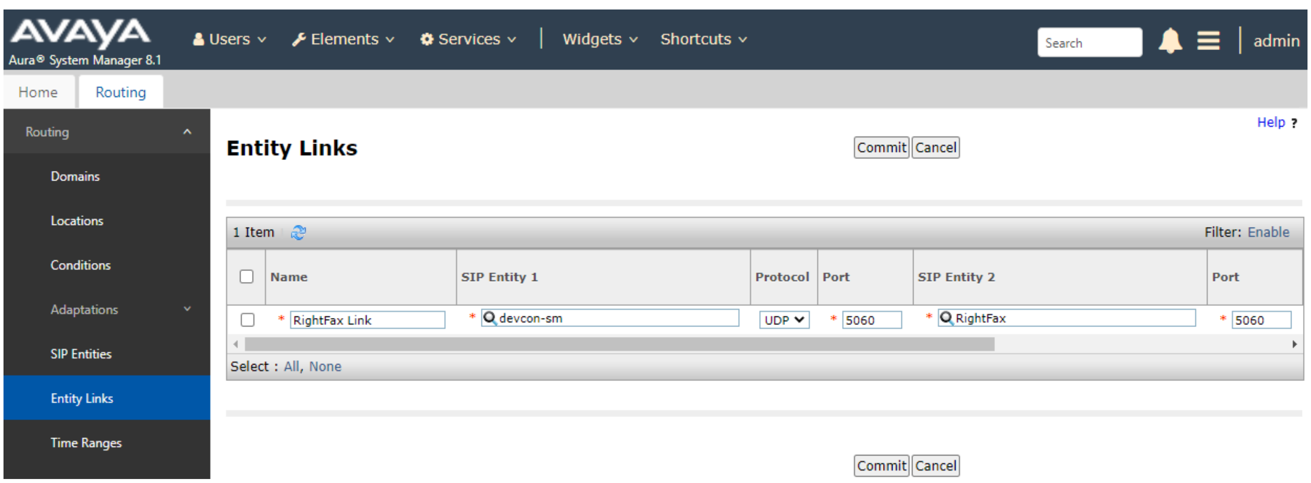
6.2. Add Entity Link for RightFax

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.



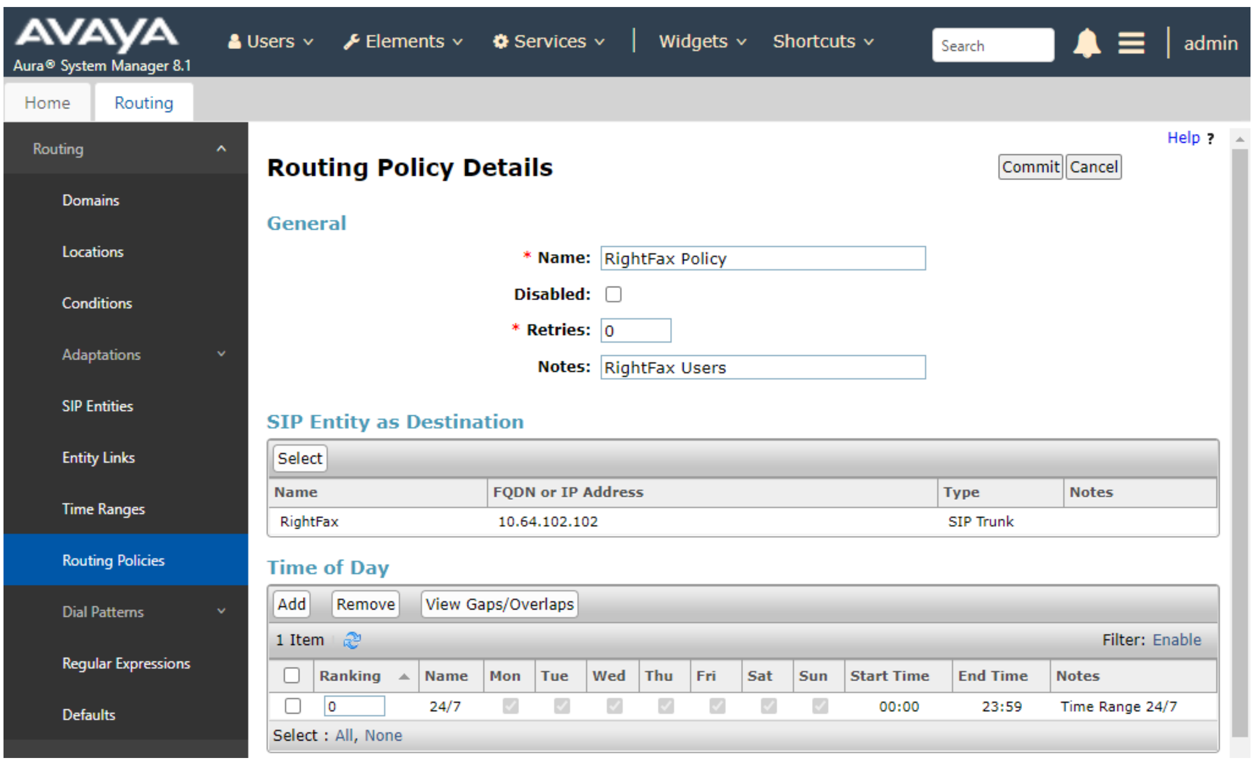
6.3. Add Routing Policies

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.

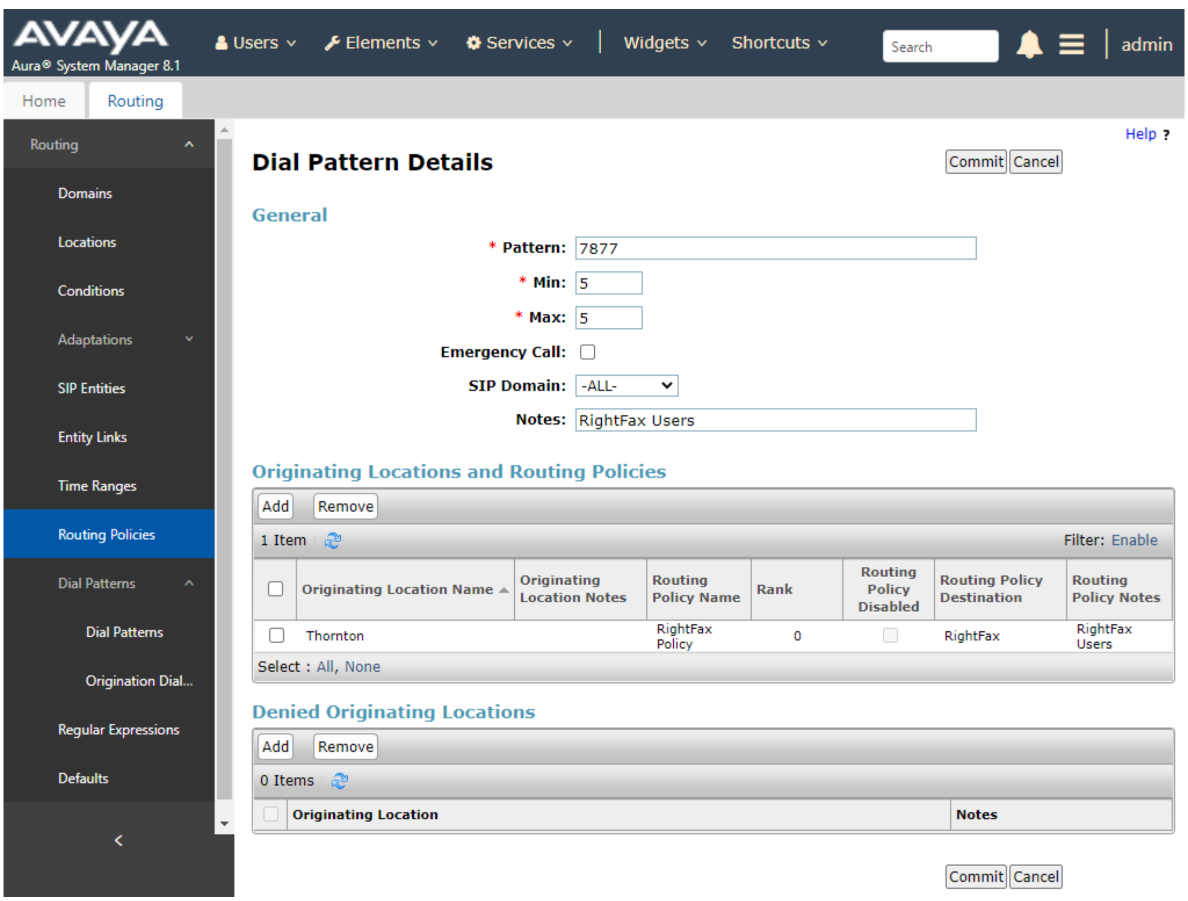
To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

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:

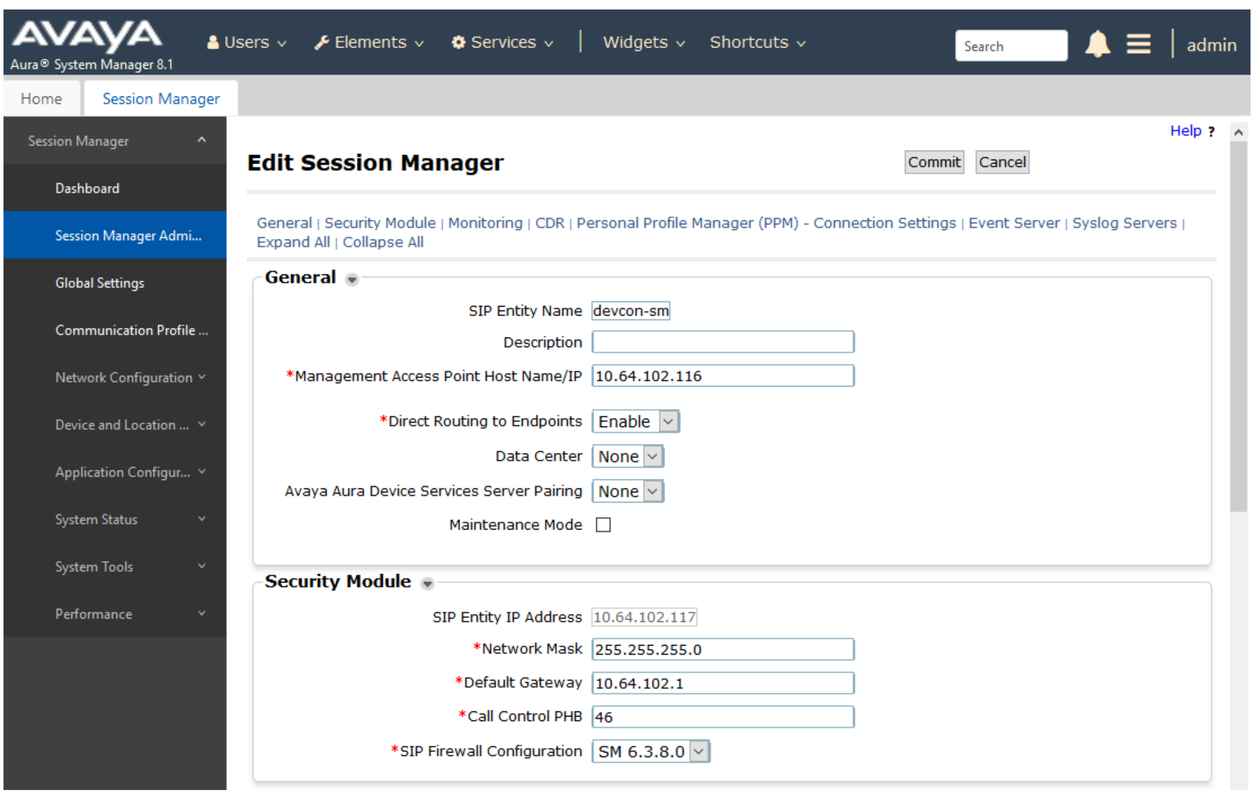
Under *General*:

- **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.

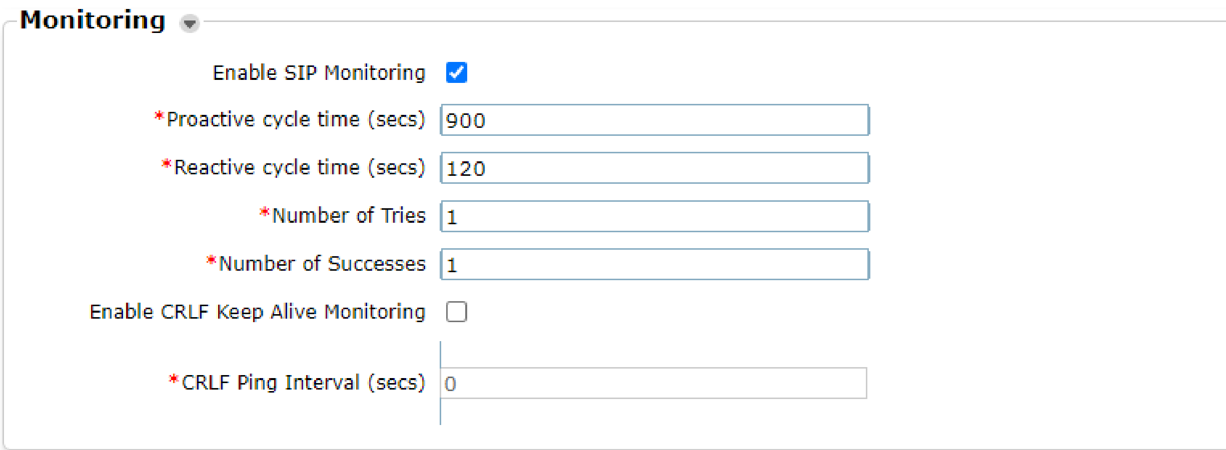
Under *Security Module*:

- **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.



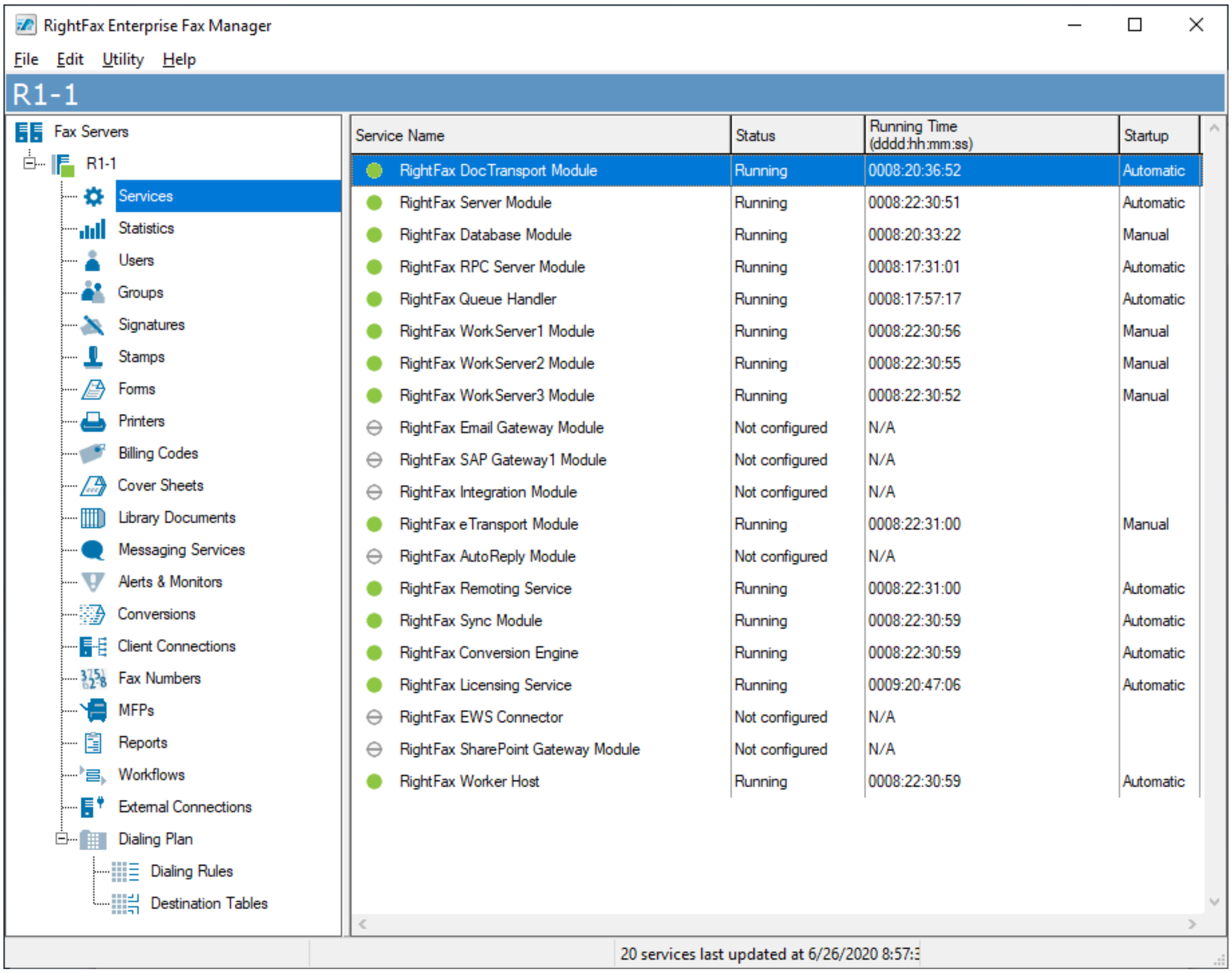
7. Configure OpenText RightFax

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:

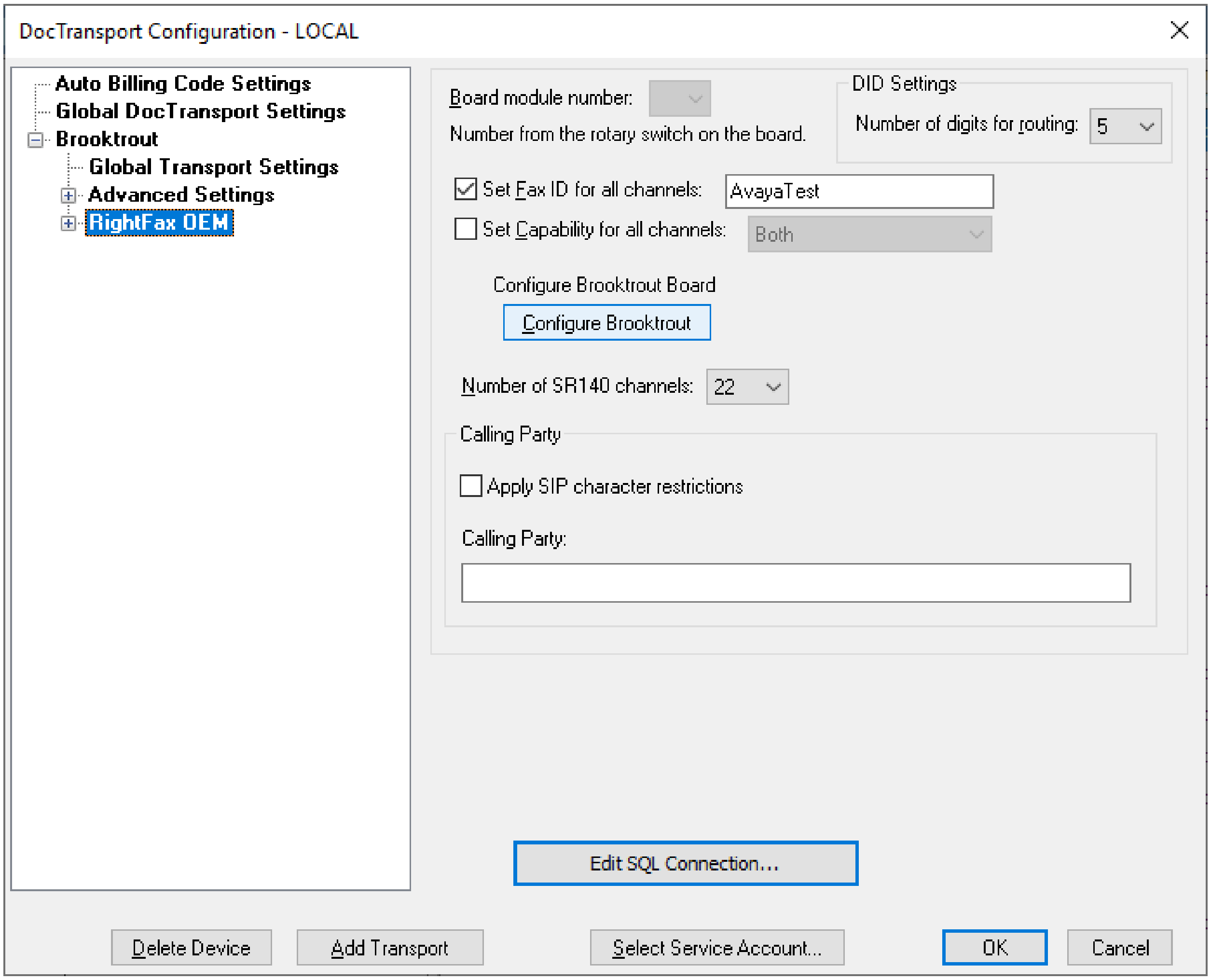
- Launch RightFax Enterprise Fax Manager and Brooktrout Configuration Tool
- Configure SIP IP Parameters
- Configure T.38 Parameters
- Configure RTP Parameters
- Configure Call Control Parameters
- Configure Other RightFax Options
- Add RightFax Users
- Restart RightFax Services

7.1. Launch RightFax Enterprise Fax Manager and Brooktrout Configuration Tool

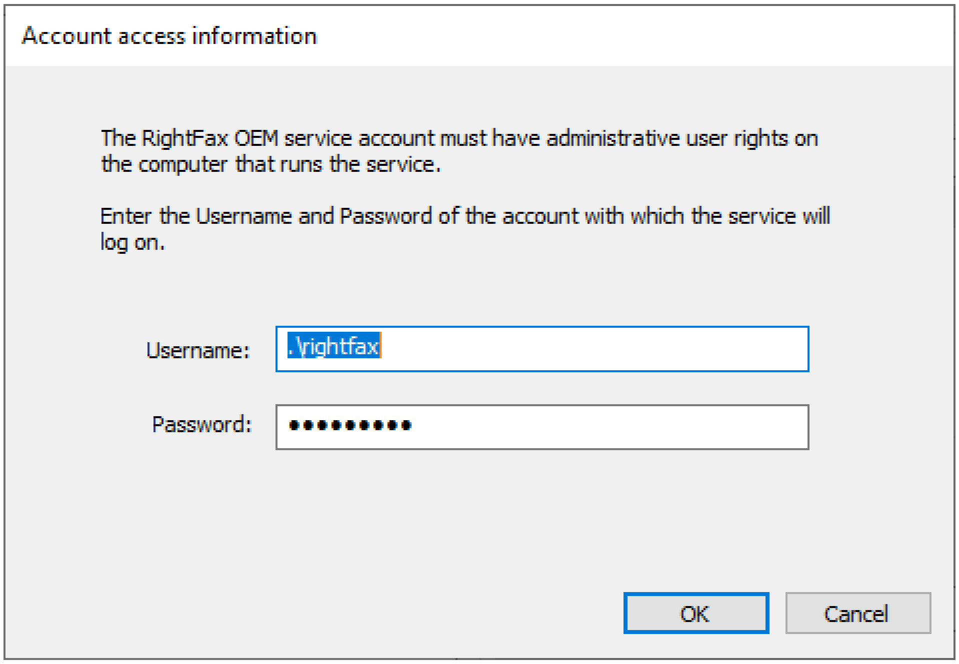
Launch RightFax Enterprise Fax Manager (EFM) to configure RightFax. In the EFM, expand the server, which is **R1-1** in this case, and select **Services** in the left pane. The list of RightFax services is displayed in the right pane as shown below. Note that the services have already been started. Double-click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



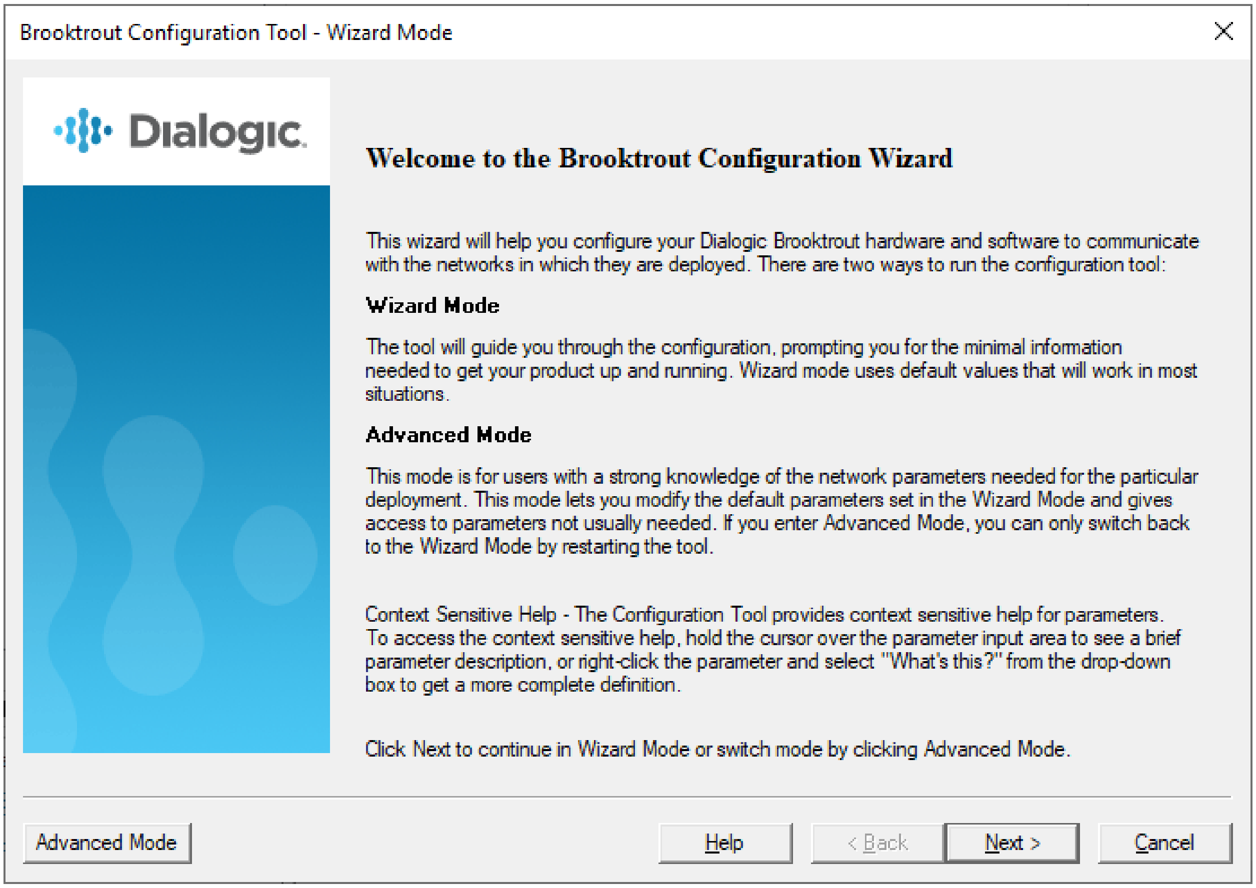
The **DocTransport Configuration** window is shown below. Expand **Brooktrout** in the left pane and select **RightFax as OEM**. Next, click on **Configure Brooktrout**.



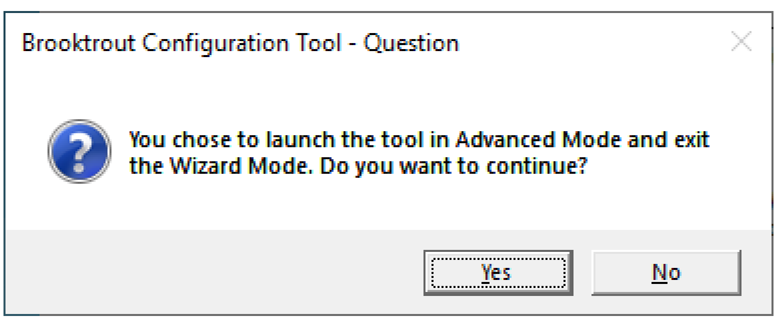
Log into the Brooktrout Configuration Tool using the appropriate access credentials as shown below.



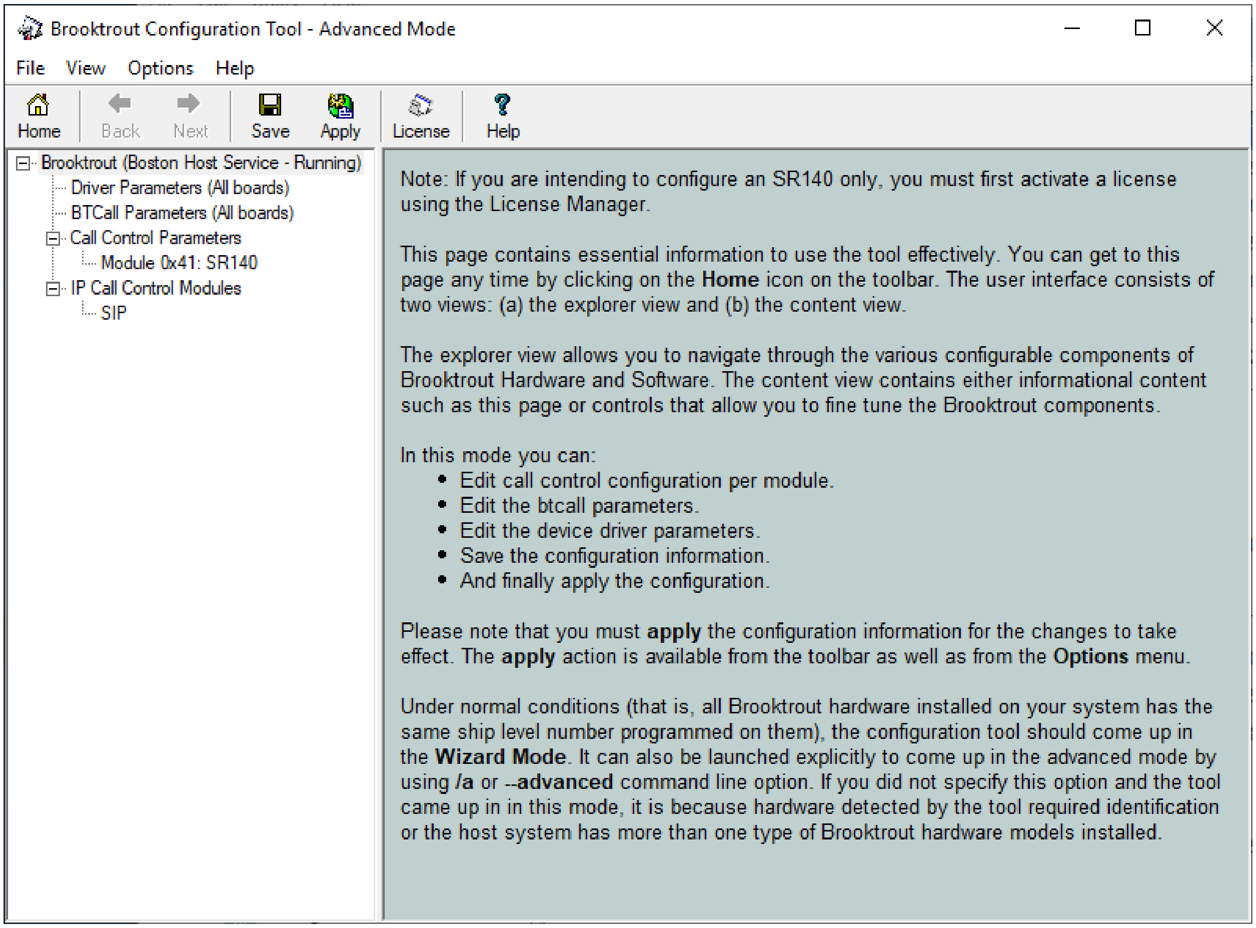
The **Brooktrout Configuration Tool – Wizard Mode** window is displayed. Click **Advanced Mode**.



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

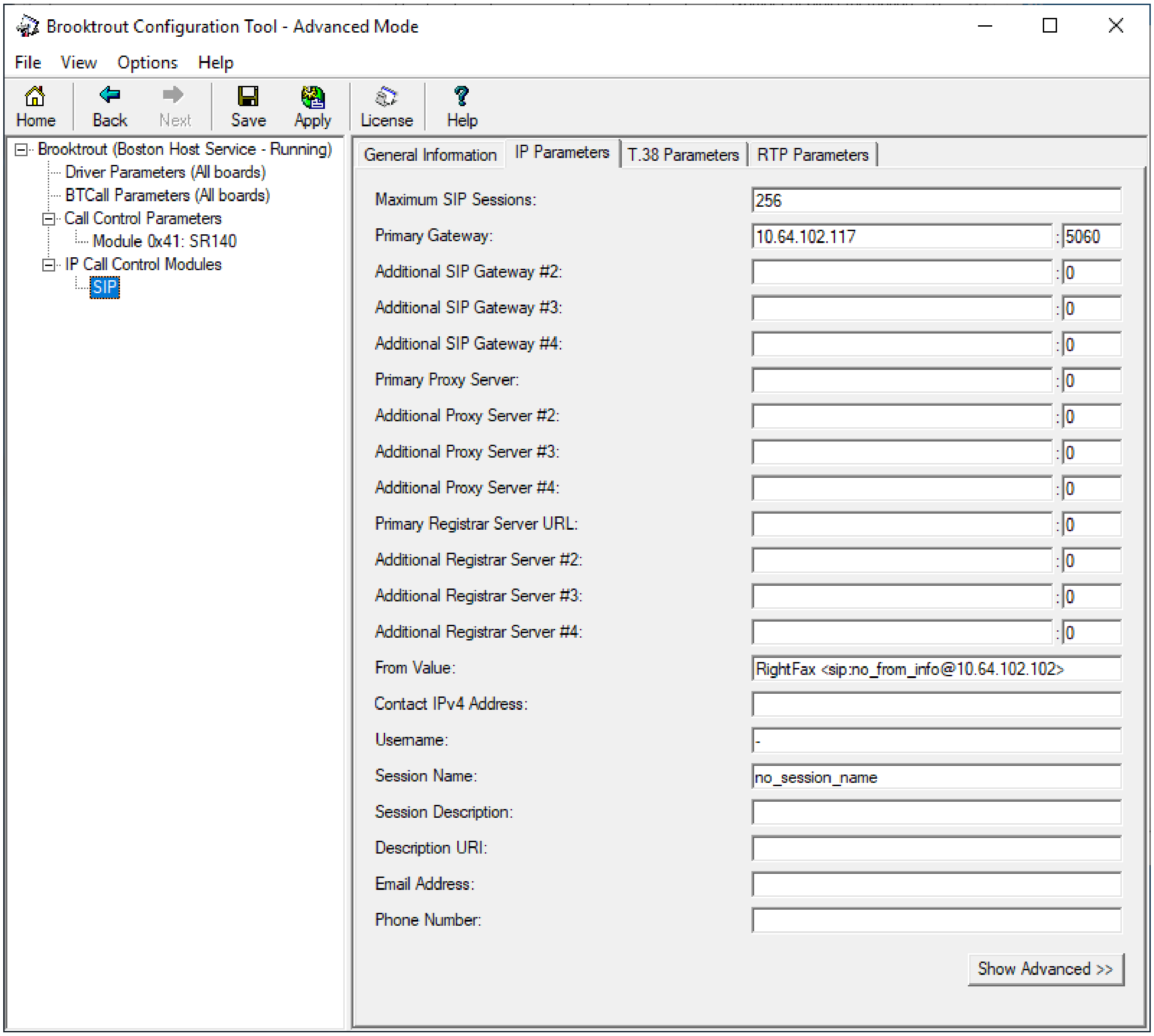


Brooktrout Configuration Tool – Advanced Mode window is displayed as shown below.



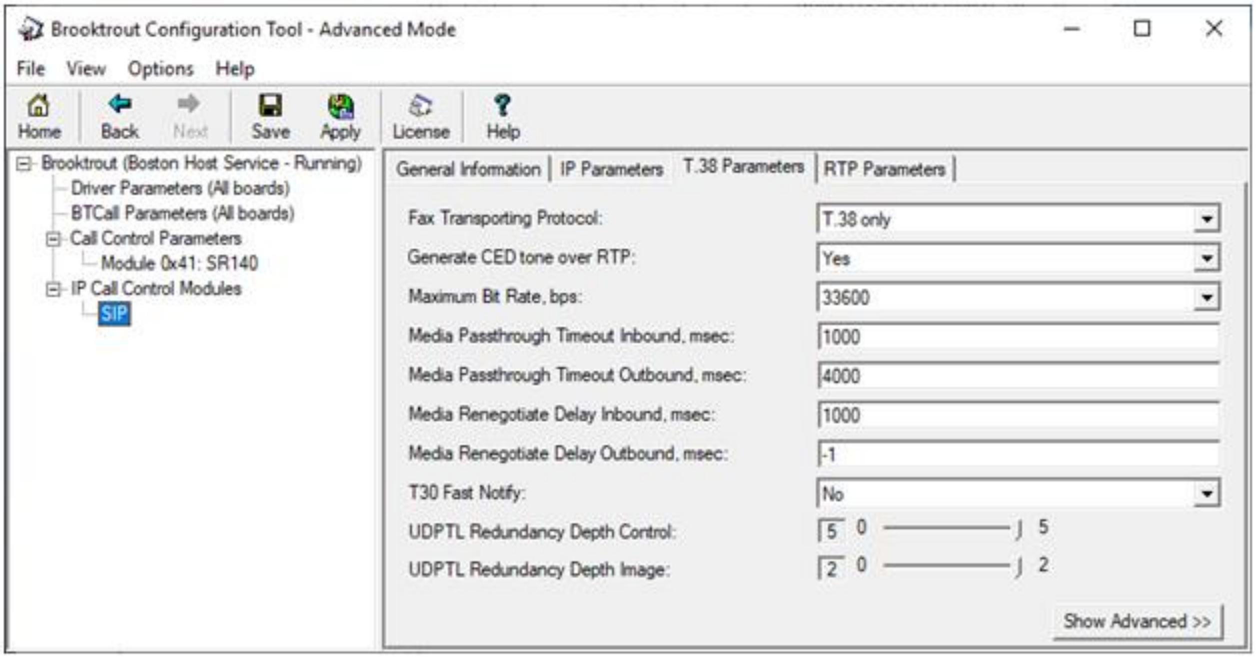
7.2. Configure SIP IP Parameters

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.



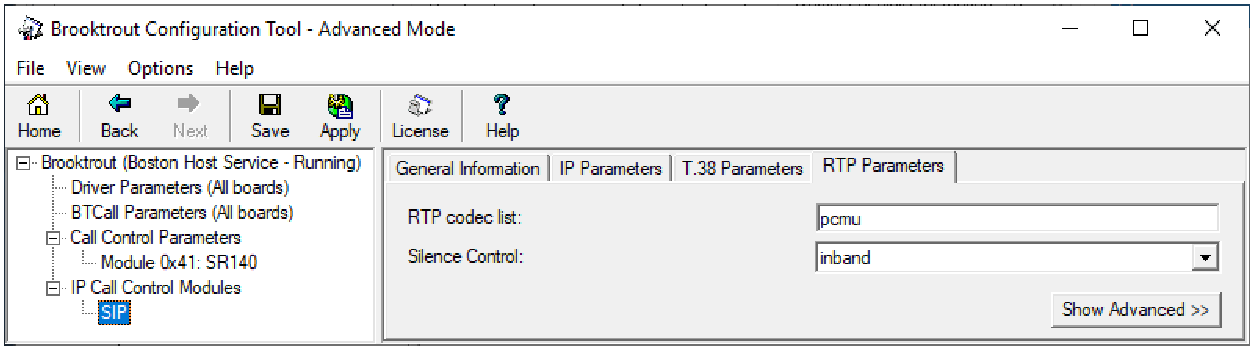
7.3. Configure T.38 Parameters

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

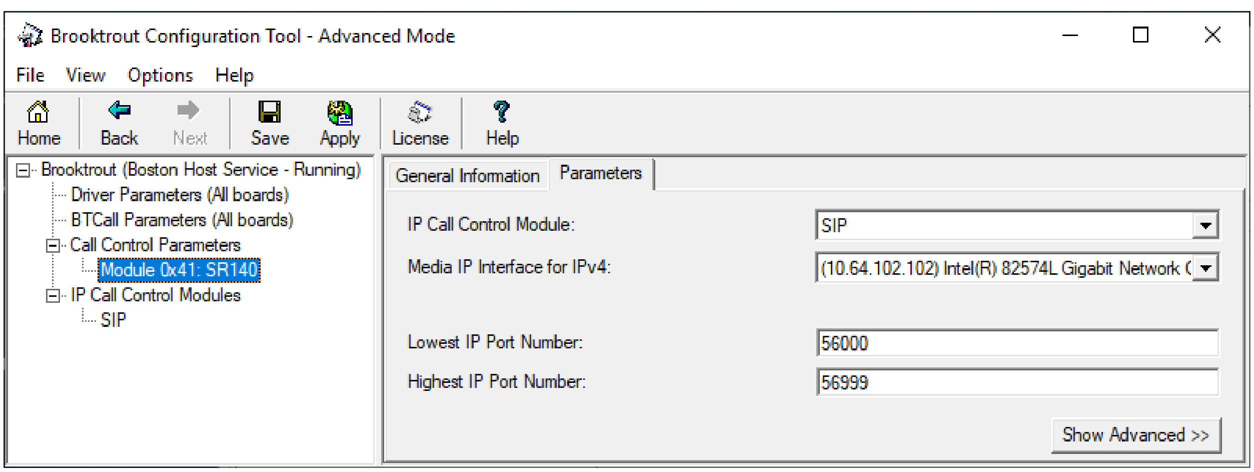
In the **RTP Parameters** tab, **RTP codec list** was set to *pcmu*.



7.5. Configure Call Control Parameters

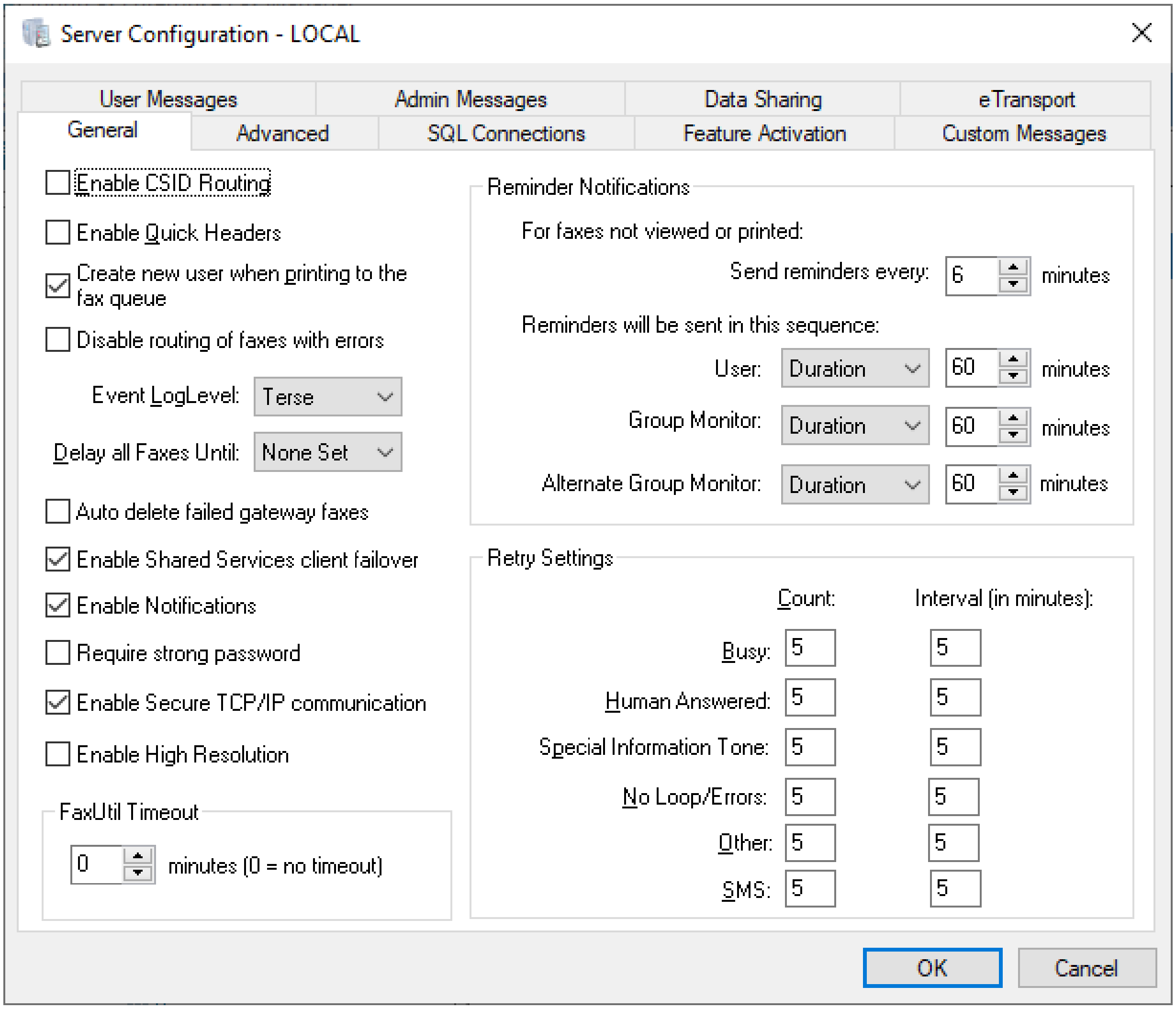
Navigate to **Brooktrout → Call Control Parameters → Module 0x41: SR140** in the left pane. Ensure the following settings in the **Parameters** tab are correct for your environment.

- **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.*



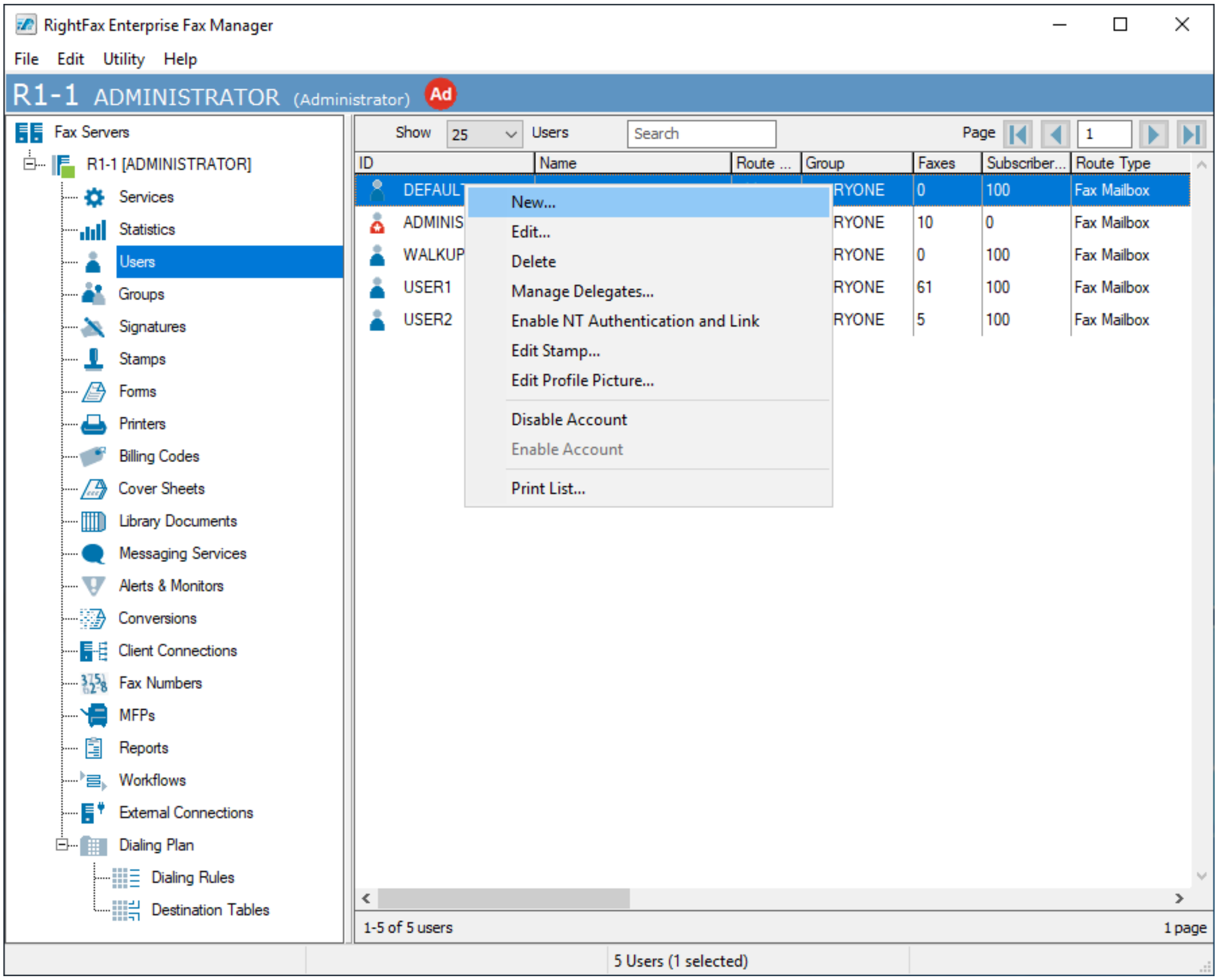
7.6. Configure Other RightFax Options

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.

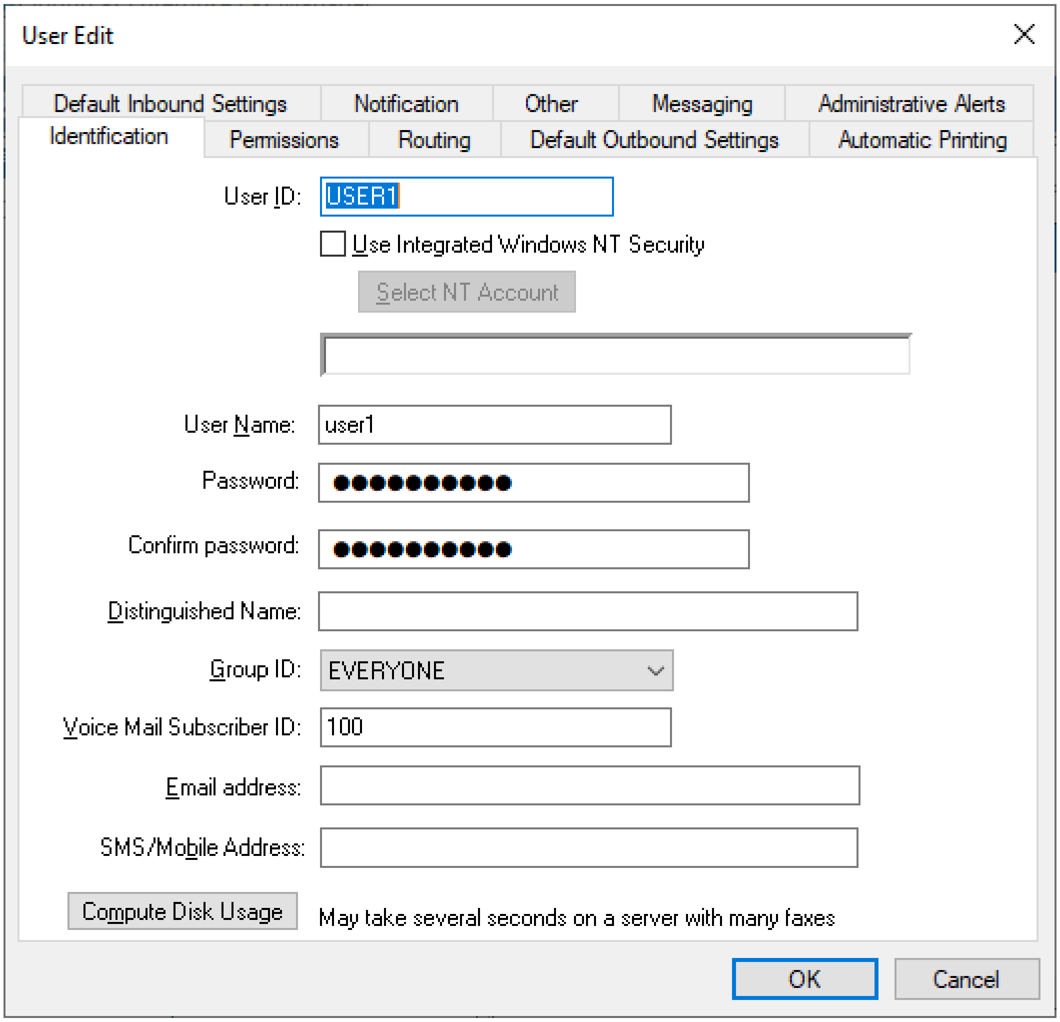


7.7. Add RightFax Users

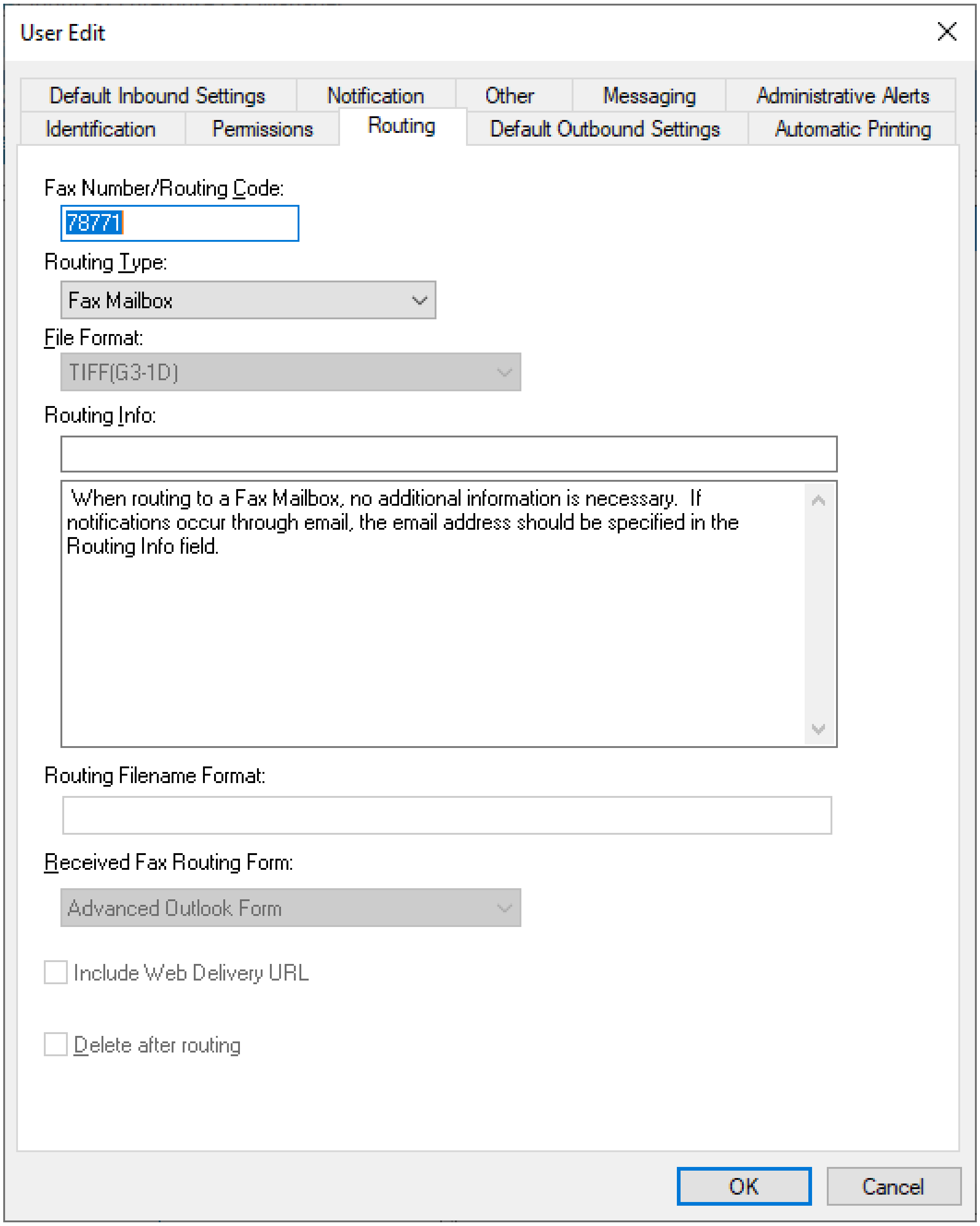
A user is created on the RightFax server an associated with a fax number. From a user's account, a fax may be received or sent. To view the list of users, select **Users** in the left pane of EFM. To add a new user, right-click on one of the users in the right pane and select **New** from the pop-up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



The 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.

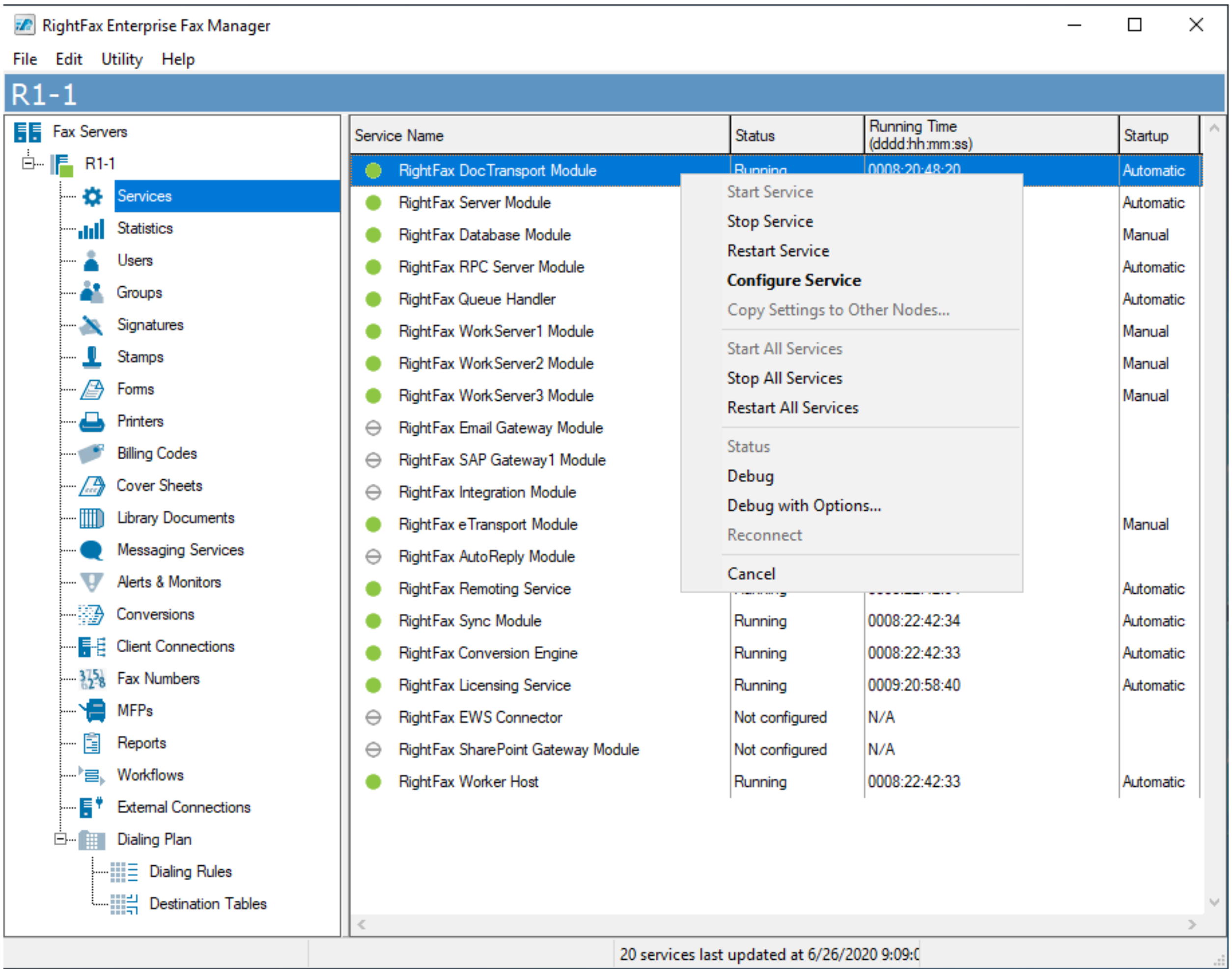


In the **Routing** tab, **Fax Number/Routing Code** is set to the user's fax number (e.g., *78771*).



7.8. Restart RightFax Services

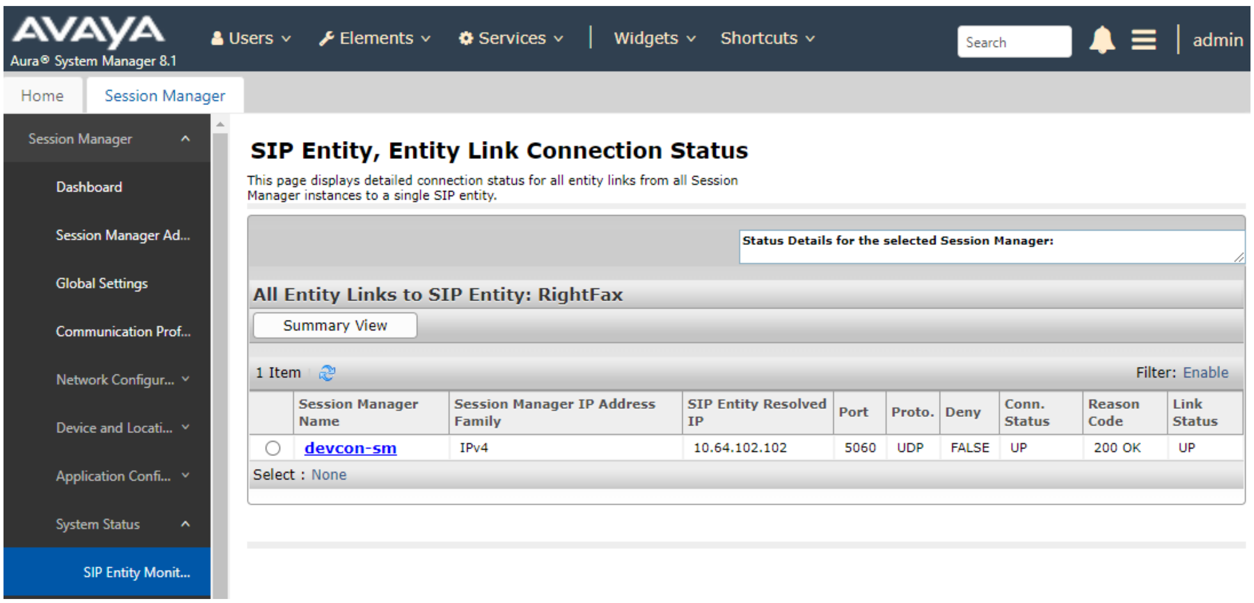
After the configuration is complete, click **Save** and exit the Brooktrout Configuration Tool. In the **DocTransport Configuration** window, click the **OK** button (shown in **Section 7.1**). Restart all RightFax service modules by right-clicking the **RightFax DocTransport Module** in EFM and select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



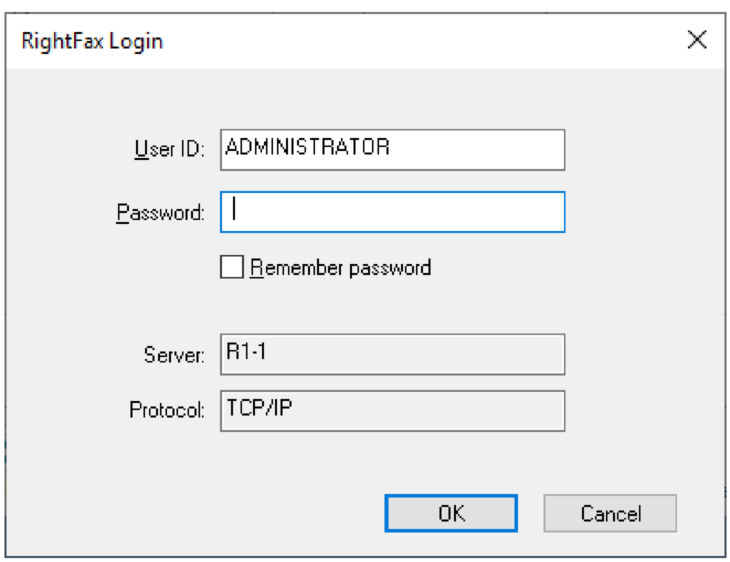
8. Verification Steps

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.

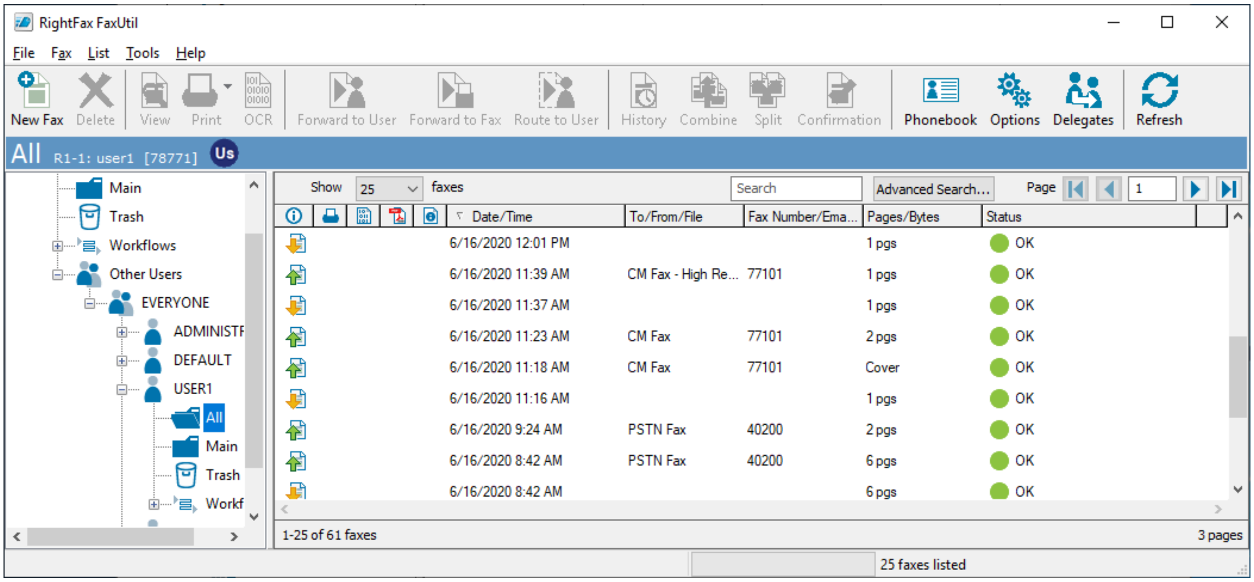
1. Verify that the SIP trunk between Session Manager and the RightFax server is up by navigating to **Home→Elements→Session Manager→System Status→SIP Entity Monitoring** on System Manager. The status of the SIP trunk to the RightFax server is shown below.



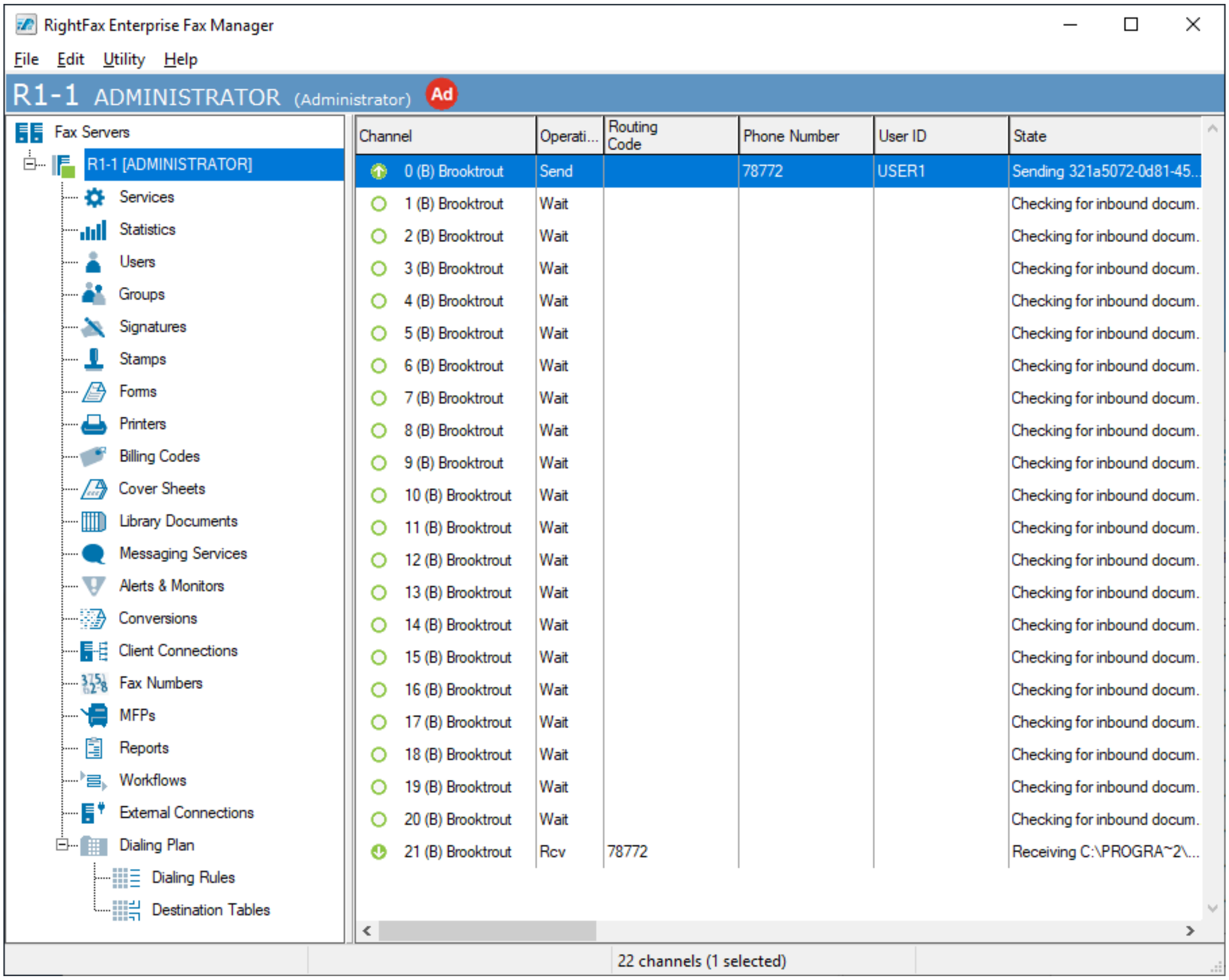
2. Verify that fax messages can be sent and received by RightFax. Fax messages can be sent or received via **RightFax FaxUtil** or accessing FaxUtil from a web browser. The following steps uses the FaxUtil application. Launch **RightFax FaxUtil** and log in as shown below.



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).

[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] *OpentText RightFax 20.2 Installation Guide*, April 22, 2020.
[7] *OpenText RightFax 20.2 Administrator Guide*, April 22, 2020.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.