Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Open Text RightFax with Avaya Aura® Communication Manager and Avaya Aura® Session Manager via SIP Trunk Interface - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring the Open Text RightFax with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using a SIP trunk interface.

Open Text RightFax is a software based fax server that sends and receives fax calls over an IP network. In the tested configuration, Open Text RightFax interoperated with Avaya Aura® Session Manager to send/receive faxes using SIP trunk facilities.

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.

Introduction

These Application Notes describe the procedures for configuring Open Text RightFax with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks.

Open Text RightFax is a software based fax server that sends and receives fax calls over an IP network. Open Text RightFax utilizes the Brooktrout SR140 T.38 Fax over Internet Protocol (FoIP) virtual fax board software from Dialogic. In the tested configuration, Open Text RightFax interoperated with Avaya Aura® Session Manager to send/receive faxes using a SIP trunk interface.

General Test Approach and Test Results

This section describes the compliance test approach used to verify interoperability of Open Text RightFax with Session Manager.

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.

1.1. Interoperability Compliance Testing

The compliance test tested interoperability between RightFax and Session Manager by making intra-site fax calls between a RightFax server and an analog fax machine that was connected to a Communication Manager via Session Manager using SIP trunks. For inter-site fax, calls were made between a RightFax server and an analog fax machine or another RightFax server that was connected on a remote site. The remote site connection used SIP or ISDN trunks.
Specifically, the following fax operations were tested in the setup for the compliance test:

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

In the compliance test, SIP or ISDN/T1 trunks directly connecting two Communication Manager Systems connected the Main Site, and Remote Site.

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 SIP or ISDN/T1 trunks between the sites. Faxes were sent with various page lengths and resolutions. For capacity, a large number of 2-page faxes were continuously sent between the two RightFax servers simultaneously. Serviceability testing included verifying proper operation/recovery from failed cables, unavailable resources, and RightFax restarts. Fax calls were also tested with different Avaya Media Gateway media resources used to process the fax data between sites. 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 G450 Media Gateway and the Avaya MM760 Media Module installed in the Avaya G450 Media Gateway.

1.2. Test Results

OpenText RightFax successfully passed all compliance testing with the following observations,

- RightFax server transmission rate was set to 9600 for all test cases. The actual transmission rate depends on the Avaya Media Gateway or Media processor card being used. TN2302 in G650 supports 14.4 K and G450 and TN2602 in G650 support 9.6 K.
- Not all services of RightFax start automatically after a reboot of the fax server. Some services need to be manually restarted. Also if the fax server reboots in the midst of a fax transmission, the status of the fax does not change after the reboot. Open Text indicates this is abnormal behavior, not attributable to telephony components of the product. Customers experiencing similar problems should contact OpenText Technical Support for further assistance.
- A small percentage of faxes failed when sending simultaneous 100 two page faxes, from each site, between the main and remote site RightFax servers. Open Text indicates that a significant number of factors contribute to fax transmission failures, including network and line conditions. A small percentage of fax transmission failures are considered normal for most production telephony environments.

Note 1: 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 packs in the Avaya G650 Media Gateway and the integrated Voice over Internet Protocol (VoIP) engine of the Avaya G450 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 G450 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, G450, MM760	64	4
TN2602	64	1

Note 2: The SIP trunk group on Communication Manager for connecting to Session Manager at each site, as well as the SIP trunk group for connecting the two sites must be configured with adequate number of trunk group members to support the number of simultaneous fax calls intended. On RightFax, an adequate number of fax channels must also be appropriately configured for the intended capacity.

Note 3: The ISDN/T1 link between the two sites should be clean with no clock synchronizing errors. Any errors in the link will cause the fax transmission to fail. Use the command **list measurements ds1 log <card slot>** to provide DS-1 link performance measurements detailed log report.

1.3. Support

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

- Phone: (800) 540-7292
- Email: support@opentext.com

For other locations go to http://www.opentext.com/2/global/company/company-contact.html

Reference Configuration

The test configuration was designed to emulate two separate sites with a single Port Network at one site, and modular Gateway resources at the other site. **Figure 1** illustrates the configuration used in these Application Notes.

1.4. Configuration Details

In the sample configuration, Communication Manager Servers and Gateways at the two sites were connected via SIP or ISDN/T1 trunks. Faxes were alternately sent between the two sites using the SIP or ISDN/T1 facilities. Connections to Session Manager were via SIP trunk facilities, and the RightFax servers communicated directly with Session Manager via SIP.

Two separate Session Manager Servers were used to connect the RightFax Servers to each site.

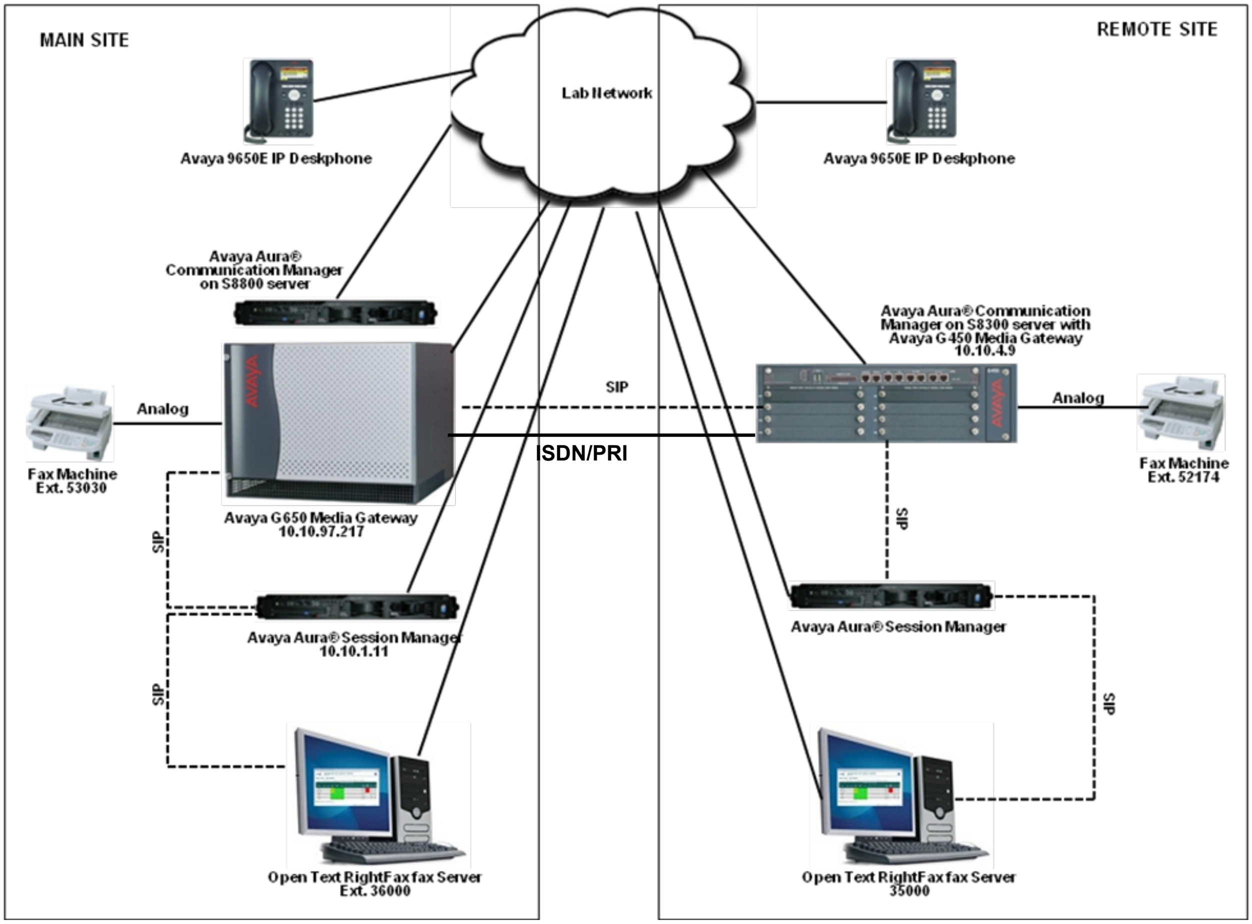


Figure 1: RightFax interoperating with Avaya Aura® Session Manager via SIP Trunk

The Main Site had an Avaya S8800 Server running Communication Manager with an Avaya G650 Media Gateway. The RightFax server at this site communicated with Session Manager via SIP. In turn, Communication Manager used a SIP Trunk which terminated on a CLAN circuit pack in port network 1 to communicate with Session Manager. IP media resources were provided by Media Processor (MedPro) circuit packs. Two versions of the MedPro circuit pack were tested in this configuration: TN2302AP and TN2602AP. Endpoints at this site included an Avaya 9600 Series IP Telephone (with H.323 firmware), and an analog fax machine.

The Remote Site had an Avaya S8300 Server running Communication Manager in an Avaya G450 Media Gateway. The RightFax server at this site communicated with Session Manager via SIP. On the Avaya G450 Media Gateway, the signaling and media resources supporting a SIP trunk connected to Session Manager were integrated directly on the media gateway processor. Endpoints at this site included Avaya 9600 Series IP Telephones (with H.323 firmware), and an analog fax machine.

The IP telephones were not involved in the faxing operations, they were present in the configuration to verify the effect VoIP telephone calls had on the FoIP faxing operations.

Outbound fax calls originating from RightFax were sent to Session Manager first, then to Communication Manager, via the configured SIP trunks. Based on the dialed digits, Communication Manager directed the calls to the local fax machine, or the inter-site trunks (SIP or ISDN/T1) to reach the Remote Site. Inbound fax calls to RightFax were first received by Communication Manager from the local fax machine or from across the SIP or ISDN/T1 trunks connected to the Remote Site. Communication Manager then directed the calls to RightFax via the configured Session Manager SIP trunks.

Equipment and Software Validated

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

Equipment	Release/Version
Avaya S8800 Servers (at both sites)	Avaya Aura® Session Manager 6.2 Avaya Aura® System Manager 6.2
Avaya S8800 Server (at Main Site)	Avaya Aura® Communication Manager 6.2 SP4
Avaya S8300D Server (at Remote Site)	Avaya Aura® Communication Manager 6.2 SP4
Avaya 9650E IP Deskphone (H.323)	3.104S
Analog Fax Machines	N/A
OpenText RightFax on Windows 2008R2 Enterprise SP1	10.5.0.895 (RightFax 10.5 Release version) with Dialogic Brooktrout SR140 SDK 6.5.2

Configure Avaya Aura® Communication Manager

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

- The examples shown in this section refer to the Main Site. Unless specified otherwise, these same steps also apply to the Remote Site using values appropriate for that location.
- These same steps also apply to the SIP trunk configuration between the Main and Remote site using appropriate values.
- The configuration of Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration, the **save translation** command was used to make the changes permanent.

1.5. Steps to Configure Communication Manager

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 – 4)
- Administer IP Node Name (Step 5)
- Administer IP Codec Set (Steps 6 – 7)
- Administer SIP Signaling Group (Step 8)
- Administer SIP Trunk Group (Steps 9 – 10)
- Administer Private Numbering (Step 11)
- Administer Route Pattern (Step 12)
- Administer Uniform Dial plan (Step 13)
- Administer AAR Analysis (Step 14)
- Administer DS1 for ISDN/T1 (Step 15)
- Administer ISDN Signaling Group (Step 16)
- Administer ISDN Trunk Group (Step 17)

Verify Communication Manager License

1. 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.

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: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 5
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 2
                   Maximum Video Capable IP Softphones: 18000 4
                       Maximum Administered SIP Trunks: 24000 130
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   1
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

2.2

Identify IP Interfaces

Use the **list ip-interface clan** and **list ip-interface medpro** commands to identify IP interfaces in the network region. 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.

Testing with the TN2302 and TN2602 circuit packs were done separately. When testing with the TN2302, the TN2602 was disabled (turned off) and vice versa as indicated in the **ON** field.

```
list ip-interface clan

                                IP INTERFACES

                                                          Skts  Net       Eth
ON  Slot   Code/Sfx Node Name/       Mask  Gateway Node     Warn  Rgn VLAN  Link
                    IP-Address
--  ----   -------- ---------------  ----  ---------------  ----  --- ---- ----
 y  01A02  TN799  D CLAN1            /26   GW               400   1   n    1
                    10.10.97.217
 y  01A03  TN799  D CLAN2            /26   GW               400   2   n    2
                    10.10.97.238
```

```
list ip-interface medpro

                                IP INTERFACES

                                                     Net
ON Slot  Code/Sfx Node Name/      Mask Gateway Node    Rgn VLAN Virtual Node
                  IP-Address
-- ----- -------- --------------- ---- --------------- --- ---- ---------------
 y 01A07 TN2302   MedPro1         /26  GW              1   n
                  10.10.97.218
 n 01A08 TN2602   MedPro2         /26  GW              1   n
                  10.10.97.233
```

3.

Administer IP Network Region 1

The configuration of the IP network regions (**Steps 3 – 4**) was already in place and is included here for clarity. At the Main Site, the Avaya G650 Media Gateway comprising port network 1 and all IP endpoints were located in IP network region 1.

Use the **display ip-network-region** command to view these settings.

A descriptive name can be entered for the **Name** field. None was used during compliance testing.

- **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.
- 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.
- The default values were used for all other fields.

At the Remote Site, 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  20
                               IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: bvwdev.com
    Name:
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
 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
```

4.

Administer IP Network Region 1 – Continued

On **Page 4**, codec sets are defined for inter-region calls. In the case of the compliance test at the Main and Remote Site, only one IP network region was used, so no inter-region settings were required and therefore only codec set 1 is used. The default values were used for all other fields.

```
display ip-network-region 1                                     Page   4 of  20

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

5.

Administer IP Node Name

Use the **change node-names ip** command to create a node name that maps to the Session Manager IP address. This node name is used in the configuration of the SIP trunk signaling group in **Step 8**.

```
display node-names ip                                           Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AES62               10.10.98.17
CLAN1               10.10.97.217
CLAN2               110.10.97.238
DevCM3              10.10.4.9
GW                  10.10.97.193
InteropSM62         10.10.1.11
MedPro1             10.10.97.218
MedPro2             10.10.97.233
SM61                10.10.97.198
default             0.0.0.0
procr               10.10.97.201
procr6              ::
```

6.

Administer IP Codec set

Use the **change ip-codec-set 1** 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: G.729            n          2        20
 3: G.722-64K                   2        20
 4:
 5:
 6:
 7:
```

7. **Administer IP Codec set – Fax settings**

On **Page 2**, set the **FAX Mode** field to ***t.38-standard.*** This is necessary to support the RightFax server. The **Modem Mode** field should be set to ***off***.

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. 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 Brooktrout SR140 configuration; otherwise Brooktrout SR140 will negotiate T.38 redundancy to the most common denominator (no redundancy in this case).

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

                          IP Codec Set

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

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

8. **Administer SIP Signaling Group**

For the compliance test, a signaling group and the associated SIP trunk group was used for routing fax calls to/from the RightFax server via Session Manager. For the compliance test at the Main Site, signaling group 4 was configured using the parameters highlighted below. For further details on other fields refer to **Section 10**.

- The **Group Type** was set to ***sip***.
- The **Transport Method** was set to ***tcp*** (Transport Layer Security). As a result, the **Near-end Listen Port** and **Far-end Listen Port** are automatically set to ***5060.***
- The **Near-end Node Name** was set to ***CLAN1***, the node name that maps to the IP address of the CLAN circuit pack used to connect to Session Manager. Node names are defined using the **change node-names ip** command (see **Step 5** above).
- The **Far-end Node Name** was set to ***InteropSM62***. This node name maps to the IP address of the Session Manager server as defined using the **change node-names ip** command.
- The **Far-end Network Region** was set to ***1***. This is the IP network region which contains Session Manager and RightFax.
- The **Far-end Domain** was set to ***bvwdev.com***. This domain is sent in the headers of SIP INVITE messages for calls originating from and terminating to Session Manager using this signaling group.
- **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**).
- The **DTMF over IP** field was set to the default value of ***in-band***.
- The default values were used for all other fields.

```
display signaling-group 4
                                SIGNALING GROUP

 Group Number: 4             Group Type: sip
  IMS Enabled? n       Transport Method: tcp
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? n  Peer Server: SM



   Near-end Node Name: CLAN1                 Far-end Node Name: InteropSM62
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1

Far-end Domain: bvwdev.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: in-band                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
```

9. Administer SIP Trunk Group

For the compliance test, trunk group 4 was used for the SIP trunk group for routing fax calls to/from Session Manager. Trunk group 4 was configured using the parameters highlighted below. For further details on other fields refer to **Section 10**.

On **Page 1**:

- The **Group Type** field was set to ***sip***.
- A descriptive name was entered for the **Group Name**.
- An available trunk access code (TAC) that was consistent with the existing dial plan was entered in the **TAC** field.
- The **Service Type** field was set to ***tie***.
- The **Signaling Group** was set to the signaling group shown in the previous step.
- 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.
- The default values were used for all other fields.

```
display trunk-group 4                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 4                  Group Type: sip           CDR Reports: y
  Group Name: G650 to InteropSM         COR: 1        TN: 1        TAC: #004
   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: 4
                                                     Number of Members: 50
```

10. **Administer SIP Trunk Group – continued**

On **Page 3**:

- Set the **Numbering Format** field to ***private***. This field specifies the format of the calling party number sent to the far-end.
- Default values may be used for all other fields.

```
display trunk-group 4                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n             Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? Y
```

11. **Administer Private Numbering**

Private numbering defines the calling party number to be sent to the far-end. Use the **change private-numbering** command to create an entry that will be used by the trunk groups defined in **Steps 9-10**. In the example shown below, all calls originating from a 5-digit extension beginning with 5 is routed across trunk group 4 is sent as a 5-digit calling number.

```
display private-numbering 0                                     Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 5  5              1                           5     Total Administered: 2
 5  5              4                           5        Maximum Entries: 540
```

12 .

Administer Route Pattern

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.

The example below shows the route pattern used for the compliance test at the Main Site. A descriptive name was entered for the **Pattern Name** field. The **Grp No** field was set to the trunk group created in **Steps 9–10**. 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.

```
display route-pattern 4                                         Page   1 of   3
                    Pattern Number: 4   Pattern Name: SIP-To-SM62
                             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
```

13 .

Administer Uniform Dial Plan

Use the **change uniform-dialplan** command to create a matching pattern that matches with the extensions used in the RightFax server. During compliance testing extensions 36xxx were used on the Main site RightFax and therefore a matching pattern of 36 with length of 5 was configured as shown below.

```
display uniform-dialplan 0                                      Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching                   Insert              Node
  Pattern       Len Del      Digits     Net Conv Num
 36              5   0                  aar  n
 5               5   0                  aar  n
```

14 .

Administer AAR Analysis

Automatic Alternate Routing (AAR) was used to route calls to RightFax via Session Manager. 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 the Main Site using the **display aar analysis 0** command. The highlighted entry specifies that 5 digit dial string 36 was to use route pattern 4 to route calls to the RightFax fax server at the Main Site via Session Manager.

```
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
    36                    5    5     4        aar          n
    5                     5    5     1        aar          n
    53                    5    5     4        aar          n
```

15. Administer DS1 for ISDN/T1

Use the **add ds1 01a13** command. Note that the actual slot number may vary. During compliance testing 01a13 was used as the slot number. The highlighted values shown below were used during compliance testing and retain the default values for the remaining fields. Submit these changes.

The **Interface** field must be complementary on both switches. For the sample configuration, Main Site was administered as the ***peer-master***, and therefore the remote site was administered as the ***user/slave***.

```
display ds1 01a13                                                Page   1 of   2
                                DS1 CIRCUIT PACK

            Location: 01A13                            Name: T1toG450
            Bit Rate: 1.544                     Line Coding: b8zs
   Line Compensation: 1                        Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                         Interface: peer-master
   TN-C7 Long Timers? n                       Peer Protocol: Q-SIG
Interworking Message: PROGress                         Side: a
Interface Companding: mulaw                             CRC? n
           Idle Code: 11111111
                               DCP/Analog Bearer Capability: 3.1kHz

                                            T303 Timer(sec): 4
                                           Disable Restarts? n

      Slip Detection? y                 Near-end CSU Type: other

   Echo Cancellation? N
```

16. Administer ISDN Signaling Group

For the compliance test, a signaling group and the associated ISDN trunk group was used for routing fax calls between the two sites. For the compliance test at the Main Site, signaling group 6 was configured using the parameters highlighted below. For further details on other fields refer to **Section 10**.

- The **Group Type** was set to ***isdn-pri***.
- The **Primary D-Channel,** enter the slot number for the DS1 circuit pack which is ***01a13*** and the port is ***24***.
- The **Trunk Group for Channel Selection** was set to ***6***, since this was the ISDN trunk group number configured during compliance testing (see **Step 17** below).
- For the **TSC Supplementary Service Protocol** field, enter ***b*** for QSIG.
- The default values were used for all other fields.

```
display signaling-group 6
                                SIGNALING GROUP

 Group Number: 6              Group Type: isdn-pri
                    Associated Signaling? y          Max number of NCA TSC: 0
                       Primary D-Channel: 01A1324     Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 6       X-Mobility/Wireless Type: NONE
      TSC Supplementary Service Protocol: b          Network Call Transfer? N
```

17.

Administer ISDN Trunk Group

For the compliance test, trunk group 4 was used for the SIP trunk group for routing fax calls to/from Session Manager. Trunk group 4 was configured using the parameters highlighted below. For further details on other fields refer to **Section 10**.

On **Page 1**:

- The **Group Type** field was set to isdn.
- A descriptive name was entered for the **Group Name**.
- An available trunk access code (TAC) that was consistent with the existing dial plan was entered in the **TAC** field.
- The **Direction** was set to ***two-way***.
- The **Service Type** field was set to ***tie***.
- The **Carrier Medium** was set to ***PRI/BRI***.
- The default values were used for all other fields.

```
display trunk-group 6                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 6                   Group Type: isdn          CDR Reports: y
  Group Name: T1-to-G450                 COR: 1        TN: 1        TAC: #006
   Direction: two-way        Outgoing Display? n         Carrier Medium: PRI/BRI
 Dial Access? n              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

Administer ISDN Trunk Group – continued

On **Page 2**:

- Set the **Supplementary Service Protocol** to ***b*** for QSIG.
- For the **Format** field, enter ***unk-unk***.
- Default values may be used for all other fields.

```
display trunk-group 6                                           Page   2 of  21
Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: b     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical
                                                 Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:              Format: unk-unk
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
          Administer Timers? n        CONNECT Reliable When Call Leaves ISDN? n
             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

Repeat **Steps 12 – 14** to configure the values required for the ISDN routing and dialing plan between the two sites.

Configure Avaya Aura® Session Manager

This section provides the procedures for configuring routing using Avaya Aura ® System Manager. The procedures include the following areas:

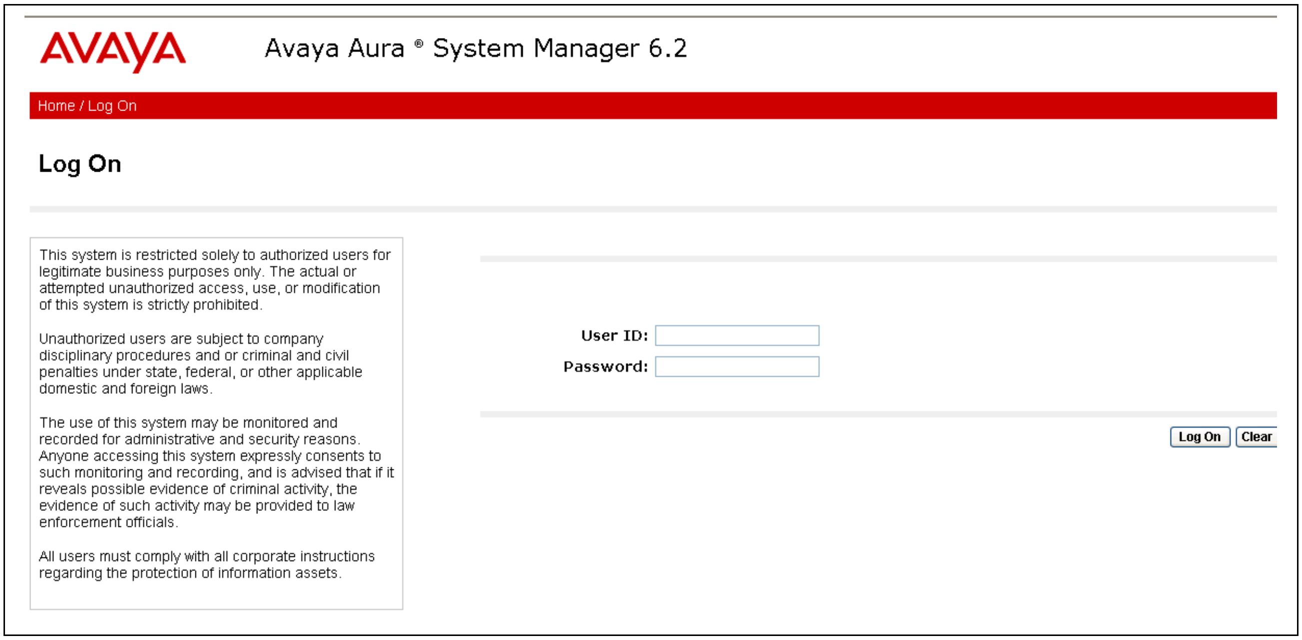
- Logging into the System Manager.
- Adding Domain.
- Adding Location.
- Adding SIP entities.
- Adding Entity Links.
- Adding Routing Policies.
- Adding Dial Patterns.

Examples shown in this section refer to the Main Site. Unless specified otherwise, these same steps also apply to the Remote Site using values appropriate for that location. For detail configuration details of the Session Manager refer to **Section 10**

1.6. Logging into the Avaya Aura® System Manager

This section explains the steps to launch the login screen of the System Manager and accessing the Network Routing Policy.

To launch the System Manager Login screen, start an IE browser and type the IP address of the System Manager in the URL (not shown). Screen below shows the Log On Screen. Type the required **User ID** and **Password** credentials and click on **Log On** to continue.



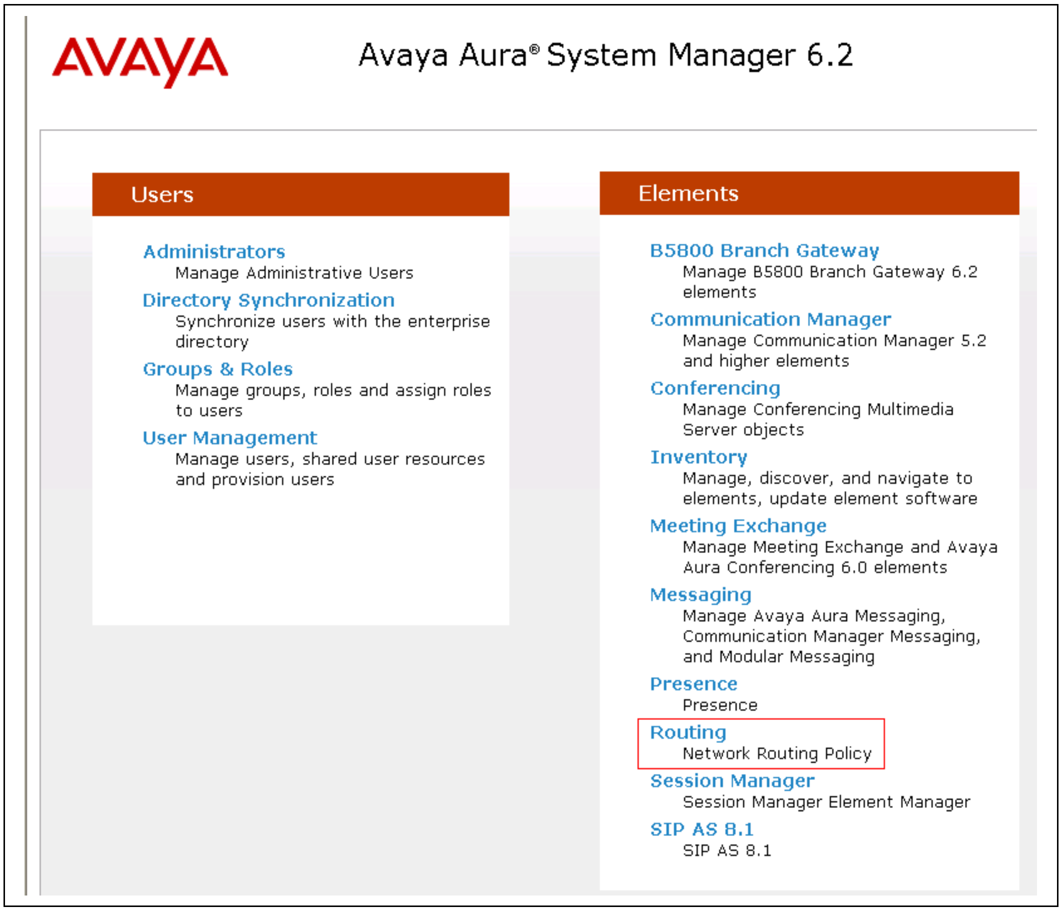
This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

From the main screen of System Manager access the Network Routing Policy by selecting **Routing** as shown below.



Avaya Aura® System Manager 6.2

Users

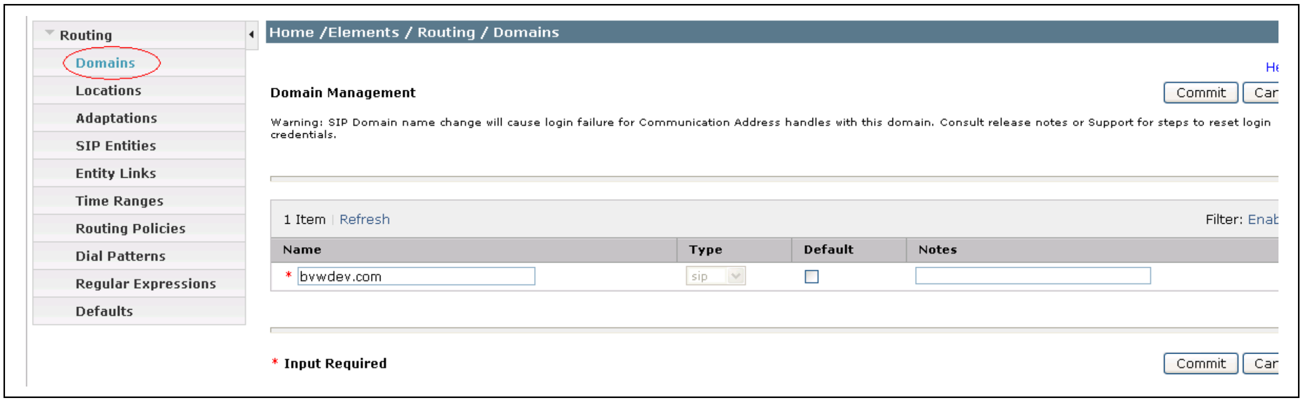
Administrators
Manage Administrative Users
Directory Synchronization
Synchronize users with the enterprise directory
Groups & Roles
Manage groups, roles and assign roles to users
User Management
Manage users, shared user resources and provision users

Elements

B5800 Branch Gateway
Manage B5800 Branch Gateway 6.2 elements
Communication Manager
Manage Communication Manager 5.2 and higher elements
Conferencing
Manage Conferencing Multimedia Server objects
Inventory
Manage, discover, and navigate to elements, update element software
Meeting Exchange
Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements
Messaging
Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging
Presence
Presence
Routing
Network Routing Policy
Session Manager
Session Manager Element Manager
SIP AS 8.1
SIP AS 8.1

1.7. Adding Domain

To add a domain, select **Domains** from the left hand window of the Routing screen and click on **New** (not shown). Configure the **Name** as shown in screen below and click on **Commit** to complete adding a domain. During compliance testing a domain name of **bvwdev.com** was used. Additional domains can be added in a similar fashion.



Domain Management

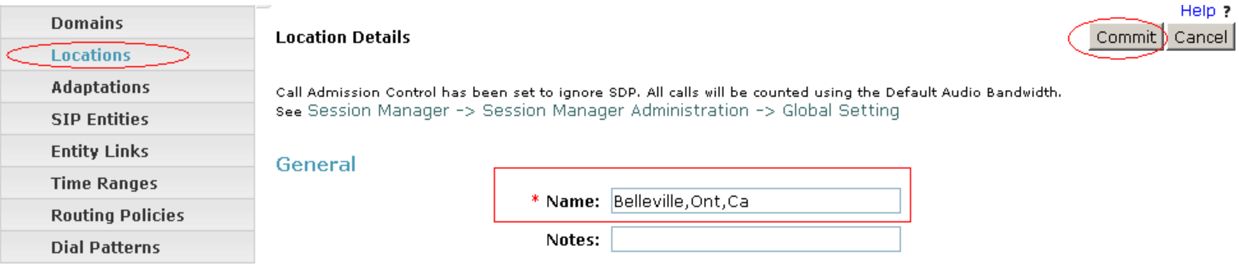
Warning: SIP Domain name change will cause login failure for Communication Address handles with this domain. Consult release notes or Support for steps to reset login credentials.

Name	Type	Default	Notes
* bvwdev.com	sip	☐	

* Input Required

1.8. Adding Location

To add a location, select **Locations** from the left hand window of the Routing screen and click on **New** (not shown). Configure the **Name** as shown in screen below and click on **Commit** to add a Domain. During compliance testing a location name of **Belleville,Ont,Ca** was used. Click on **Commit** to complete adding a location. Additional locations can be added in a similar fashion.



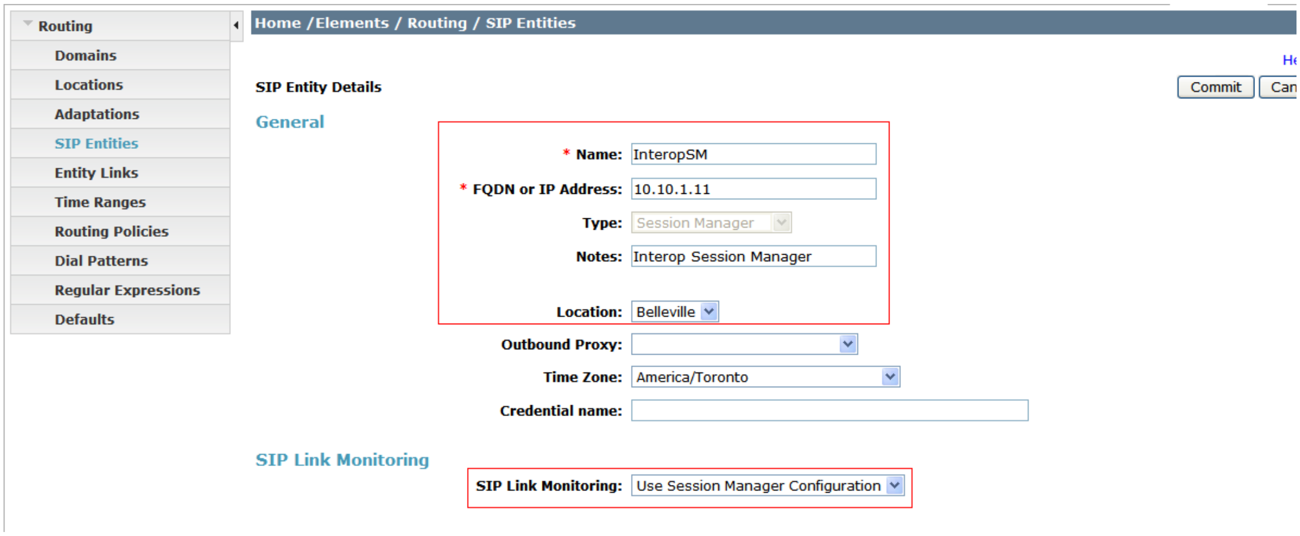
Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth.
See Session Manager -> Session Manager Administration -> Global Setting

* Name: Belleville,Ont,Ca
Notes:

1.9. Adding SIP Entities

This section explains the adding of SIP entities for the Session Manager, RightFax server and the Communication Manager routing. To add SIP Entities, select **SIP Entities** from the left hand window of the Routing screen and click on **New** (not shown).

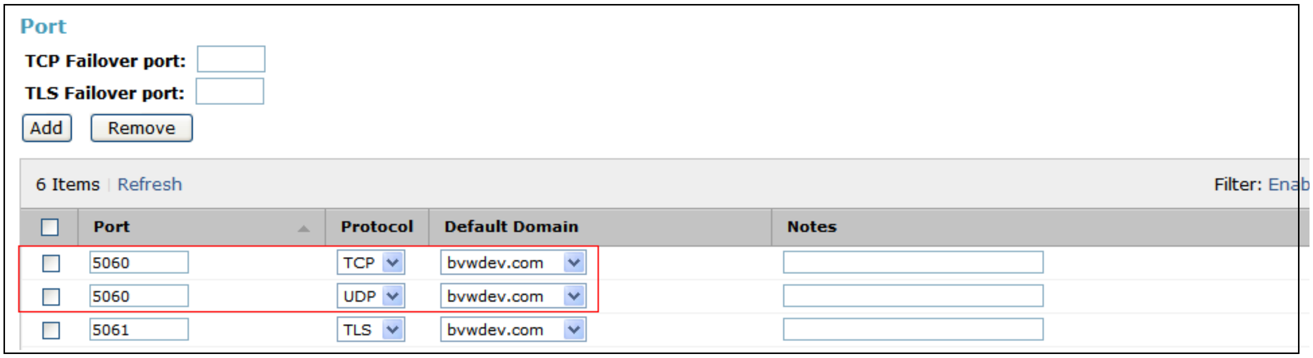
Next two screens show the SIP Entity Details for the Session Manager routing.
Enter a descriptive name for the **Name** field.
Populate the **FQDN or IP Address** field with **10.10.1.11** , which is the IP address of the Session Manager.
Select **Type** as **Session Manager**.
Enter some descriptive notes in the **Notes** field if required.
Select the location configured in **Section 6.3** in the **Location** field.
Select **Use Session Manager Configuration** option under the **SIP Link Monitoring** field.
Click on



* Name: InteropSM
* FQDN or IP Address: 10.10.1.11
Type: Session Manager
Notes: Interop Session Manager
Location: Belleville
Outbound Proxy:
Time Zone: America/Toronto
Credential name:
SIP Link Monitoring: Use Session Manager Configuration

Under the Port section, add both TCP and UDP protocol along with the Port value and the Default Domain value.

Click on **Commit** to complete adding the SIP Entity.

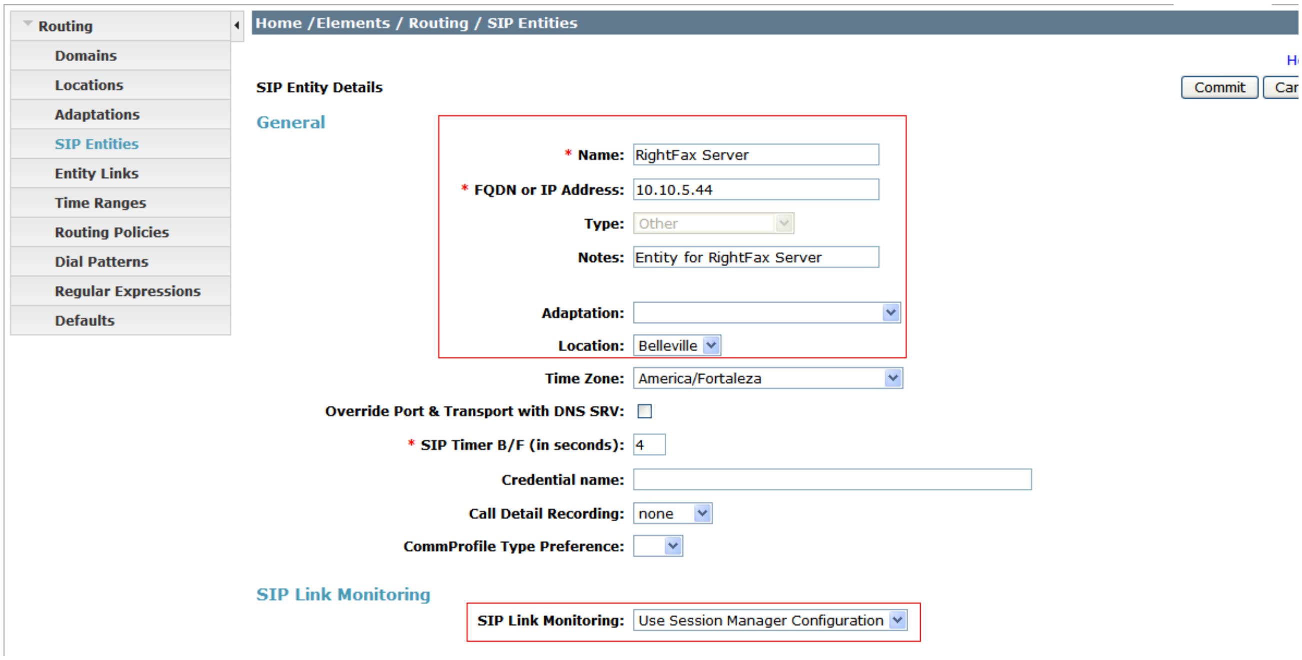


TCP Failover port:

TLS Failover port:

☐	Port	Protocol	Default Domain	Notes
☐	5060	TCP	bvwdev.com	
☐	5060	UDP	bvwdev.com	
☐	5061	TLS	bvwdev.com	

Next two screens show the SIP Entity Details for the RightFax server routing.
Enter a descriptive name for the **Name** field.
Populate the **FQDN or IP Address** field with **10.10.5.44**, which is the IP address of the RightFax server.
Enter some descriptive notes in the **Notes** field if required.
Select the location configured in **Section 6.3** in the **Location** field.
Select **Use Session Manager Configuration** option under the **SIP Link Monitoring** field.



* Name: RightFax Server

* FQDN or IP Address: 10.10.5.44

Type: Other

Notes: Entity for RightFax Server

Adaptation:

Location: Belleville

Time Zone: America/Fortaleza

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds): 4

Credential name:

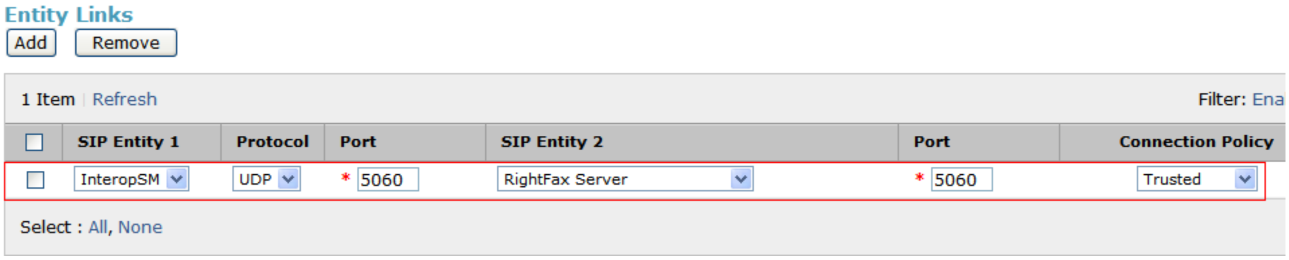
Call Detail Recording: none

CommProfile Type Preference:

SIP Link Monitoring: Use Session Manager Configuration

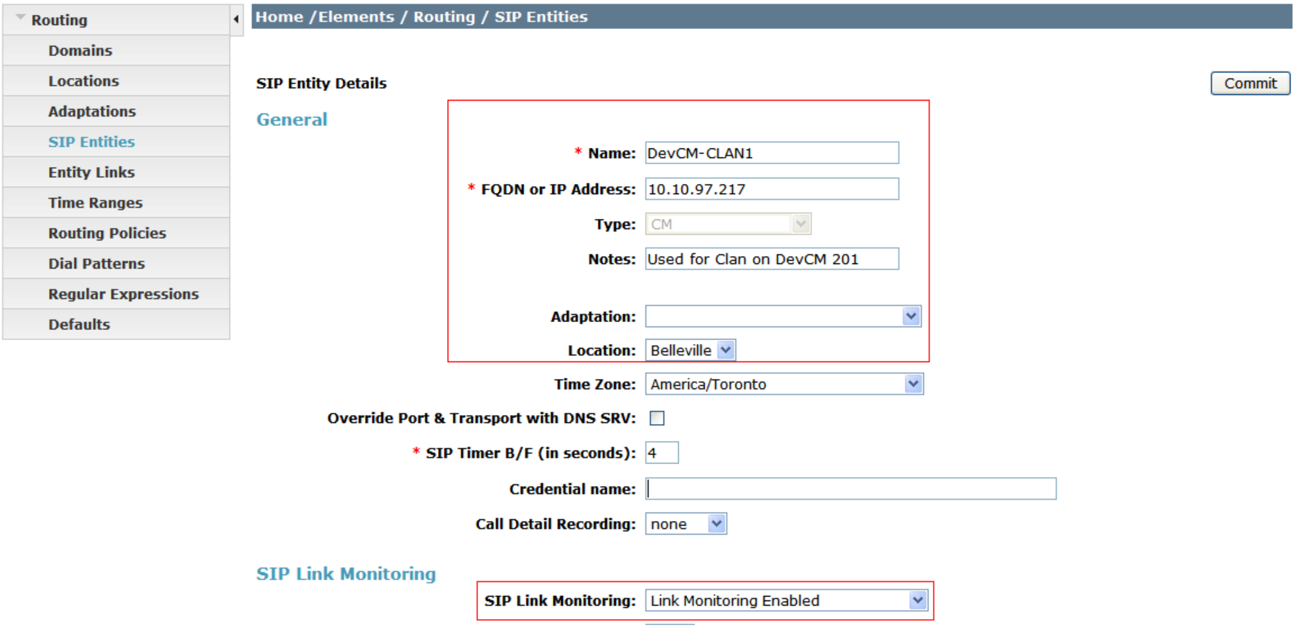
Under the **Entity Links** section, add **InteropSM** as **SIP Entity 1** and **RightFax** as **SIP Entity 2** with **UDP Protocol** and **5060** as **Port**.

Click on **Commit** (not shown) to complete adding the SIP Entity.



☐	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
☐	InteropSM	UDP	* 5060	RightFax Server	* 5060	Trusted

Next two screens show the SIP Entity Details for the Communication Manager routing.
Enter a descriptive name for the **Name** field.
Populate the FQDN **or IP Address** field with **10.10.97.217**, which is the CLAN IP address of the G650 Media Gateway of the Communication Manager.
Enter some descriptive notes in the **Notes** field if required.
Select the location configured in **Section 6.3** in the **Location** field.
Select **Link Monitoring Enabled** option under the **SIP Link Monitoring** field. This was the value used during compliance testing however **Use Session Manager Configuration** option can also be used here.



SIP Entity Details

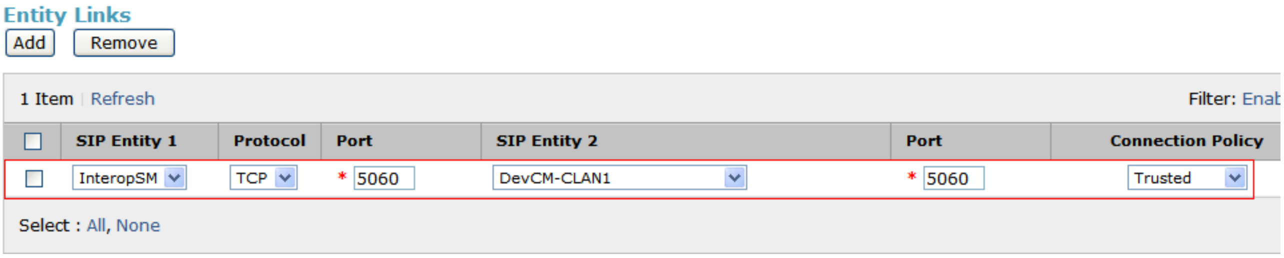
General

* Name: DevCM-CLAN1
* FQDN or IP Address: 10.10.97.217
Type: CM
Notes: Used for Clan on DevCM 201
Adaptation:
Location: Belleville
Time Zone: America/Toronto
Override Port & Transport with DNS SRV: ☐
* SIP Timer B/F (in seconds): 4
Credential name:
Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Link Monitoring Enabled

Under the **Entity Links** section, add **InteropSM** as **SIP Entity 1** and **DevCM-CLAN1** as **SIP Entity 2** with **TCP Protocol** and **5060** as **Port**.
Click on **Commit** to complete adding the SIP Entity.

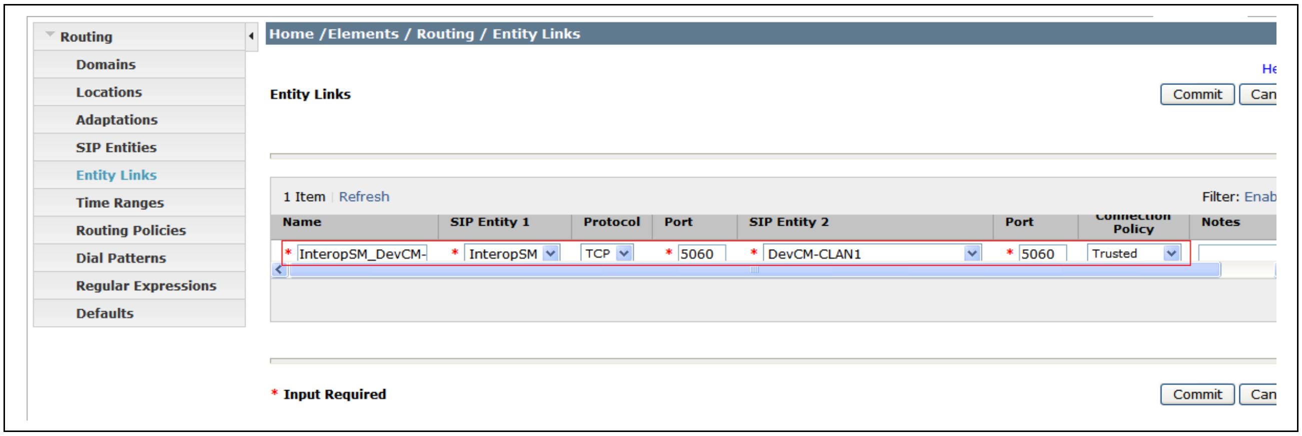


☐	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
☐	InteropSM	TCP	* 5060	DevCM-CLAN1	* 5060	Trusted

1.10. Adding Entity Links

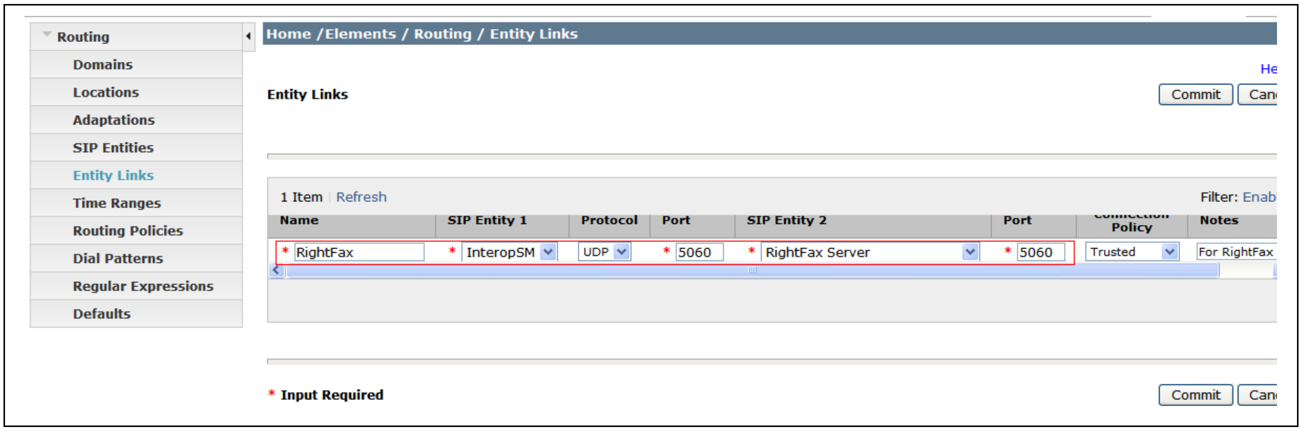
This section explains the adding of Entity Links for RightFax server and the Communication Manager routing. To add Entity Links, select **Entity Links** from the left hand window of the Routing screen and click on **New** (not shown).

Next two screens show the Entity Links for Communication Manager and RightFax server. Enter a descriptive name under the **Name** field. Select **InteropSM** under **SIP Entity 1**. Select **DevCM-CLAN1** for Communication Manager and **RightFax Server** for RightFax server under **SIP Entity 2**. Select the required **Protocol** and enter the **Port** value of **5060**. Click on **Commit** to complete adding an Entity Link.



Entity Links

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* InteropSM_DevCM-	* InteropSM	TCP	* 5060	* DevCM-CLAN1	* 5060	Trusted	



Entity Links

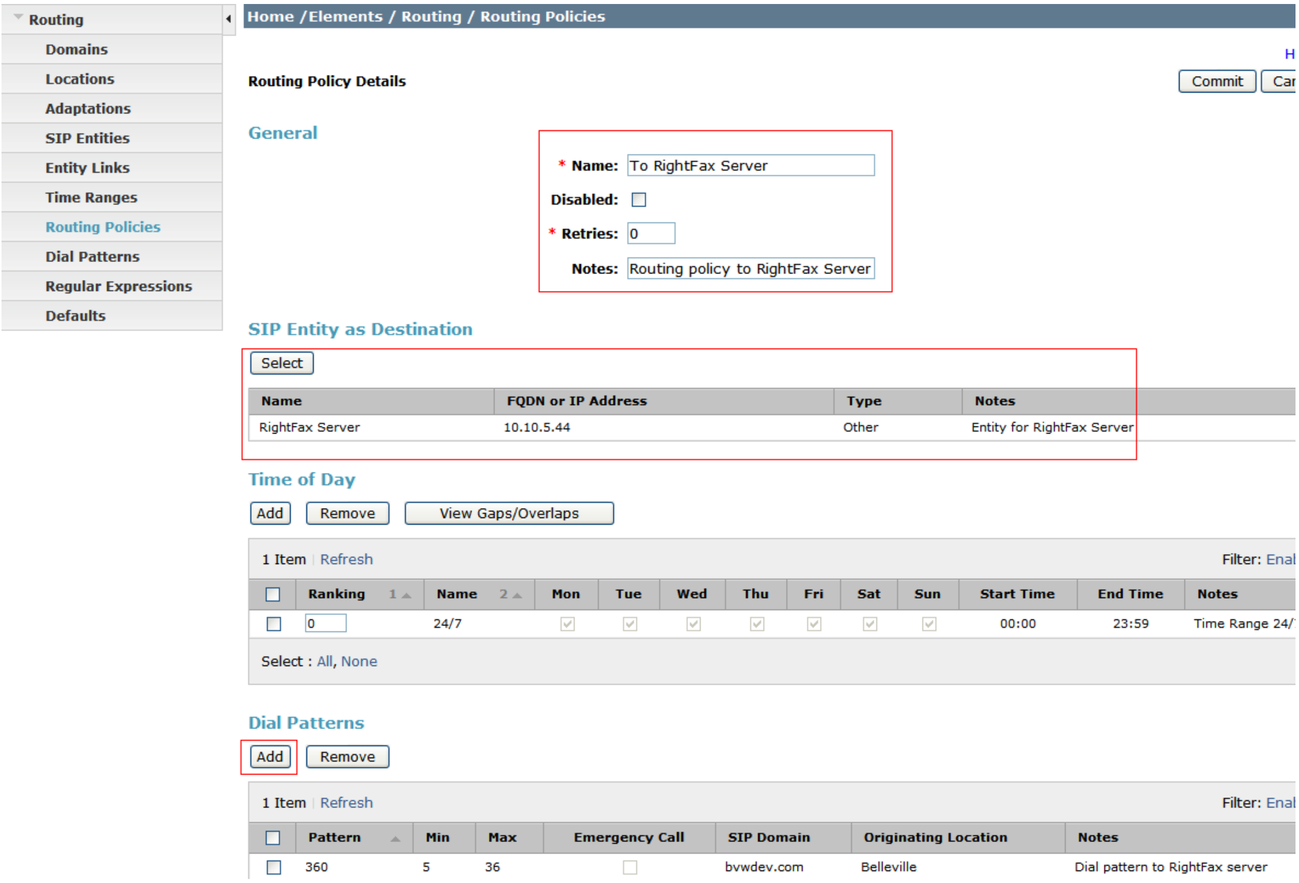
Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* RightFax	* InteropSM	UDP	* 5060	* RightFax Server	* 5060	Trusted	For RightFax

1.11. Adding Routing Policies

This section explains the Routing Policy configuration for RightFax server and Communication Manager. To add a routing policy, select **Routing Policies** from the left hand window of the Routing screen and click on **New** (not shown).

Screen below shows the Routing Policy Details for the RightFax server. Enter a descriptive name in the **Name** field and include some notes in the **Notes** field if required. Leave the rest of the values at default.

Click on the **Select** button and various SIP Entities configured are displayed (not shown). Select the **RightFax Server** as the SIP Entity Destination. To add a dial pattern, click on **Add** and various dial patterns that are configured is displayed (not shown). Select the dial pattern that needs to be associated with RightFax server. A dial pattern can be added once it has been configured as explained in **Section 6.7** below. Click on **Commit** to complete adding a routing policy.



Routing Policy Details

General

* Name: To RightFax Server

Disabled: ☐

* Retries: 0

Notes: Routing policy to RightFax Server

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
RightFax Server	10.10.5.44	Other	Entity for RightFax Server

Time of Day

1 Item | Refresh

☐	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
☐	0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/

Select : All, None

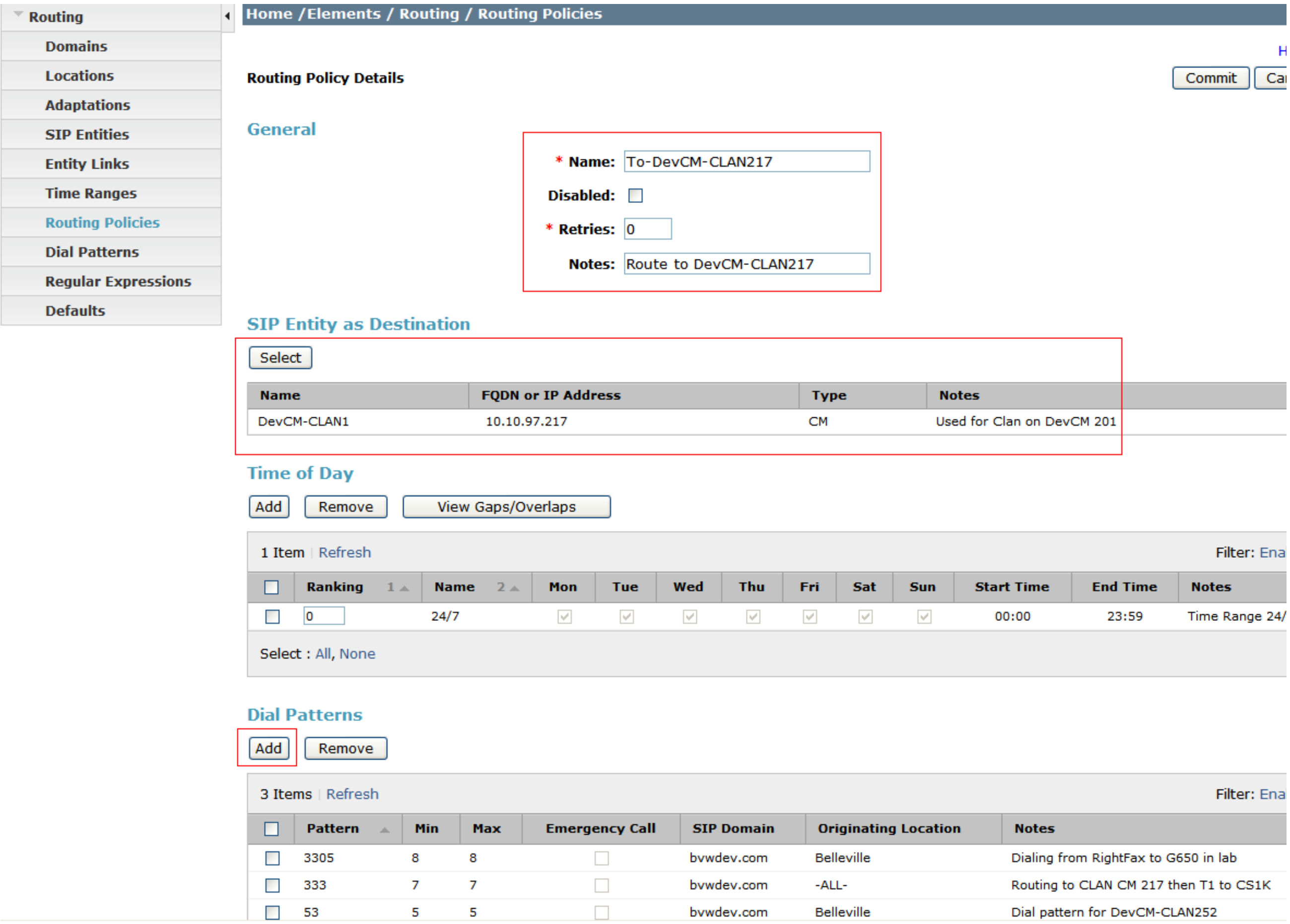
Dial Patterns

1 Item | Refresh

☐	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
☐	360	5	36	☐	bvwdev.com	Belleville	Dial pattern to RightFax server

Screen below shows the Routing Policy Details for the Communication Manager. Enter a descriptive name in the **Name** field and include some notes in the **Notes** field if required. Leave the rest of the values at default. Click on the **Select** button and various SIP Entities configured are displayed (not shown). Select the **DevCM-CLAN1** as the SIP Entity Destination. To add a dial pattern, click on **Add** and various dial patterns that are configured is displayed (not shown). Select the dial pattern that needs to be associated with Communication Manager. t. A dial pattern can be added once it has been configured as explained in **Section 6.7** below. Click on **Commit** to complete adding a routing policy.

Additional routing policies can be configured as required in a similar fashion.



Routing Policy Details

General

* Name: To-DevCM-CLAN217

Disabled: ☐

* Retries: 0

Notes: Route to DevCM-CLAN217

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
DevCM-CLAN1	10.10.97.217	CM	Used for Clan on DevCM 201

Time of Day

1 Item | Refresh

☐	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
☐	0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/

Select : All, None

Dial Patterns

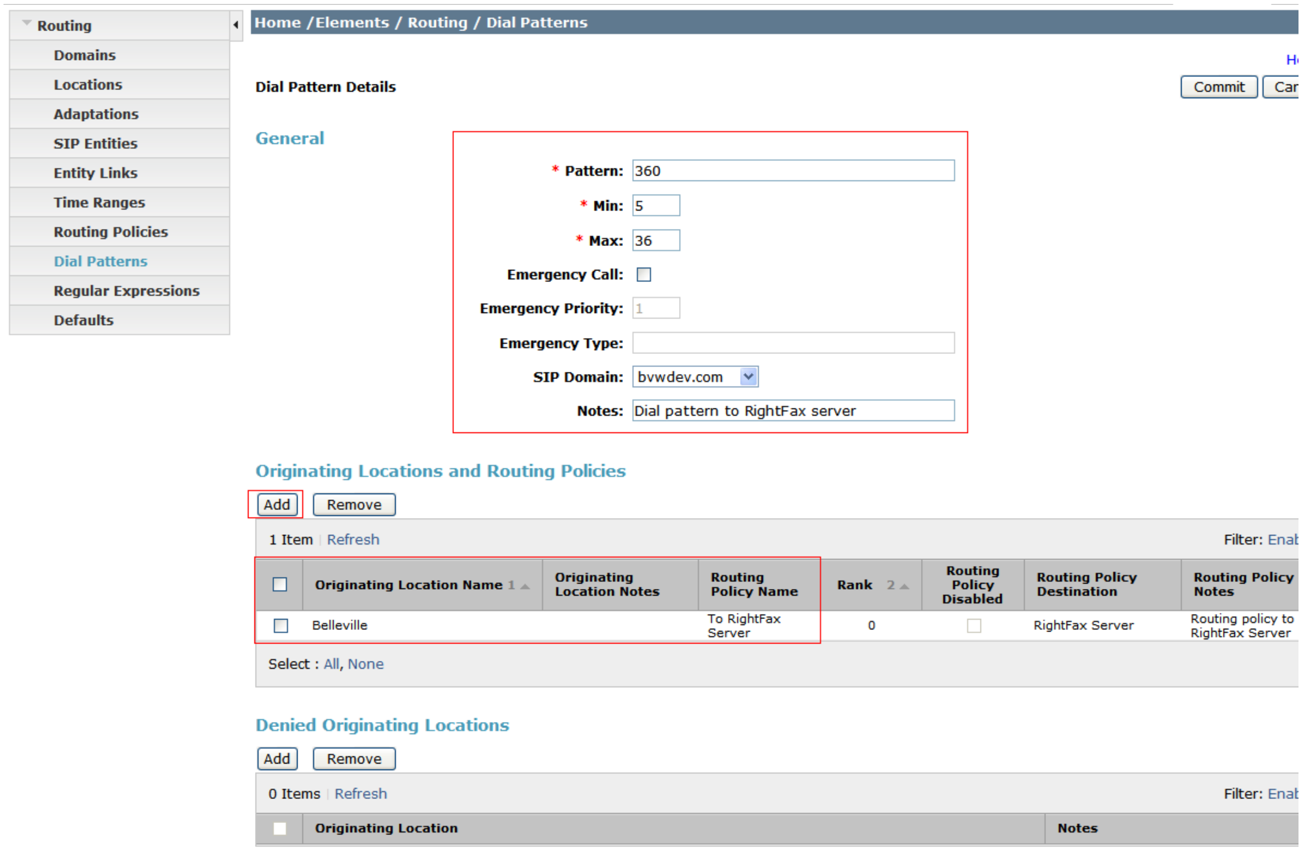
3 Items | Refresh

☐	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
☐	3305	8	8	☐	bvwdev.com	Belleville	Dialing from RightFax to G650 in lab
☐	333	7	7	☐	bvwdev.com	-ALL-	Routing to CLAN CM 217 then T1 to CS1K
☐	53	5	5	☐	bvwdev.com	Belleville	Dial pattern for DevCM-CLAN252

1.12. Adding Dial Patterns

This section explains the steps to add a dial pattern for the RightFax and Communication Manager. To add a dial pattern, select **Dial Patterns** from the left hand window of the Routing screen and click on **New** (not shown).

Screen below shows the Dial Pattern Details for the RightFax server. During compliance testing extensions range on RightFax server started with 360xx and therefore **360** are used in the **Pattern** field. The minimum and maximum size of the extension is defined as **5** to **36**. Add the **To RightFax Server** policy as configured in **Section 6.6** above. Click on **Commit** to complete adding the dial pattern. Additional dial patterns can be configured as required in a similar fashion.



Dial Pattern Details

General

* **Pattern:** 360

* **Min:** 5

* **Max:** 36

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: bvwdev.com

Notes: Dial pattern to RightFax server

Originating Locations and Routing Policies

1 Item | Refresh

☐	Originating Location Name 1	Originating Location Notes	Routing Policy Name	Rank 2	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
☐	Belleville		To RightFax Server	0	☐	RightFax Server	Routing policy to RightFax Server

Select : All, None

Denied Originating Locations

0 Items | Refresh

☐	Originating Location	Notes

Configure Open Text RightFax

This section describes the configuration of OpenText RightFax and the embedded RightFax Original Equipment Manufacturer (OEM) or Brooktrout SR140 virtual fax board software from Dialogic (hereafter referred to as "SR140"). It assumes that the application and all required software components, including Brooktrout SR140 and the database software (Microsoft SQL 2012), have been installed and properly licensed. For instructions on installing RightFax, refer to **Section 10**.

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, refer to **Section 10**.

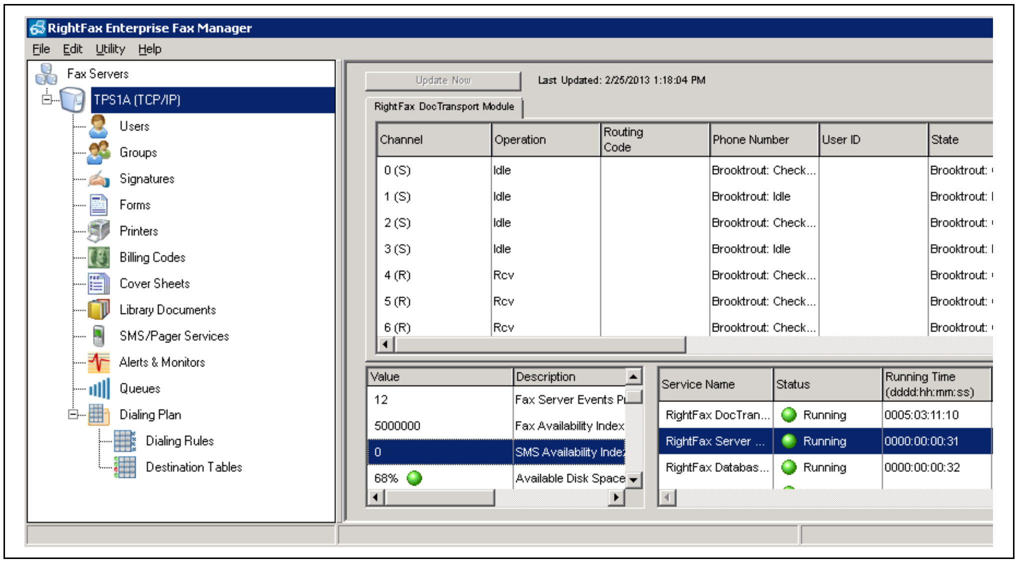
The configuration procedures covered in this section include the following:

- Launch RightFax Enterprise Fax Manager and Brooktrout Configuration Tool (Steps 1 – 6)
- Configure IP stack (Step 7)
- Configure BTCall parameters (Steps 8 – 9)
- Configure Call Control parameters (Step 10)
- Configure SIP IP parameters (Step 11)
- Configure T.38 parameters (Step 12)
- Configure RTP parameters (Steps 13 – 14)
- Administer RightFax dialing rules (Steps 15 – 16)
- Administer RightFax users (Steps 17 – 20)

The examples shown in this section refer to the Main Site. Unless specified otherwise, these same steps also apply to Remote Site using values appropriate from **Figure 1**.

1. **Launch RightFax Enterprise Fax Manager**

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:



Last Updated: 2/25/2013 1:18:04 PM

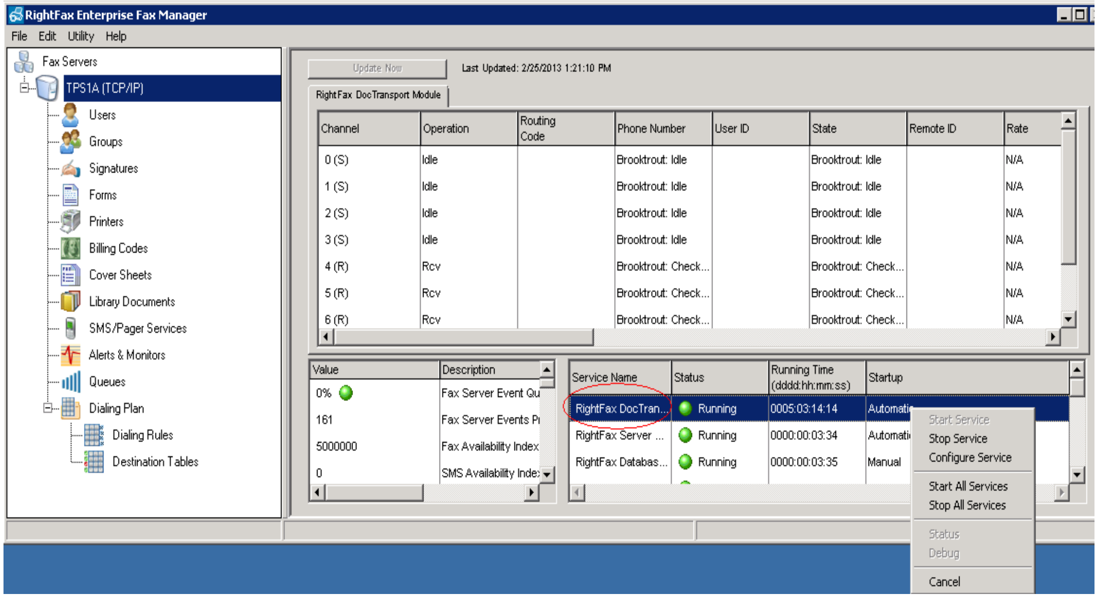
Channel	Operation	Routing Code	Phone Number	User ID	State
0 (S)	Idle		Brooktrout: Check...		Brooktrout:
1 (S)	Idle		Brooktrout: Idle		Brooktrout:
2 (S)	Idle		Brooktrout: Check...		Brooktrout:
3 (S)	Idle		Brooktrout: Idle		Brooktrout:
4 (R)	Rcv		Brooktrout: Check...		Brooktrout:
5 (R)	Rcv		Brooktrout: Check...		Brooktrout:
6 (R)	Rcv		Brooktrout: Check...		Brooktrout:

Value	Description
12	Fax Server Events P
5000000	Fax Availability Index
0	SMS Availability Inde
68% ●	Available Disk Space

Service Name	Status	Running Time (dddd:hh:mm:ss)
RightFax DocTran...	● Running	0005:03:11:10
RightFax Server ...	● Running	0000:00:00:31
RightFax Databas...	● Running	0000:00:00:32

2. **RightFax DocTransport Module**

The Brooktrout 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**.



Last Updated: 2/25/2013 1:21:10 PM

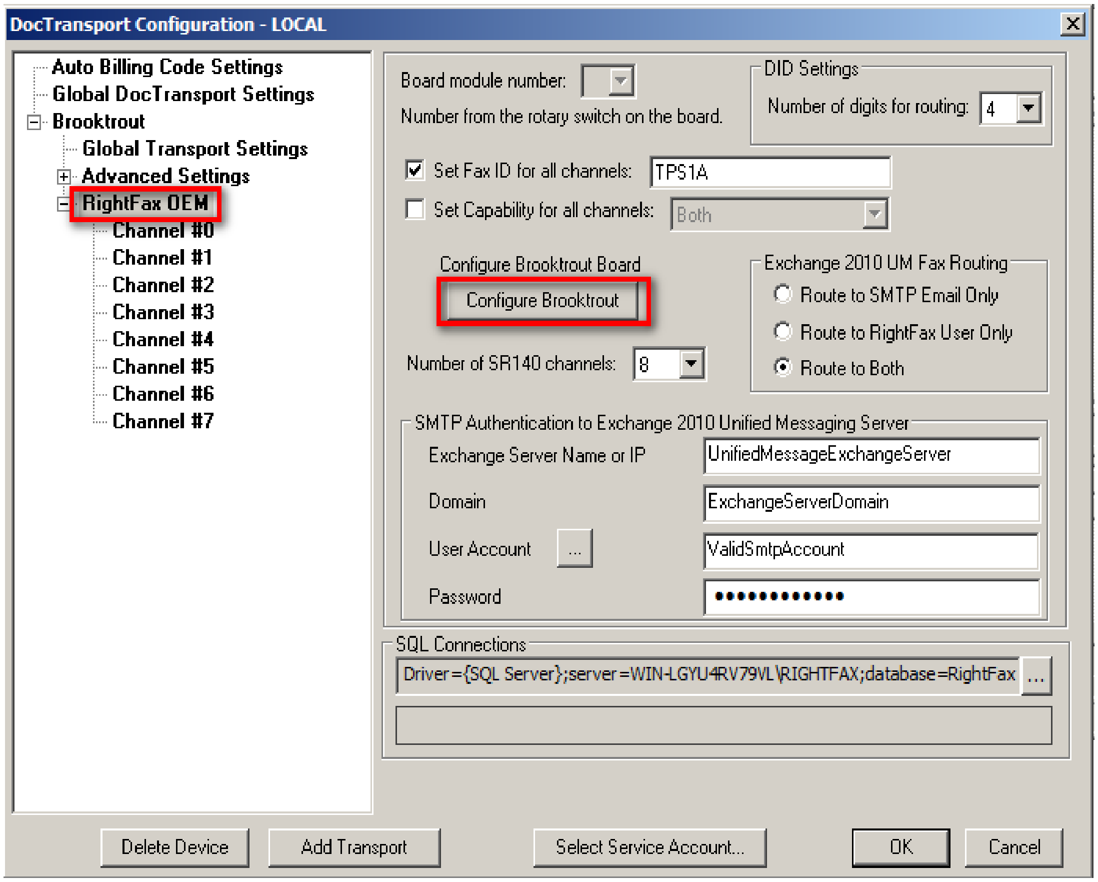
Channel	Operation	Routing Code	Phone Number	User ID	State	Remote ID	Rate
0 (S)	Idle		Brooktrout: Idle		Brooktrout: Idle		N/A
1 (S)	Idle		Brooktrout: Idle		Brooktrout: Idle		N/A
2 (S)	Idle		Brooktrout: Idle		Brooktrout: Idle		N/A
3 (S)	Idle		Brooktrout: Idle		Brooktrout: Idle		N/A
4 (R)	Rcv		Brooktrout: Check...		Brooktrout: Check...		N/A
5 (R)	Rcv		Brooktrout: Check...		Brooktrout: Check...		N/A
6 (R)	Rcv		Brooktrout: Check...		Brooktrout: Check...		N/A

Value	Description
0% ●	Fax Server Event Qu
161	Fax Server Events Pr
5000000	Fax Availability Index
0	SMS Availability Inde:

Service Name	Status	Running Time (dddd:hh:mm:ss)	Startup
RightFax DocTran...	● Running	0005:03:14:14	Automatic
RightFax Server ...	● Running	0000:00:03:34	Automati
RightFax Databas...	● Running	0000:00:03:35	Manual

3. **RightFax DocTransport Module - Continued**

In the **DocTransport Configuration** window that appears, click **RightFax OEM** (left side of screen), then click on the Configure **Brooktrout** button.



Board module number:

Number from the rotary switch on the board.

DID Settings

Number of digits for routing: 4

☑ Set Fax ID for all channels: TPS1A

☐ Set Capability for all channels: Both

Configure Brooktrout Board

Number of SR140 channels: 8

Exchange 2010 UM Fax Routing

○ Route to SMTP Email Only

○ Route to RightFax User Only

● Route to Both

SMTP Authentication to Exchange 2010 Unified Messaging Server

Exchange Server Name or IP: UnifiedMessageExchangeServer

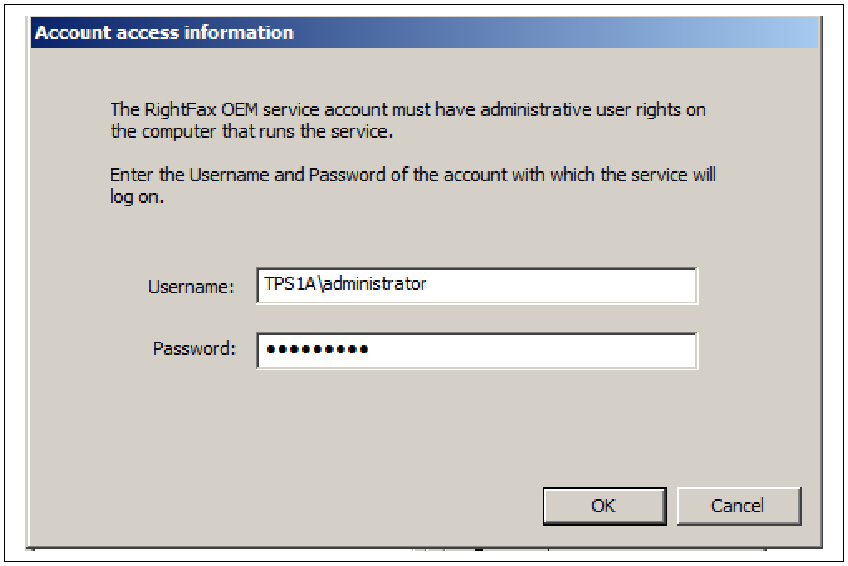
Domain: ExchangeServerDomain

User Account: ValidSmtpAccount

Password: •••••••••••••

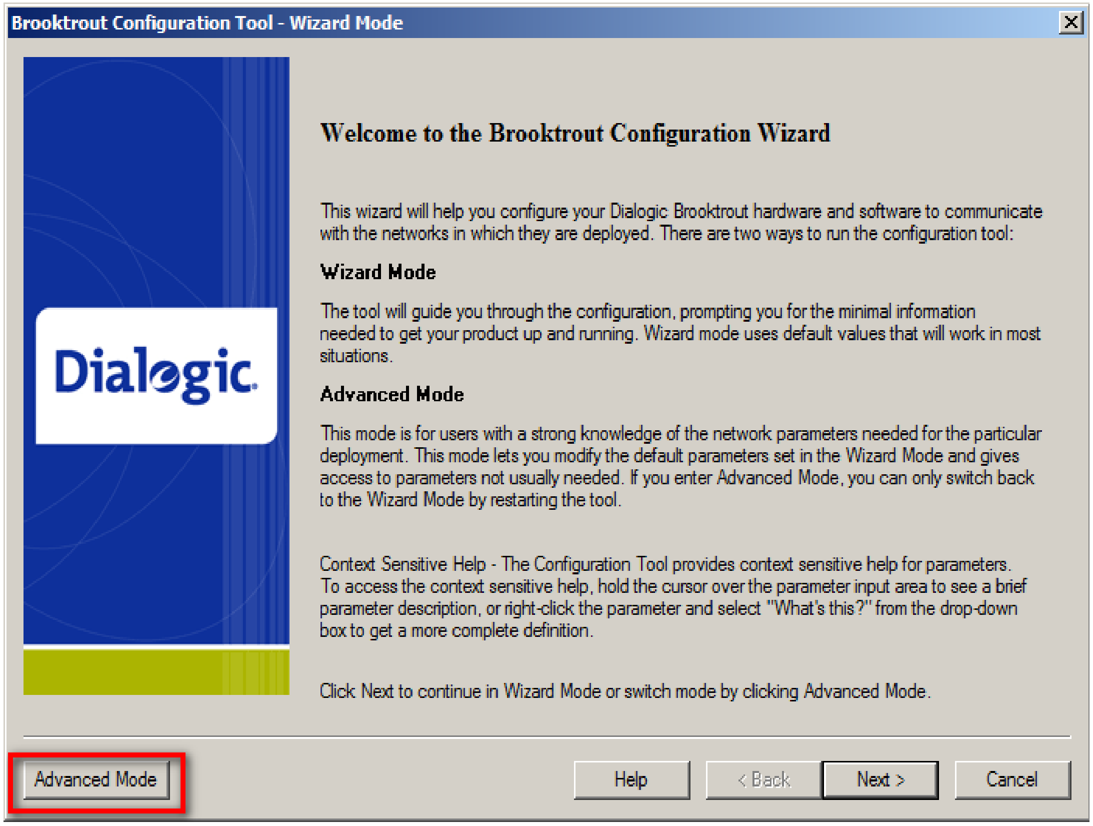
SQL Connections

Driver={SQL Server};server=WIN-LGYU4RV79VL\RIGHTFAX;database=RightFax

4.	**Account Access Information** Enter the credentials for the RightFax Service account used for the RightFax DocTransport Module. This account must have administrative user rights on the computer that runs the service.  The RightFax OEM service account must have administrative user rights on the computer that runs the service. Enter the Username and Password of the account with which the service will log on. Username: TPS1A\administrator Password: •••••••••

5. **Brooktrout Configuration Tool**

The **Brooktrout 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.

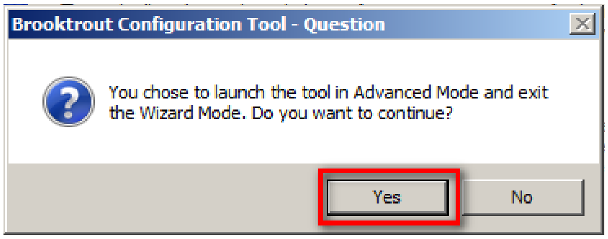
Advanced Mode

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.

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

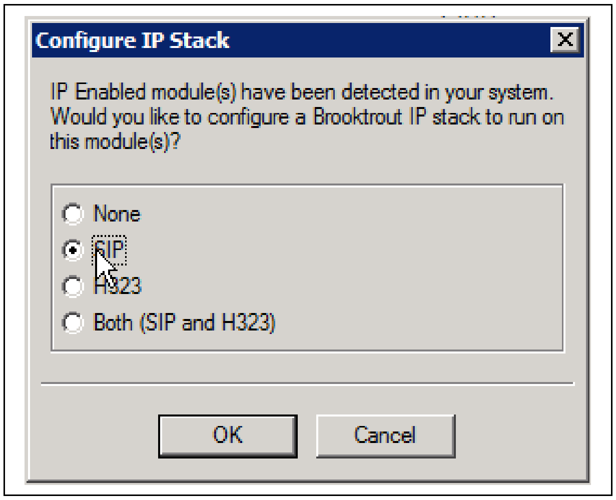


You chose to launch the tool in Advanced Mode and exit the Wizard Mode. Do you want to continue?

7.

Configure IP Stack

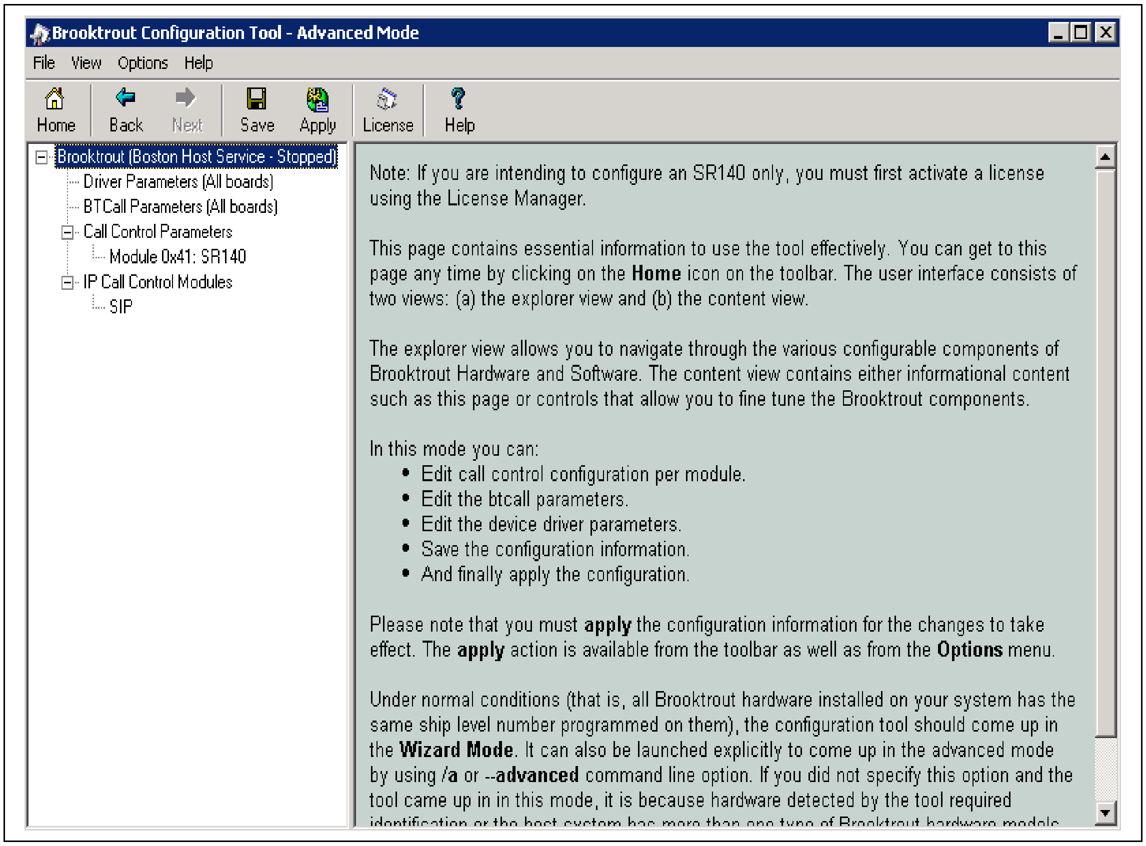
A Configure IP Stack window is displayed on first invocation of the Brooktrout configuration tool:



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)?

○ None
● SIP
○ H323
○ Both (SIP and H323)

Choose **SIP** and click **OK**. The following Brooktrout Configuration Tool window is displayed.



Note: If you are intending to configure an SR140 only, you must first activate a license using the License Manager.

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.

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.

In this mode you can:

- Edit call control configuration per module.
- Edit the btcall parameters.
- Edit the device driver parameters.
- Save the configuration information.
- And finally apply the configuration.

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.

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

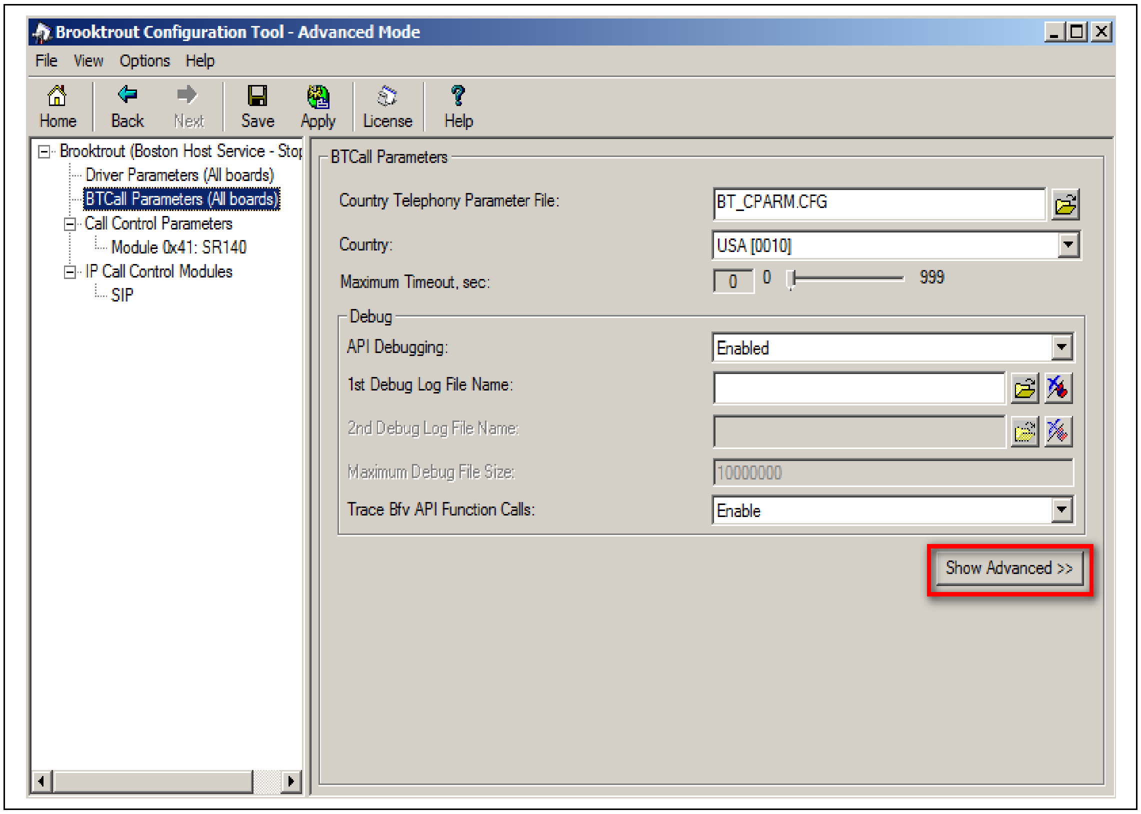
Note that IP Stack can be viewed/reconfigured from the Brooktrout Configuration Tool menu **Options** → **Configure IP Stack**.

8.

Configure BTCall Parameters

Note: *During the compliance testing, the following settings were configured differently than the default settings. In practice, these settings may not be required for full functionality.*

Navigate to **Brooktrout → BTCall Parameters (All boards)** in the left navigation menu. Click the **Show Advanced** button.



BTCall Parameters

Country Telephony Parameter File: BT_CPARM.CFG

Country: USA [0010]

Maximum Timeout, sec: 0 0 999

Debug

API Debugging: Enabled

1st Debug Log File Name:

2nd Debug Log File Name:

Maximum Debug File Size: 10000000

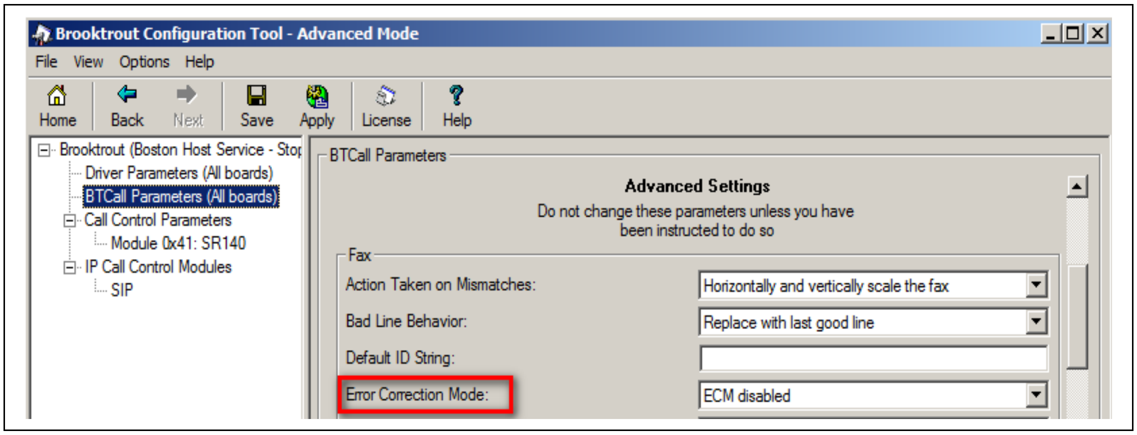
Trace Bfv API Function Calls: Enable

9. **Configure BTCall Parameters (continued)**

Under Advanced Settings, configure the fields as follows:

- **Error Correction Mode**: ECM Disabled
- **Permitted Compression Types**: MH only
- **V.34 Enable Send Call Indicator:** Disable
- **V.34 High Speed Control Signaling:** Disable
- **V.34 Modulation Capability:** Disable

Use default values for other fields.



Advanced Settings

Do not change these parameters unless you have been instructed to do so

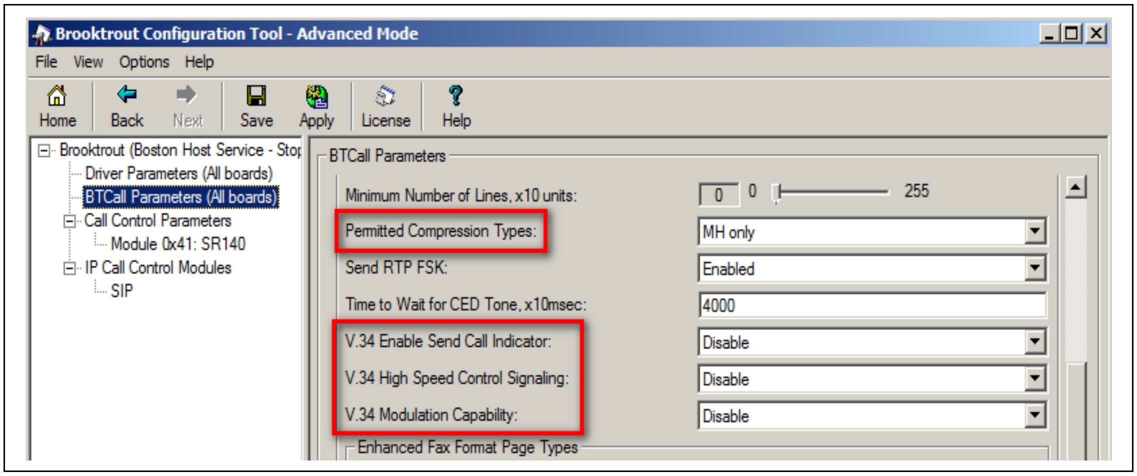
Fax

Action Taken on Mismatches: Horizontally and vertically scale the fax

Bad Line Behavior: Replace with last good line

Default ID String:

Error Correction Mode: ECM disabled



Minimum Number of Lines, x10 units: 0 0 255

Permitted Compression Types: MH only

Send RTP FSK: Enabled

Time to Wait for CED Tone, x10msec: 4000

V.34 Enable Send Call Indicator: Disable

V.34 High Speed Control Signaling: Disable

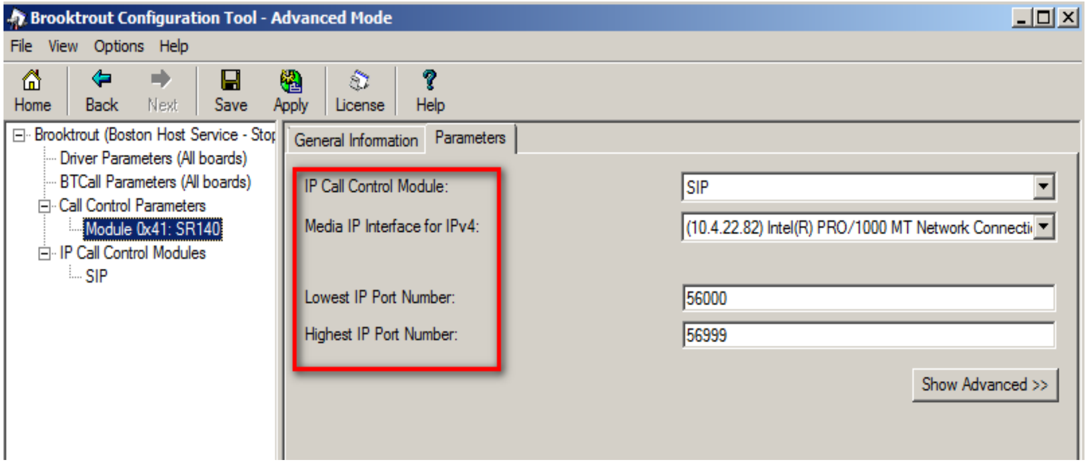
V.34 Modulation Capability: Disable

Enhanced Fax Format Page Types

10. **Configure Call Control Parameters**

Navigate to **Brooktrout → Call Control Parameters → Module 0x41: SR140** in the left navigation menu. Ensure the following configuration parameters are correct for your environment:

- **IP Call Control Module:** SIP
- **Media IP Interface for IPv4:** If the server contains multiple network interface cards (NICs), ensure you have selected an interface that is able to communicate with the Avaya Session Manager.
- **Lowest/Highest IP Port Numbers:** Ensure your RTP range matches the port range configured on the Avaya SIP infrastructure. *By default, the port range for SR140 is 56000 to 56999. A maximum range of 1000 ports may be specified. When you change the Lowest IP Port Number value, the Highest IP Port Number value will adjust automatically.*



IP Call Control Module: SIP

Media IP Interface for IPv4: (10.4.22.82) Intel(R) PRO/1000 MT Network Connecti

Lowest IP Port Number: 56000

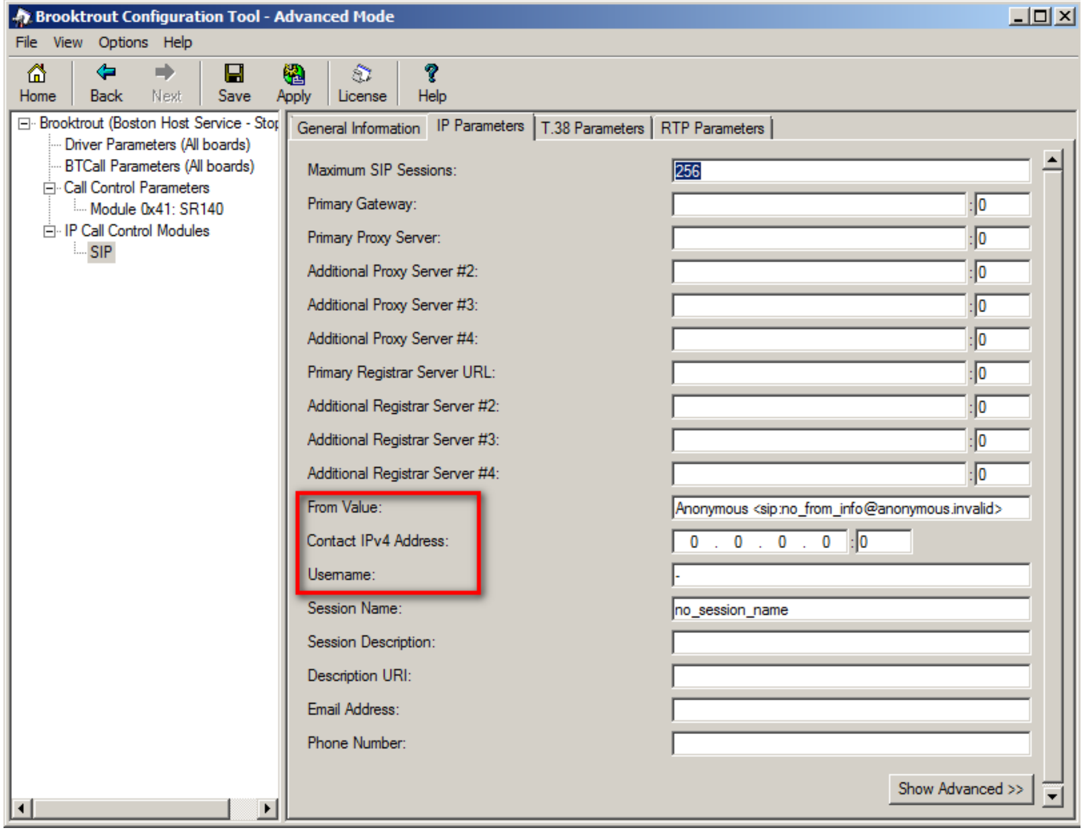
Highest IP Port Number: 56999

11. **Configure SIP IP Parameters**

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:

- **From Value** – If required by the Avaya environment, set this to an appropriate ***UserInfo@DomainName.*** The *DomainName* should be set to the authoritative domain as configured in Session Manager. During compliance testing this value was left at default.
- **Contact Address** – If required by the Avaya environment, set enter the IP address assigned to RightFax and the port number ***5060***. During compliance testing this value was left at default.
- **Username** – Required. Default value is a dash (‘-‘) character.

Use default values for all other fields.



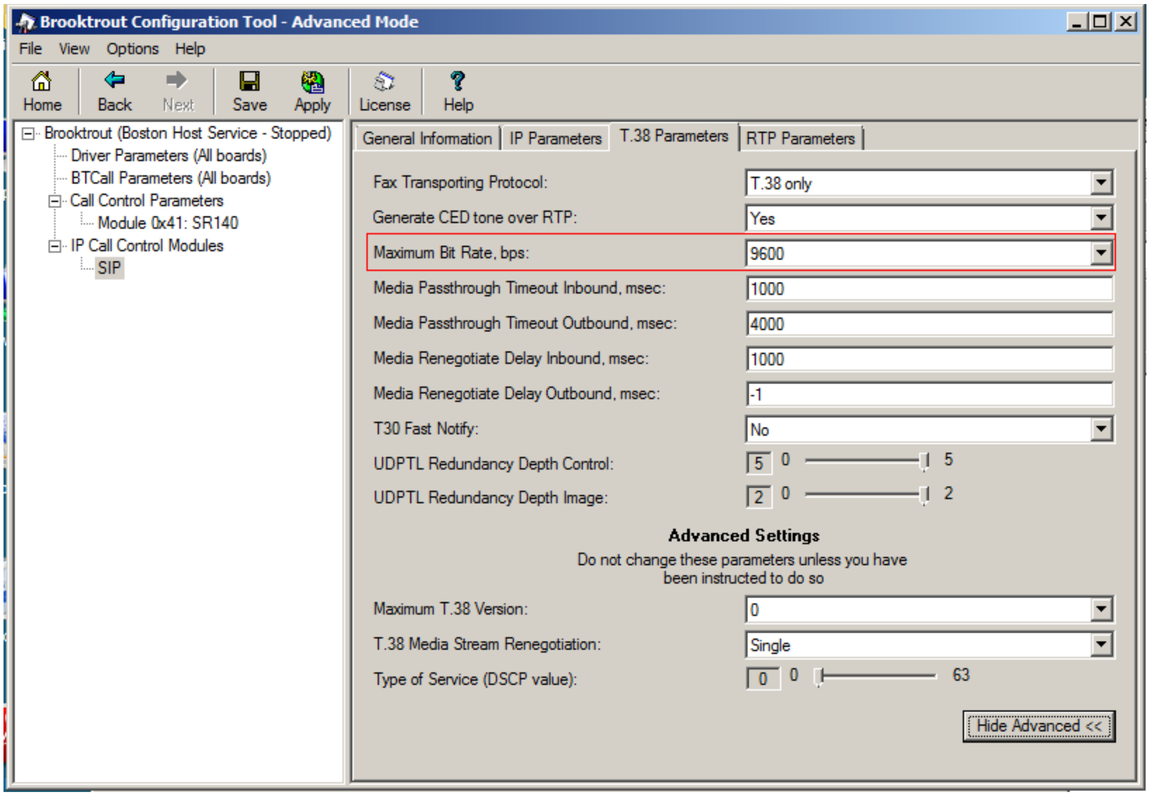
Maximum SIP Sessions: 256
Primary Gateway: :0
Primary Proxy Server: :0
Additional Proxy Server #2: :0
Additional Proxy Server #3: :0
Additional Proxy Server #4: :0
Primary Registrar Server URL: :0
Additional Registrar Server #2: :0
Additional Registrar Server #3: :0
Additional Registrar Server #4: :0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

12. **Configure T.38 Parameters**

Select the **T.38 Parameters** tab. Configure the fields as shown below in the screenshot.

Note: *During the compliance testing, the following settings were configured differently than the default settings. In practice, these settings may not be required for full functionality.*

- *"Maximum Bit Rate, bps" is set to maximum, 9600 (default is 14400).*



Fax Transporting Protocol: T.38 only
Generate CED tone over RTP: Yes
Maximum Bit Rate, bps: 9600
Media Passthrough Timeout Inbound, msec: 1000
Media Passthrough Timeout Outbound, msec: 4000
Media Renegotiate Delay Inbound, msec: 1000
Media Renegotiate Delay Outbound, msec: -1
T30 Fast Notify: No
UDPTL Redundancy Depth Control: 5 0 5
UDPTL Redundancy Depth Image: 2 0 2

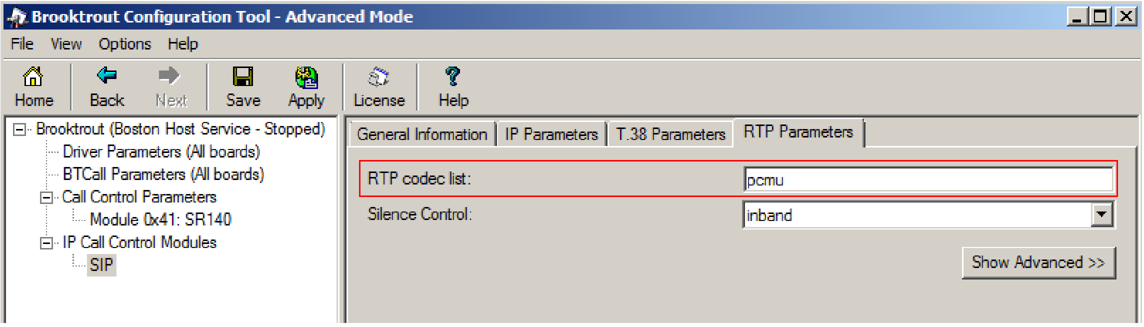
Advanced Settings

Do not change these parameters unless you have been instructed to do so

Maximum T.38 Version: 0
T.38 Media Stream Renegotiation: Single
Type of Service (DSCP value): 0 0 63

13. **Configure RTP Parameters**

Select the **RTP Parameters** tab. Set the **RTP codec list** value to use only a single codec, either ***pcmu*** or ***pcma*** to match the codec used in your region.



RTP codec list: pcmu
Silence Control: inband

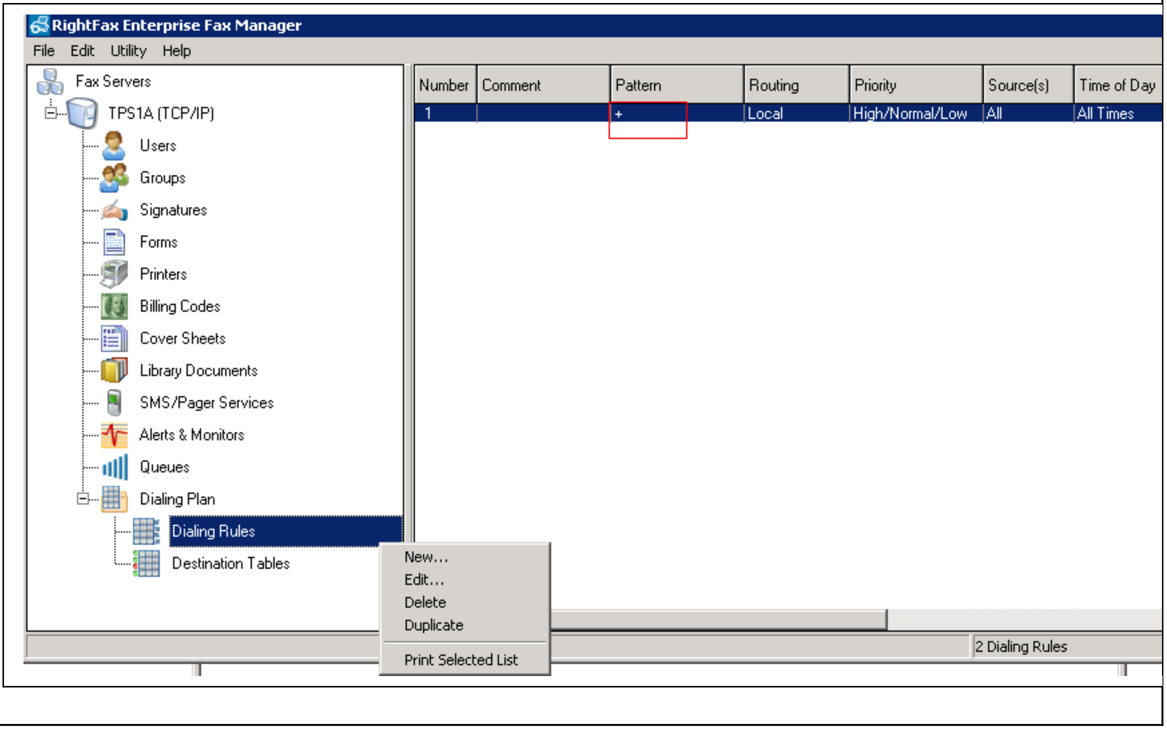
14.	**Complete Brooktrout SR140 Configuration** After verifying all the above parameters are properly set, click **Save** in the button menu (not shown). Exit the Brooktrout Configuration Tool. In the **DocTransport Configuration** screen, click the **OK** button (See screen shot in Step 3 above). Restart 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 screen shot in Step 2 above).

15. **Configure Dialing Rules**

Dialing Rules are used by RightFax to route calls. In the compliance test, a dialing rule was created to route outbound fax calls to the Session Manager.

In the left navigation menu under the host name of the fax server, navigate to **Dialing Plan**, right-click **Dialing Rules** and select **New** to create a new rule.

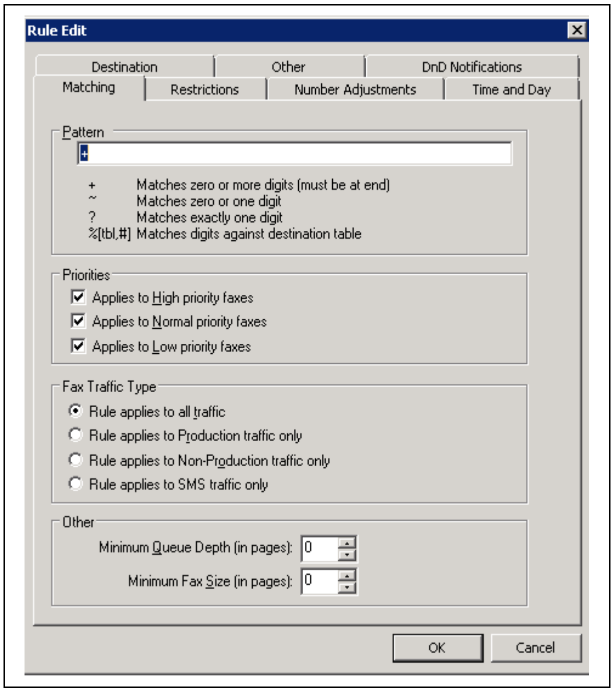
The example below shows the single rule created for the compliance test at Main site. The + in the **Pattern** field indicates that this rule applies to all dialed numbers. To view the details, double click on the rule in the right pane.



Number	Comment	Pattern	Routing	Priority	Source(s)	Time of Day
1		+	Local	High/Normal/Low	All	All Times

16. **Configure Dialing Rules - Continued**

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 outbound fax phone number is appended with ***@10.10.1.11*** as indicated in the **Append this** field. This IP address is for the Session Manager server at the Main site.



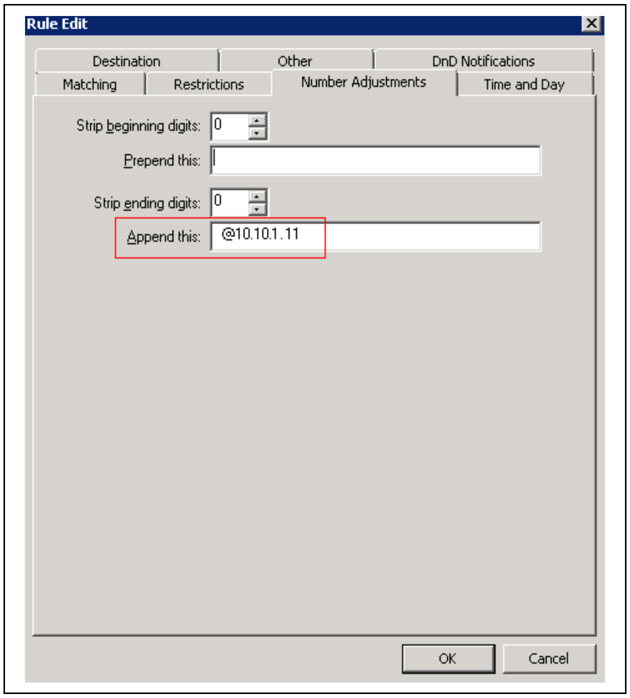
Pattern: +

\+ Matches zero or more digits (must be at end)
~ Matches zero or one digit
? Matches exactly one digit
%[tbl,#] Matches digits against destination table

Priorities
☑ Applies to High priority faxes
☑ Applies to Normal priority faxes
☑ Applies to Low priority faxes

Fax Traffic Type
● Rule applies to all traffic
○ Rule applies to Production traffic only
○ Rule applies to Non-Production traffic only
○ Rule applies to SMS traffic only

Other
Minimum Queue Depth (in pages): 0
Minimum Fax Size (in pages): 0

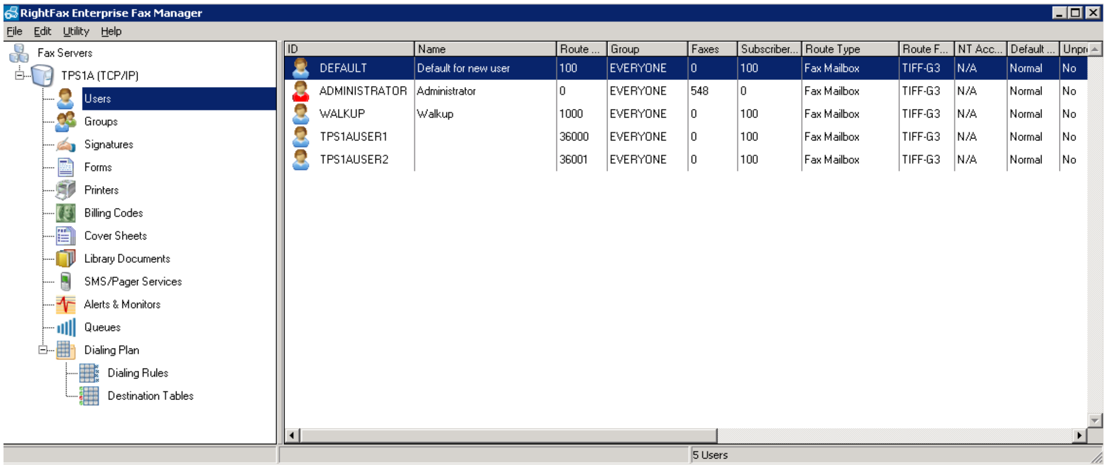


Strip beginning digits: 0
Prepend this:
Strip ending digits: 0
Append this: @10.10.1.11

17.

Configure Users

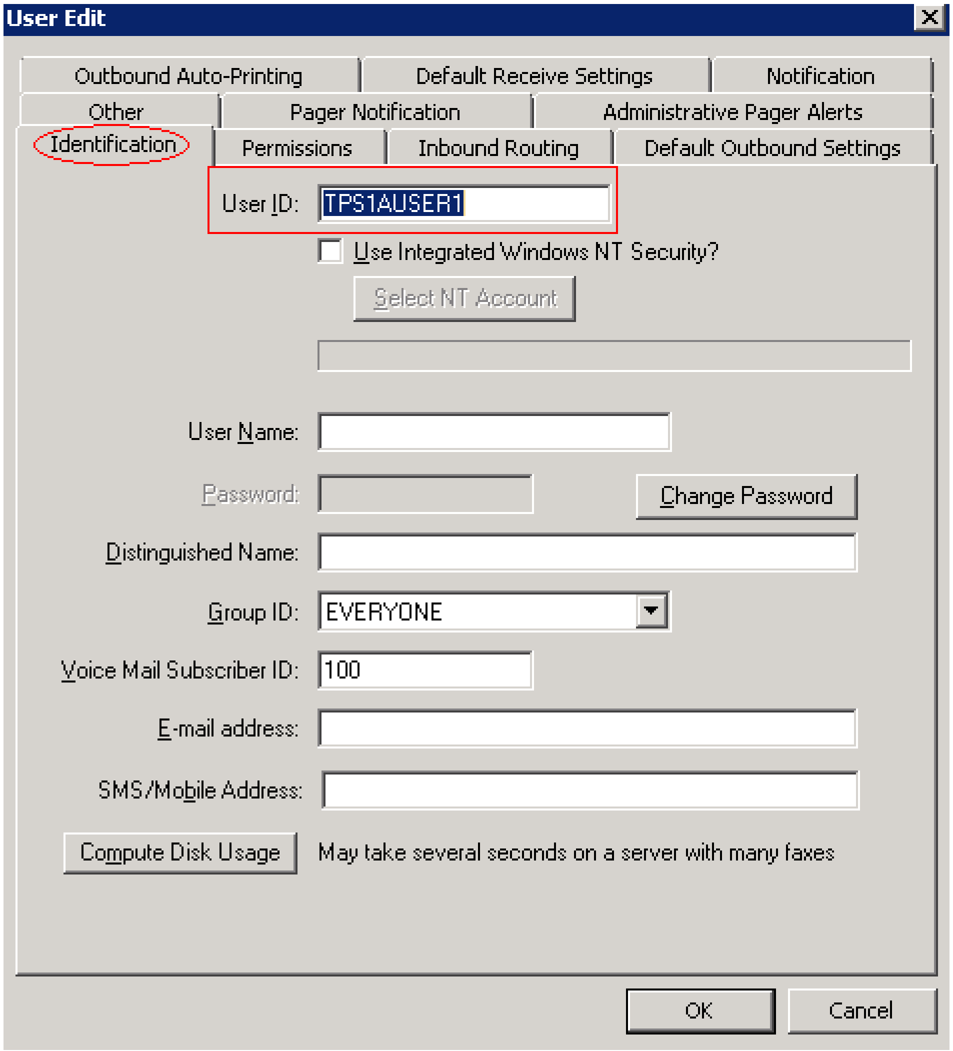
A user is created on the RightFax server for each incoming fax number. The user represents the fax recipient. To view the list of users, navigate to **Users** in the left navigation menu under the host name of the fax server. The example below shows a list of five users. To view the details of a user, double-click on the user entry in the right pane.



ID	Name	Route ...	Group	Faxes	Subscriber...	Route Type	Route F...	NT Acc...	Default ...	Unpr
DEFAULT	Default for new user	100	EVERYONE	0	100	Fax Mailbox	TIFF-G3	N/A	Normal	No
ADMINISTRATOR	Administrator	0	EVERYONE	548	0	Fax Mailbox	TIFF-G3	N/A	Normal	No
WALKUP	Walkup	1000	EVERYONE	0	100	Fax Mailbox	TIFF-G3	N/A	Normal	No
TPS1AUSER1		36000	EVERYONE	0	100	Fax Mailbox	TIFF-G3	N/A	Normal	No
TPS1AUSER2		36001	EVERYONE	0	100	Fax Mailbox	TIFF-G3	N/A	Normal	No

18. **Configure Users – Identification**

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 Main site. The **User ID** field is set to a descriptive name. Appropriate values should be entered or selected for other fields.



User ID: TPS1AUSER1

☐ Use Integrated Windows NT Security?

User Name:

Password:

Distinguished Name:

Group ID: EVERYONE

Voice Mail Subscriber ID: 100

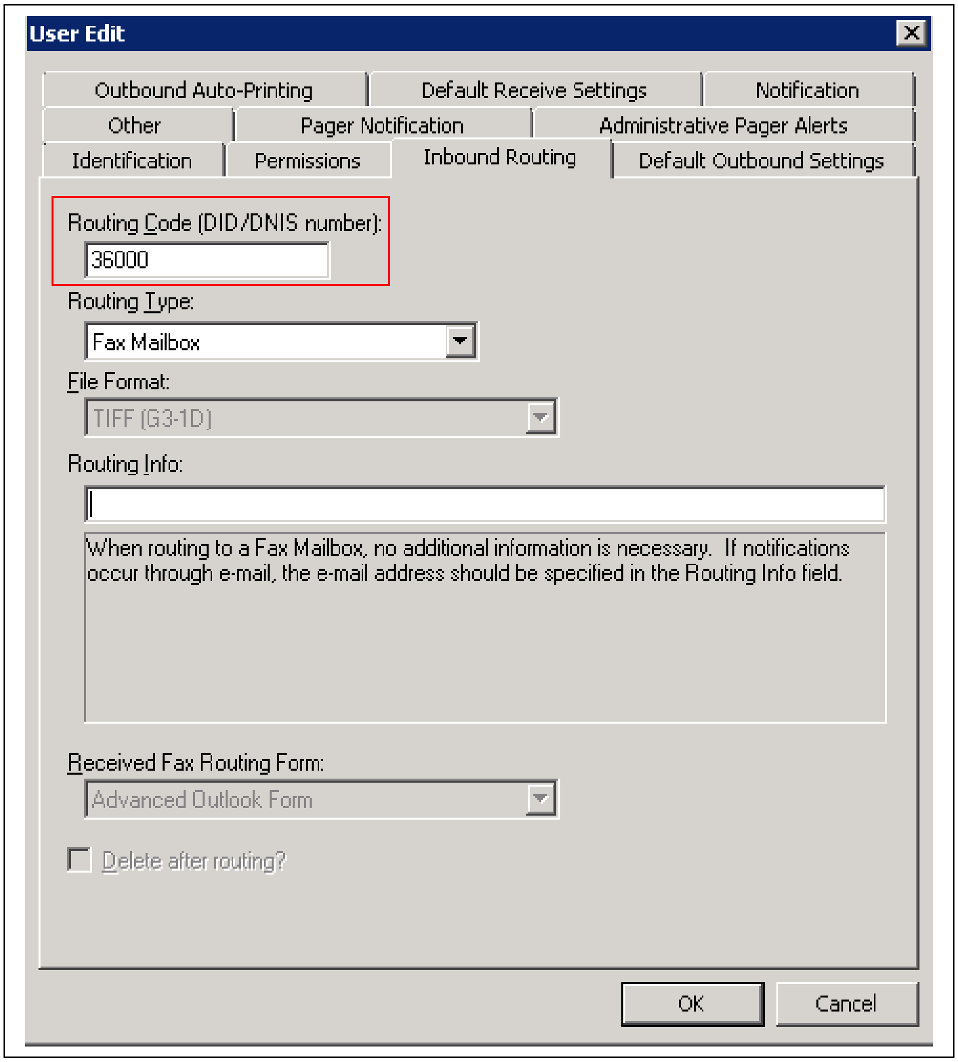
E-mail address:

SMS/Mobile Address:

May take several seconds on a server with many faxes.

19. **Configure Users – Inbound Routing**

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 ***36000*** for Main site as shown below. Default values may be used for all other fields.



Routing Code (DID/DNIS number): 36000

Routing Type: Fax Mailbox

File Format: TIFF (G3-1D)

Routing Info:

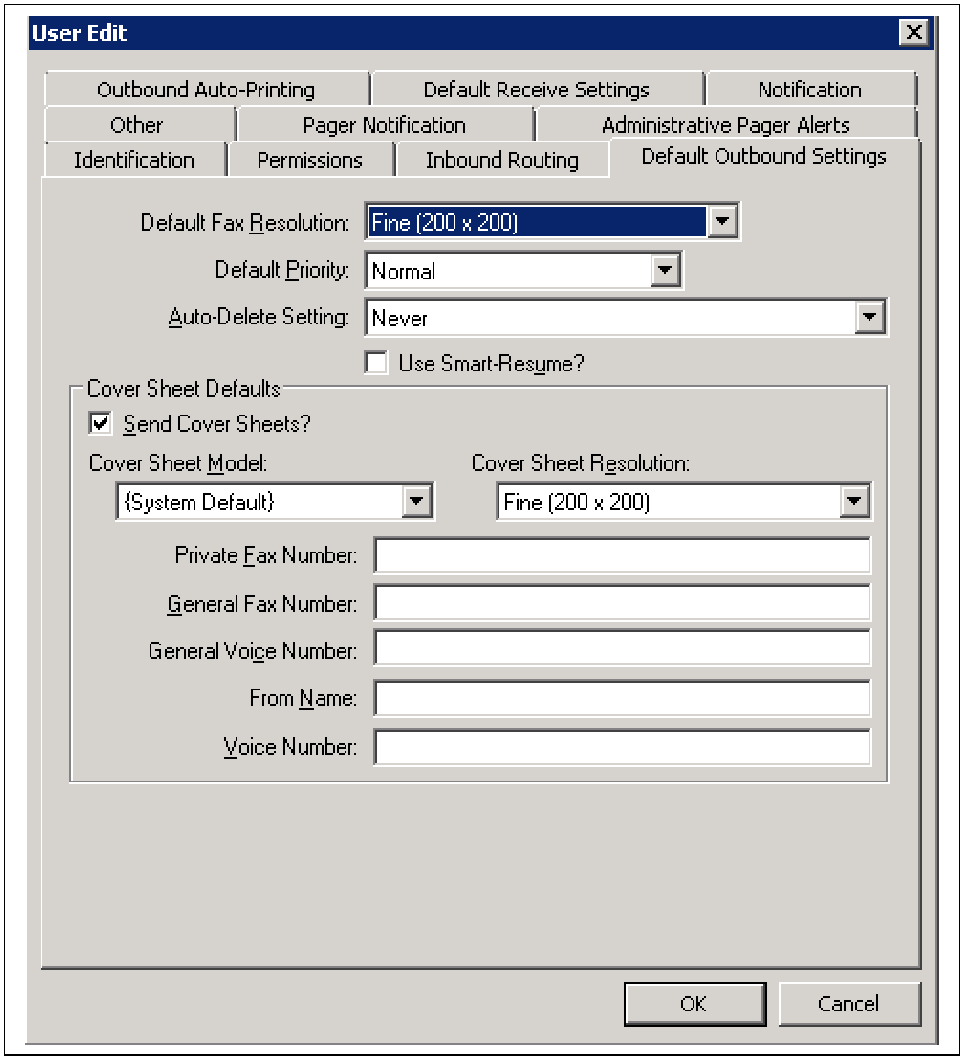
When routing to a Fax Mailbox, no additional information is necessary. If notifications occur through e-mail, the e-mail address should be specified in the Routing Info field.

Received Fax Routing Form: Advanced Outlook Form

☐ Delete after routing?

20. **Configure Users – Outbound Settings**

The **Default Outbound Settings** tab configures various outbound fax call settings. Configure these settings as appropriate.



User Edit

Default Fax Resolution: Fine (200 x 200)

Default Priority: Normal

Auto-Delete Setting: Never

☐ Use Smart-Resume?

Cover Sheet Defaults

☑ Send Cover Sheets?

Cover Sheet Model: {System Default}

Cover Sheet Resolution: Fine (200 x 200)

Private Fax Number:

General Fax Number:

General Voice Number:

From Name:

Voice Number:

Verification Steps

The following steps may be used to verify the configuration:

- From Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling groups configured in **Step 8** of **Section 5.1** are in-service.
- From Communication Manager SAT, use the **status signaling-group** command to verify that the ISDN signaling groups configured in **Step 16** of **Section 5.1** are in-service.
- From Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group configured in **Section 5.1**, **Steps 9 - 10** is in-service.
- From Communication Manager SAT, use the **status trunk-group** command to verify that the ISDN trunk group configured in **Section 5.1**, **Step 16 - 17** is in-service.
- Verify that fax calls can be placed to/from Open Text RightFax server at each site.
- From Communication Manager SAT, use the **list trace tac** command to verify that fax calls are routed to the expected trunks.
- From System Manager, confirm that the Entity Link between Session Manager and the Open Text RightFax server is in service.

Conclusion

These Application Notes describe the procedures required to configure Open Text RightFax server to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Open Text RightFax successfully passed compliance testing with the observations and notes mentioned in **Section 2.2**.

Additional References

[1] *Avaya Aura® Communication Manager Feature Description and Implementation*, Doc # 555-245-205, Release 6.2, Issue 9.0, December, 2012.
[2] *Administering Avaya Aura® Communication Manager*, Doc # 03-300509, Release 6.2, Issue 7.0, December, 2012.
[3] *Administering Avaya Aura® Session Manager*, Doc # 03-603324, Release 6.2, July, 2012.
[4] *Administering Avaya Aura® System Manager*, Release 6.2, Issue 2.0, July 2012

[5] *OpenText RightFax 10.5 Administrator's Guide,* July, 2012.
[6] *OpenText RightFax 10.5 Installation Guide,* June, 2012.

Documentation for:
Avaya products may be found at http://support.avaya.com.
RightFax products may be found at https://knowledge.opentext.com. (Valid login required).

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.