**Avaya Solution & Interoperability Test Lab**

# Application Notes for Configuring Open Text Fax Server (RightFax) with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services via SIP Trunking Interface - Issue 1.0

## Abstract

These Application Notes describe the procedures for configuring the Open Text Fax Server (RightFax) with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services (SES) using SIP trunking interface.

RightFax is a software based fax server that sends and receives fax calls over an IP network. In the tested configuration, RightFax interoperates with the Communication Manager and the SIP Enablement Services to send/receive faxes using SIP trunks between RightFax and the Avaya SIP infrastructure.

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 procedures for configuring Open Text Fax Server (RightFax) with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services (SES) using SIP trunks.

RightFax is a software based fax server that sends and receives fax calls over an IP network. RightFax includes the Brooktrout SR140 T.38 Fax over Internet Protocol (FoIP) virtual fax board software from Dialogic. In the tested configuration, RightFax interoperates with the Communication Manager and the SIP Enablement Services to send/receive faxes using the SIP trunking interface between RightFax and the Avaya SIP infrastructure.

## 1.1. Interoperability Compliance Testing

The compliance test tested interoperability between RightFax and the Communication Manager and the SIP Enablement Services by making intra-site and inter-site fax calls to and from RightFax that is connected (at each of the two sites in the test configuration) to the Communication Manager and the SIP Enablement Services via SIP trunks (see **Section 2** for detailed configuration). Specifically, the following fax operations were tested in the setup for the compliance test:

- Fax from/to RightFax to/from fax machine at local site
- Fax from/to RightFax to/from fax machine at remote site
- Fax from/to RightFax to/from RightFax server at remote site

In the compliance test, Site 1 and Site 2 were connected by both ISDN-PRI trunks and H.323 trunks. The inter-site calls were tested by using either of these 2 types of trunks between sites.

Faxes were sent with various page lengths, resolutions and at various fax data speeds. For capacity, a large number of 3-page faxes were continuously sent between the two RightFax servers across sites. Serviceability testing included verifying proper operation/recovery from failed cables, unavailable resources, restarts of the Communication Manager and the SIP Enablement Services as well as RightFax reboots. Fax calls were also tested with different Avaya Media Gateway media resources to process the fax data. This included the TN2302AP IP Media Processor (MedPro) circuit pack and the TN2602AP IP Media Processor circuit pack in the Avaya G650 Media Gateway; the integrated Voice over Internet Protocol (VoIP) engine of the Avaya G700 Media Gateway, and the Avaya MM760 Media Module installed in the Avaya G700 Media Gateway.

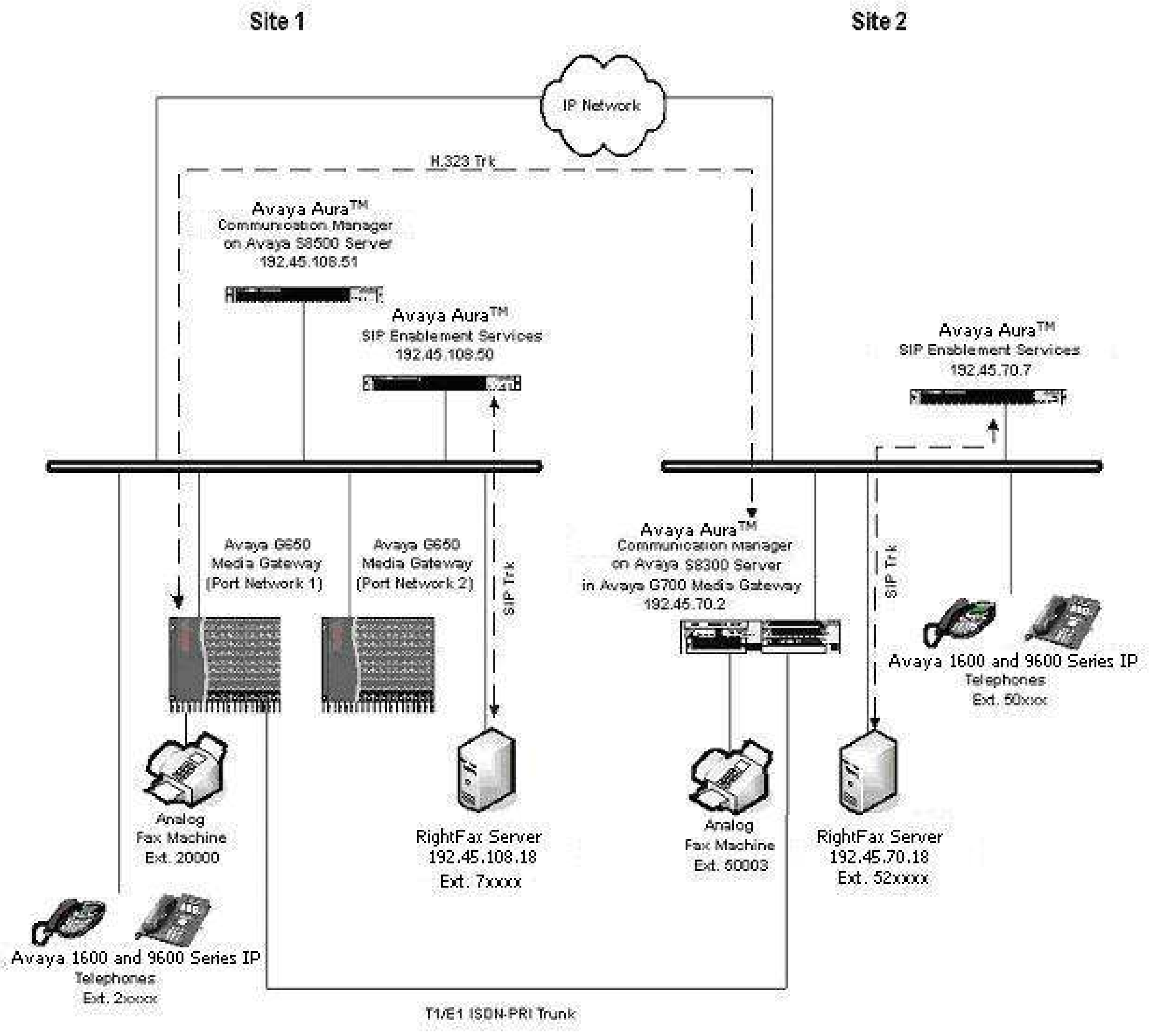
## 1.2. Support

Technical support for RightFax can be obtained by contacting Open Text at

- Phone: (800) 540-7292
- Email: support@opentext.com
- https://cslogin.opentext.com/login/

## 2. Configuration

**Figure 1** illustrates the configuration used in these Application Notes. In the sample configuration, two sites are connected via H.323 trunks, as well as ISDN-PRI trunks. Faxes can be sent between the two sites using either of these two trunk groups.



| Port Network 1 | |
|---|---|
| CLAN: | 192.45.108.55 |
| MedPro TN2302: | 192.45.108.54 |
| MedPro TN2602: | 192.45.108.58 |

| Port Network 2 | |
|---|---|
| CLAN: | 192.45.108.57 |
| MedPro TN2302: | 192.45.108.56 |
| MedPro TN2602 | 192.45.108.59 |

**Figure 1: RightFax interoperating with Communication Manager and SIP Enablement Services**

Located at Site 1 is an SIP Enablement Services server and an Avaya S8500 Server running Communication Manager with two Avaya G650 Media Gateways. Each media gateway is configured as a separate port network in separate IP network regions. RightFax at this site is running on a Windows Server 2008 laptop PC and communicates to the Avaya SIP infrastructure (Communication Manager and SIP Enablement Services) via SIP trunks whose signaling is terminated on a CLAN circuit pack in port network 2. The media resources required by the trunk are provided by the IP Media Processor (MedPro) circuit pack. Two versions of the IP MedPro circuit pack were tested in this configuration: TN2302AP and TN2602AP. Endpoints at this site include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 9600 Series IP Telephone (with SIP firmware), and a fax machine.

Located at Site 2 is an SIP Enablement Services server and an Avaya S8300 Server running Communication Manager in an Avaya G700 Media Gateway. RightFax at this site is also running on a Windows Server 2008 laptop PC and communicates to the Avaya SIP infrastructure (Communication Manager and SIP Enablement Services) via SIP trunks. On the Avaya G700 Media Gateway, the signaling and media resources needed to support SIP and H.323 trunks are integrated directly on the media gateway processor. Endpoints at this site include an Avaya 1600 Series IP Telephone (with H.323 firmware), Avaya 9600 Series IP Telephones (with H.323 firmware and SIP firmware), and a fax machine.

Although the IP telephones are not involved in the faxing operations, they are present in the configuration to verify VoIP telephone calls are not affected by the FoIP faxing operations and vice versa.

Outbound fax calls originating from RightFax are sent to the SIP Enablement Services server first, then from the SIP Enablement Services to the Communication Manager, via the configured SIP trunks. Based on the dialed digits, the Communication Manager will direct the calls to the local fax machine, or the inter-site trunks (ISDN-PRI or H.323) to reach the remote site. Inbound fax calls terminating to RightFax are first received by the Communication Manager from the local fax machine or from across either ISDN-PRI or H.323 trunks connected to the remote site. The Communication Manager then directs the calls to RightFax via the configured SIP trunks.

# 3. Equipment and Software Validated

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

| Equipment | Software/Firmware |
|---|---|
| Avaya S8500 Server (site 1) | Communication Manager 5.2<br>Service Pack R015x.02.0.947.3-17250 |
| Avaya G650 Media Gateway (Site 1)<br>- 2 CLANs<br>- 2 IP MedPros – TN2302AP<br>- 2 IP MedPros – TN2602AP | TN799DP - HW01 FW26<br>TN2302AP - HW20 FW118<br>TN2602AP - HW02 FW047 |
| SIP Enablement Services (Site 1) | SES-5.2.0.0-947.3b |
| Avaya S8300 Server (Site 2) | Communication Manager 5.2<br>Service Pack R015x.02.0.947.3-17250 |
| Avaya G700 Media Gateway (Site 2) | 28.18.0 |
| SIP Enablement Services (Site 2) | SES-5.2.0.0-947.3b |
| Avaya 1608 IP Telephone (H.323)<br>Avaya 1616 IP Telephone (H.323) | Avaya one-X® Deskphone Value Edition 1.100 |
| Avaya 9620 IP Telephone (SIP)<br>Avaya 9630 IP Telephone (SIP)<br>Avaya 9630 IP Telephone (H.323) | Avaya one-X® Deskphone Edition SIP 2.2<br>Avaya one-X® Deskphone Edition SIP 2.2<br>Avaya one-X® Deskphone Edition H.323 3.0 |
| Analog Fax Machines | - |
| Open Text Fax Server (RightFax) on Windows Server 2008 Laptop PC | 9.4 Service Release 2 |
| Dialogic Brooktrout SR140 Fax Software<br>– Boston Bfv API<br>– Boston Driver<br>– Boston SDK<br>– Boot Rom | v6.0.00 (Build 11)<br>v6.0.00 (Build 7)<br>v6.0.00 (Build 11)<br>6.0.0B4 |

# 4. Configure Avaya Aura™ Communication Manager

This section describes the Communication Manager configuration necessary to interoperate with RightFax. It focuses on the configuration of the SIP trunks connecting RightFax to the Avaya SIP infrastructure with the following assumptions:

- Procedures necessary to support SIP and connectivity to Avaya SES have been performed as described in [3], including all SIP phones at each site.
- All other components are assumed to be in place and previously configured, including the H.323 and ISDN-PRI trunk groups that connect both sites.

The procedures for configuring Communication Manager include the following areas:

- Verify Communication Manager license (Step 1)
- Identify IP Interfaces (Step 2)
- Administer IP network regions (Steps 3 – 6)
- Administer IP node name (Step 7)
- Administer IP network map (Step 8)
- Administer IP codec set (Steps 9 – 10)
- Administer SIP signaling group (Step 11)
- Administer SIP trunk group (Steps 12 – 13)
- Administer public unknown numbering (Step 14)
- Administer route pattern (Step 15)
- Administer AAR analysis (Steps 16 – 17)
- Turn on Media Shuffling on cross-site H.323 trunks (Step 18)

The configuration of the Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration, perform a **save translation** command to make the changes permanent.

The examples shown in this section refer to site 1. Unless specified otherwise, these same steps also apply to site 2 using values appropriate for site 2 from **Figure 1**.

| Step | Description |
|---|---|
| 1. | **Communication Manager License**<br>Use the **display system-parameters customer-options** command to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.<br>The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes |

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 800   100
           Maximum Concurrently Registered IP Stations: 18000 1
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 800   212
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   2
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

| Step | Description |
|---|---|
| 2. | **IP Interfaces**<br>Use the **list ip-interface all** command to identify which IP interfaces are located in which network region. The example below shows the IP interfaces used in the compliance test. All interfaces in cabinet 01 (port network 1) as indicated in the **Slot** field are in IP network region 1 as indicated in the **Net Rgn** field. These interfaces are highlighted below. Testing with the TN2302AP and TN2602AP circuit packs were done separately. When testing with the TN2302AP, the TN2602AP was disabled (turned off) and vice versa as indicated in the **ON** field. |

```
list ip-interface all

                                IP INTERFACES
                                                                     Net
ON Type   Slot   Code Sfx Node Name/       Subnet Mask     Gateway Address Rgn VLAN
                          IP-Address
-- ----   ----   ---- --- --------------- --------------- --------------- --- ----
y MEDPRO 01A02 TN2302    MEDPRO1A        255.255.255.0   192.45.108.1    1   n
                          192.45.108.54
y C-LAN  01A03 TN799  D CLAN1A          255.255.255.0   192.45.108.1    1   n
                          192.45.108.55
y MEDPRO 02A02 TN2302    MEDPRO2A        255.255.255.0   192.45.108.1    2   n
                          192.45.108.56
y C-LAN  02A03 TN799  D CLAN2A          255.255.255.0   192.45.108.1    2   n
                          192.45.108.57
n MEDPRO 01A04 TN2602    MEDPRO1A-2      255.255.255.0   192.45.108.1    1   n
                          192.45.108.58
n MEDPRO 02A04 TN2602    MEDPRO2A-2      255.255.255.0   192.45.108.1    2   n
                          192.45.108.59
```

| Step | Description |
|---|---|
| 3. | **IP Network Region – Region 1**<br>The configuration of the IP network regions (**Steps 3 – 6**) is assumed to be already in place and is included here for clarity. At site 1, the Avaya S8500 Server, the Avaya G650 Media Gateway comprising port network 1, the Avaya SES, and the RightFax fax server were located in IP network region 1 using the parameters described below. Use the **display ip-network-region** command to view these settings. The example below shows the values used for the compliance test.<br>▪ The **Authoritative Domain** field was configured to match the domain name configured on Avaya SES. In this configuration, the domain name is ***avayatest.com***. This name appears in the "From" header of SIP messages originating from this IP region.<br>▪ A descriptive name was entered for the **Name** field.<br>▪ **IP-IP Direct Audio** (Media Shuffling) was enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. This was done for both intra-region and inter-region IP-IP Direct Audio. This is the default setting. Media Shuffling can be further restricted at the trunk level on the **Signaling Group** form.<br>▪ The **Codec Set** field was set to the IP codec set to be used for calls within this IP network region. In this case, IP codec set 1 was selected.<br>▪ The default values were used for all other fields.<br><br>At site 2, all IP components were located in IP network region 1 and the IP network region was configured in the same manner as shown below. |

```
display ip-network-region 1                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avayatest.com
    Name: PN1
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: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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
```

| Step | Description |
|---|---|
| 4. | **IP Network Region 1 – Continued**<br>On **Page 3**, codec sets are defined for inter-region calls. In the case of the compliance test at site 1, calls from IP network Source Region 1 to IP network region 2 (**dst rgn 2**) used codec set 1. The default values were used for all other fields. At site 2, only one IP network region exists so no inter-region settings were required. |

```
display ip-network-region 1                                     Page   3 of  19

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    1                                                                all
 2    1     y    NoLimit                                             n
 3    3     y    NoLimit                                             n all
```

| Step | Description |
|---|---|
| 5. | **IP Network Region – Region 2**<br>At site 1, IP network region 2 was created for Port Netowrk 2 in a similar manner as IP network region 1 shown in **Step 3** but with a different name. |

```
display ip-network-region 2                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain: avayatest.com
    Name: PN2
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: 3329
DIFFSERV/TOS PARAMETERS                 RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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
```

| Step | Description |
|---|---|
| 6. | **IP Network Region 2 – Continued**<br>The inter-region codec setting was created similarly to **Step 4**. |

```
display ip-network-region 2                                     Page   3 of  19

 Source Region: 2     Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    1     y    NoLimit                                             n all
 2    1                                                                all
 3    3     y    NoLimit                                             n all
```

| Step | Description |
|---|---|
| 7. | **IP Node Names**<br>This step is optional. Use the **change node-names ip** command to create a node name that maps to the RightFax server IP address. The example below shows the entry on the Avaya Communication Manager at site 1. Note that this configuration step is not required but will add clarity to the site configuration. |

```
change node-names ip                                            Page   1 of   2
                              IP NODE NAMES
    Name              IP Address
CLAN1A              192.45.108.55
CLAN2A              192.45.108.57
CMnorth             192.45.70.2
MEDPRO1A            192.45.108.54
MEDPRO1A-2          192.45.108.58
MEDPRO2A            192.45.108.56
MEDPRO2A-2          192.45.108.59
RightFax            192.45.108.18
SES                 192.45.108.50
default             0.0.0.0
procr               192.45.108.51
```

| Step | Description |
|---|---|
| 8. | **IP Network Map**<br>If the RightFax server is to be located in an IP network region other than the default region 1, then the region is assigned using the **change ip-network-map** command. In the case of the compliance test, the RightFax IP address at site 1 is assigned to IP network region 2 as shown in the example below. At site 2, the RightFax server is located in the default IP network region 1, so it does not require an IP address map entry. |

```
change ip-network-map                                           Page   1 of  63
                           IP ADDRESS MAPPING

                                             Subnet Network     Emergency
 IP Address                                  Bits   Region VLAN Location Ext
 ------------------------------------------- ------ ------ ---- -------------
 FROM: 192.45.108.18                          /      2      n
   TO: 192.45.108.18
 FROM:                                        /      2      n
   TO:
 FROM:                                        /             n
   TO:
```

| Step | Description |
|---|---|
| 9. | **Codecs**<br>Use the **change ip-codec-set** command to verify that G.711MU or G.711A is contained in the codec list. The example below shows the value used in the compliance test. |

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
```

| Step | Description |
|---|---|
| 10. | **Fax**<br>On **Page 2**, set the **FAX Mode** field to ***t.38-standard*.** This is necessary to support the RightFax server assigned to IP network region 2. The **Modem Mode** field should be set to ***off***.<br>Leave the **FAX Redundancy** setting at its default value of 0. A packet redundancy level can be assigned to improve packet delivery and robustness of FAX transport over the network (with increased bandwidth as trade-off). Avaya uses IETF RFC-2198 and ITU-T T.38 specifications as redundancy standard. With this standard, each Fax over IP packet is sent with additional (redundant) 0 to 3 previous fax packets based on the redundancy setting. A setting of 0 (no redundancy) is suited for networks where packet loss is not a problem. This setting should match the redundancy settings in Brooktroute SR140 configuration; otherwise Brooktroute SR140 will negotiate T.38 redundancy to the most common denominator (no redundancy in this case). |

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n



                    Mode                Redundancy
    FAX             t.38-standard         0
    Modem           off                   0
    TDD/TTY         US                    3
    Clear-channel   n                     0
```

| Step | Description |
|---|---|
| 11. | **Signaling Group for Fax Calls**<br>For the compliance test, this signaling group and the associated SIP trunk group are used for routing fax calls to/from the RightFax server. For the compliance test at site 1, signaling group 7 was configured using the parameters highlighted below. All other fields were set as described in [3].<br>▪ The **Group Type** was set to ***sip***.<br>▪ The **Transport Method** was set to the recommended default value of ***tls*** (Transport Layer Security). As a result, the **Near-end Listen Port** and **Far-end Listen Port** are automatically set to ***5061.***<br>▪ The **Near-end Node Name** was set to ***CLAN2A***, the node name that maps to the IP address of the CLAN circuit pack used to connect to RightFax. Node names are defined using the **change node-names ip** command (see **Step 7** above).<br>▪ The **Far-end Node Name** was set to ***SES***. This node name maps to the IP address of the SIP Enablement Services server as defined using the **change node-names ip** command.<br>▪ The **Far-end Network Region** was set to ***2***. This is the IP network region which contains RightFax.<br>▪ The **Far-end Domain** was set to the IP address assigned to RightFax. This domain is sent in the headers of SIP INVITE messages for calls originating from and terminating to the fax server using this signaling group.<br>▪ **Direct IP-IP Audio Connections** was set to ***y***. This field must be set to ***y*** to enable Media Shuffling on the trunk level (see **Step 3** on **IP-IP Direct Audio**).<br>▪ The **DTMF over IP** field was set to the default value of ***rtp-payload***. This value enables the Communication Manager to send DTMF transmissions using RFC 2833.<br>▪ The default values were used for all other fields. |

```
add signaling-group 7
                                SIGNALING GROUP

 Group Number: 7              Group Type: sip
                        Transport Method: tls






   Near-end Node Name: CLAN2A              Far-end Node Name: SES
 Near-end Listen Port: 5061              Far-end Listen Port: 5061
                                      Far-end Network Region: 2
       Far-end Domain: 192.45.108.18

                                           Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
                                                      IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3              Alternate Route Timer(sec): 6
```

| Step | Description |
|---|---|
| 12. | **Trunk Group for Fax Calls**<br>For the compliance test, trunk group 7 was used for the SIP trunk group for routing fax calls to/from RightFax. Trunk group 7 was configured using the parameters highlighted below. All other fields were set as described in [3].<br><br>On **Page 1**:<br>▪ The **Group Type** field was set to ***sip***.<br>▪ A descriptive name was entered for the **Group Name**.<br>▪ An available trunk access code (TAC) that was consistent with the existing dial plan was entered in the **TAC** field.<br>▪ The **Service Type** field was set to ***tie***.<br>▪ The **Signaling Group** was set to the signaling group shown in the previous step.<br>▪ The **Number of Members** field contained the number of trunks in the SIP trunk group. It determines how many simultaneous SIP calls can be supported by the configuration. Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call.<br>▪ The default values were used for all other fields. |

```
add trunk-group 7                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 7                   Group Type: sip           CDR Reports: y
  Group Name: RightFax                   COR: 1        TN: 1        TAC: *007
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 7
                                                    Number of Members: 10
```

| Step | Description |
|---|---|
| 13. | **Trunk Group for Fax Calls – continued**<br>On **Page 3**:<br>▪ Set the **Numbering Format** field to ***public***. This field specifies the format of the calling party number sent to the far-end.<br>▪ Default values may be used for all other fields. |

```
add trunk-group 7                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
```

| Step | Description |
|---|---|
| 14. | **Public Unknown Numbering**<br>Public unknown numbering defines the calling party number to be sent to the far-end. Use the **change public-unknown-numbering** command to create an entry that will be used by the trunk groups defined in **Steps 12-13**. In the example shown below, all calls originating from a 5-digit extension beginning with 2 and routed across any trunk group (**Trk Grp** column is blank) will be sent as a 5-digit calling number. |

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                     Total Administered: 1
 5  2                                       5           Maximum Entries: 9999
```

| Step | Description |
|---|---|
| 15. | **Route Pattern**<br>Use the **change route-pattern** command to create a route pattern that will route fax calls to the SIP trunk that connects to the RightFax server.<br>The example below shows the route pattern used for the compliance test at site 1. A descriptive name was entered for the **Pattern Name** field. The **Grp No** field was set to the trunk group created in **Steps 12–13**. The Facility Restriction Level (**FRL**) field was set to a level that allows access to this trunk for all users that require it. The value of ***0*** is the least restrictive level. The default values were used for all other fields. |

```
change route-pattern 7                                       Page   1 of   3
                    Pattern Number: 7   Pattern Name: RightFax
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 7    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  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                      none
 2: y y y y y n  n            rest                                      none
 3: y y y y y n  n            rest                                      none
```

| Step | Description |
|---|---|
| 16. | **Routing Calls to RightFax**<br>Automatic Alternate Routing (AAR) was used to route calls to RightFax. Use the **change aar analysis** command to create an entry in the AAR Digit Analysis Table for this purpose. The example below shows entries previously created for site 1 using the **display aar analysis 0** command. The 3$^{rd}$ highlighted entry specifies that numbers that begin with 7 and are 5 digits long use route pattern 7. Route pattern 7 routes calls to the RightFax fax server at Site 1. |

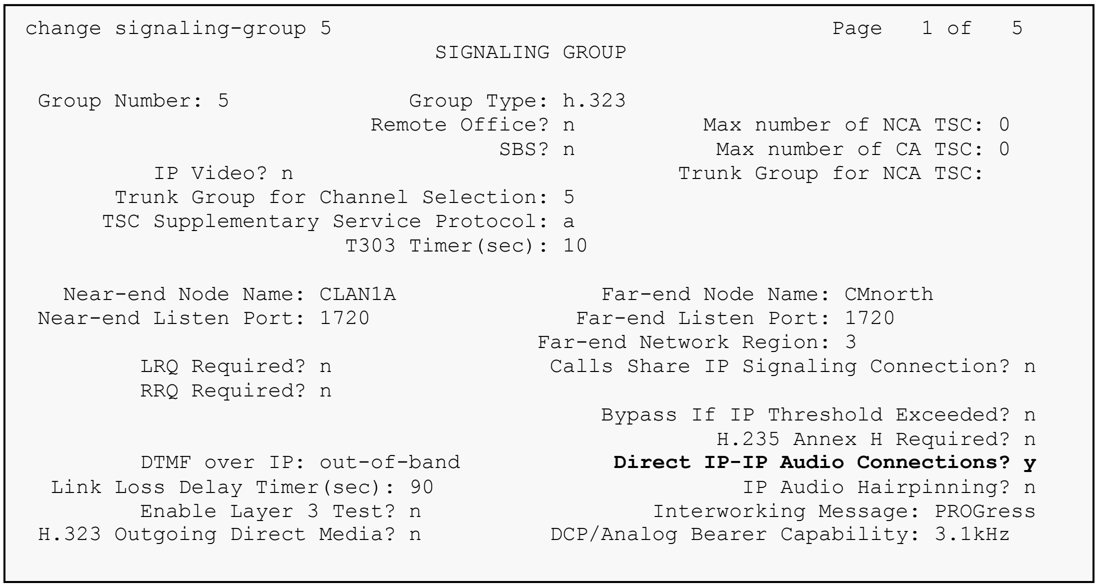
```
display aar analysis 0                                        Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    1

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max   Pattern    Type   Num   Reqd
    50                    5    5      4         aar          n
    52                    5    5      4         aar          n
    7                     5    5      7         aar          n
```

| Step | Description |
|---|---|
| 17. | **Routing Calls From Site 1 to Site 2**<br>The AAR Digit Analysis Table in **Step 16** also shows that a 5-digit dialed number starting with 50 or 52 will use route pattern 4 by AAR. The previously created route pattern 4 as displayed below specifies that a call from Site 1 to the fax machine at 50003 or the RightFax server at 52xxx at Site 2 will be routed to trunk group 4 which is an administered ISDN-PRI trunk. In the same way, this trunk group can be changed to an H.323 trunk group for fax calls from Site 1 to Site 2 to go over an H.323 trunks. |

```
display route-pattern 4                                         Page   1 of   3
                    Pattern Number: 4   Pattern Name: CMnorth RP
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               QSIG
                             Dgts                                      Intw
 1: 4    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  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                         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
```

| Step | Description |
|---|---|
| 18. | **Turn On Media Shuffling on H.323 Trunk between Sites**<br>Use the **change signaling-group** command to turn on Media Shuffling on the previously administered H.323 trunks between Site 1 and Site 2 (in this compliance test, trunk group 5 was used at Site 1).<br> |

```
change signaling-group 5                                      Page   1 of   5
                                SIGNALING GROUP

 Group Number: 5             Group Type: h.323
                          Remote Office? n         Max number of NCA TSC: 0
                                    SBS? n          Max number of CA TSC: 0
          IP Video? n                            Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 5
      TSC Supplementary Service Protocol: a
                        T303 Timer(sec): 10

   Near-end Node Name: CLAN1A                Far-end Node Name: CMnorth
 Near-end Listen Port: 1720                Far-end Listen Port: 1720
                                        Far-end Network Region: 3
         LRQ Required? n                 Calls Share IP Signaling Connection? n
         RRQ Required? n
                                             Bypass If IP Threshold Exceeded? n
                                                      H.235 Annex H Required? n
         DTMF over IP: out-of-band             Direct IP-IP Audio Connections? y
  Link Loss Delay Timer(sec): 90                         IP Audio Hairpinning? n
         Enable Layer 3 Test? n                   Interworking Message: PROGress
 H.323 Outgoing Direct Media? n          DCP/Analog Bearer Capability: 3.1kHz
```

# 5. Configure Avaya Aura™ SIP Enablement Services

This section covers the configuration of the SIP Enablement Services at site 1. The SIP Enablement Services are configured via an Internet browser using the administration web interface. It is assumed that the SIP Enablement Services software and the license file have already been installed on the server. During the software installation, an installation script is run from the Linux shell of the server to specify the IP network properties of the server along with other parameters. In addition, it is assumed that the setup screens of the administration web interface have been used for initial configurations. For additional information on these installation tasks, refer to [4].

Each SIP endpoint used in the compliance test that registers with the SIP Enablement Services requires that a user and media server extension be created in the SIP Enablement Services. This configuration is not directly related to the interoperability between RightFax, and the Avaya SIP infrastructure (Communication Manager and SIP Enablement Services), so it is not included here. These procedures are covered in [4].

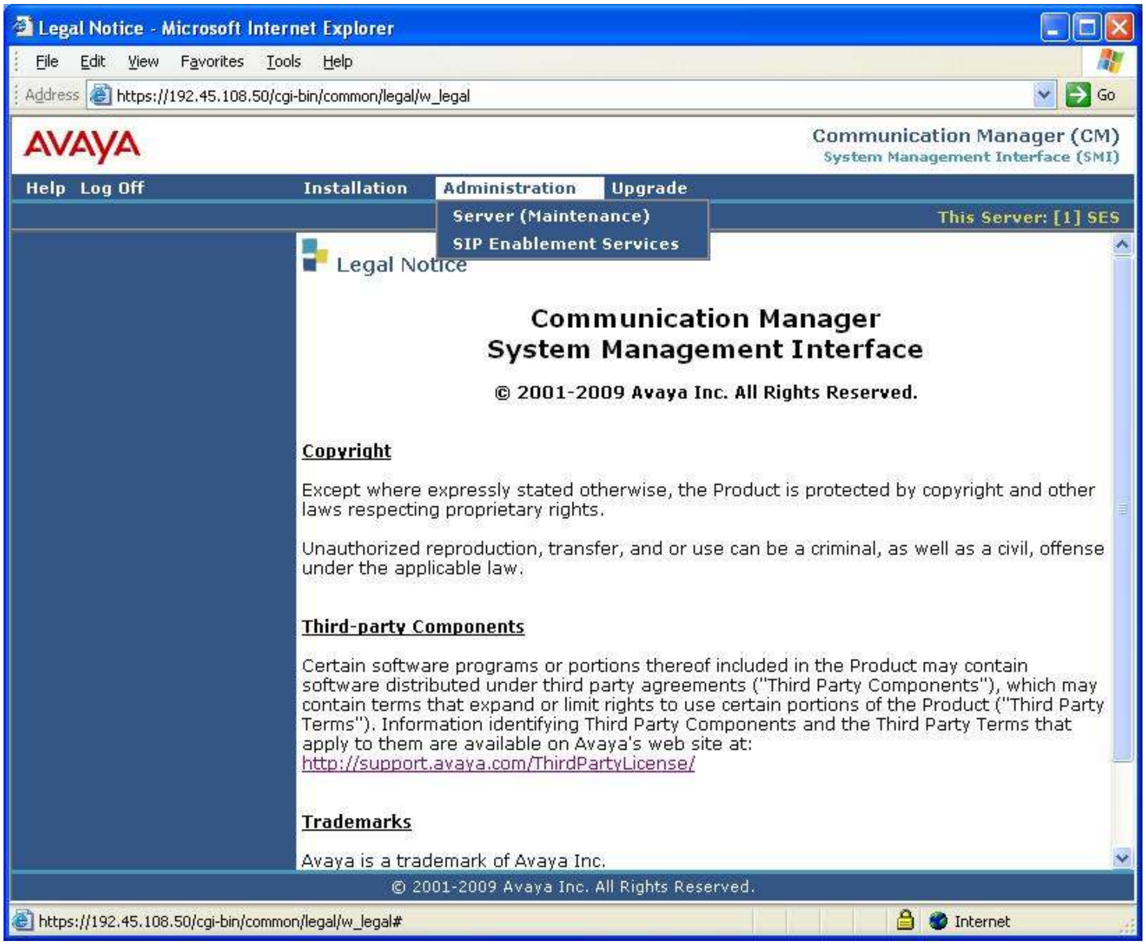
This section is divided into two parts. **Section 5.1** summarizes the user-defined parameters used in the SIP Enablement Services installation procedures that are important for the understanding of the solution as a whole. It does not attempt to show the installation procedures in their entirety. It also describes any deviations from the standard procedures, if any.

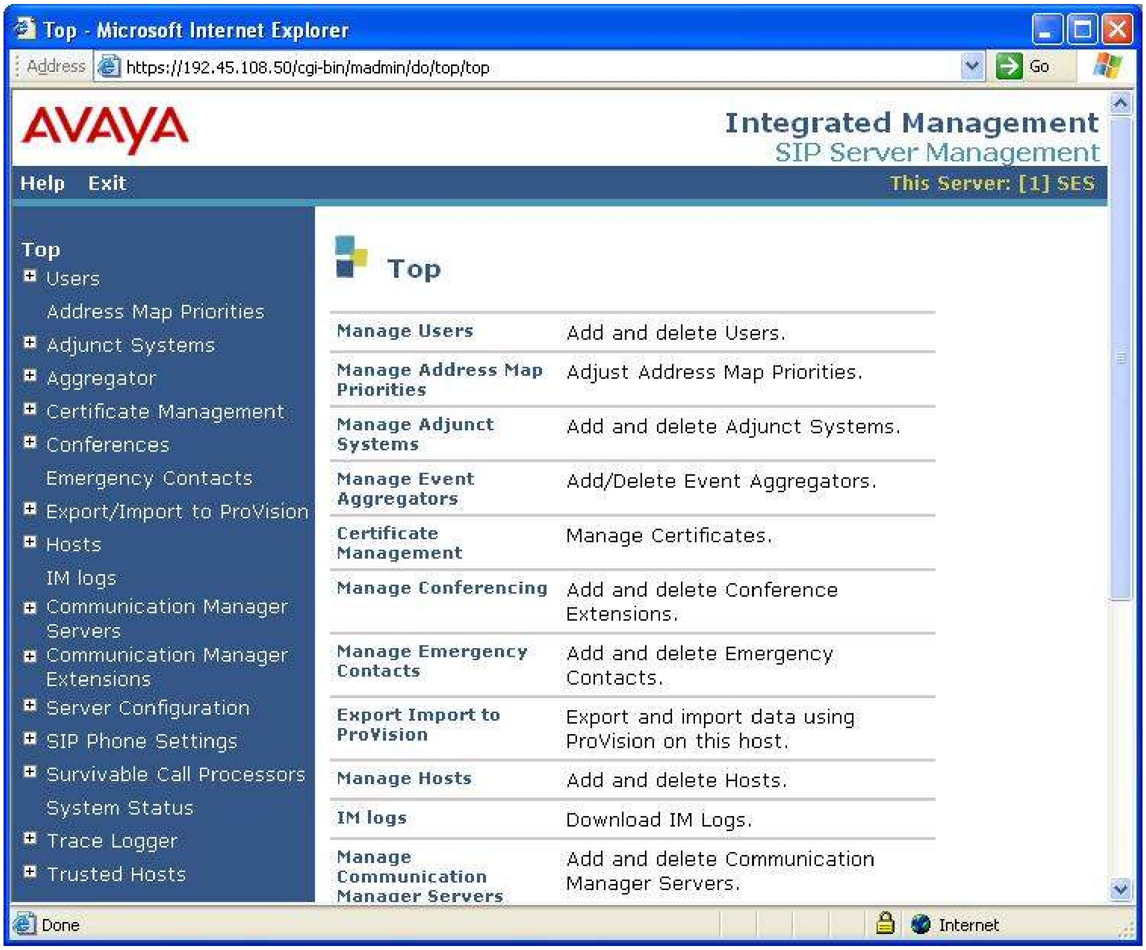
**Section 5.2** describes configurations beyond those covered in **Section 5.1** that are necessary for interoperating with RightFax.

The documented configurations must be repeated for the SIP Enablement Services at site 2 using values appropriate for site 2 from **Figure 1**. This includes but is not limited to the IP addresses, SIP domain and user extensions.

## 5.1. Summarize Initial Configuration Parameters

This section summarizes the applicable user-defined parameters used during the SIP installation procedures.

| Step | Description |
|---|---|
| 1. | **Login**<br>Access the Avaya SES administration web interface by entering http://<*ip-addr*>/admin as the URL in an Internet browser, where <*ip-addr*> is the IP address of the Avaya SES server.<br><br>Log in with the appropriate credentials and select **Administration → SIP Enablement Services** from the top menu on the main page as shown below.<br><br> <br>**Communication Manager**<br>**System Management Interface**<br>**© 2001-2009 Avaya Inc. All Rights Reserved.**<br><br>**Copyright**<br>Except where expressly stated otherwise, the Product is protected by copyright and other laws respecting proprietary rights.<br>Unauthorized reproduction, transfer, and or use can be a criminal, as well as a civil, offense under the applicable law.<br><br>**Third-party Components**<br>Certain software programs or portions thereof included in the Product may contain software distributed under third party agreements ("Third Party Components"), which may contain terms that expand or limit rights to use certain portions of the Product ("Third Party Terms"). Information identifying Third Party Components and the Third Party Terms that apply to them are available on Avaya's web site at: http://support.avaya.com/ThirdPartyLicense/<br><br>**Trademarks**<br>Avaya is a trademark of Avaya Inc.<br>© 2001-2009 Avaya Inc. All Rights Reserved. |

| Step | Description |
|---|---|
| 2. | **Top Page**<br>The Avaya SES **Top** page will be displayed as shown below.<br> |

| | |
|---|---|
| **Manage Users** | Add and delete Users. |
| **Manage Address Map Priorities** | Adjust Address Map Priorities. |
| **Manage Adjunct Systems** | Add and delete Adjunct Systems. |
| **Manage Event Aggregators** | Add/Delete Event Aggregators. |
| **Certificate Management** | Manage Certificates. |
| **Manage Conferencing** | Add and delete Conference Extensions. |
| **Manage Emergency Contacts** | Add and delete Emergency Contacts. |
| **Export Import to ProVision** | Export and import data using ProVision on this host. |
| **Manage Hosts** | Add and delete Hosts. |
| **IM logs** | Download IM Logs. |
| **Manage Communication Manager Servers** | Add and delete Communication Manager Servers. |

| Step | Description |
|---|---|
| 3. | **Initial Configuration Parameters**<br>As part of the Avaya SES installation and initial configuration procedures, the following parameters were defined. Although these procedures are out of the scope of these Application Notes, the values used in the compliance test are shown below for reference. After each group of parameters is a brief description of the required steps to view the values for that group from the Avaya SES administration home page shown in the previous step. Note that for Site 2, the **SIP Trunk IP Address** should be set to the IP assigned to the Avaya Communication Manager (***procr***) since there is no separate CLAN circuit pack in the Avaya G700 Media Gateway.<br>• SIP Domain: ***avayatest.com***<br>(To view, navigate to **Server Configuration→System Properties**)<br>• Host IP Address (SES IP address): ***192.45.108.50***<br>• Host Type: ***SES combined home-edge***<br>(To view, navigate to **Hosts→List**; click **Edit**)<br>• Communication Manager Interface Name: ***CLAN2A***<br>• SIP Trunk Link Type: ***TLS***<br>• SIP Trunk IP Address (CLAN2A IP address): ***192.45.108.57***<br>(To view, navigate to **Communication Manger Servers→List**; click **Edit**) |

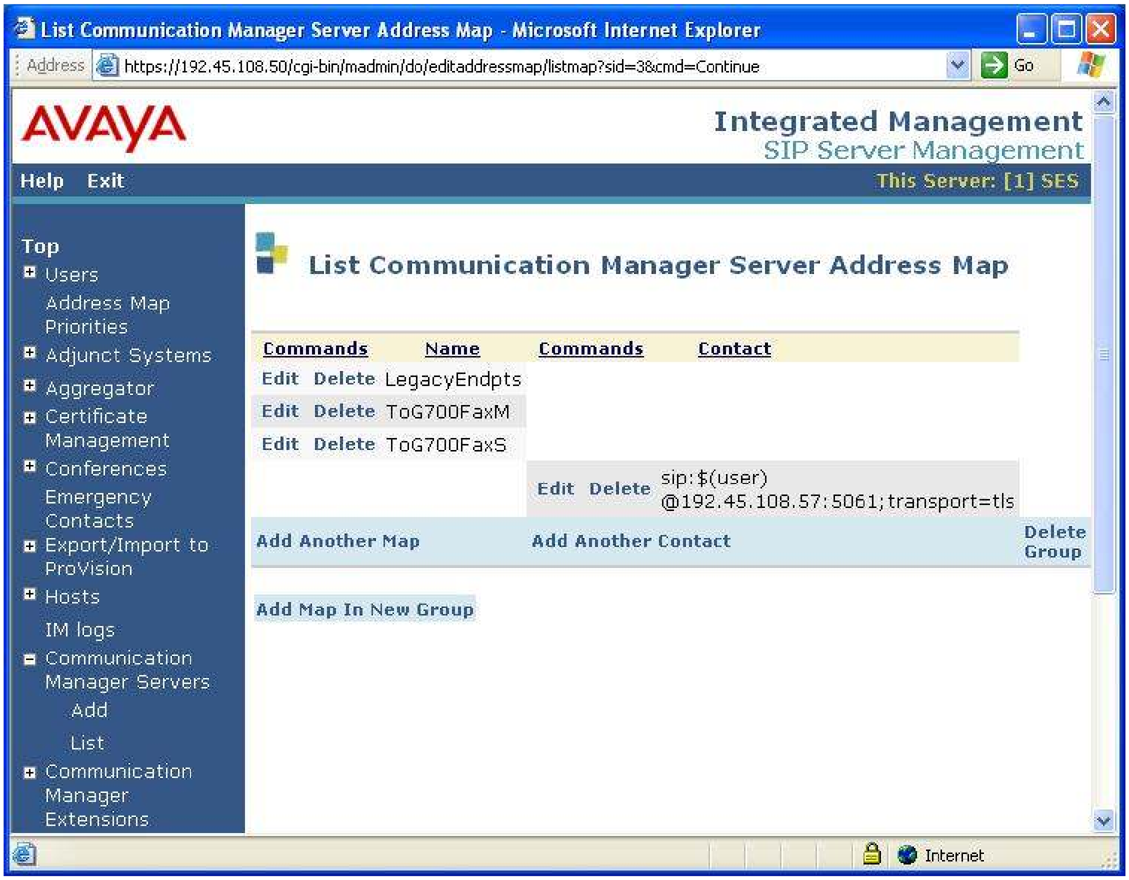
## 5.2. RightFax Specific Configuration

This section describes additional SIP Enablement Services configurations necessary for interoperating with RightFax. These specific configurations include the following:

- Administer Communication Manager servers map (Steps 1 – 4)
- Administer trusted host (Step 5)

| Step | Description |
|---|---|
| 1. | **Communication Manager Server Address Map**<br>A Communication Manager Server Address Map is needed to route calls to the fax machines (local or remote) or the RightFax fax server at the remote site. This is because neither the caller nor the called party is a registered user on the local Avaya SES with a media server extension assigned to it. Thus, Avaya SES does not know how to route this call to Avaya Communication Manager. To accomplish this task, a Communication Manager Server Address Map is needed.<br>To view the Communication Manager Server Address Maps, navigate to **Communication Manager Servers → List** in the left pane.<br> |

| Commands | | | | | Interface | Host |
|---|---|---|---|---|---|---|
| Edit | Extensions | Map | Test-Link | Delete | CLAN1A | 192.45.108.50 |
| Edit | Extensions | Map | Test-Link | Delete | CLAN2A | 192.45.108.50 |

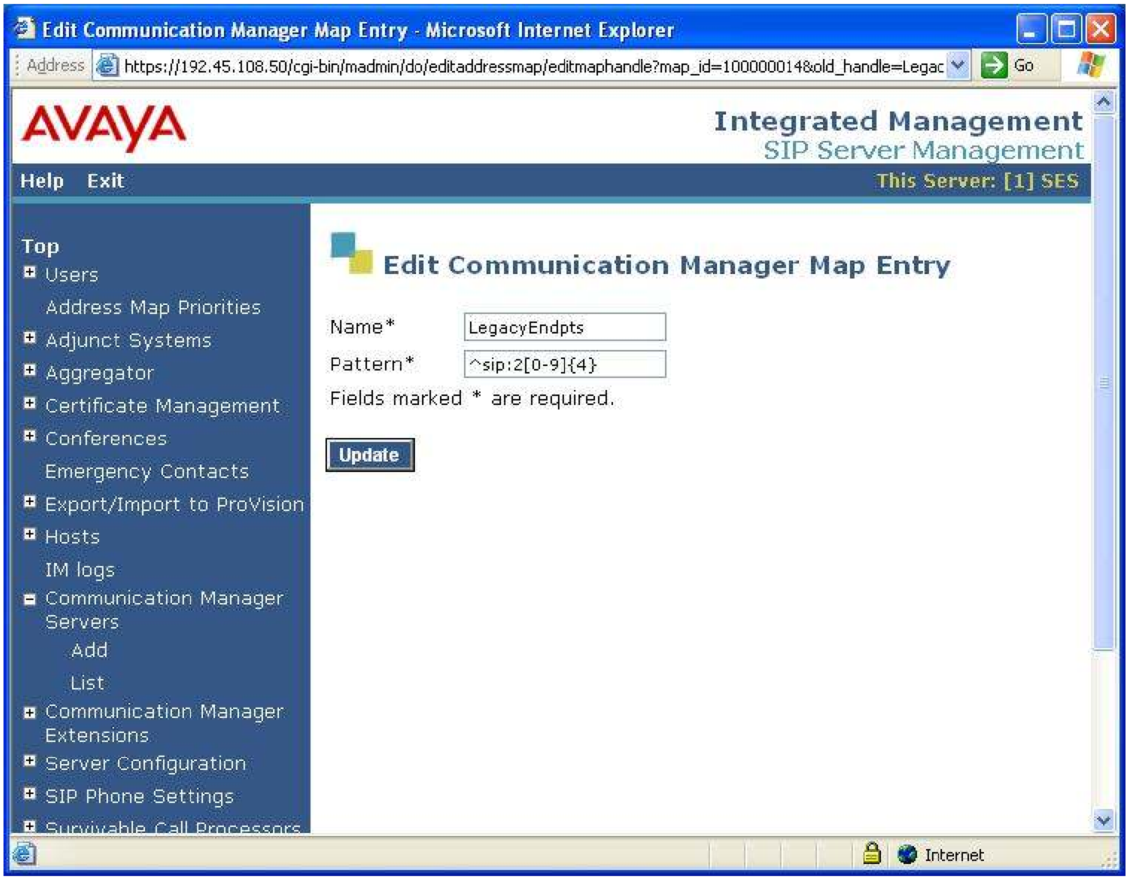
| Step | Description |
|---|---|
| 2. | **Communication Server Address Map – Continued**<br>In the displayed window above, click the **Map** link next to the **CLAN2A** interface name. The list of Communication Manager Server Address Maps will appear as shown below. Each map defines criteria for matching calls to the Avaya SES based on the contents of the SIP Request-URI of the call<br><br>In the example below, three configured maps are shown for the compliance test:<br>– ***LegacyEndpts*** was used for mapping calls to the fax machine at local site<br>– ***ToG700FaxM*** was used for mapping calls to the fax machine at remote site<br>– ***ToG700FaxS*** was used for mapping calls to the RightFax fax server at remote site<br><br>All 3 maps were associated to a **Contact** that directs the calls to the IP address of the ***CLAN2A*** interface using port ***5061*** and ***TLS*** as the transport protocol. The user portion in the original request URI is substituted for ***$(user)*** in the **Contact** expression shown below and in the screenshot:<br><br>`sip:$(user)@192.45.108.57:5061;transport=tls`<br><br> <br>**List Communication Manager Server Address Map**<br>Commands \| Name \| Commands \| Contact<br>Edit Delete \| LegacyEndpts<br>Edit Delete \| ToG700FaxM<br>Edit Delete \| ToG700FaxS<br>\| \| Edit Delete \| sip:$(user)@192.45.108.57:5061;transport=tls<br>Add Another Map \| Add Another Contact \| Delete Group<br>Add Map In New Group |

| Step | Description |
|---|---|
| 3. | **Communication Server Address Map– Continued** |

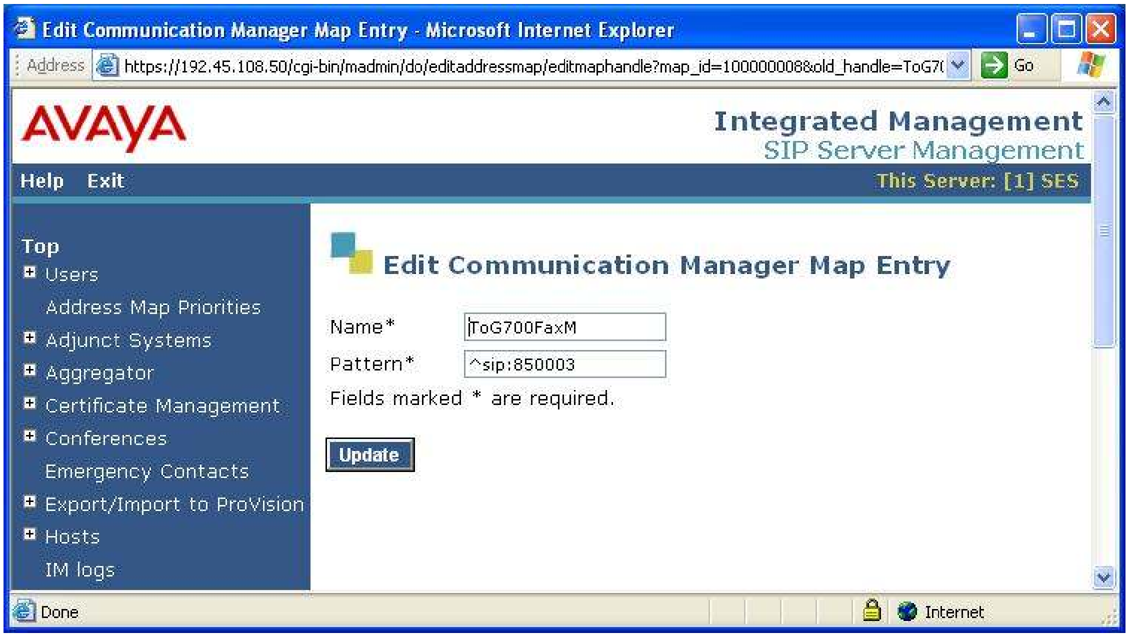
To view or edit the call matching criteria of the map, click the **Edit** link next to the map name. The content of the Communication Server Address Map is described below.

- **Name**: Contains any descriptive name
- **Pattern**: Contains an expression to define the matching criteria for calls to be routed to this Avaya Communication Manager. For the address map named ***LegacyEndpts***, the expression will match any URI that begins with ***sip:2*** followed by any digit between ***0-9*** for the next ***4*** digits. Additional information on the syntax used for address map patterns can be found in [4].

If any changes are made, click **Update**.



Name* LegacyEndpts
Pattern* ^sip:2[0-9]{4}
Fields marked * are required.

| Step | Description |
|---|---|
| 4. | **Communication Server Address Map– Continued**<br>Displayed below are the address maps configured in the compliance test for routing calls to the fax machine and fax server at remote site.<br><br>Edit Communication Manager Map Entry - Microsoft Internet Explorer<br>Address https://192.45.108.50/cgi-bin/madmin/do/editaddressmap/editmaphandle?map_id=100000008&old_handle=ToG7(<br>Edit Communication Manager Map Entry<br>Name* ToG700FaxM<br>Pattern* ^sip:850003<br>Fields marked * are required.<br> <br><br>Edit Communication Manager Map Entry - Microsoft Internet Explorer<br>Address https://192.45.108.50/cgi-bin/madmin/do/editaddressmap/editmaphandle?map_id=100000016&old_handle=ToG7(<br>Edit Communication Manager Map Entry<br>Name* ToG700FaxS<br>Pattern* ^sip:852000<br>Fields marked * are required.<br> |

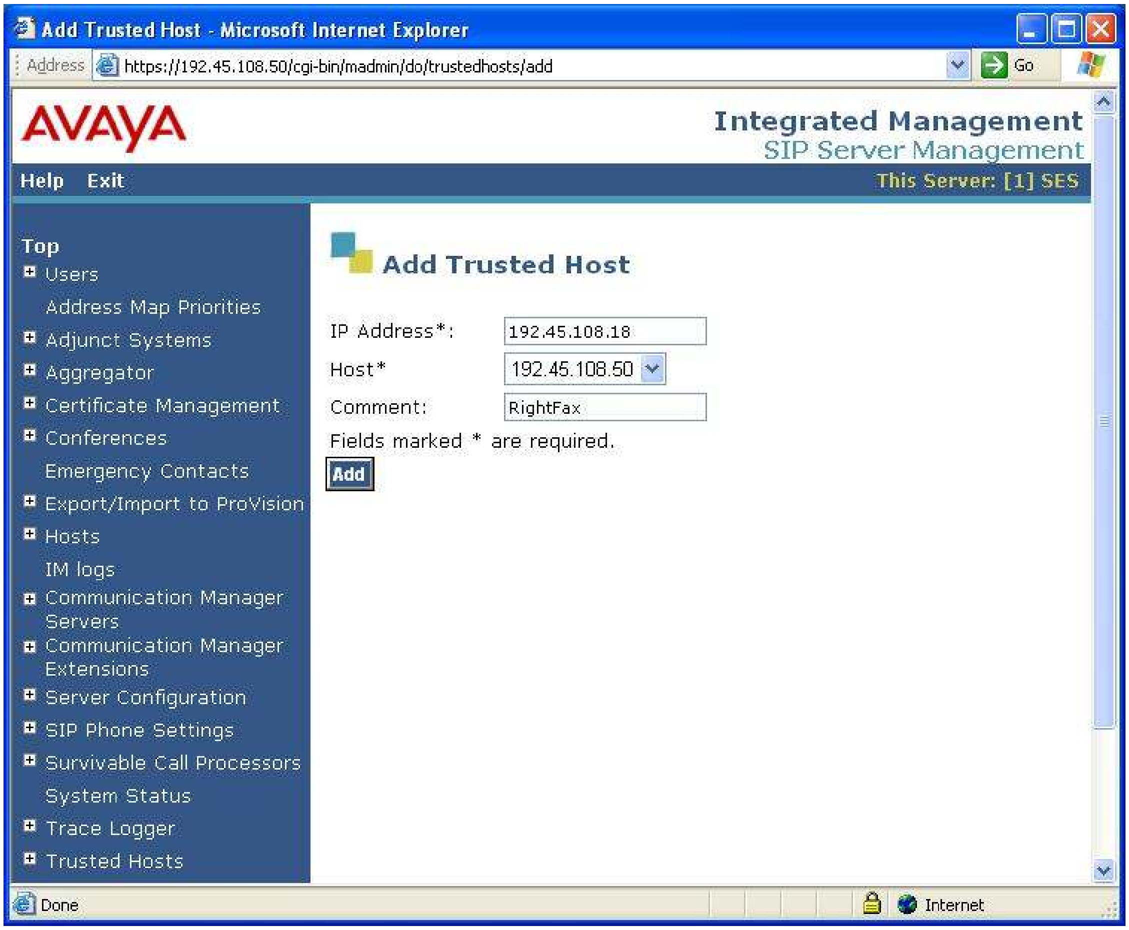
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>

**Trusted Host**

RightFax fax server must be added as a Trusted Host (to the SIP Enablement Services). To add a new Trusted Host, navigate to **Trusted Hosts → Add Trusted Host** in the left pane. In the displayed window, configure the following fields:

- **IP Address**: Enter IP address assigned to the RightFax server
- **Host**: Select the IP address for the Avaya SES
- **Comments**: Enter a descriptive text

After the fields are properly set, click **Add**.



Add Trusted Host

IP Address*: 192.45.108.18
Host* 192.45.108.50
Comment: RightFax
Fields marked * are required.

</td></tr>
</table>

# 6. Configure RightFax

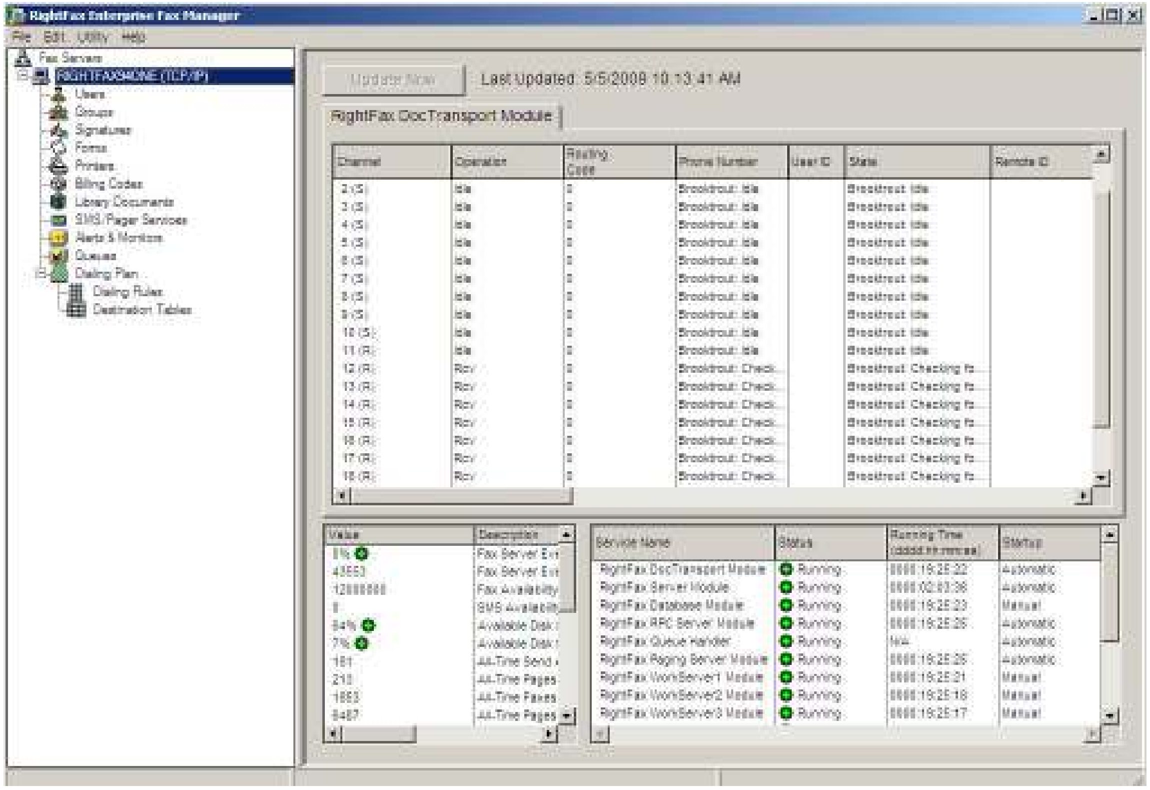
This section describes the configuration of RightFax and the embedded Brooktrout SR140 virtual fax board software from Dialogic. It assumes that the application and all required software components, including Brooktrout SR140 and the database software (MSSQL 2008), have been installed and properly licensed. For instructions on installing RightFax, consult the RightFax Installation Guide [6].

Note that the configurations documented in this section pertain to interoperability between RightFax and the Avaya SIP infrastructure. The standard configurations pertaining to RightFax itself (e.g., administering fax channels) are not covered. For instructions on administering and operating RightFax, consult the RightFax Administrator's Guide [5].

The configuration procedures covered in this section include the following:

- Launch RightFax Enterprise Fax Manager and SR140 Configuration Tool (Steps 1 – 4)
- Configure SR140 IP stack (Step 5)
- Configure SR140 SIP IP parameters (Step 6)
- Configure SR140 T.38 parameters (Step 7)
- Update SR140 configuration file (Step 8)
- Administer RightFax dialing rules (Steps 9 – 11)
- Administer RightFax users (Steps 12 – 15)

The examples shown in this section refer to site 1. Unless specified otherwise, these same steps also apply to site 2 using values appropriate for site 2 from **Figure 1**.

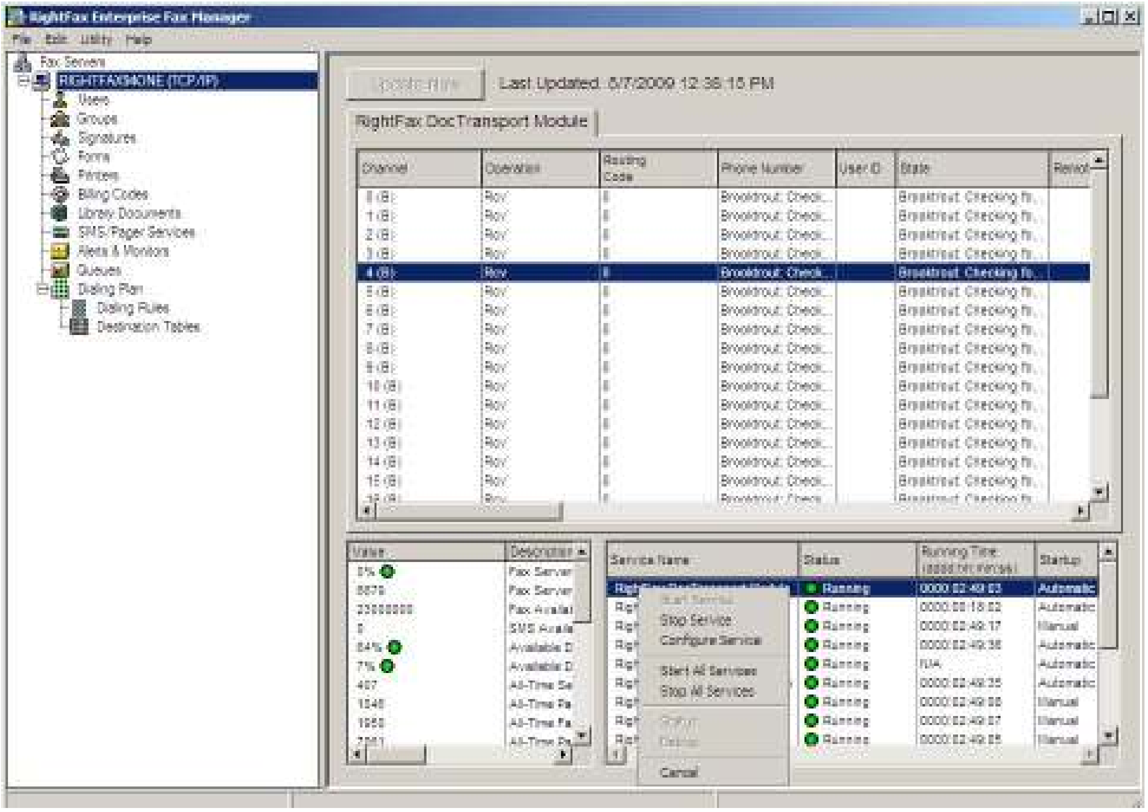
| Step | Description |
|---|---|
| 1. | **Launch RightFax Enterprise Fax Manager**<br>The RightFax configuration is performed using the RightFax Enterprise Fax Manager. Launch the RightFax Enterprise Fax Manager from the Windows Start menu. At the main window, highlight the host name of the fax server (created during the installation process) from the navigation menu in the left pane:<br>.<br> |

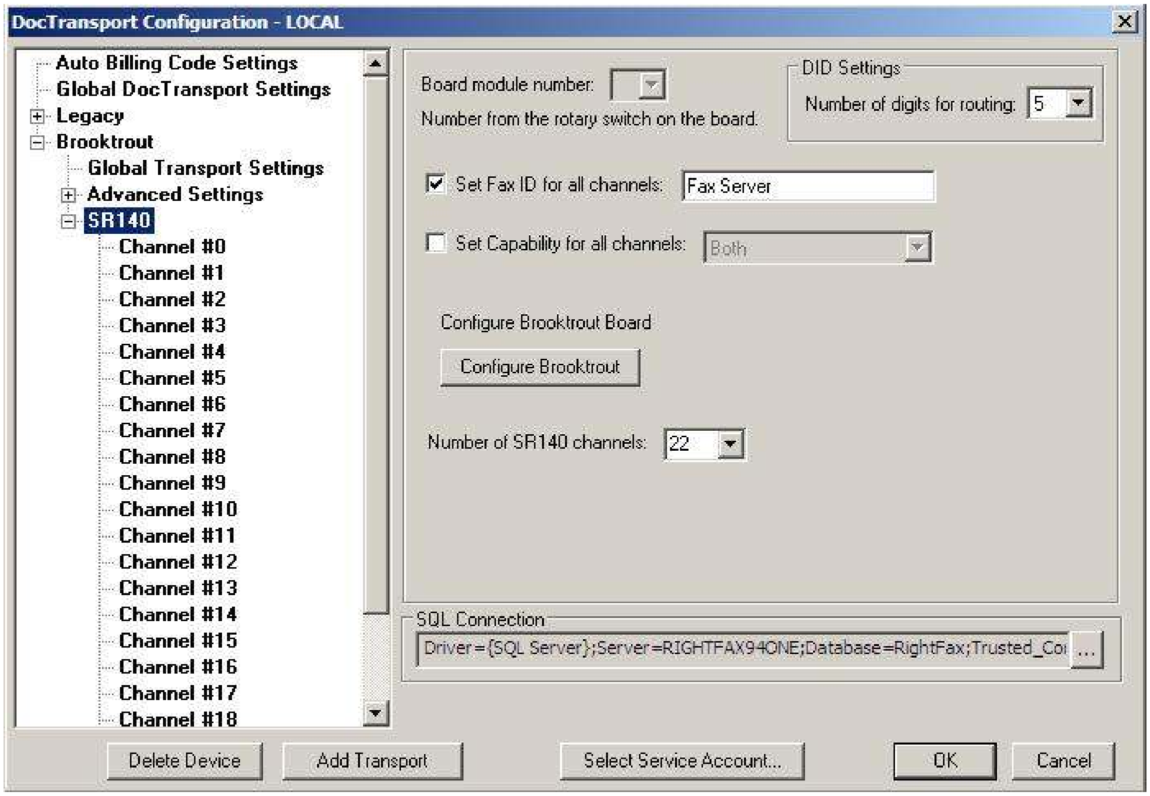
Last Updated: 5/5/2009 10:13:41 AM

| Channel | Operation | Routing Code | Phone Number | User ID | State | Remote ID |
|---|---|---|---|---|---|---|
| 2 (S) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 3 (S) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 4 (S) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 5 (S) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 6 (S) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 7 (S) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 8 (S) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 9 (S) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 10 (S) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 11 (R) | Idle | 0 | Brooktrout: Idle | | Brooktrout: Idle | |
| 12 (R) | Rcv | 0 | Brooktrout: Check... | | Brooktrout: Checking fo... | |
| 13 (R) | Rcv | 0 | Brooktrout: Check... | | Brooktrout: Checking fo... | |
| 14 (R) | Rcv | 0 | Brooktrout: Check... | | Brooktrout: Checking fo... | |
| 15 (R) | Rcv | 0 | Brooktrout: Check... | | Brooktrout: Checking fo... | |
| 16 (R) | Rcv | 0 | Brooktrout: Check... | | Brooktrout: Checking fo... | |
| 17 (R) | Rcv | 0 | Brooktrout: Check... | | Brooktrout: Checking fo... | |
| 18 (R) | Rcv | 0 | Brooktrout: Check... | | Brooktrout: Checking fo... | |

| Value | Description |
|---|---|
| 0% ● | Fax Server Ev |
| 43553 | Fax Server Ev |
| 12000000 | Fax Availability |
| 0 | SMS Availability |
| 94% ● | Available Disk |
| 7% ● | Available Disk |
| 181 | All-Time Send |
| 213 | All-Time Pages |
| 1863 | All-Time Faxes |
| 8487 | All-Time Pages |

| Service Name | Status | Running Time (dddd:hh:mm:ss) | Startup |
|---|---|---|---|
| RightFax DocTransport Module | ● Running | 0000:19:25:22 | Automatic |
| RightFax Server Module | ● Running | 0000:02:03:38 | Automatic |
| RightFax Database Module | ● Running | 0000:19:25:23 | Manual |
| RightFax RPC Server Module | ● Running | 0000:19:25:26 | Automatic |
| RightFax Queue Handler | ● Running | N/A | Automatic |
| RightFax Paging Server Module | ● Running | 0000:19:25:26 | Automatic |
| RightFax WorkServer1 Module | ● Running | 0000:19:25:21 | Manual |
| RightFax WorkServer2 Module | ● Running | 0000:19:25:18 | Manual |
| RightFax WorkServer3 Module | ● Running | 0000:19:25:17 | Manual |

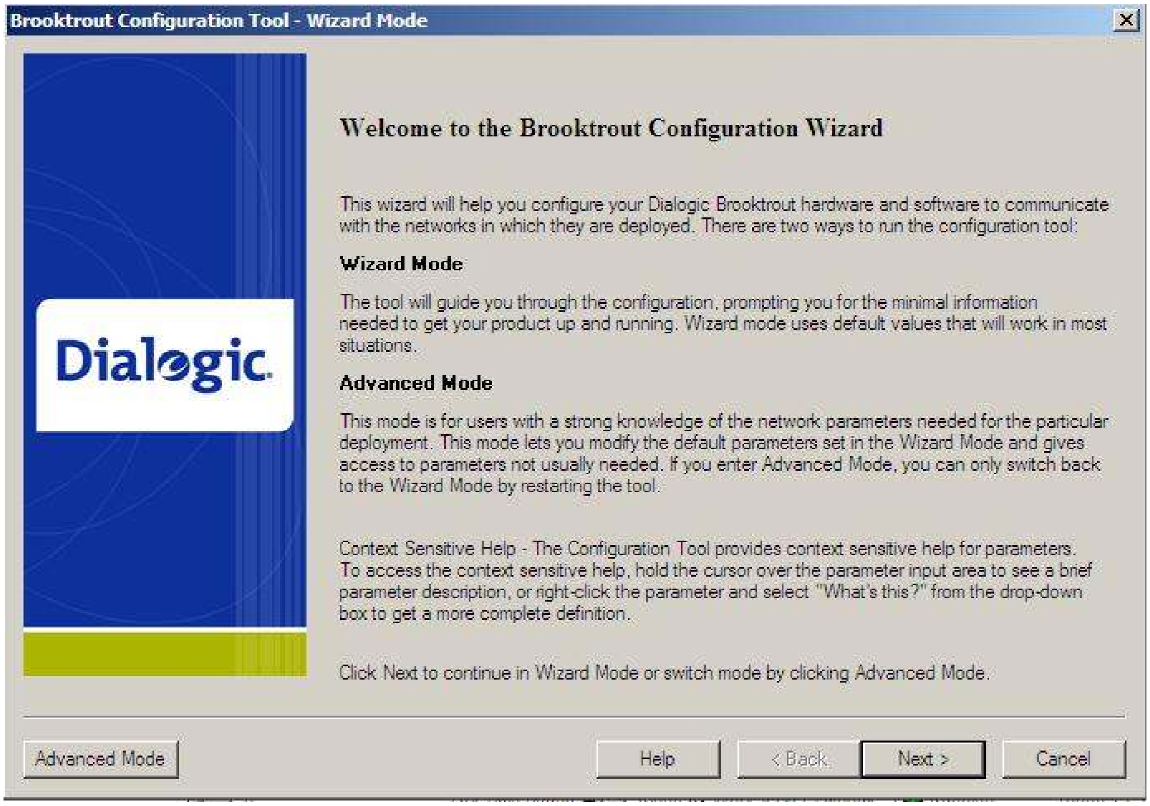
| Step | Description |
|---|---|
| 2. | **RightFax DocTransport Module**<br>The Brooktroute SR140 was configured during installation. To view or modify the settings, the RightFax DocTransport Module must be stopped. Right-click this module in the lower right pane and select **Stop All Services**. After all the service modules indicate the stopped status, right-click the **RightFax DocTransport Module** name again to select **Configure Service**.<br>.<br> |

| Step | Description |
|---|---|
| 3. | **RightFax DocTransport Module - Continued**<br>In the **DocTransport Configuration** window that appears, click the **Configure Brooktrout** button.<br>.<br><br>DocTransport Configuration - LOCAL<br>Board module number:<br>Number from the rotary switch on the board.<br>DID Settings<br>Number of digits for routing: 5<br>☑ Set Fax ID for all channels: Fax Server<br>☐ Set Capability for all channels: Both<br>Configure Brooktrout Board<br>Number of SR140 channels: 22<br>SQL Connection<br>Driver={SQL Server};Server=RIGHTFAX94ONE;Database=RightFax;Trusted_Co<br> |

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>

**Brooktrout Configuration Tool**

The **Brooktroute Configuration Tool – Wizard Mode** window gets displayed. Click the **Advanced Mode** button in this window.



Welcome to the Brooktrout Configuration Wizard

This wizard will help you configure your Dialogic Brooktrout hardware and software to communicate with the networks in which they are deployed. There are two ways to run the configuration tool:

**Wizard Mode**

The tool will guide you through the configuration, prompting you for the minimal information needed to get your product up and running. Wizard mode uses default values that will work in most situations.

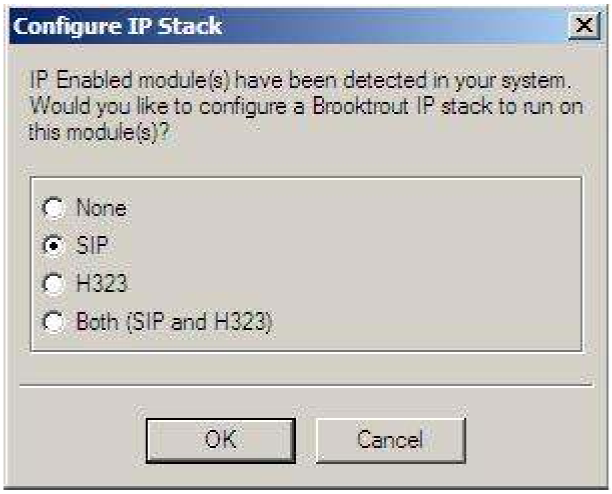
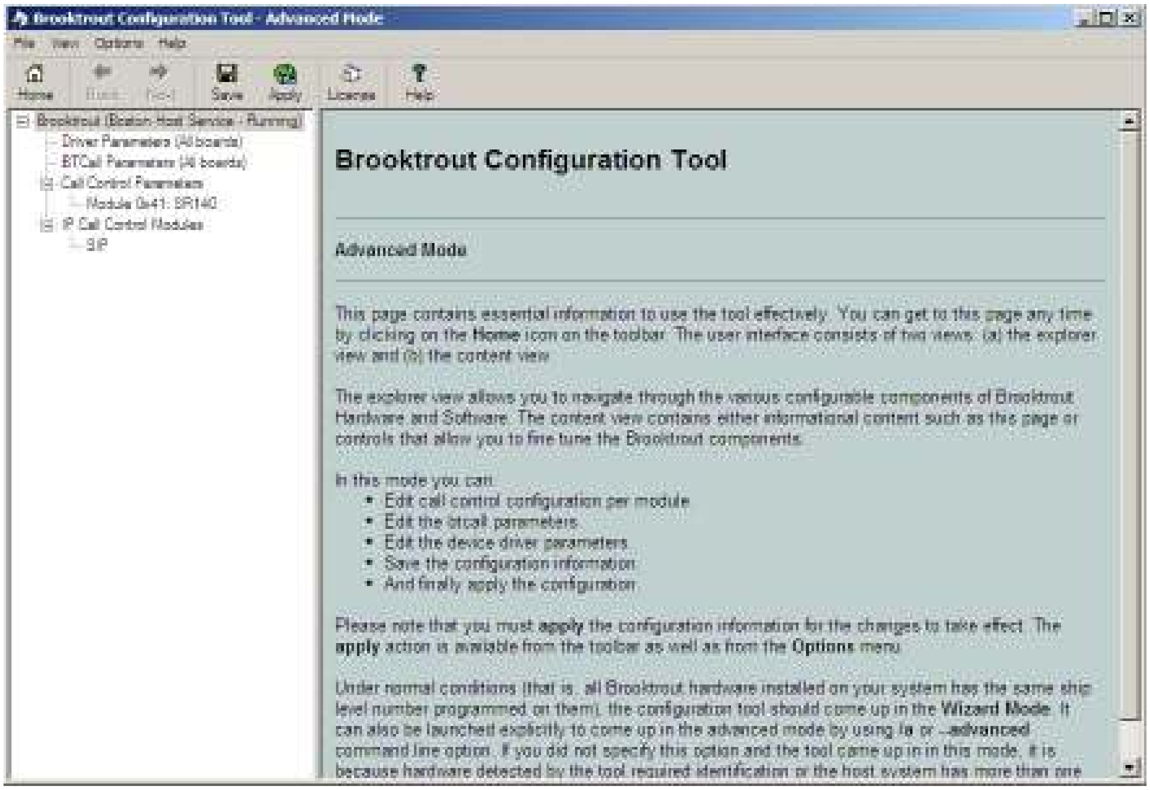
**Advanced Mode**

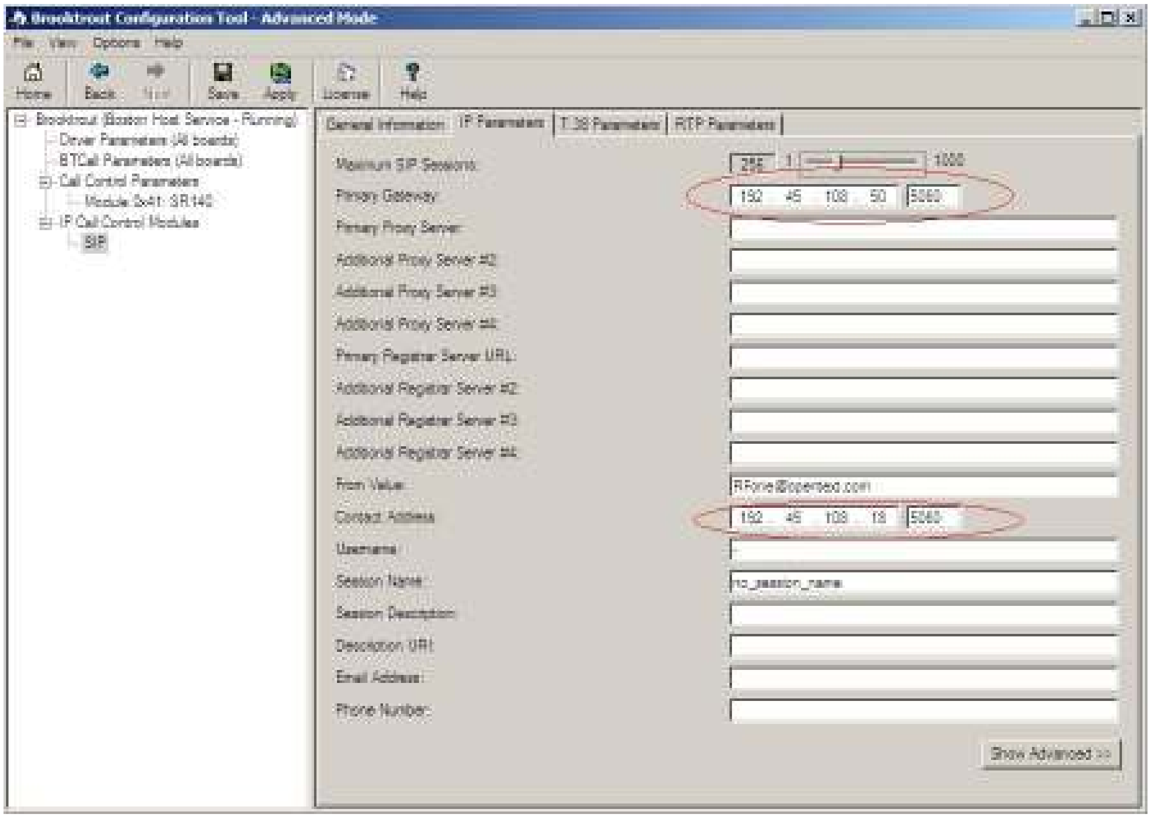
This mode is for users with a strong knowledge of the network parameters needed for the particular deployment. This mode lets you modify the default parameters set in the Wizard Mode and gives access to parameters not usually needed. If you enter Advanced Mode, you can only switch back to the Wizard Mode by restarting the tool.

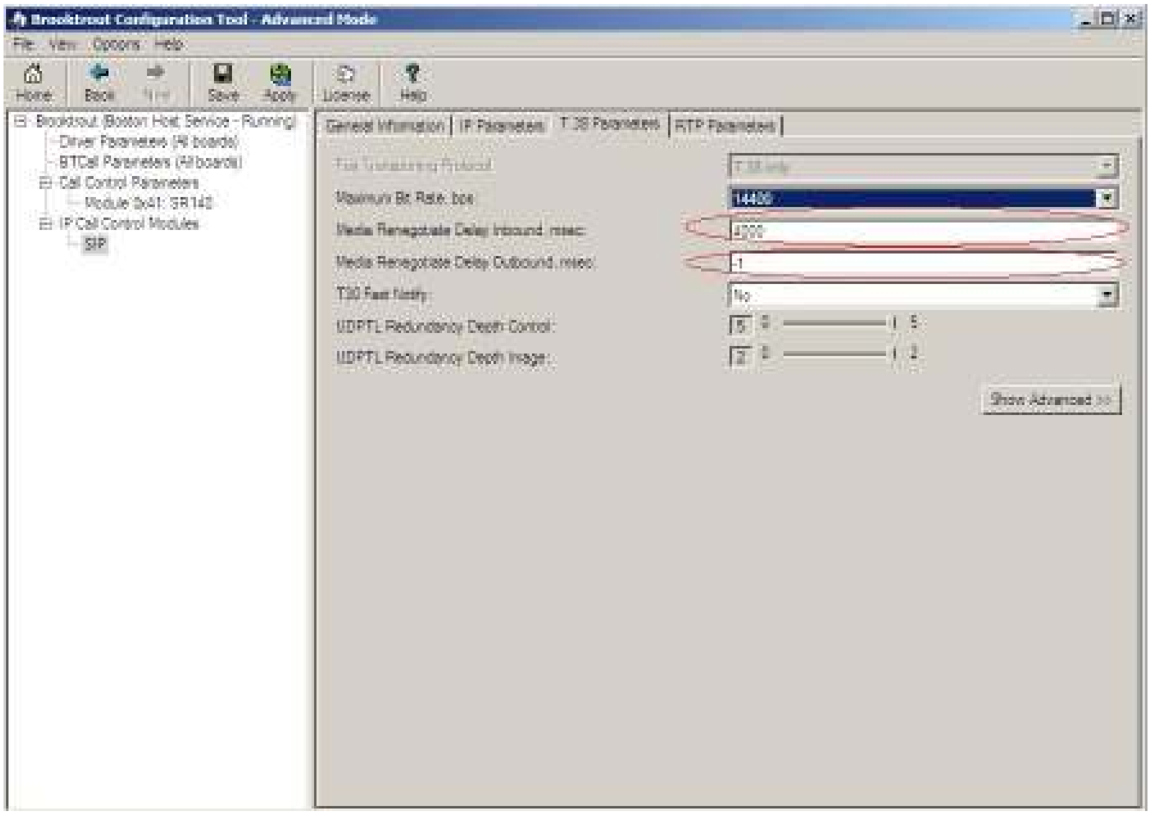
Context Sensitive Help - The Configuration Tool provides context sensitive help for parameters. To access the context sensitive help, hold the cursor over the parameter input area to see a brief parameter description, or right-click the parameter and select "What's this?" from the drop-down box to get a more complete definition.

Click Next to continue in Wizard Mode or switch mode by clicking Advanced Mode.

</td></tr>
</table>

| Step | Description |
|---|---|
| 5. | **Configure IP Stack**<br>A Configure IP Stack window is displayed on first invocation of the Brooktrout SR140 configuration tool (assuming the Brooktrout SR-140 licenses were installed):<br><br> <br>IP Enabled module(s) have been detected in your system. Would you like to configure a Brooktrout IP stack to run on this module(s)?<br>○ None<br>● SIP<br>○ H323<br>○ Both (SIP and H323)<br> <br>Choose **SIP** and click **OK**. The following SR140 configuration tool window is displayed.<br><br> <br>Brooktrout Configuration Tool<br>Advanced Mode<br>This page contains essential information to use the tool effectively. You can get to this page any time by clicking on the **Home** icon on the toolbar. The user interface consists of two views: (a) the explorer view and (b) the content view.<br>The explorer view allows you to navigate through the various configurable components of Brooktrout Hardware and Software. The content view contains either informational content such as this page or controls that allow you to fine tune the Brooktrout components.<br>In this mode you can:<br>• Edit call control configuration per module<br>• Edit the btcall parameters<br>• Edit the device driver parameters<br>• Save the configuration information<br>• And finally apply the configuration<br>Please note that you must **apply** the configuration information for the changes to take effect. The **apply** action is available from the toolbar as well as from the **Options** menu.<br>Under normal conditions (that is, all Brooktrout hardware installed on your system has the same ship level number programmed on them), the configuration tool should come up in the **Wizard Mode**. It can also be launched explicitly to come up in the advanced mode by using **/a** or **--advanced** command line option. If you did not specify this option and the tool came up in in this mode, it is because hardware detected by the tool required identification or the host system has more than one<br>Note that IP Stack can be viewed/reconfigured from the Brooktroute Configuration Tool menu **Options → Configure IP Stack**. |

| Step | Description |
|---|---|
| 6. | **Configure SIP IP Parameters**<br>On the main screen, navigate to **Brooktrout → IP Call Control Modules → SIP** in the left navigation menu. Select the **IP Parameters** tab in the right pane. Configure the fields as follows:<br>• **Primary Gateway** –set to the IP address of the SIP Enablement Services server, and port number ***5060***.<br>• **From Value** – set to ***RFone@avayatest.com*** or some other appropriate value.<br>• **Contact Address** – set to the IP address assigned to RightFax and the port number ***5060***.<br>Use default values for all other fields.<br>.<br><br>Maximum SIP Sessions: 256<br>Primary Gateway: 192 . 45 . 108 . 50 5060<br>From Value: RFone@opentest.com<br>Contact Address: 192 . 45 . 108 . 18 5060<br>Session Name: no_session_name |

| Step | Description |
|---|---|
| 7. | **Configure T.38 Parameters**<br>Select the **T.38 Parameters** tab.  Configure the fields as shown below in the screenshot.<br> <br>Fax Transmission Protocol: T.38 only<br>Maximum Bit Rate, bps: 14400<br>Media Renegotiate Delay Inbound, msec: 4000<br>Media Renegotiate Delay Outbound, msec: -1<br>T30 Fast Notify: No<br>UDPTL Redundancy Depth Control: 5<br>UDPTL Redundancy Depth Image: 2 |

| Step | Description |
|---|---|
| 8. | **Complete Brooktrout SR140 Configuration**<br>After verifying all the above parameters are properly set, click **Save** in the button menu. Then in the command menu, navigate to **File → Exit** to exit the Brooktroute Configuration Tool.<br><br>From Windows explorer, navigate to the Brooktrout folder in the RightFax install directory (typically Program Files\RightFax\DecTransport\Brooktrout), Open and edit the **callctrl.cfg** file as follows, then save the updates:<br><br>• Verify that the following configuration segment is present; add it if necessary. |

```
[host_module.1/rtp]
     rtp_codec=pcmu
```

- Change **rtp_ced_enable** setting to ***true*** under the **[host_modele.1/t.38parameters]** header (. . . below indicates other entries under the header)

```
[host_module.1/t.38parameters]
     ...
     rtp_ced_enable=true
     ...
```

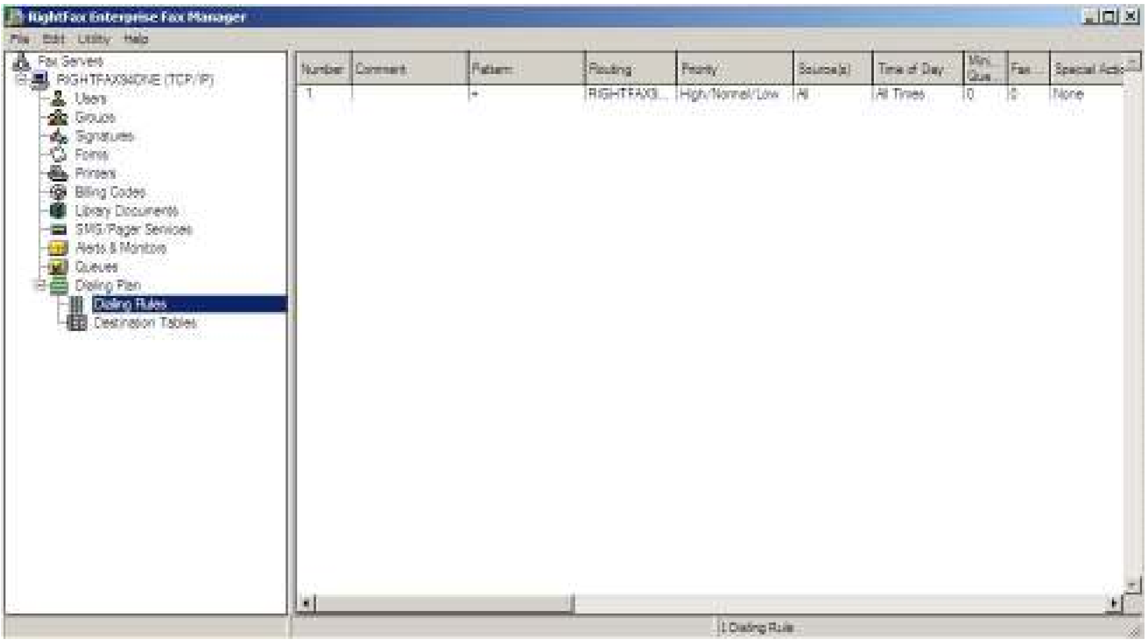
After making and saving the above updates in the **callctrl.cfg** file, restart all RightFax service modules by right clicking the **RightFax DocTransport Module** name in the lower right pane of the RightFax Enterprise Fax Manager window and select **Start All Services** (see **Step 2**).

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>9.</td><td>

**Configure Dialing Rules**

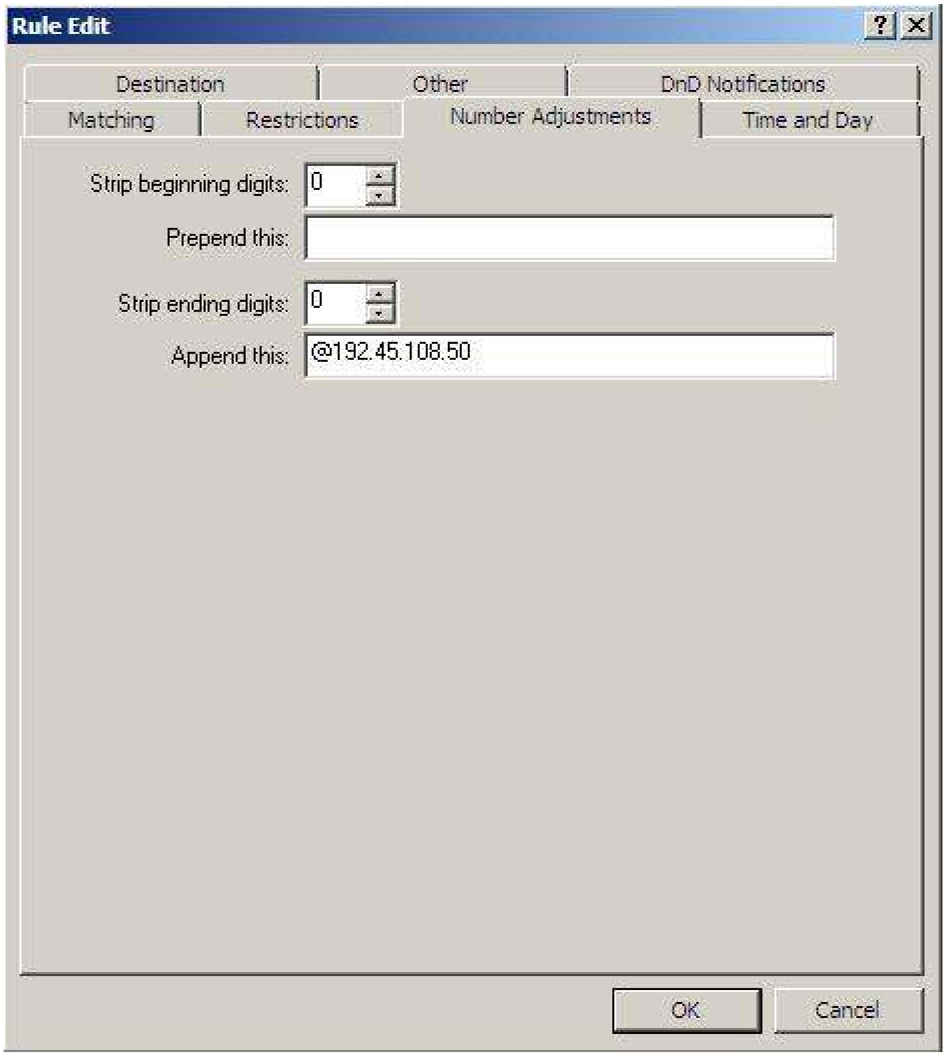
Dialing Rules are used by RightFax to route calls. In the compliance test, a dialing rule is created to route outbound fax calls to the Avaya Media Gateway. In the left navigation menu under the host name of the fax server, navigate to **Dialing Plan → Dialing Rules** to view the existing rules.

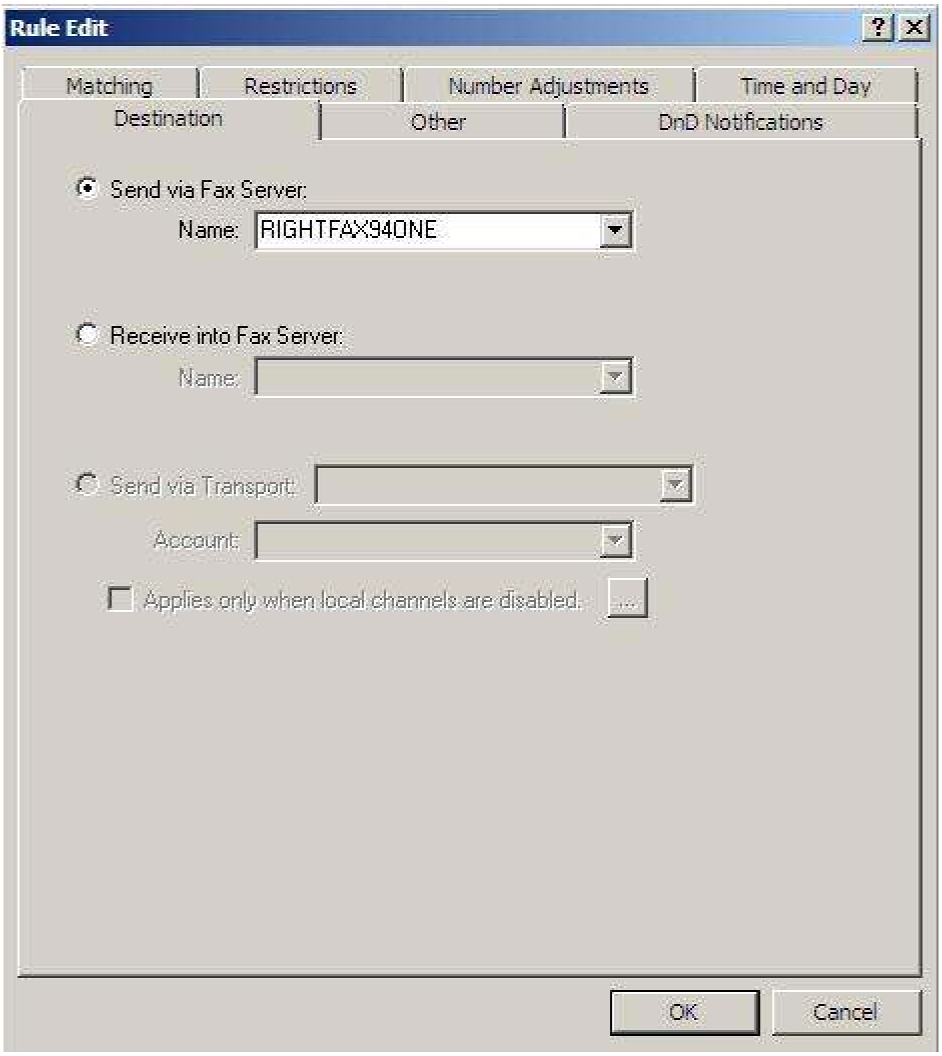
The example below shows the single rule created for the compliance test at site 1. The * in the Pattern field indicates that this rule applies to all dialed numbers. To view the details, double click on the rule in the right pane.



| Number | Comment | Pattern | Routing | Priority | Source(s) | Time of Day | Min. Que... | Fax... | Special Actio |
|---|---|---|---|---|---|---|---|---|---|
| 1 | | * | RIGHTFAX3... | High/Normal/Low | All | All Times | 0 | 0 | None |

</td></tr>
</table>

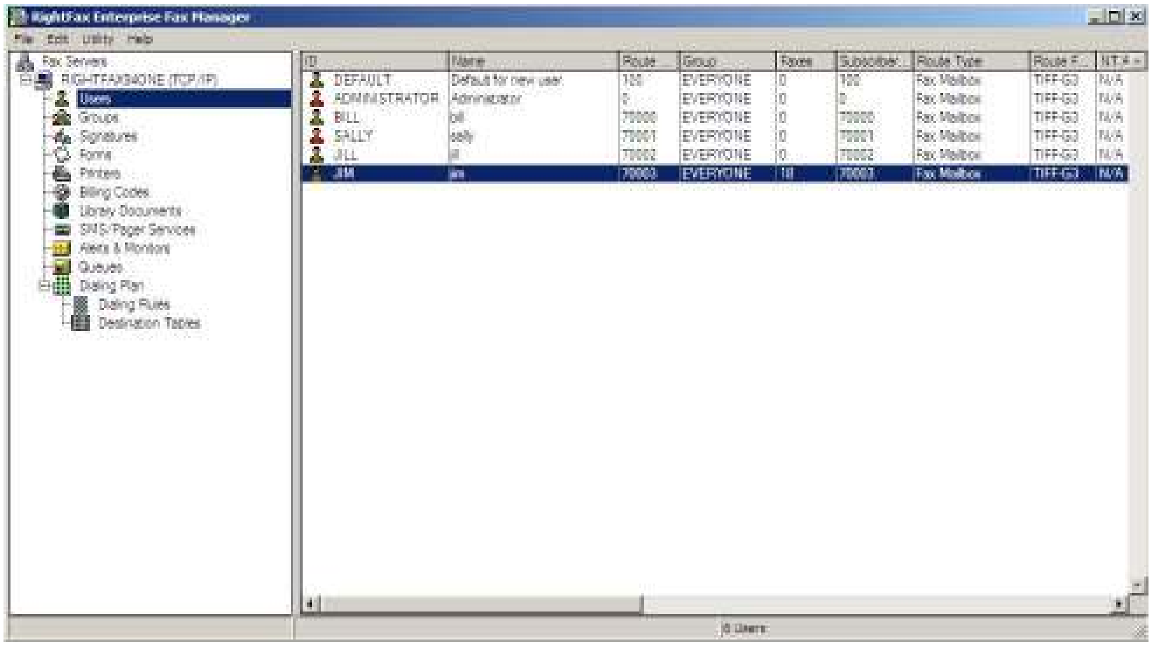
| Step | Description |
|---|---|
| 10. | **Configure Dialing Rules - Continued**<br>The **Rule Edit** window will appear as shown below. The **Number Adjustments** tab shows the digit string manipulation that is done to each dialed number. In the example below, each number is appended with ***@192.45.108.50*** as indicated in the **Append this** field. This IP address is for the SIP Enablement Services server at site 1.<br><br> <br>Strip beginning digits: 0<br>Prepend this:<br>Strip ending digits: 0<br>Append this: @192.45.108.50 |

| Step | Description |
|---|---|
| 11. | **Configure Dialing Rules - Continued**<br>In the **Destination** tab of the **Rule Edit** window, select the correct host name of the fax server for **Send via Fax Server**.<br><br>● Send via Fax Server:<br>Name: RIGHTFAX94ONE<br>○ Receive into Fax Server:<br>Name:<br>○ Send via Transport:<br>Account:<br>☐ Applies only when local channels are disabled. |

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>12.</td><td>

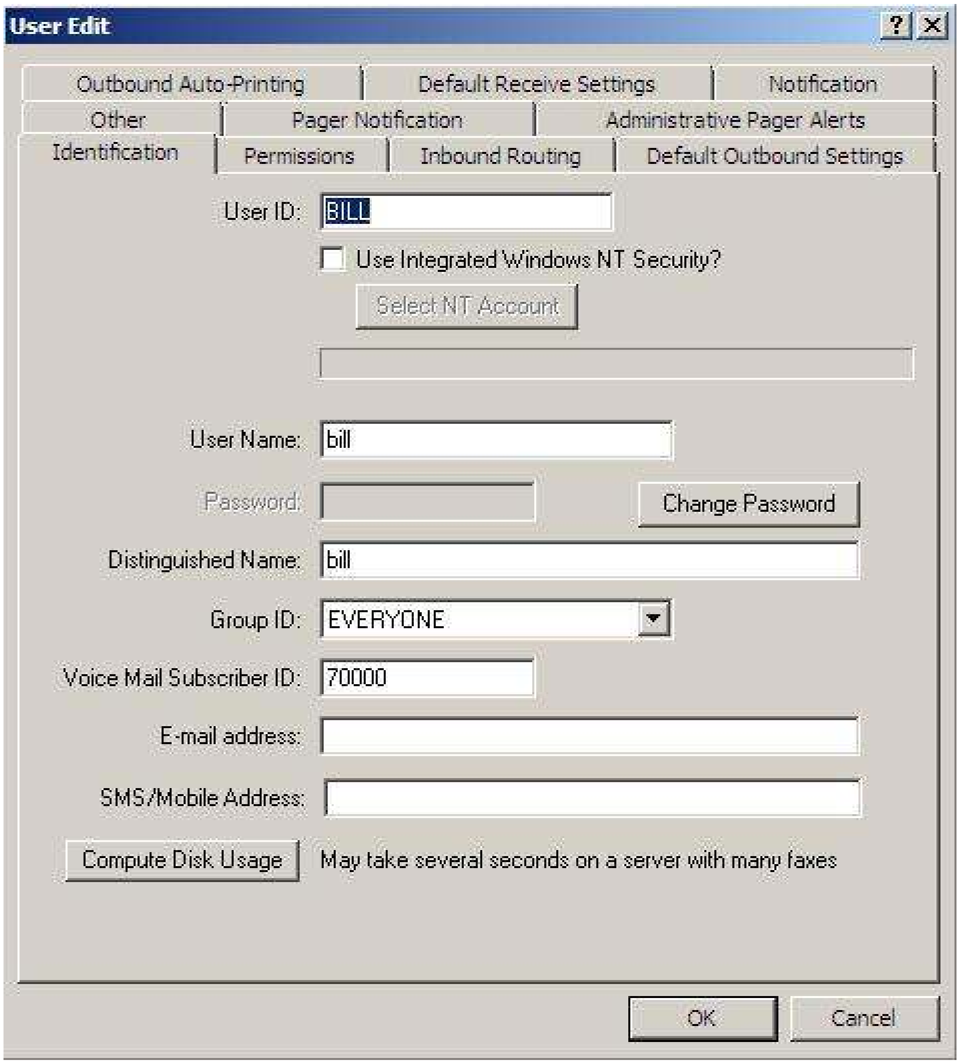
**Configure Users**

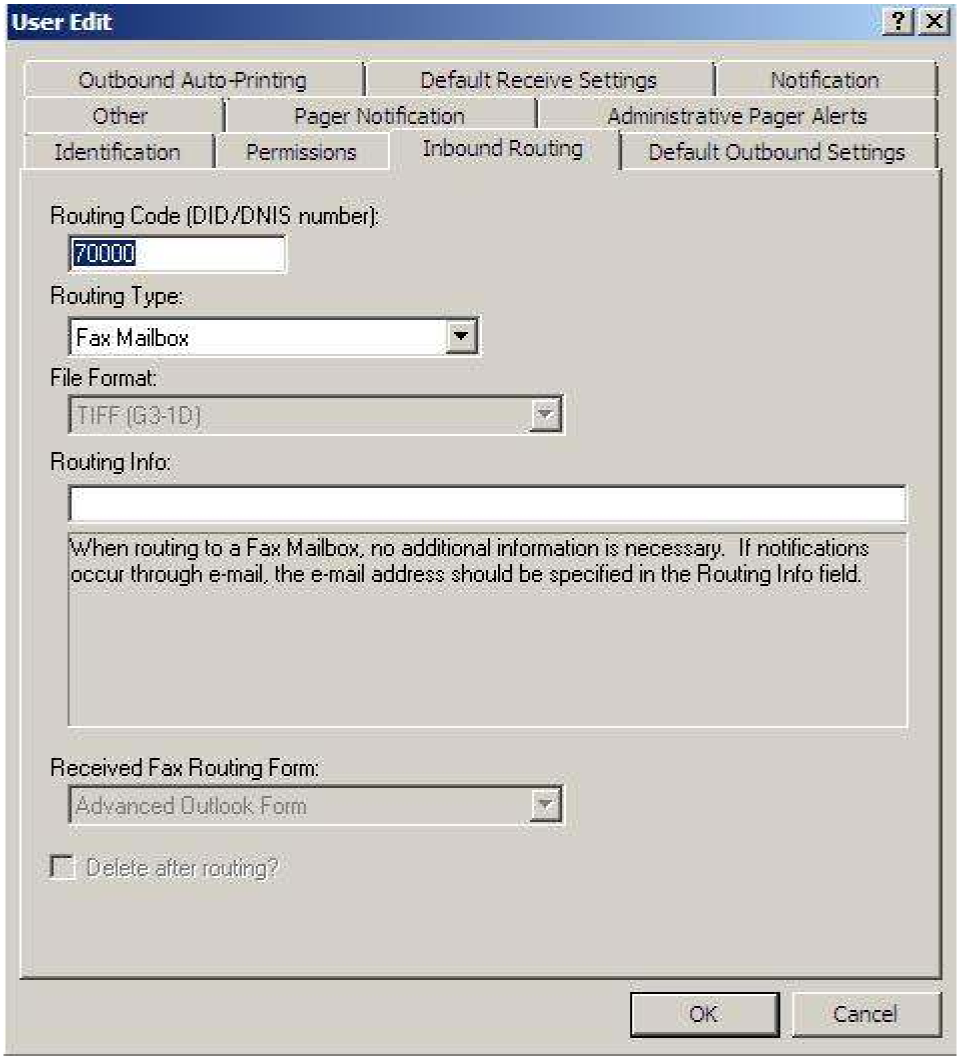
A user is created on RightFax for each incoming fax number. The user represents the fax recipient. To view the list of users, navigate to **Users** in the left navigation menu under the host name of the fax server. The example below shows a list of 6 users, two of which are created by default. The users, named ***BILL***, ***SALLY***, ***JILL*** and ***JIM***, were created at site 1 for the compliance test. To view the details of ***BILL***, double-click on the user entry for ***BILL*** in the right pane.



<table>
<tr><th>ID</th><th>Name</th><th>Route ...</th><th>Group</th><th>Faxes</th><th>Subscriber...</th><th>Route Type</th><th>Route F...</th><th>NT A...</th></tr>
<tr><td>DEFAULT</td><td>Default for new user</td><td>100</td><td>EVERYONE</td><td>0</td><td>100</td><td>Fax Mailbox</td><td>TIFF-G3</td><td>N/A</td></tr>
<tr><td>ADMINISTRATOR</td><td>Administrator</td><td>0</td><td>EVERYONE</td><td>0</td><td>0</td><td>Fax Mailbox</td><td>TIFF-G3</td><td>N/A</td></tr>
<tr><td>BILL</td><td>bill</td><td>70000</td><td>EVERYONE</td><td>0</td><td>70000</td><td>Fax Mailbox</td><td>TIFF-G3</td><td>N/A</td></tr>
<tr><td>SALLY</td><td>sally</td><td>70001</td><td>EVERYONE</td><td>0</td><td>70001</td><td>Fax Mailbox</td><td>TIFF-G3</td><td>N/A</td></tr>
<tr><td>JILL</td><td>jill</td><td>70002</td><td>EVERYONE</td><td>0</td><td>70002</td><td>Fax Mailbox</td><td>TIFF-G3</td><td>N/A</td></tr>
<tr><td>JIM</td><td>jim</td><td>70003</td><td>EVERYONE</td><td>18</td><td>70003</td><td>Fax Mailbox</td><td>TIFF-G3</td><td>N/A</td></tr>
</table>

</td></tr>
</table>

| Step | Description |
|---|---|
| 13. | **Configure Users – Identification**<br>The **User Edit** window will appear as shown below. Select the **Identification** tab. The example below shows the settings used for the compliance test at site 1. The **User ID** field is set to a descriptive name. Appropriate values should be entered or selected for other fields.<br><br>User ID: BILL<br>☐ Use Integrated Windows NT Security?<br>User Name: bill<br>Password:<br>Distinguished Name: bill<br>Group ID: EVERYONE<br>Voice Mail Subscriber ID: 70000<br>E-mail address:<br>SMS/Mobile Address:<br>May take several seconds on a server with many faxes<br> |

| Step | Description |
|---|---|
| 14. | **Configure Users – Inbound Routing**<br>On the **Inbound Routing** tab, the **Routing Code** field is set to the fax number of the recipient. In the case of the compliance test, this was extension ***70000*** for site 1. Default values may be used for all other fields.<br> |

User Edit

Routing Code (DID/DNIS number): 70000

Routing Type: Fax Mailbox

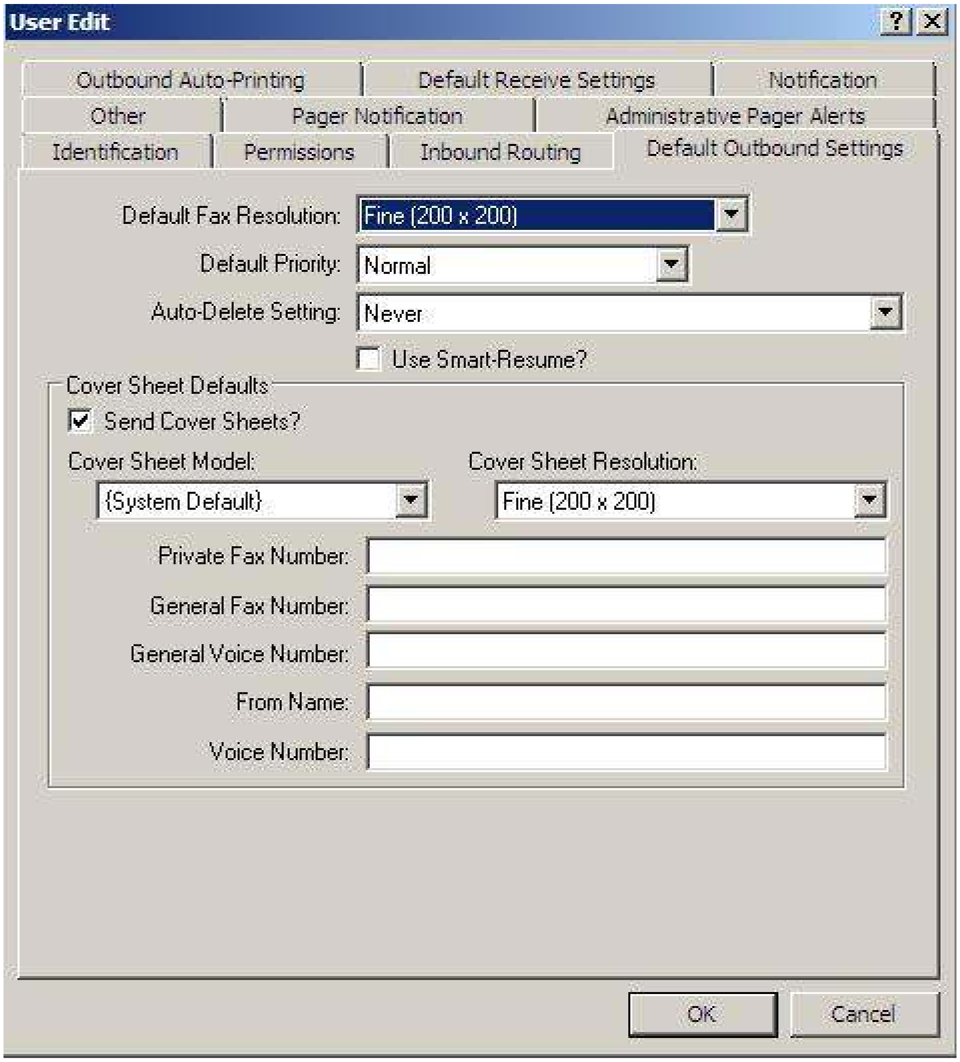
File Format: TIFF (G3-1D)

Routing Info:

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

Received Fax Routing Form: Advanced Outlook Form

☐ Delete after routing?

| Step | Description |
|---|---|
| 15. | **Configure Users – Outbound Settings**<br>The **Default Outbound Settings** tab configures various outbound fax call settings. Configure these settings as appropriate.<br><br>Default Fax Resolution: Fine (200 x 200)<br>Default Priority: Normal<br>Auto-Delete Setting: Never<br>☐ Use Smart-Resume?<br>Cover Sheet Defaults<br>☑ Send Cover Sheets?<br>Cover Sheet Model: {System Default}<br>Cover Sheet Resolution: Fine (200 x 200)<br>Private Fax Number:<br>General Fax Number:<br>General Voice Number:<br>From Name:<br>Voice Number: |

# 7. General Test Approach and Test Results

This section describes the compliance testing used to verify the interoperability of RightFax with the Avaya SIP infrastructure (Communication Manager and SIP Enablement Services). This section covers the general test approach and the test results.

## 7.1. General Test Approach

The general test approach was to make intra-site and inter-site fax calls to and from RightFax. The inter-site calls were made using H.323 trunks or ISDN-PRI trunks between the sites. Faxes were sent with various page lengths, resolutions, and at various fax data speeds. For capacity, a large number of 3-page faxes were continuously sent between the two RightFax servers simultaneously. Serviceability testing included verifying proper operation/recovery from failed cables, unavailable resources, and Communication Manager and RightFax restarts. Fax calls were also tested with different Avaya Media Gateway media resources to process the fax data. This included the TN2302 MedPro circuit pack, the TN2602 MedPro circuit pack in the Avaya G650 Media Gateway; the integrated VoIP engine of the Avaya G700 Media Gateway and the Avaya MM760 Media Module installed in the Avaya G700 Media Gateway.

## 7.2. Test Results

RightFax successfully passed compliance testing. The following observation was made during the compliance test:

Fax calls consume DSP (Digital Signal Processing) resources for processing fax data on the TN2302AP IP Media Processor (MedPro) circuit pack and the TN2602AP IP Media Processor circuit pack in the Avaya G650 Media Gateway, and the integrated Voice over Internet Protocol (VoIP) engine of the Avaya G700 Media Gateway. To increase the capacity to support simultaneous fax calls, additional TN2302AP and/or TN2602AP MedPro circuit packs need to be installed in the Avaya G650 Gateway, and additional Avaya MM760 Media Module or Modules need to be installed in the Avaya G700 Media Gateway. The information contained in the table below indicates DSP capacities/usage in the Avaya media processors. Customers should work with their Avaya sales representatives to ensure that their fax solutions have adequate licenses and DSP resources to match the intended Fax capacity/usage.

| **Platform Device** | **DSP Resources per Platform Device** | **DSP Resources per FoIP Call** |
|---|---|---|
| TN2302, G700, MM760 | 64 | 4 |
| TN2602 | 64 | 1 |

Note that the SIP trunk group on the Communications Manager for connecting RightFax at each site, as well as the H.323 trunk group or ISDN-PRI trunk group for connecting the 2 sites must be configured with adequate number of trunk group members to support the number of simultaneous fax calls intended. On RightFax, adequate number of fax channels must also be appropriately configured for the intended capacity.

# 8. Verification Steps

The following steps may be used to verify the configuration:

- From the Avaya Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling groups configured in **Step 11** of **Section 4** are in-service.
- From the Avaya Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group configured in **Section 4**, **Steps 12 - 13** is in-service.
- Verify that fax calls can be placed to/from RightFax server at each site.
- From the Avaya Communication Manager SAT, use the **list trace tac** command to verify that fax calls are routed to the expected trunks.

# 9. Conclusion

These Application Notes describe the procedures required to configure RightFax to interoperate with Avaya SIP infrastructure (Communication Manager and SIP Enablement Services). RightFax successfully passed compliance testing with the observations documented in **Section 7.2**.

# 10. Additional References

[1] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Doc # 555-245-205, May 2009.

[2] *Administering Avaya Aura™ Communication Manager*, Doc # 03-300509, May 2009.

[3] *SIP support in Avaya Aura™ Communication Manager Running on the Avaya S8xxx Servers,* Doc # 555-245-206, May 2009.

[4] *Administering Avaya Aura™ SIP Enablement Services on the Avaya S8300 Server*, Doc # 03-602508, May 2009.

[5] *RightFax Version 9.4 Administrator's Guide*, v1.0, October 29, 2008.

[6] *RightFax Version 9.4 Installation Guide*, v1.0, November 18, 2008.

Product documentation for Avaya products may be found at http://support.avaya.com.

Documentation for RightFax version 9.4 may be found at http://www.captaris.com/support/documentation/rightfax/index.html.

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.