Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Sagemcom XMediusFAX Service Provider Edition with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Sagemcom XMediusFAX Service Provider (SP) Edition with Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

XMediusFAX is a software based fax server that sends and receives fax calls over an IP network. In the configuration tested, XMediusFAX interoperates with Avaya Aura® Session Manager and Avaya Aura® Communication Manager to send/receive faxes using SIP trunks and the T.38 fax protocol between XMediusFAX 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 Sagemcom XMediusFAX Service Provider (SP) Edition with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunks.

XMediusFAX is a software based fax server that sends and receives fax calls over an IP network. In the configuration tested, XMediusFAX interoperates with the Session Manager and Communication Manager to send/receive faxes using SIP trunks and the T.38 protocol between XMediusFAX and the Avaya SIP infrastructure. The compliance testing focused on fax calls to and from the XMediusFAX fax server using various page lengths, resolutions, and fax data speeds.

2. General Test Approach and Test Results

This section describes the general test approach used to verify the interoperability of Sagemcom XMediusFAX SP Edition with the Avaya SIP infrastructure (Session Manager and Communication Manager). This section also covers the test results.

2.1. Interoperability Compliance Testing

The general test approach was to make intra-site and inter-site fax calls to and from the XMediusFAX fax server. The compliance tested configuration contained two sites. Site 2 served as the main enterprise site and Site 1 served as a simulated PSTN or a remote enterprise site. Inter-site calls and simulated PSTN calls were made using SIP trunks and ISDN-PRI trunks between the sites. By using two Communication Managers and two port networks at Site 1, fax calls across multiple TDM/IP hops were able to be tested. Faxes were sent with various page lengths, resolutions, and at various fax data speeds. For capacity testing, 100 2-page faxes were continuously sent between the two XMediusFAX fax servers. Serviceability testing included verifying proper operation/recovery from network outages, unavailable resources, and Communication Manager and XMediusFAX 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; and the integrated VoIP engine of the Avaya G450 Media Gateway.

2.2. Test Results

XMediusFAX successfully passed compliance testing.

2.3. Support

For technical support on XMediusFAX, contact Sagemcom at:

- Web: http://xmediusfax.sagemcom.com/support/
- Phone: (888) 766-1668
- Email: xmediusfax.support.americas@sagemcom.com

3. Reference Configuration

Figure 1 illustrates the reference configuration used during testing. In the reference configuration, the two sites are connected via a direct SIP trunk and an ISDN-PRI trunk. Faxes were sent between the two sites using either of these two trunks, as dictated by each individual test case.

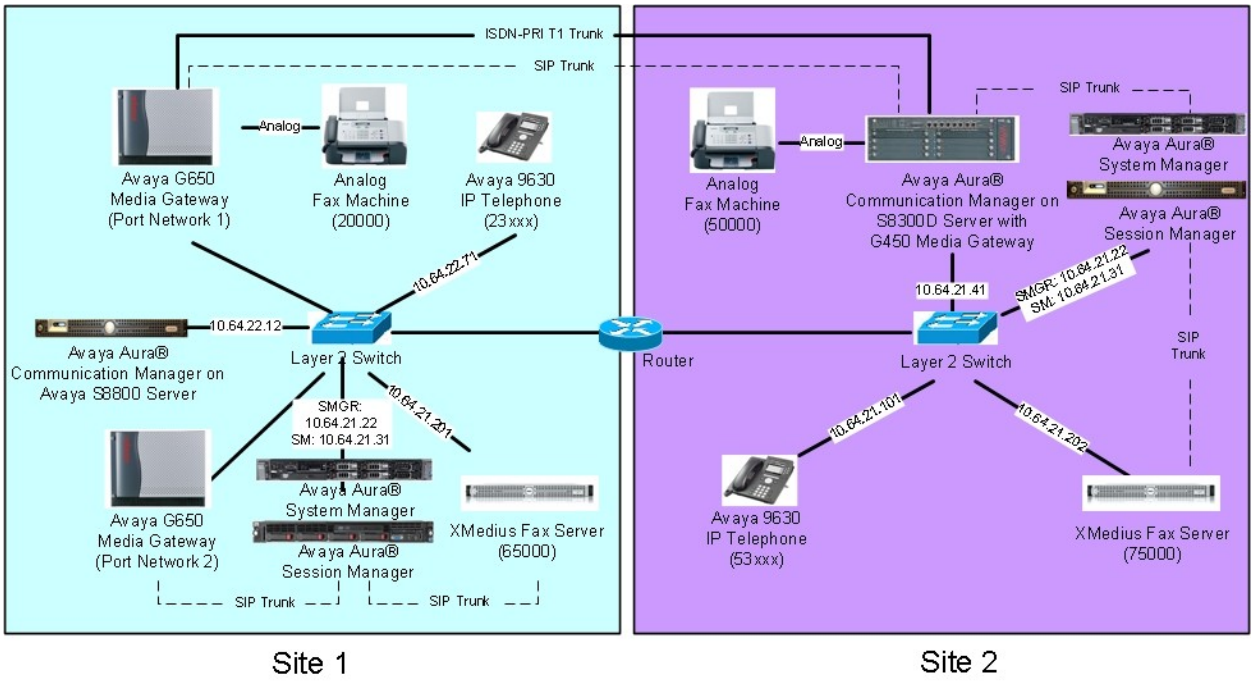


Figure 1: XMediusFAX with Session Manager and Communication Manager

At Site 1 consists of the following equipment:

- An Avaya S8800 Server running Avaya Aura® Communication Manager with two Avaya G650 Media Gateways. Each media gateway is configured as a separate port network in separate IP network regions. The media resources required are provided by the IP Media Processor (MedPro) circuit packs. Two versions of the IP MedPro circuit pack were tested in the configuration: the TN2302AP and the TN2602AP.
- An Avaya S8800 Server running Avaya Aura® System Manager. System Manager provides management functions for Session Manager.
- An Avaya S8800 Server running Avaya Aura® Session Manager.
- XMediusFAX running on a Windows 2008 R2 Enterprise Server (SP1).
- An analog fax machine.
- Various Avaya IP endpoints (not all shown).

At Site 2 consists of the following equipment:

- An Avaya S8300D Server running Avaya Aura® Communication Manager in an Avaya G450 Media Gateway. The signaling and media resources needed to support SIP trunks are integrated directly on the media gateway processor.

- A Dell™ PowerEdge™ R610 Server running Avaya Aura® System Manager. System Manager provides management functions for Session Manager.
- An HP ProLiant DL360 G7 Server running Avaya Aura® Session Manager.
- XMediusFAX running on a Windows 2008 R2 Enterprise Server (SP1).
- An analog fax machine
- Various Avaya IP endpoints (not all shown).

Although the IP endpoints (H.323 and SIP telephones) are not involved in the faxing operations, they are present at both sites to verify that VoIP telephone calls are not affected by the FoIP faxing operations and vice versa.

Outbound fax calls originating from the XMediusFAX fax server are sent to Session Manager first, and then from Session Manager to Communication Manager via SIP trunks. Based on the dialed digits, Communication Manager will either direct the calls to the local fax machine, or to the other site via an ISDN-PRI or SIP trunk. Inbound fax calls terminating to the XMediusFAX fax server are sent from the local fax machine or from the remote site are received by Communication Manager. The calls are then directed to Session Manager for onward routing to the XMediusFAX fax server via SIP trunks.

4. Equipment and Software Validated

The following equipment and software were used for the reference configuration:

Equipment	Software
Site 1	
Avaya S8800 Server with a Avaya G650 Media Gateways: - 2 CLANs – TN799DP - 2 IP MedPros – TN2302AP - 2 IP MedPros – TN2602AP	Avaya Aura® Communication Manager 6.0.1, R016x.00.1.510.1, Patch 19009 : - HW01, FW038 - HW20, FW120 - HW02, FW57
Avaya S8800 Server	Avaya Aura® System Manager: 6.0.0 (Build No. – 6.0.0.0.688-3.0.7.2) (Avaya Aura® System Platform: 6.0.2.1.5)
Avaya S8800 Server	Avaya Aura® Session Manager 6.0.2.0.602004
XMediusFAX fax server (Windows 2008 R2 Enterprise Server, SP1)	6.5.5 with patch XMFSP_6.5.5.213
Fax Machine	-
Various Avaya SIP and H.323 endpoints	-
Site 2	
Avaya S8300D Server with a Avaya G450 Media Gateway	Avaya Aura® Communication Manager 6.0.1, R016x.00.1.510.1, Patch 19009 (Avaya Aura® System Platform: 6.0.3.0.3)
Dell™ PowerEdge™ R610 Server	Avaya Aura® System Manager: 6.1.0 (Build No. – 6.1.0.0.7345-6.1.5.106), Software Update Revision No : 6.1.6.1.1087 (Avaya Aura® System Platform: 6.0.3.0.3)
HP ProLiant DL360 G7 Server	Avaya Aura® Session Manager 6.1.2.0.612004
XMediusFAX fax server (Windows 2008 R2 Enterprise Server, SP1)	6.5.5 with patch XMFSP_6.5.5.213
Fax Machine	-
Various Avaya SIP and H.323 endpoints	-

5. Configure Communication Manager

This section describes the Communication Manager configuration at Site 2 to support the network shown in **Figure 1**. Although not shown is this document, a similar Communication Manager configuration would be required at Site 1.

The configuration of 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.

Step	Description
1.	**License** 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 32
           Maximum Concurrently Registered IP Stations: 18000 15
             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: 18000 0
                   Maximum Video Capable IP Softphones: 18000 1
                      Maximum Administered SIP Trunks: 24000 170
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                            Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
          Maximum TN2602 Boards with 80 VoIP Channels: 128   0
         Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

       (NOTE: You must logoff & login to effect the permission changes.)
```

Step	Description
2.	**IP network region** Use the **display ip-network-region** command to view the network region settings. The values shown below are the values used during compliance testing. ▪ **Authoritative Domain**: ***avaya.com*** This field was configured to match the domain name configured on Session Manager. The domain will appear in the "From" header of SIP messages originating from this IP region. ▪ **Name**: Any descriptive name may be used (if desired). ▪ **Intra-region IP-IP Direct Audio**: ***yes*** **Inter-region IP-IP Direct Audio**: ***yes*** By default, IP-IP direct audio (media shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Shuffling can be further restricted at the trunk level on the **Signaling Group** form. ▪ **Codec Set**: ***1*** The codec set contains the list of codecs available for calls within this IP network region.

```
Display ip-network-region 1                                   Page   1 of  20
                             IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name: FAX testing
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
```

Step	Description
3.	**Codecs** IP codec set 1 was used during compliance testing. Multiple codecs can be listed in priority order to allow the codec used by a specific call to be negotiated during call establishment. The example below shows the values used during compliance testing.

```
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:
```

On **Page 2**, set the **FAX Mode** field to ***t.38-standard***. 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. 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.

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

                         IP Codec Set

                              Allow Direct-IP Multimedia? y
            Maximum Call Rate for Direct-IP Multimedia:  2048:Kbits
    Maximum Call Rate for Priority Direct-IP Multimedia:  2048:Kbits

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

Step	Description
4.	**Node Names** Use the **change node-names ip** command to create a node name for the IP address of Session Manager. Enter a descriptive name in the **Name** column and the IP address assigned to Session Manager in the **IP address** column.

```
change node-names ip                                          Page   1 of   2
                                   IP NODE NAMES
    Name                  IP Address
AES_21_46              10.64.21.46
CM_20_40               10.64.20.40
CM_22_12_CLAN1A        10.64.22.16
CM_22_12_CLAN2A        10.64.22.19
IPO_21_64              10.64.21.64
SM_20_31               10.64.20.31
SM_21_31               10.64.21.31
default                0.0.0.0
msgserver              10.64.21.41
procr                  10.64.21.41
procr6                 ::
```

Step	Description
5.	**Signaling Group** Signaling group 1 was used for the signaling group associated with the SIP trunk group between Communication Manager and Session Manager. The signaling groups and trunk groups between the two sites of the reference configuration is assumed to already be in place and not described in this document. Signaling group 1 was configured using the parameters highlighted below. ▪ **Near-end Node Name**: ***procr*** This node name maps to the IP address of the Avaya S8300D Server. Node names are defined using the **change node-names ip** command. ▪ **Far-end Node Name**: ***SM_21_31*** This node name maps to the IP address of Session Manager. ▪ **Far-end Network Region**: ***1*** This defines the IP network region which contains Session Manager. ▪ **Far-end Domain**: ***avaya.com*** This domain is sent in the "To" header of SIP messages of calls using this signaling group. ▪ **Direct IP-IP Audio Connections**: ***y*** This field must be set to ***y*** to enable media shuffling on the SIP trunk.

```
display signaling-group 1
                               SIGNALING GROUP

 Group Number: 1               Group Type: sip
  IMS Enabled? n         Transport Method: tls
       Q-SIP? n                                          SIP Enabled LSP? n
    IP Video? y           Priority Video? n        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: SM_21_31
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain: avaya.com
                                            Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                 RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? y                 Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```

Step	Description
6.	**Trunk Group** Trunk group 1 was used for the SIP trunk group between Communication Manager and Session Manager. The signaling groups and trunk groups between the two sites of the reference configuration is assumed to already be in place and not described in this document. Trunk group 1 was configured using the parameters highlighted below. ▪ **Group Type: *sip*** This field sets the type of the trunk group. ▪ **TAC: *101*** Enter an valid value consistent with the Communication Manager dial plan ▪ **Member Assignment Method: *auto*** Set to Auto. ▪ **Signaling Group**: ***1*** This field is set to the signaling group shown in the previous step. ▪ **Number of Members: *50*** This field represents the number of trunk group members 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. Thus, a call from a SIP telephone to another SIP telephone will use two SIP trunks. A call between a non-SIP telephone and a SIP telephone will only use one trunk.

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

Group Number: 1                     Group Type: sip           CDR Reports: y
  Group Name: to SM_21_31                 COR: 1        TN: 1        TAC: 101
   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: 1
                                                        Number of Members: 50
```

Step	Description
	Trunk Group – continued On **Page 3**: ▪ The **Numbering Format** field was set to ***unk-pvt***. This field specifies the format of the calling party number sent to the far-end. ▪ The default values may be retained for the other fields.
	display trunk-group 1 **Page 3** of 21 TRUNK FEATURES ACA Assignment? n Measured: none Maintenance Tests? y **Numbering Format: unk-pvt** UUI Treatment: service-provider Replace Restricted Numbers? n Replace Unavailable Numbers? n Modify Tandem Calling Number: no Show ANSWERED BY on Display? y
7.	**Private Numbering** Private Numbering defines the calling party number to be sent to the far-end. In the example shown below, all calls originating from a ***5***-digit extension beginning with ***5*** and routed across any trunk group will be sent as a ***5*** digit calling number. The calling party number is sent to the far-end in the SIP "From" header.
	display private-numbering 0 Page 1 of 2 NUMBERING - PRIVATE FORMAT Ext Ext Trk Private Total Len Code Grp(s) Prefix Len **5 5** **5** Total Administered: 1 Maximum Entries: 540

Step	Description
8.	**Automatic Alternate Routing** Automatic Alternate Routing (AAR) was used to route calls either to Session Manager or to the Communication Manager at the other site. Use the **change aar analysis** command to create an entry in the AAR Digit Analysis Table. The example below shows numbers that begin with ***75*** and are ***5*** digits long use route pattern ***1*** (to Session Manager). Numbers that begin with ***20000*** or ***65*** and are ***5*** digits long use route pattern *7*, which routes calls to Communication Manager at the other site via a SIP trunk (route pattern 8 was also used at times to route calls to Communication Manager at the other site via an ISDN-PRI trunk).

```
display aar analysis 2                                      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
    2                     3    3    5         aar          n
    20000                 5    5    7         aar          n
    23                    5    5    8         aar          n
    531                   5    5    1         unku         n
    532                   5    5    1         unku         n
    59997                 5    5    99        aar          n
    65                    5    5    7         aar          n
    75                    5    5    1         aar          n
```

Step	Description
9.	**Route Pattern** Route pattern 1 was used for calls destined for the XMediusFAX fax server through Session Manager. Route patterns 7 and 8 (not shown) were used for calls destined for the other site in the reference configuration. Route pattern 1 was configured using the parameters highlighted below. ▪ **Pattern Name**: Any descriptive name. ▪ **Grp No**: ***1*** This field is set to the trunk group number defined in **Step 5**. ▪ **FRL**: ***0*** This field sets the Facility Restriction Level of the trunk. It must be set to an appropriate level to allow authorized users to access the trunk. The level of 0 is the least restrictive.

```
change route-pattern 1                                        Page   1 of   3
                   Pattern Number: 1   Pattern Name: to SM_21_31
                                 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: 1    0                         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                                 lev0-pvt  none
 2: y y y y y n  n           rest                                           none
 3: y y y y y n  n           rest                                           none
 4: y y y y y n  n           rest                                           none
 5: y y y y y n  n           rest                                           none
 6: y y y y y n  n           rest                                           none
```

6. Configure Session Manager

This section provides the procedures for configuring Session Manager (version 6.1) as provisioned at Site 2 in the reference configuration. Although not shown is this document, a similar Session Manager configuration would be required at Site 1 (using the appropriate version 6.0 screens). All provisioning for Session Manager is performed via the System Manager web interface.

The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

The Session Manager server provides the network interface for all inbound and outbound SIP signaling and media transport to all provisioned SIP entities. During compliance testing, the IP address assigned to the Security Module interface is 10.64.21.31 as specified in **Figure 1**. The Session Manager server also has a separate network interface used for connectivity to System Manager for provisioning Session Manager. The IP address assigned to the Session Manager management interface is 10.64.21.30.

The procedures described in this section include configurations in the following areas:

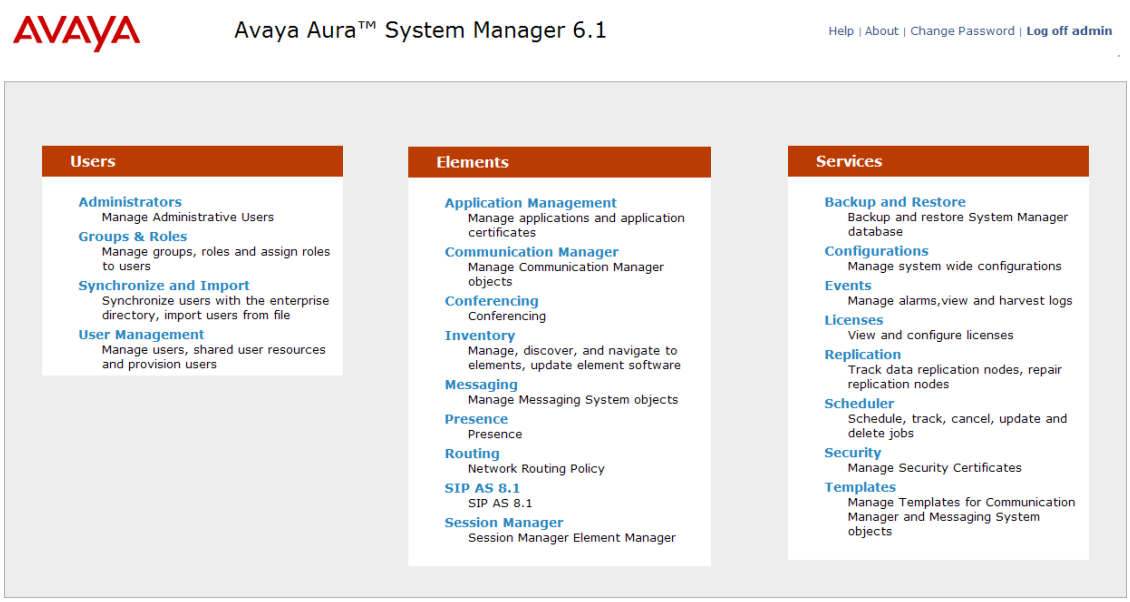
- **SIP Domains** – SIP Domains are the domains for which Session Manager is authoritative in routing SIP calls. In other words, for calls to such domains, Session Manager applies Network Routing Policies to route those calls to SIP Entities. For calls to other domains, Session Manager routes those calls to another SIP proxy (either a pre-defined default SIP proxy or one discovered through DNS).
- **Locations** – Locations define the physical and/or logical locations in which SIP Entities reside. Call Admission Control (CAC) / bandwidth management may be administered for each location to limit the number of calls to and from a particular Location.
- **Adaptations** – Adaptations are used to apply any necessary protocol adaptations, e.g., modify SIP headers, and apply any necessary digit conversions for the purpose of inter-working with specific SIP Entities.
- **SIP Entities** – SIP Entities represent SIP network elements such as Session Manager instances, Communication Manager systems, Session Border Controllers, SIP gateways, SIP trunks, and other SIP network devices.
- **Entity Links** – Entity Links define the SIP trunk/link parameters, e.g., ports, protocol (UDP/TCP/TLS), and trust relationship, between Session Manager instances and other SIP Entities.
- **Time Ranges** – Time Ranges specify customizable time periods, e.g., Monday through Friday from 9AM to 5:59PM, Monday through Friday 6PM to 8:59AM, all day Saturday and Sunday, etc. A Network Routing Policy may be associated with one or more Time Ranges during which the Network Routing Policy is in effect.
- **Routing Policies** – Routing Policies are used in conjunction with a Dial Patterns to specify a SIP Entity that a call should be routed to.
- **Dial Patterns** – A Dial Pattern specifies a set of criteria and a set of Network Routing Policies for routing calls that match the criteria. The criteria include the called party number and SIP domain in the Request-URI, and the Location from which the call originated. For

example, if a call arrives at Session Manager and matches a certain Dial Pattern, then Session Manager selects one of the Network Routing Policies specified in the Dial Pattern. The selected Network Routing Policy in turn specifies the SIP Entity to which the call is to be routed.

1. **Login**

   Access the System Manager administration web interface by entering https://<*ip-addr*>/SMGR/ as the URL in an Internet browser, where <*ip-addr*> is the IP address of the System Manager server.

   Log in with the appropriate credentials. The main page for the administrative interface is shown below.



2. **Add SIP Domain**

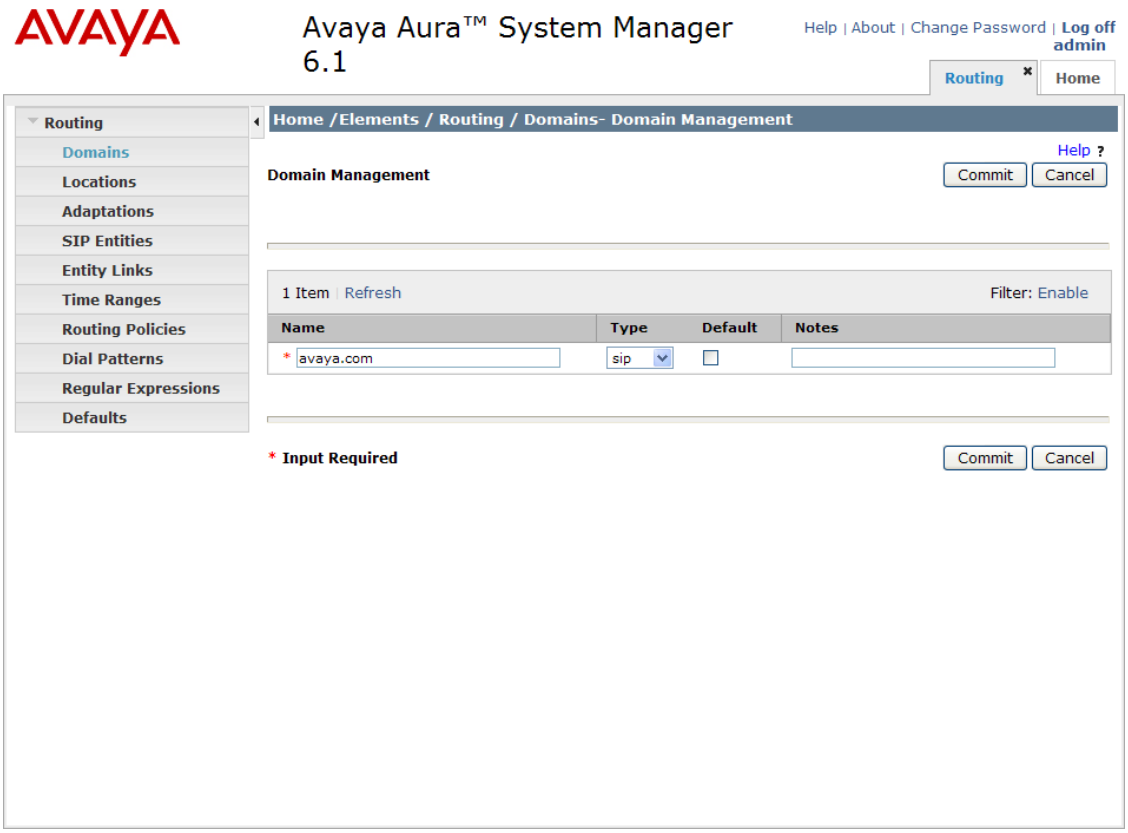
The **Routing** menu contains all the configuration tasks listed at the beginning of this section.

During compliance testing, one SIP Domain was configured.

Navigate to **Routing→Domains**, and click the **New** button (not shown) to add the SIP domain with

- **Name**: ***avaya.com*** (as set in **Section 5, Step 2**)
- **Notes**: optional descriptive text

Click **Commit** to save the configuration.



3. **Add Location**

Locations identify logical and/or physical locations where SIP entities reside. Only one Location was configured at each site for compliance testing.

Navigate to **Routing→Locations** and click the **New** button (not shown) to add the Location.

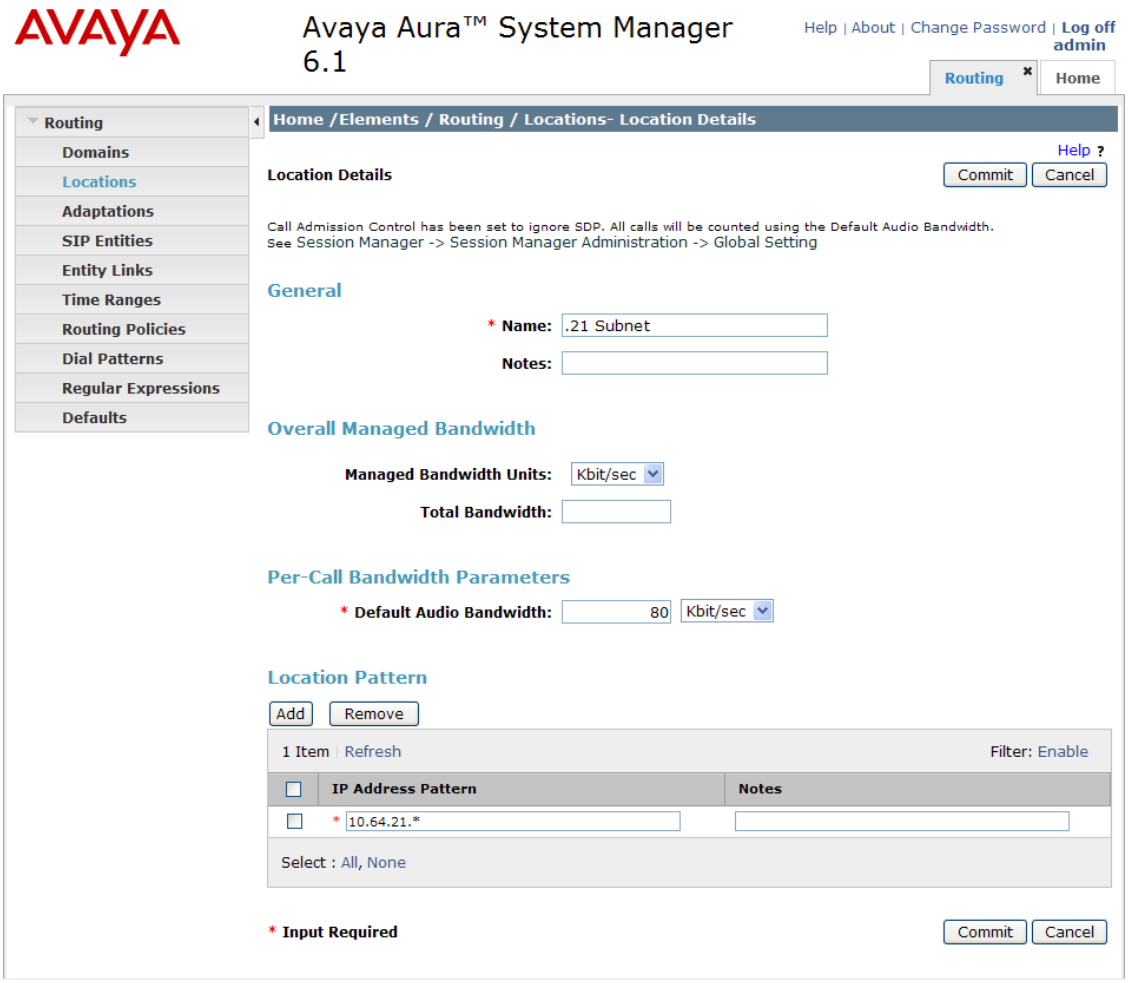
Under **General**:

- **Name**: a descriptive name
- **Notes**: optional descriptive text

Under **Location Pattern**, click the **Add** button to add a new line:

- **IP Address Pattern**: ***10.64.21.****
- **Notes**: optional descriptive text

Click **Commit** to save the configuration.

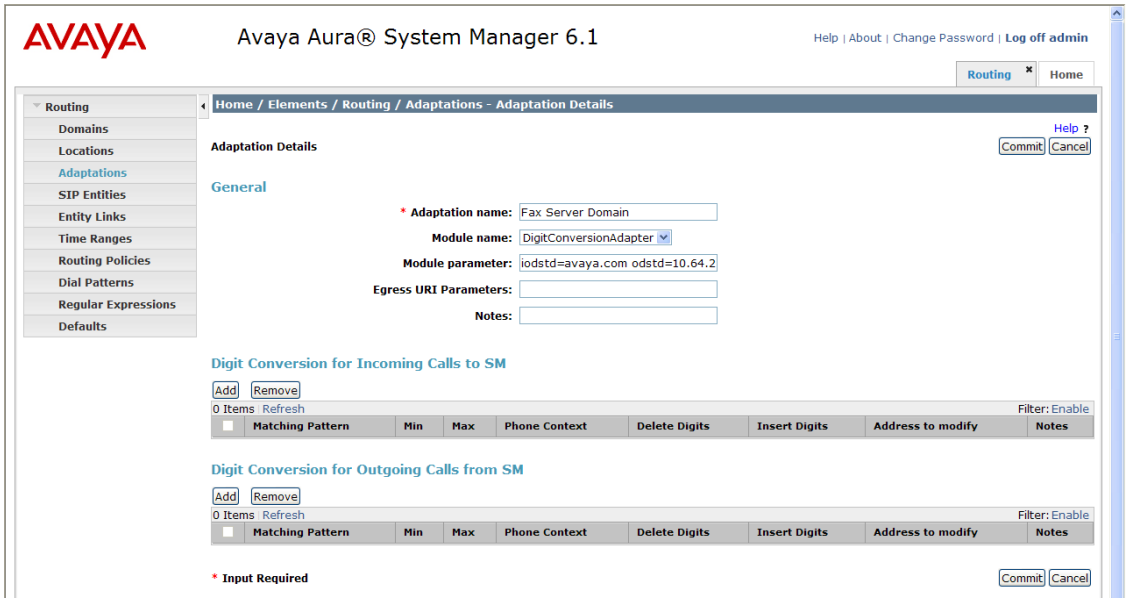


4. **Add Adaptation**

An Adaptation was created and applied to the "Fax Server" SIP entity to override the destination domain as shown below.

The ingressOverrideDestinationDomain (**iodstd**) **Module paramater** replaces the domain in the Request-URI, To Header (if administered), and Notify/message-summary body with the given value (e.g. **avaya.com**) for ingress only.

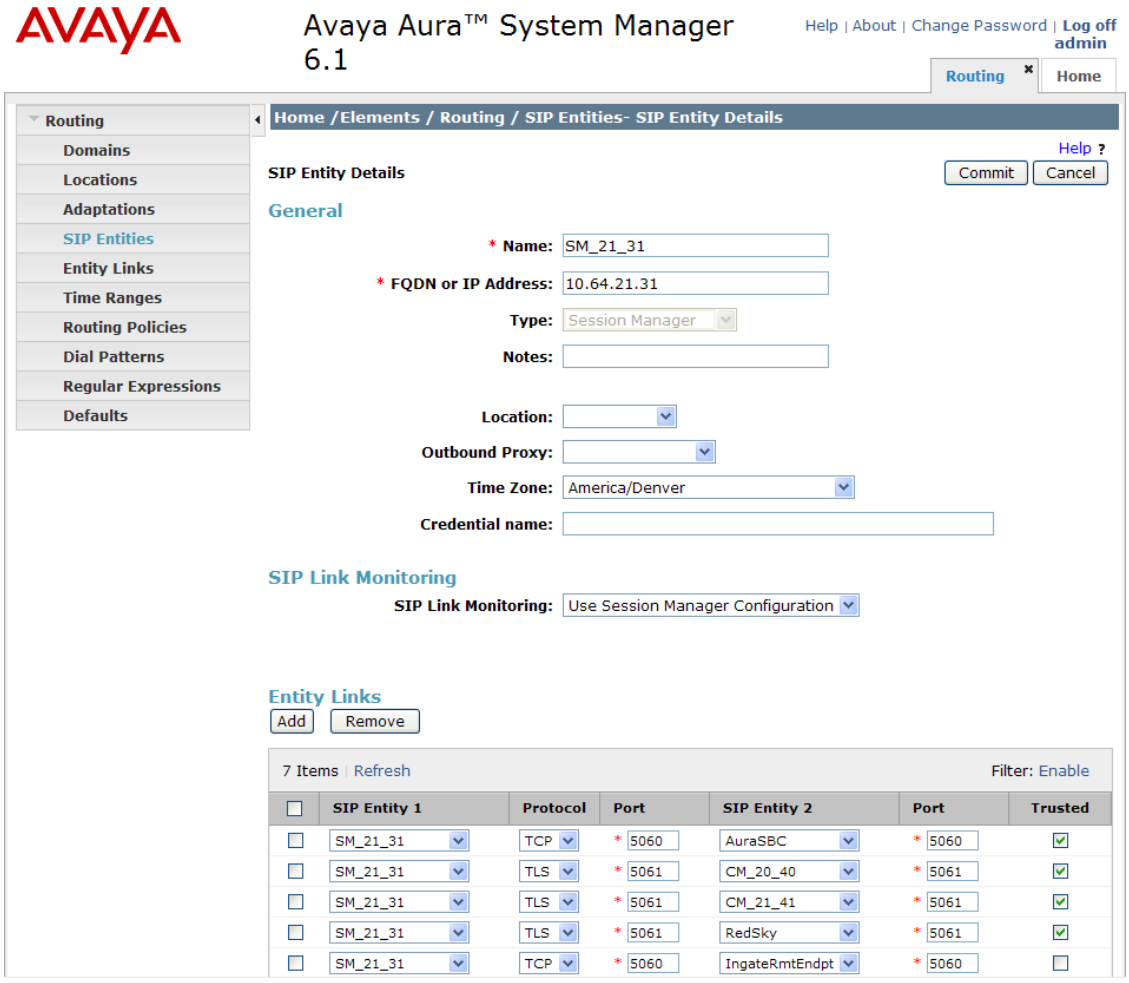
The override DestinationDomain (odstd) **Module paramater** replaces the domain in the Request-URI, To Header (if administered), Refer-To header, and Notify/message-summary body with the given value (e.g. the IP address of the fax server **10.64.21.202**) for egress only.

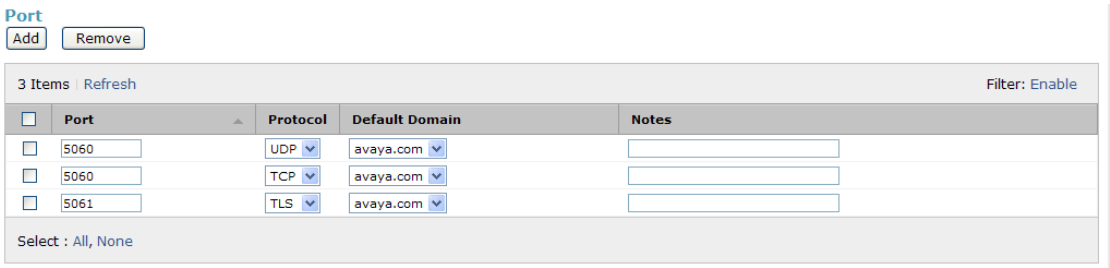


5.	**Add SIP Entities** A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. During compliance testing, a SIP Entity was added for the Session Manager itself, Communication Manager, and the XMediusFAX fax server. Navigate to **Routing→SIP Entities**, and click the **New** button (not shown) to add a SIP Entity. The configuration details for the SIP Entity defined for Session Manager are as follows: Under **General**: • **Name**: a descriptive name • **FQDN or IP Address**: ***10.64.21.31*** as specified in **Figure 1**. This is the IP address assigned to the SM-100 security module installed in the Session Manager. • **Type**: select ***Session Manager*** Under **Port**, click **Add**, then edit the fields in the resulting new row as shown below: • **Port**: ***5061***. This is the port number on which the system listens for SIP requests. • **Protocol**: ***TLS***. The TLS transport protocol was used between Session Manager and Communication Manager. • **Default Domain**: select the SIP Domain created in **Step 2**. • Repeat the three bullets above, but select ***5060*** for **Port** and ***UDP*** for **Protocol**. The UPD protocol was used between Session Manager and the XMediusFAX fax server. Default settings can be used for the remaining fields. Click **Commit** to save the SIP Entity definition.

Add SIP Entities (continued) – Session Manager

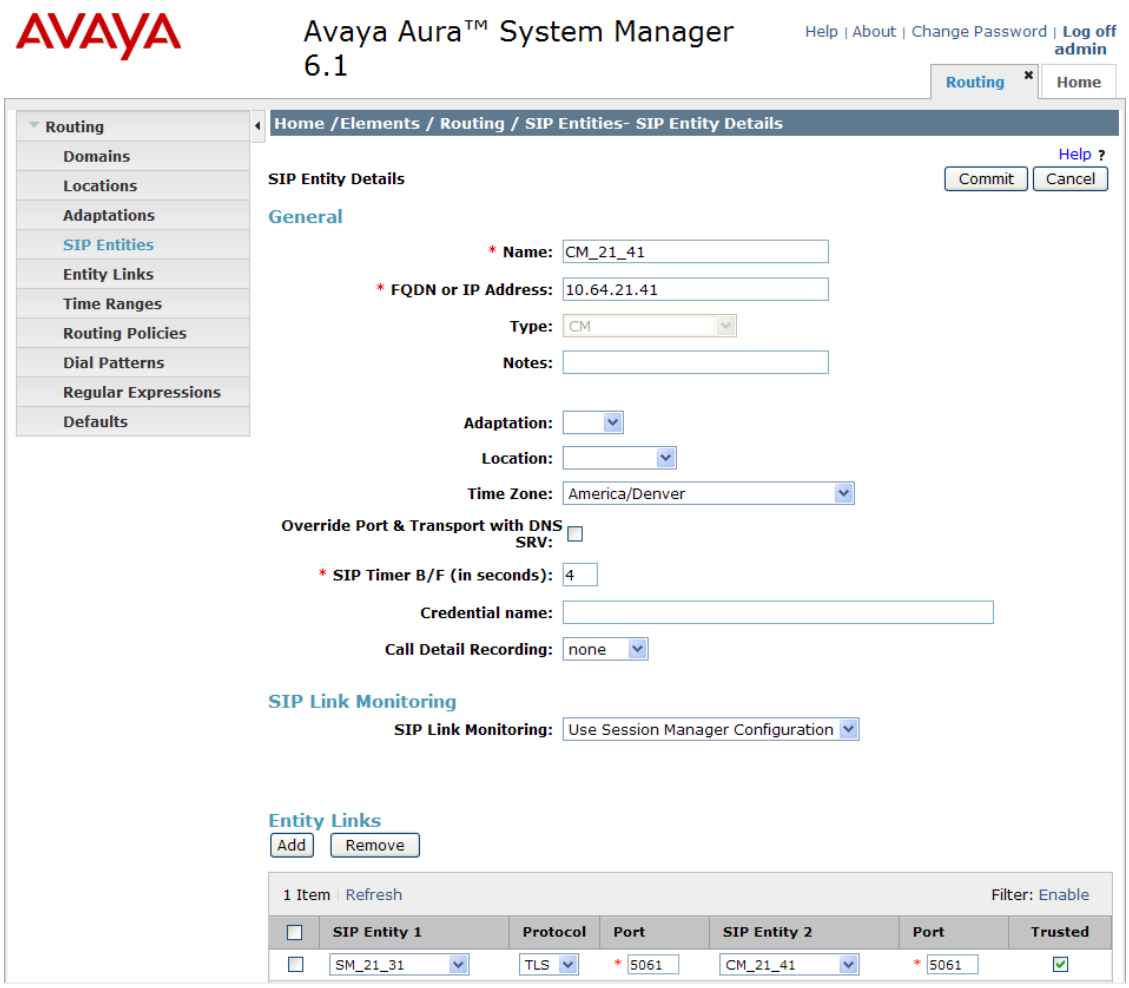
The screens below show the SIP Entity configuration details for the Session Manager.





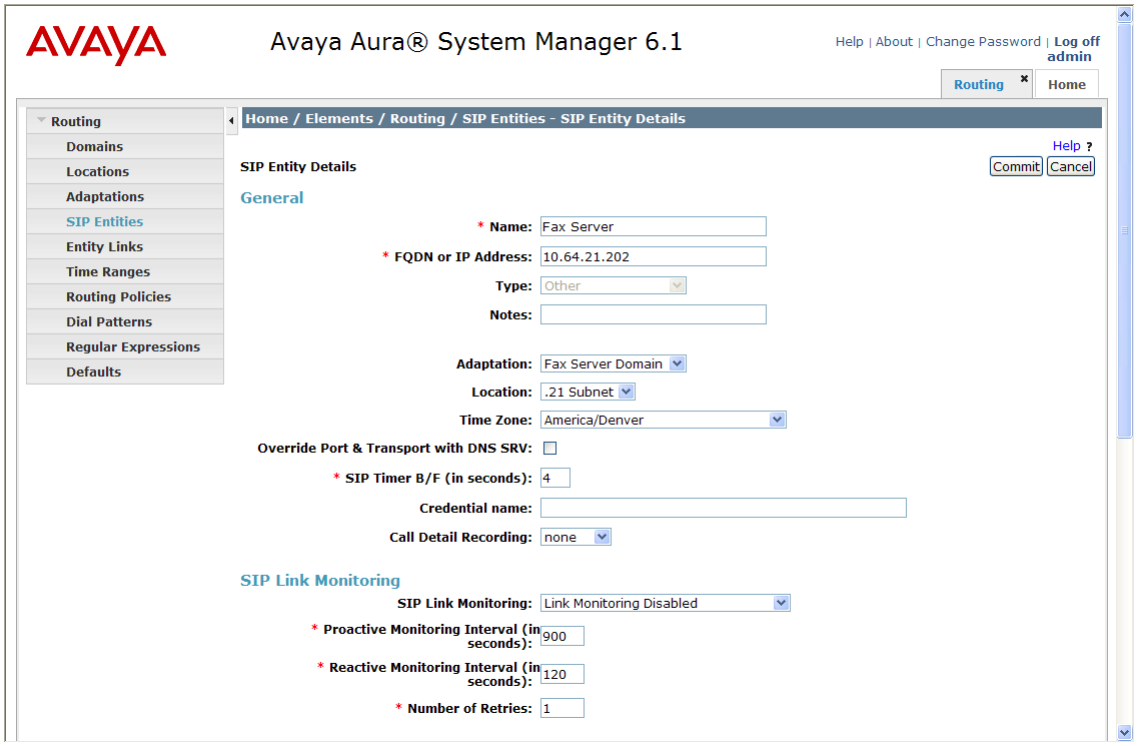
Add SIP Entities (continued) – Communication Manager

The screen below shows the SIP Entity configuration details for the Communication Manager. Note the ***CM*** selection for **Type**.



Add SIP Entities (continued) – XMediusFax

The screen below shows the SIP Entity configuration details for the XMediusFAX fax server. Note the ***Other*** selection for **Type,** and the **Adaptation** created **Step 4** of this section is selected.



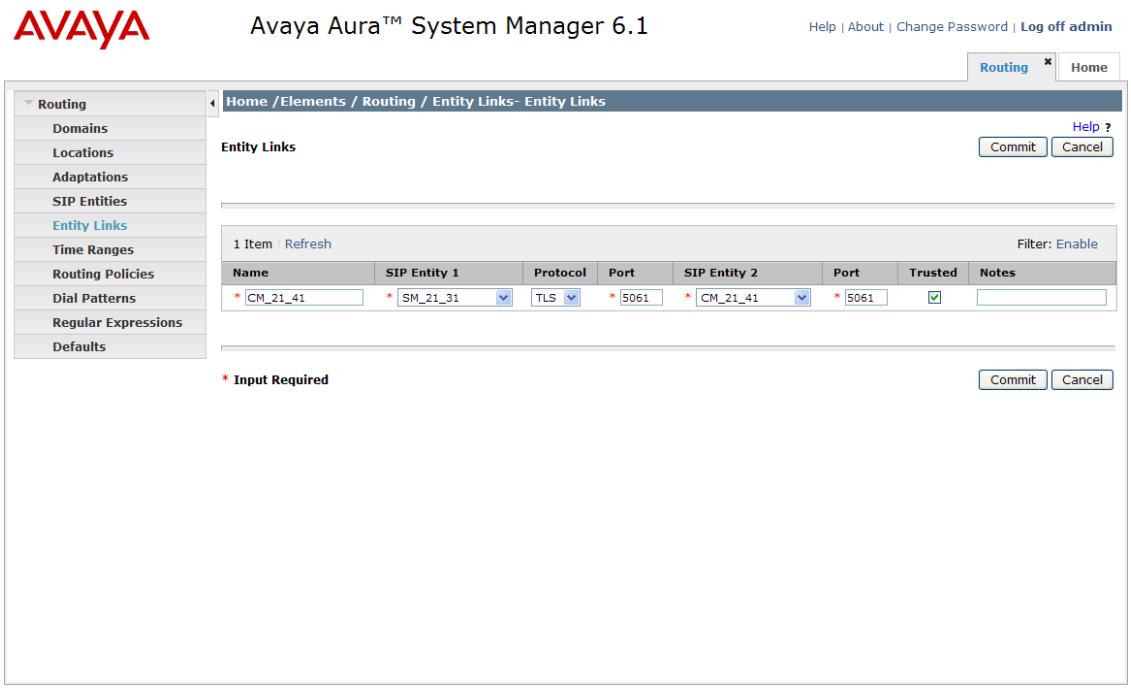
6. **Add Entity Links**

A SIP trunk between Session Manager and a telephony system is described by an Entity link. Two Entity Links were created: one between Session Manager and Communication Manger; the other between Session Manager and the XMediusFAX fax server.

Navigate to **Routing→Entity Links**, and click the **New** button (not shown) to add a new Entity Link. The screen below shows the configuration details for the Entity Link connecting Session Manager to Communication Manager.

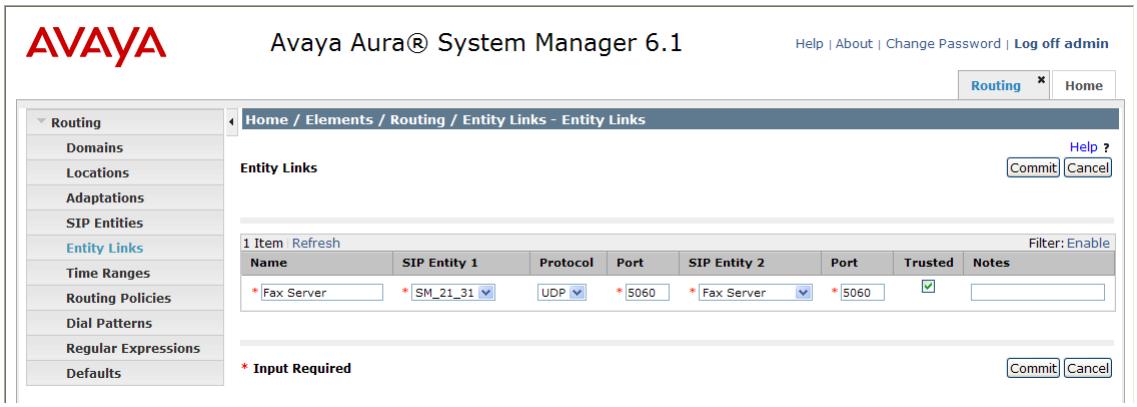
- **Name**: a descriptive name
- **SIP Entity 1**: select the Session Manager SIP Entity.
- **Port**: ***5061***. This is the port number to which the other system sends SIP requests.
- **SIP Entity 2**: select the Communication Manager SIP Entity.
- **Port**: ***5061***. This is the port number on which the other system receives SIP requests.
- **Trusted**: check this box
- **Protocol**: select ***TLS*** as the transport protocol.
- **Notes**: optional descriptive text

Click **Commit** to save the configuration.



Add Entity Links (continued)

The Entity Link for connecting Session Manager to the XMediusFAX fax server was similarly defined as shown in the screen below.



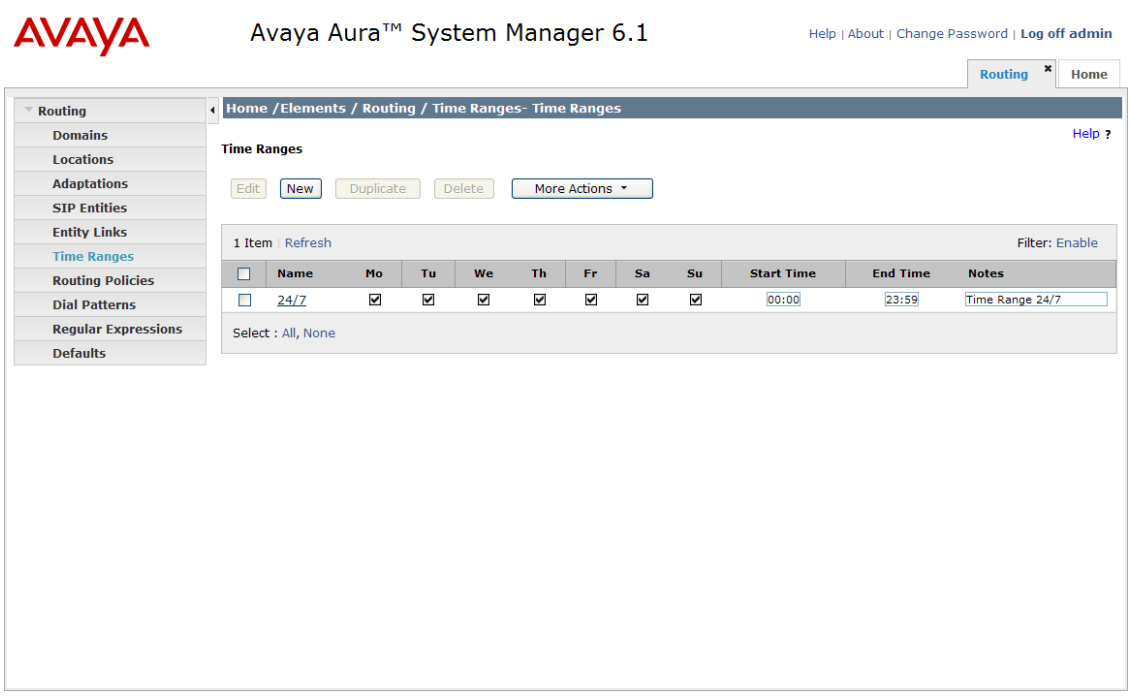
7. **Add Time Ranges**

Before adding routing policies (configured in next step), time ranges must be defined during which the policies will be active. One Time Range was defined that would allow routing to occur at anytime.

Navigate to **Routing→Time Ranges**, and click the **New** button to add a new Time Range:

- **Name**: a descriptive name
- **Mo** through **Su**: check the box under each of these headings
- **Start Time**: enter ***00:00***
- **End Time**: enter ***23:59***

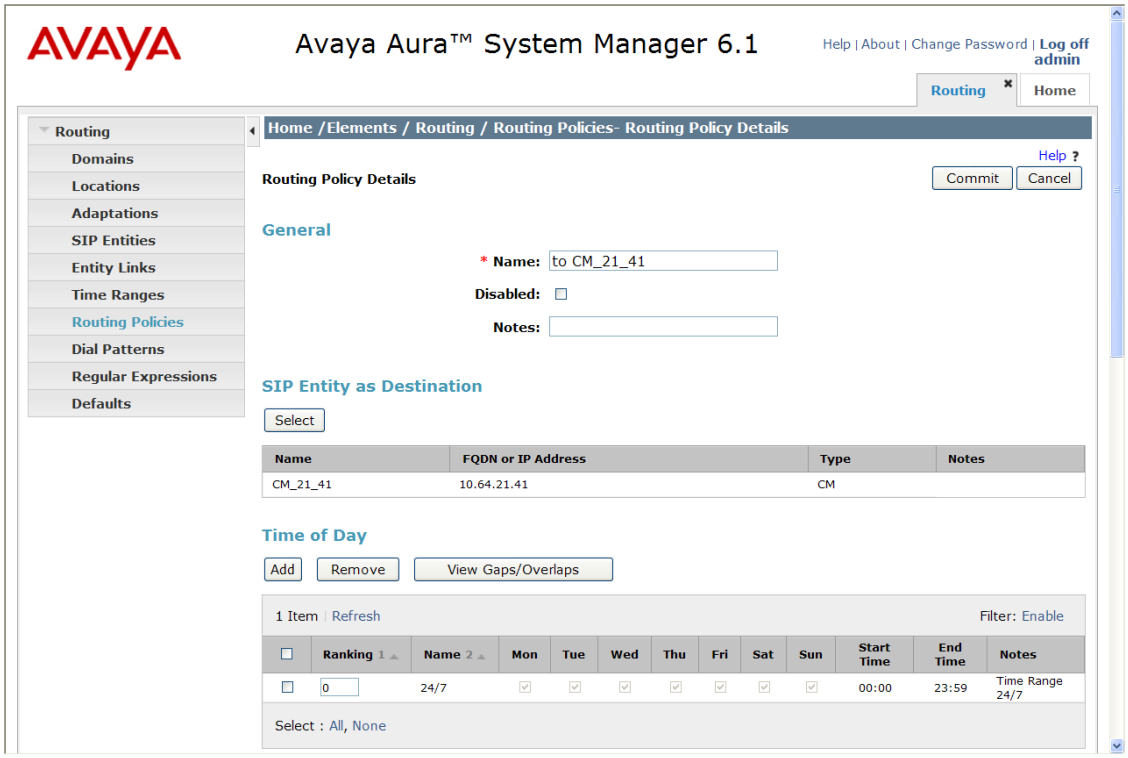
Click **Commit** to save this time range. The screen below shows the configured Time Range.

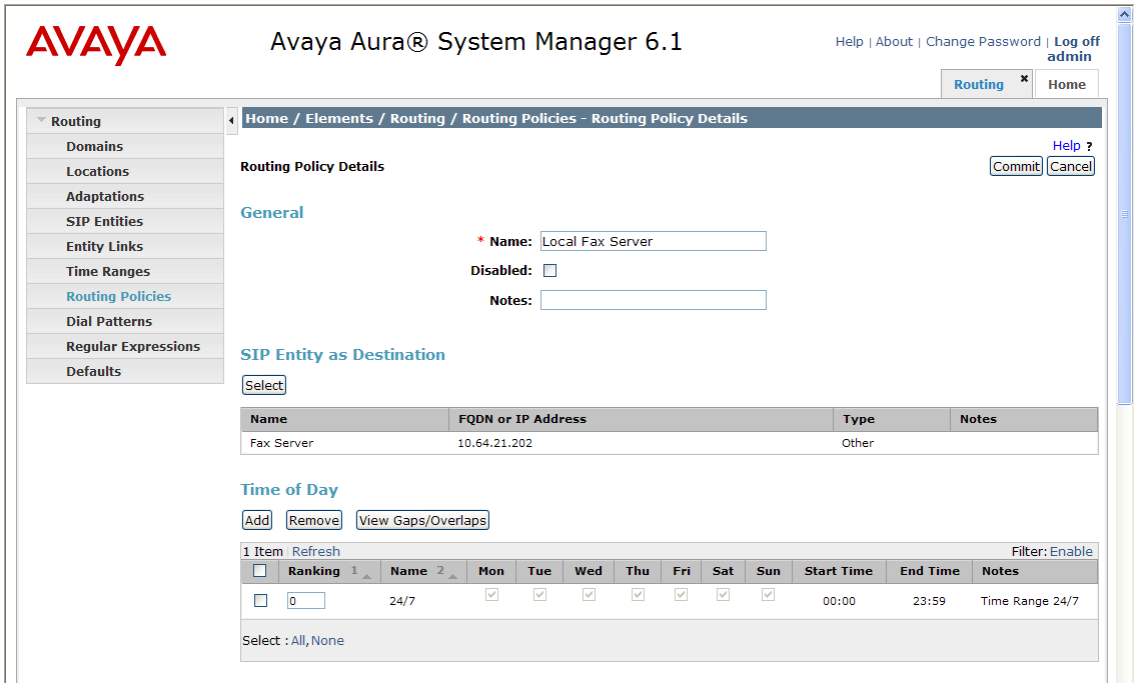


8.	**Add Routing Policies** Routing policies describe the conditions under which calls will be routed to the SIP Entities connected to the Session Manager. Two routing policies were added – one for routing calls to Communication Manager, and the other for routing calls to the XMediusFAX fax server. Navigate to **Routing→Routing Policies**, and click the **New** button (not shown) to add a new Routing Policy. Under **General**: • **Name**: a descriptive name • **Notes**: optional descriptive text Under **SIP Entity as Destination** Click **Select** to select the appropriate SIP Entity to which the routing policy applies (not shown). Under **Time of Day** Click **Add** to select the Time Range configured in the previous step (not shown). Default settings can be used for the remaining fields. Click **Commit** to save the configuration.

Add Routing Policies (continued)

The screens below show the configuration details for the two Routing Policies used during compliance testing.

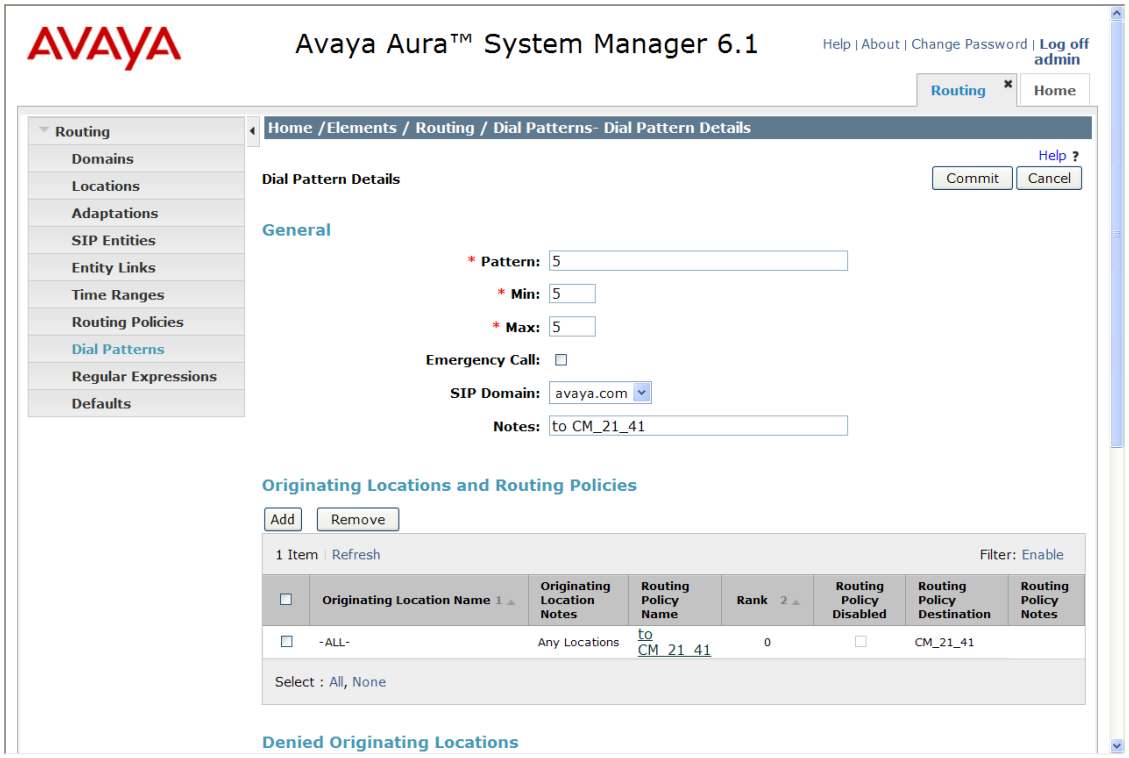


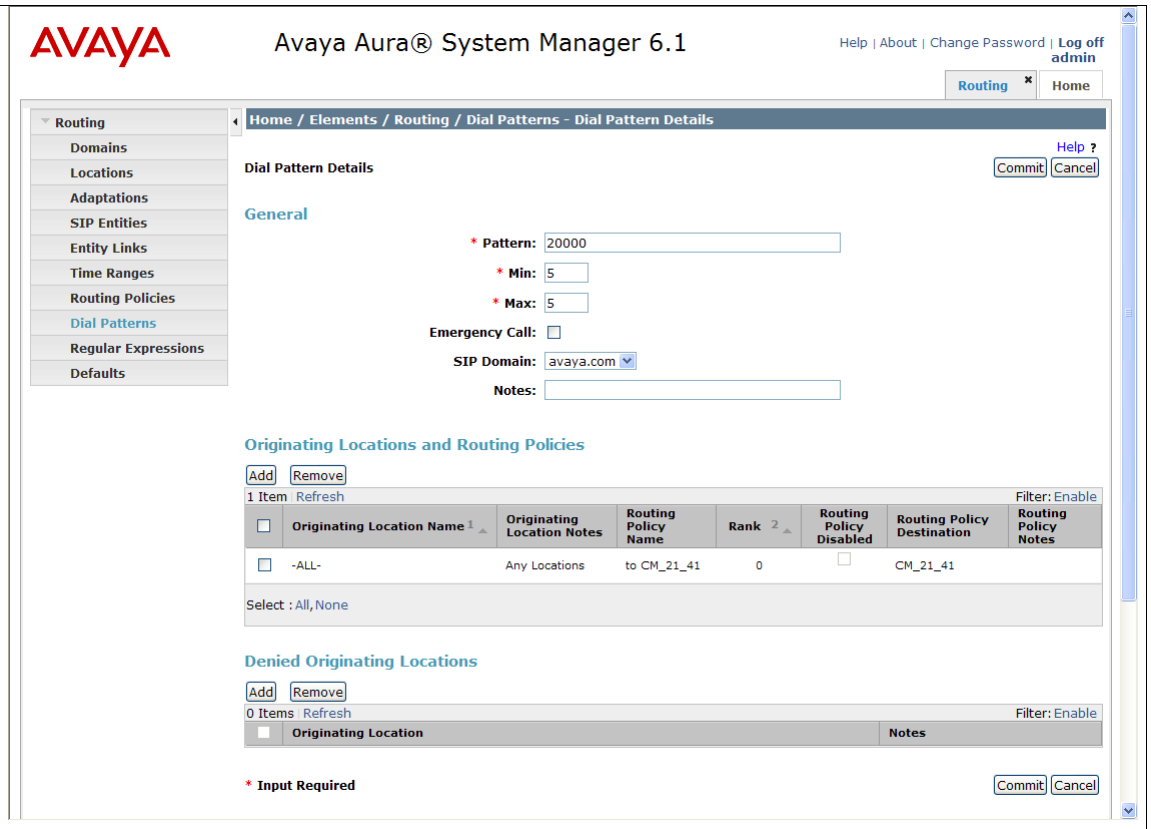


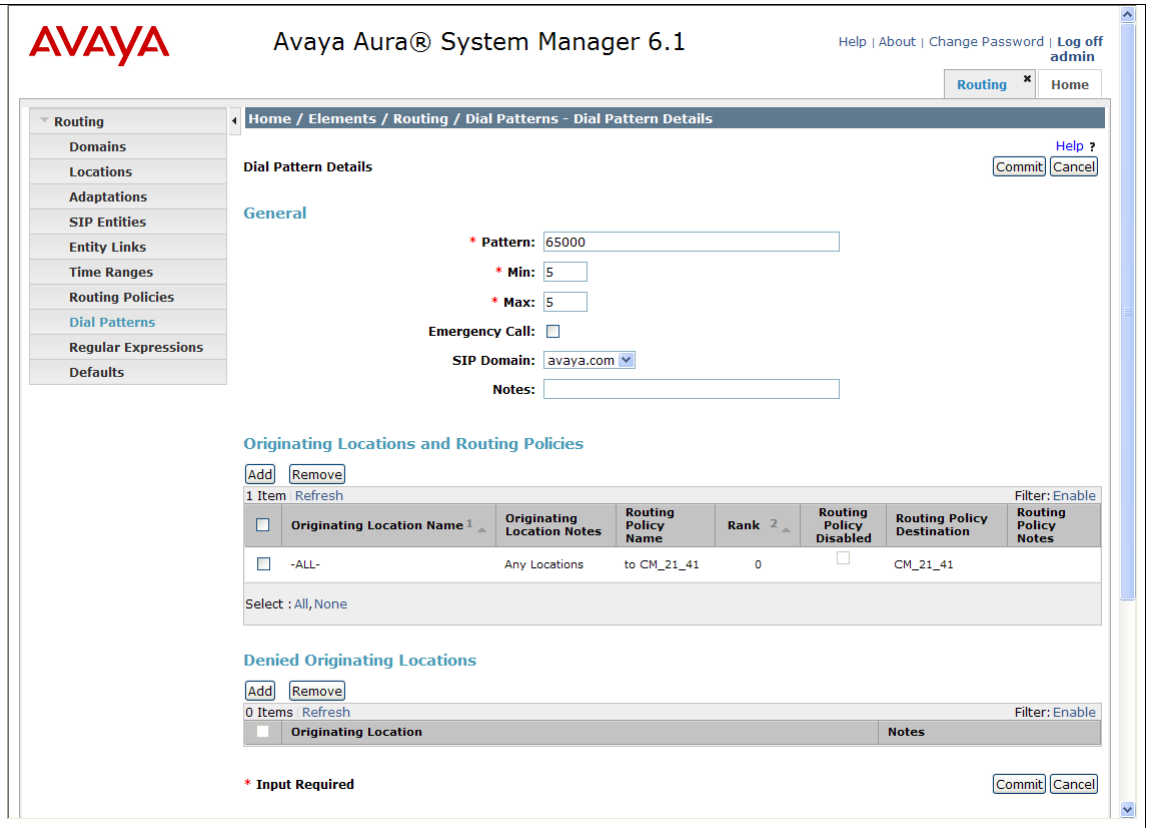
9.	**Add Dial Patterns** Dial Patterns define digit strings to be matched against dialed numbers for directing calls to the appropriate SIP Entities. 5-digit extensions beginning with "5" resided on Communication Manager at Site 2. 5-digit extensions matching "20000" or "65000" were routed to the local Communication Manager for onward routing to Site 1. 5-digit extensions beginning with "75" were routed to the XMediusFAX fax server. Therefore 4 Dial Patterns were created accordingly. Navigate to **Routing→Dial Patterns**, click the **New** button (not shown) to add a new Dial Pattern. Under **General**: • **Pattern**: dialed number or prefix • **Min**: minimum length of dialed number • **Max**: maximum length of dialed number • **SIP Domain**: select the SIP Domain created in **Step 2** (or select **–ALL–** to be less restrictive) • **Notes**: optional descriptive text Under **Originating Locations and Routing Policies** Click **Add** to select the appropriate originating Location and Routing Policy from the list (not shown). Under **Time of Day** Click **Add** to select the time range configured in **Step 7**. Default settings can be used for the remaining fields. Click **Commit** to save the configuration.

Add Dial Patterns (continued)

The screens below shows the configuration details for the Dialed Patterns defined for routing calls to Communication Manager at the main enterprise site.

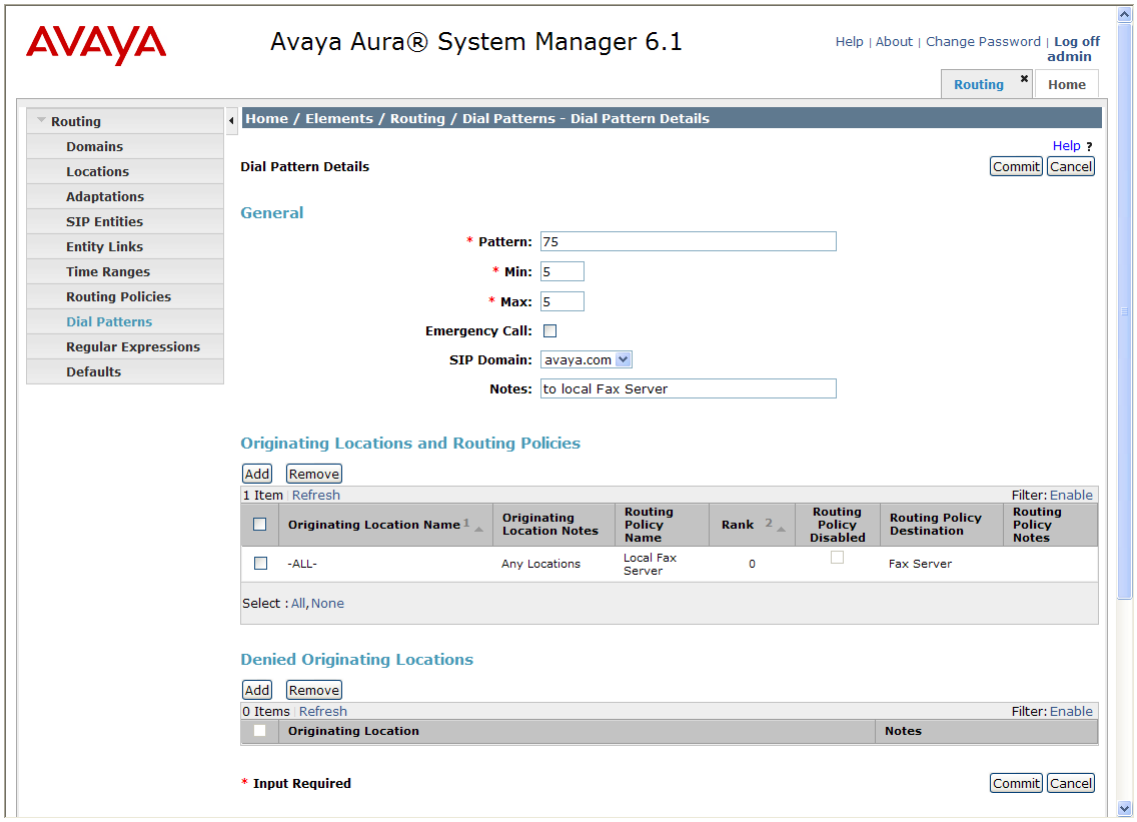






Add Dial Patterns (continued)

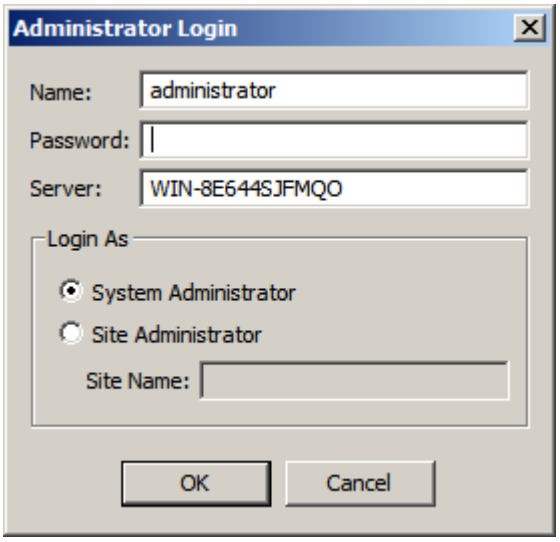
The screen below shows the configuration details for the Dialed Pattern defined for routing calls to the XMediusFAX fax server.

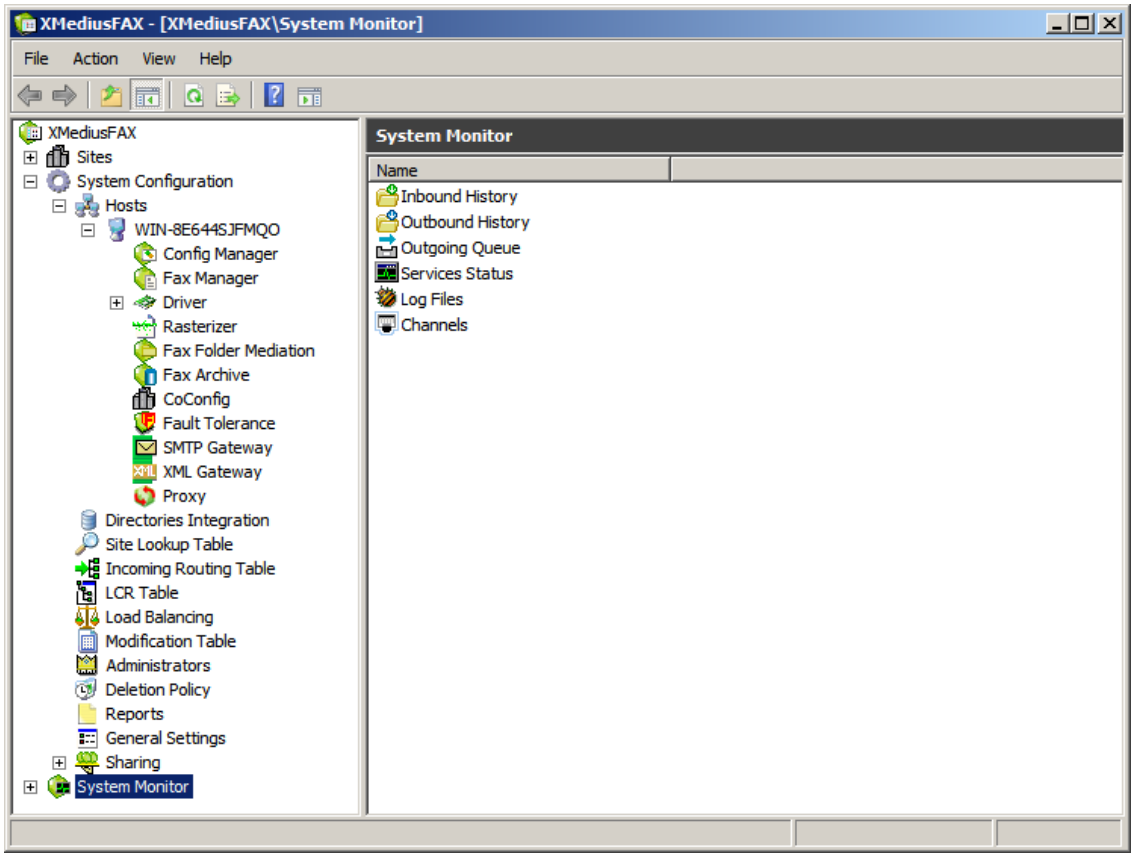


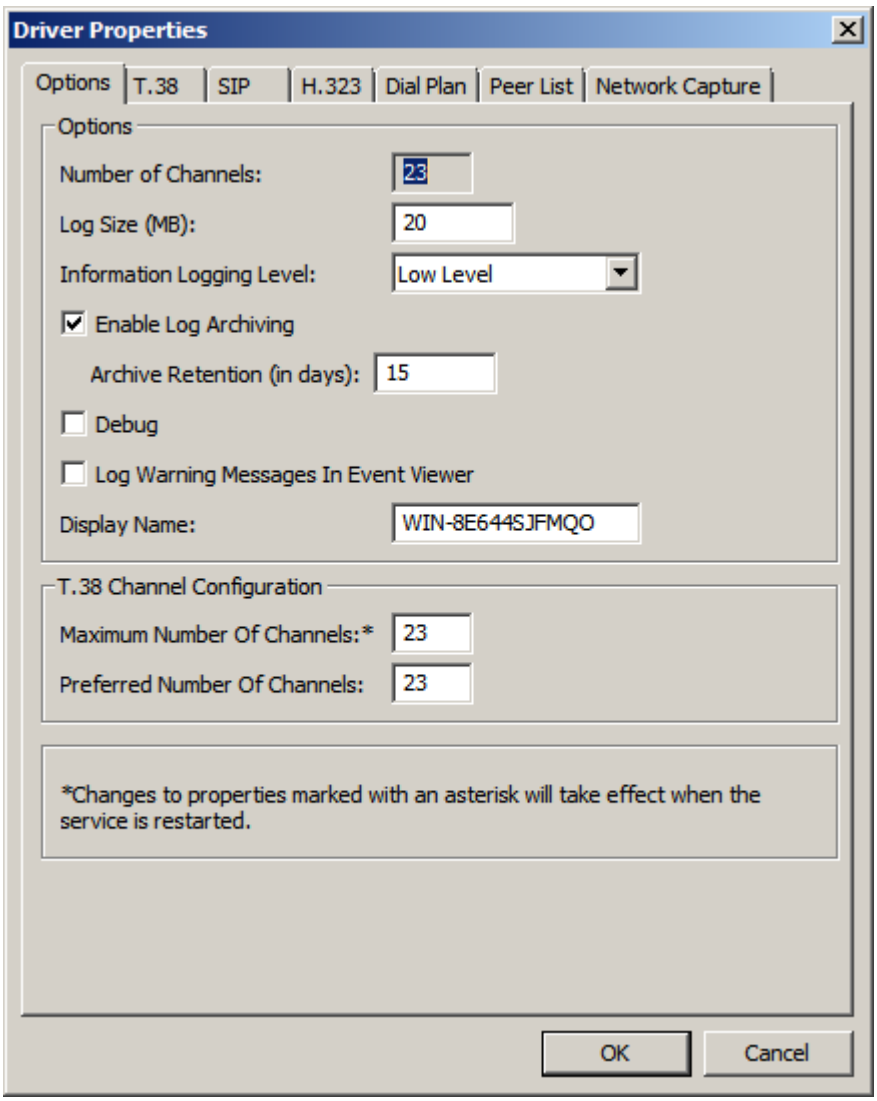
7. Configure Sagemcom XMediusFAX

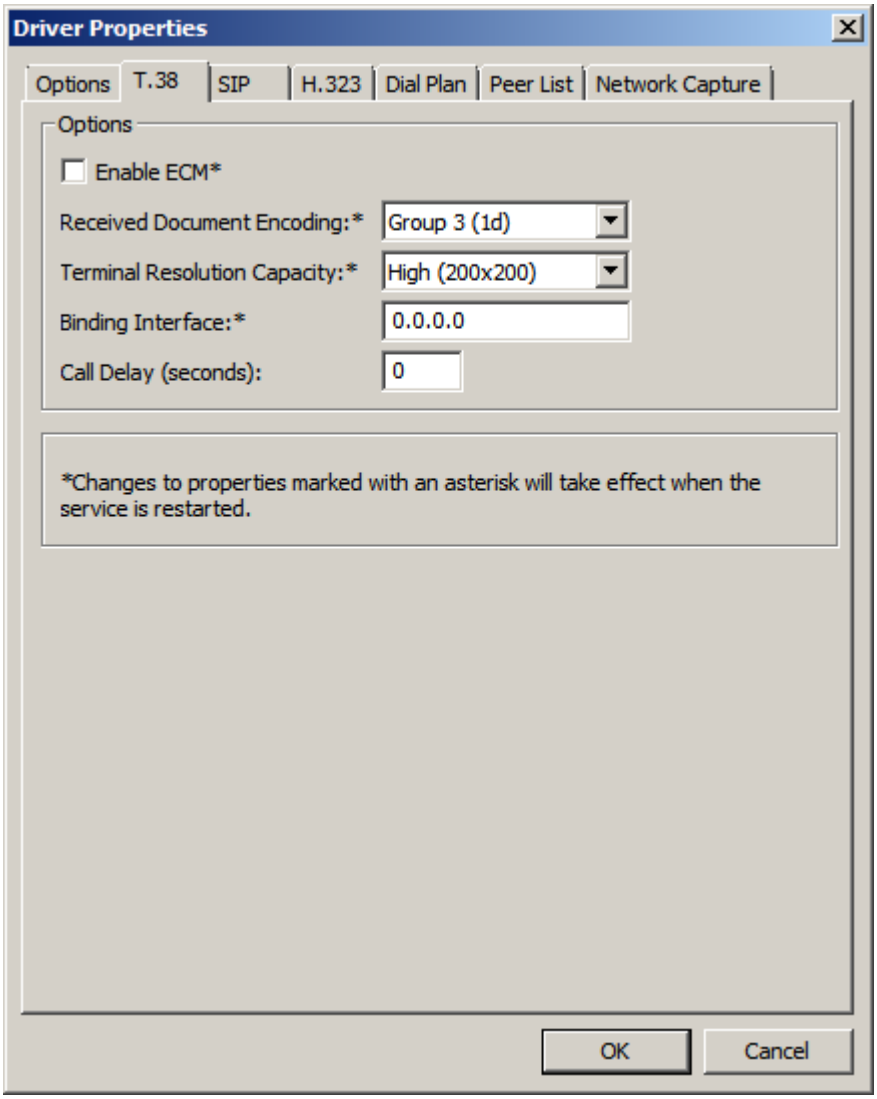
This section describes the configuration of XMediusFAX. It assumes that the application and all required software components have been installed and properly licensed. The number of channels supported by the XMediusFAX server is controlled via an XMediusFAX server license file. For instructions on sending and receiving faxes, consult the XMediusFAX Administrator Guide [5] and User Guide [7].

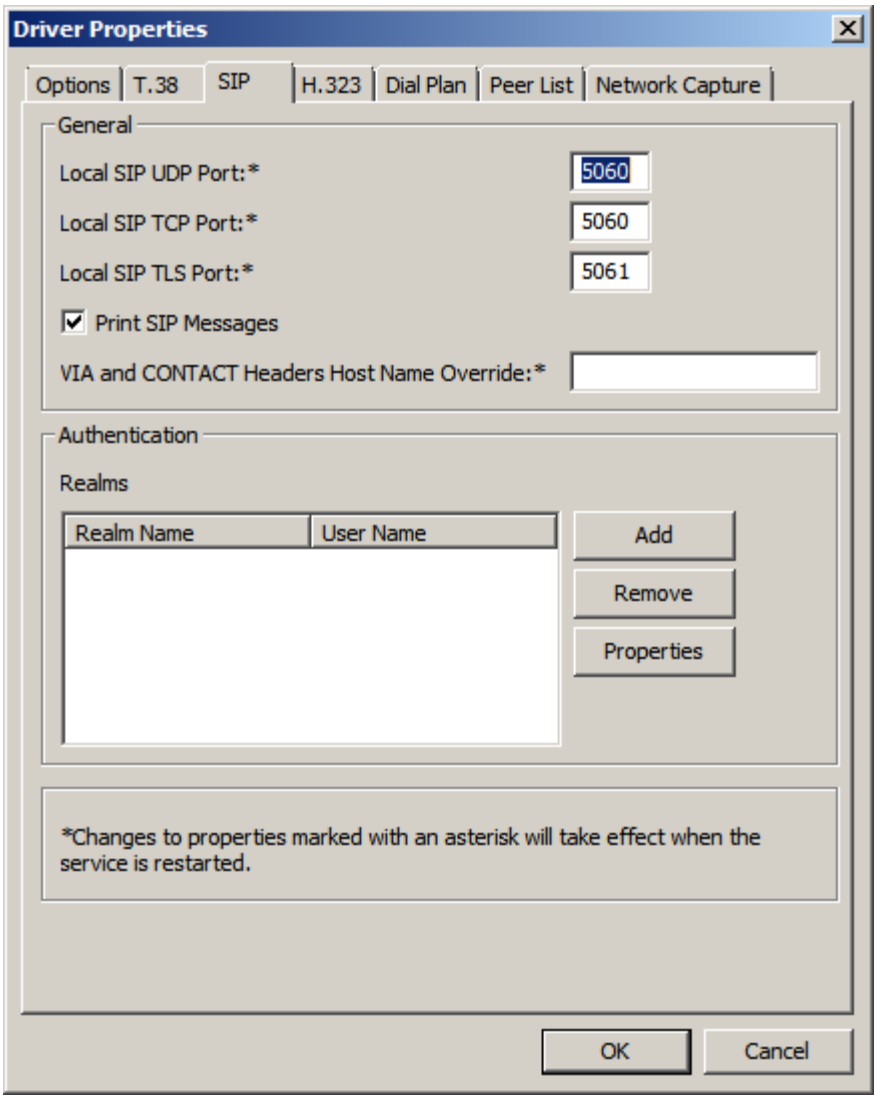
The examples shown in this section refer to Site 2. Unless specified otherwise, the same steps also apply to Site 1 using values appropriate for Site 1 from **Figure 1**.

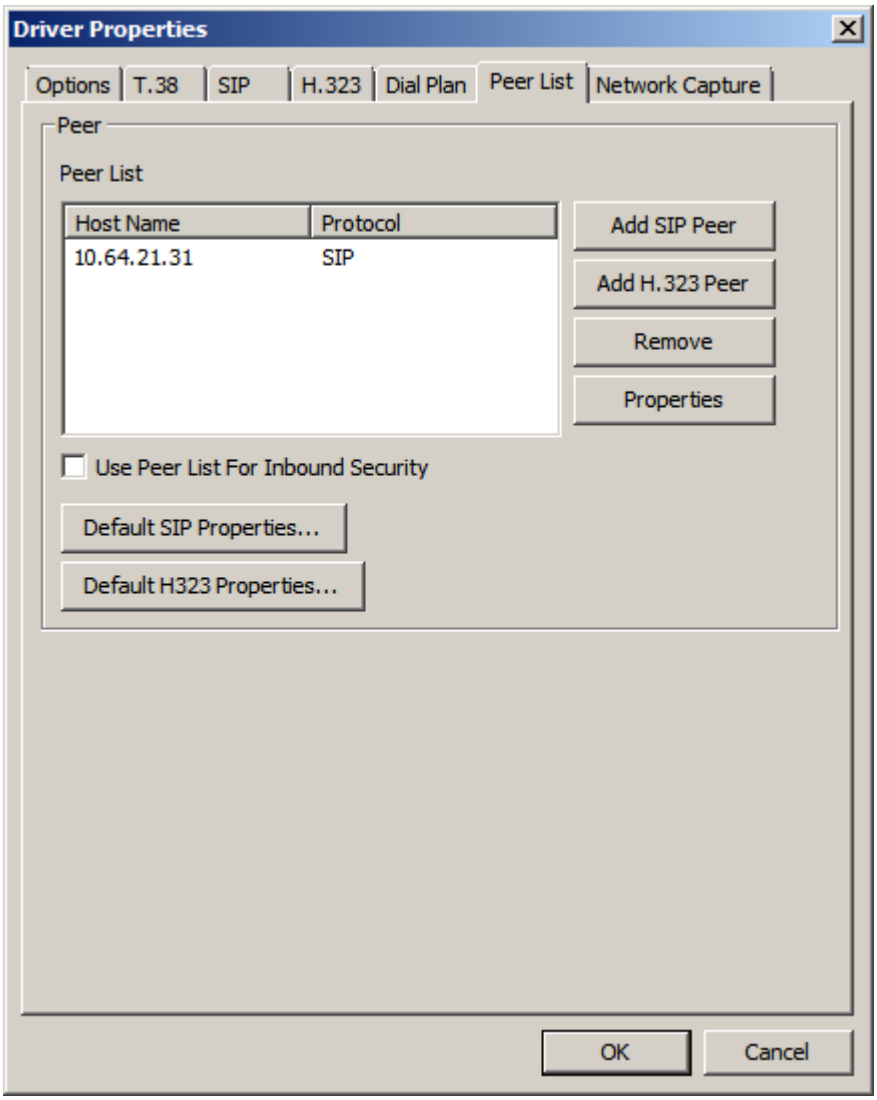
Step	Description
1.	**Prepare the fax server for launching the XMediusFAX software** Consult Sagemcom for requirements and instructions.
2.	**Launch the Application** On the XMediusFAX server, launch the XMediusFAX application from the Windows Start Menu. Navigate to **Start → All Programs → XMediusFAX → XMediusFAX**. A login screen appears. Log in with proper credentials. Click the **OK** button. 

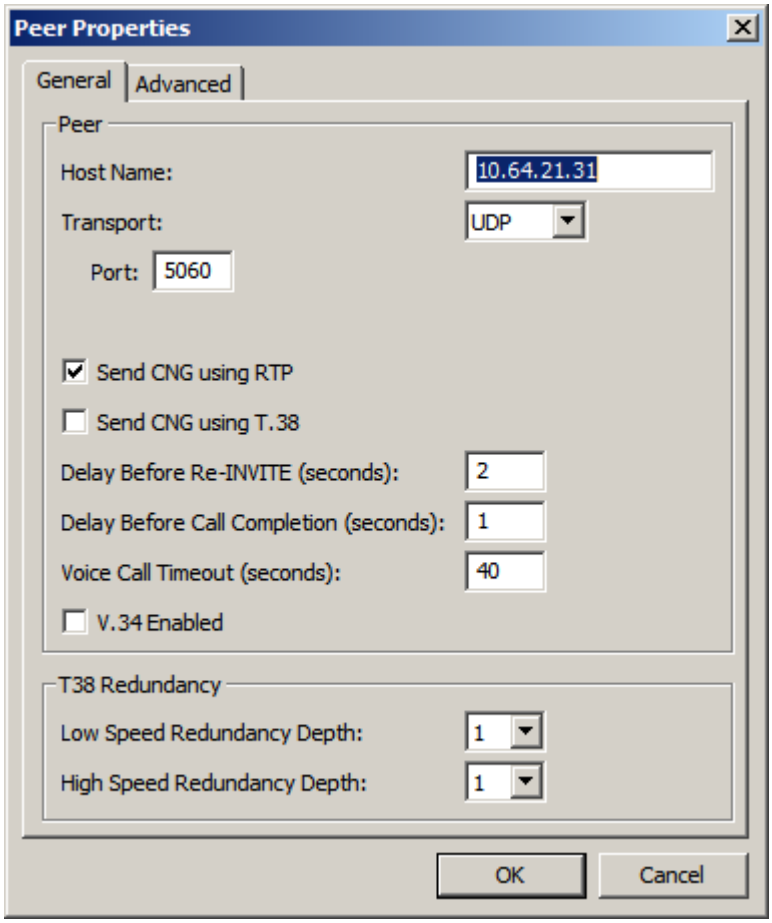
Step	Description
3.	**Configure Driver Properties** On the main screen, navigate to **XMediusFAX → System Configuration → Hosts → WIN-8E644SJFMQO → Driver** in the left hand tree menu. Right-click on **Driver** and select **Properties** (not shown). 

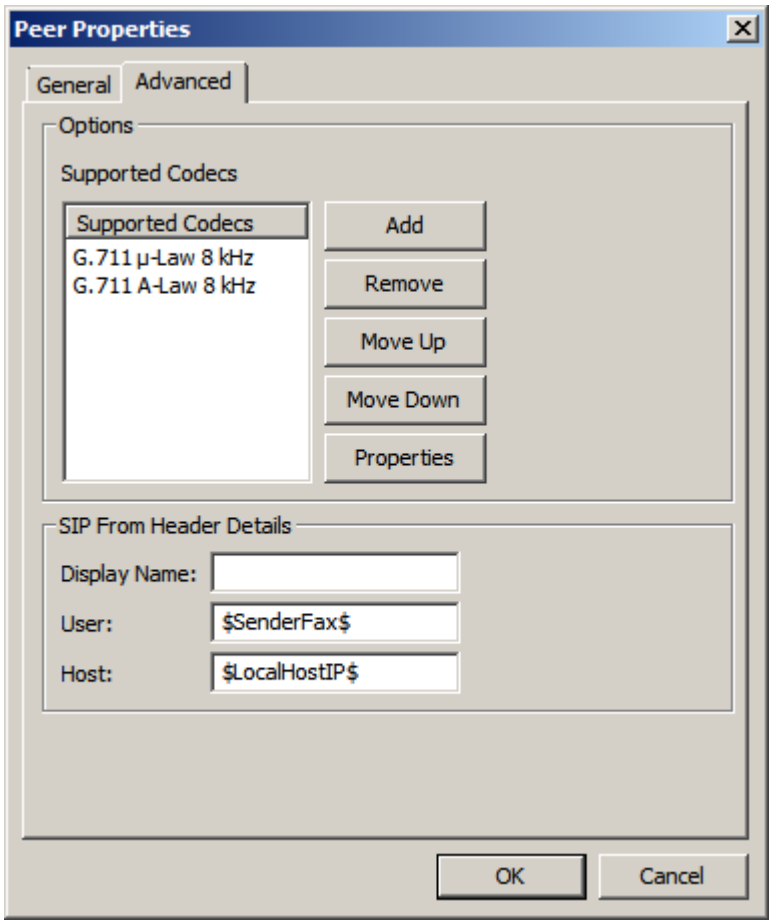
Step	Description
4.	**General Options** On the **Driver Properties** screen, select the **Options** tab. Set the **Maximum Number Of Channels** and **Preferred Number Of Channels** fields under **T.38 Channel Configuration** to the number of simultaneous faxes to be processed. 

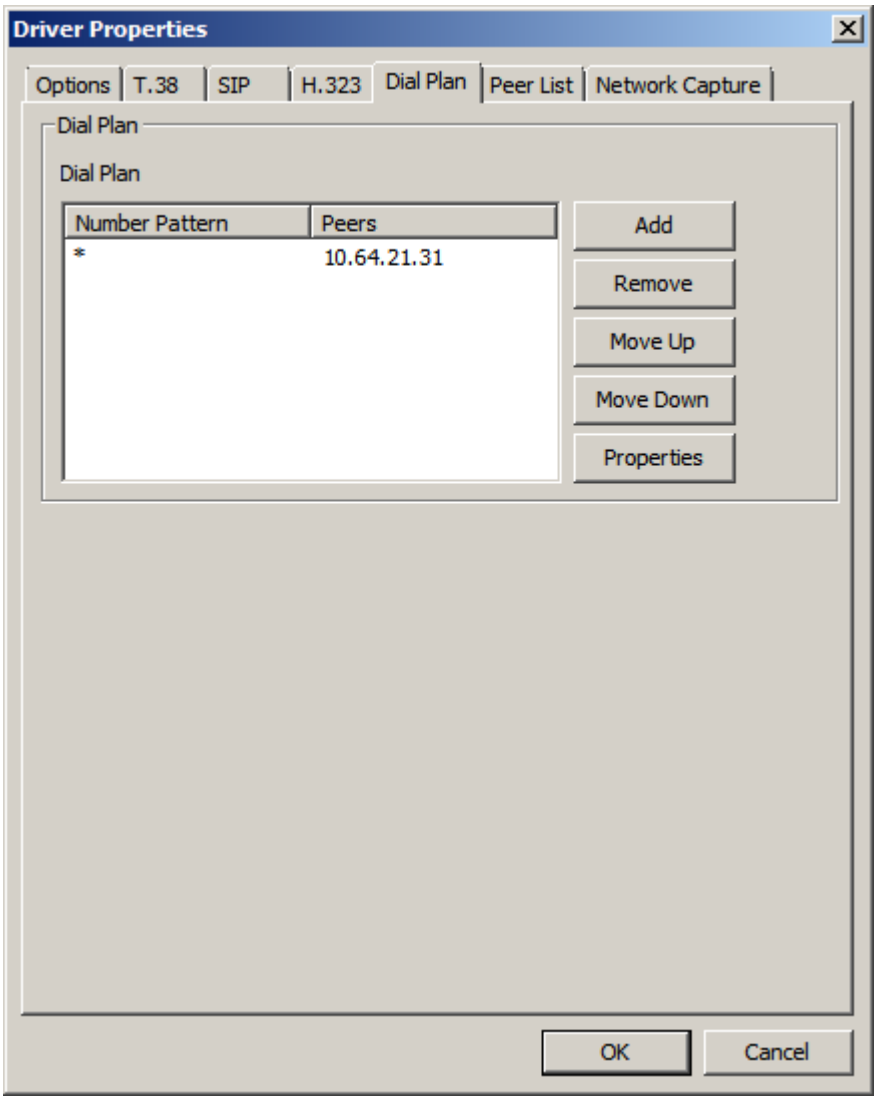
Step	Description
5.	**T.38 Parameters** On the **Driver Properties** screen, select the **T.38** tab. Configure the fields as follows: • **Received Document Encoding** – Set this field to the highest encoding allowed. For the compliance test, this value was set to ***Group 3 (1d)***. • **Terminal Resolution Capacity** – Set this field to the highest resolution allowed desired. For the compliance test, this value was set to ***High (200x200)***. 

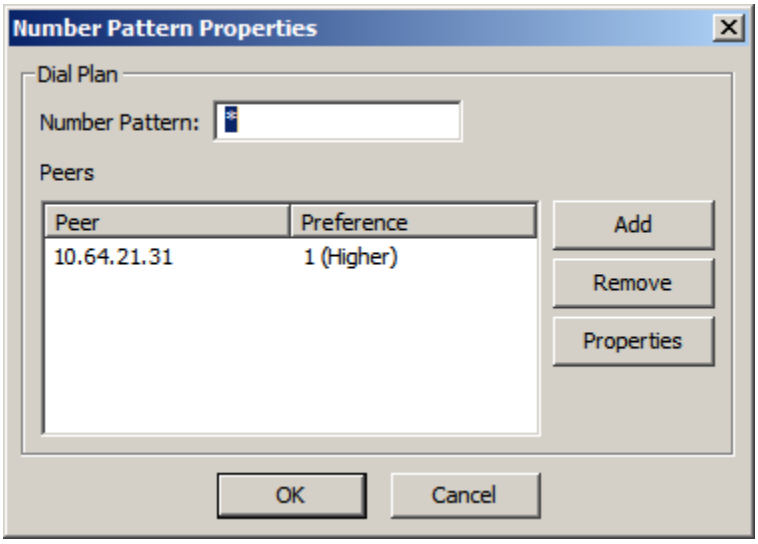
Step	Description
6.	**SIP Parameters** On the **Driver Properties** screen, select the **SIP** tab. Configure the fields as follows: • **Local SIP UDP port** – Set this field to match the first **Port** field in **Section 6**, **Step 6**. During compliance testing, UDP was used as the transport layer protocol by the XMediusFAX fax server. 

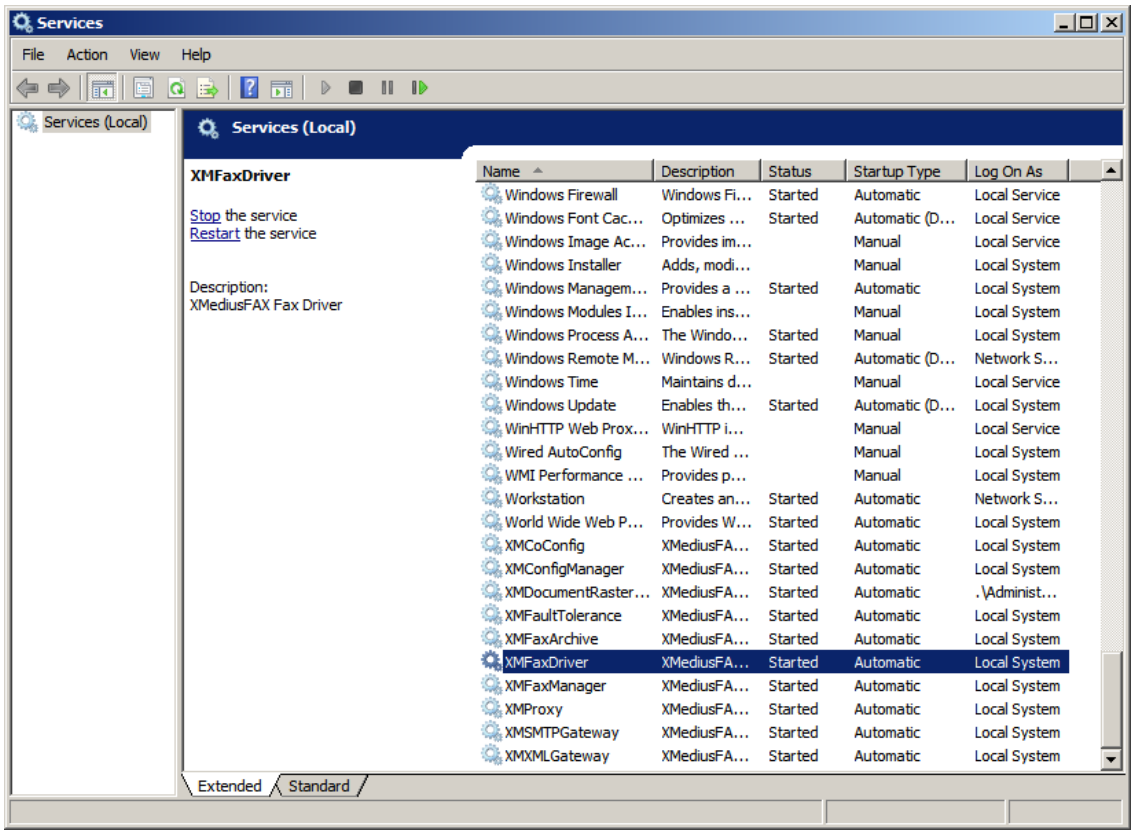
Step	Description
7.	**Peer List** On the **Driver Properties** screen, select the **Peer List** tab. To add a new SIP peer, select the **Add SIP Peer** button and enter the values shown in **Step 8**. To view an existing peer, highlight the peer in the list and click **Properties**. The example below shows the peer list after the Session Manager interface, ***10.64.21.31***, has been added to the list. 

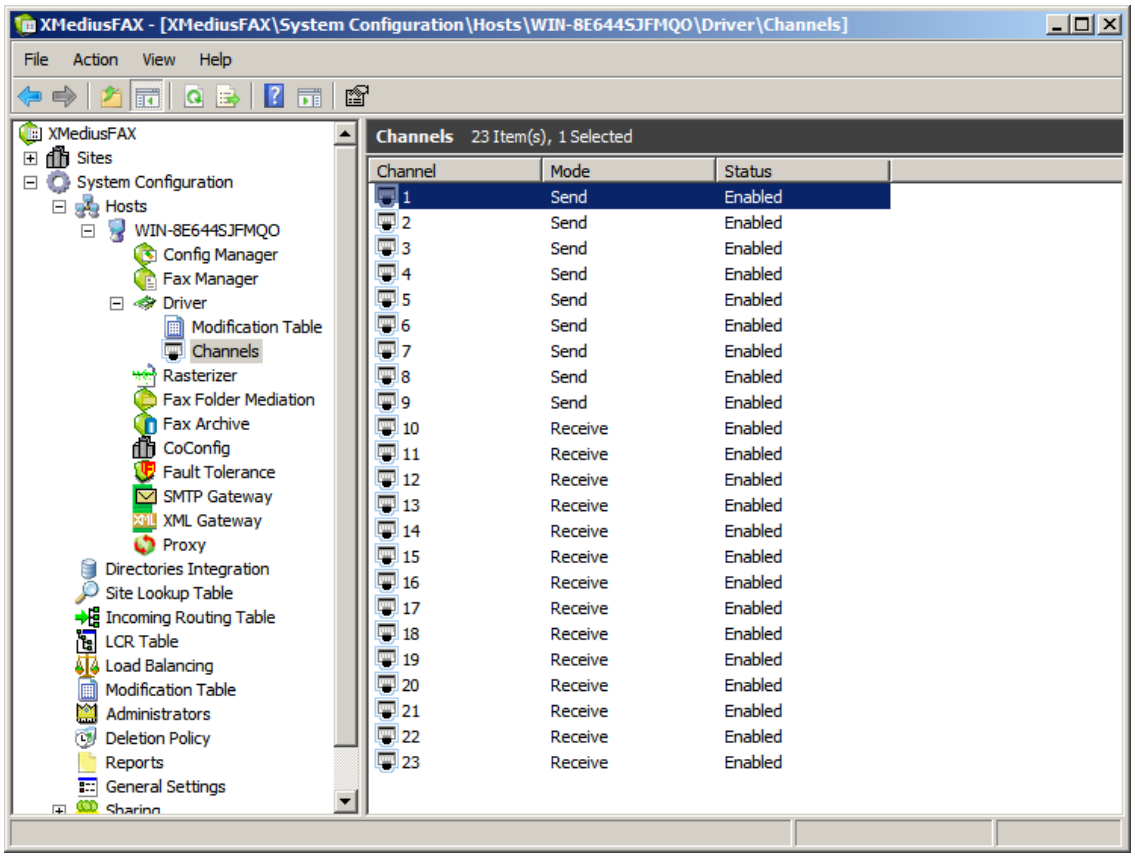
Step	Description
8.	**Peer Properties** On the **Peer Properties** screen, configure as follows: • **Host Name** – Set this field to the IP address of Session Manager. • **Transport:** Set this field to ***UDP***. During compliance testing, UDP was used as the transport layer protocol by the XMediusFAX fax server. • **Port** - Set this field to ***5060***. • Check the **Send CNG using RTP** field. 

Step	Description
9.	**Codec** On the **Peer Properties** screen, select the **Advanced** tab. To add a codec for the SIP peer, select the **Add** button and select the values from the drop-down menu. To view an existing codec, highlight the codec in the list and click **Properties**. The example below shows the codec list supported by the newly added SIP peer. 

Step	Description
10.	**Dial Plan** On the **Driver Properties** screen, select the **Dial Plan** tab. To add a new entry to the dial plan, select the **Add** button and enter the values shown in **Step 11**. To view an existing entry, highlight the entry in the list and click **Properties** to get the **Number Pattern Properties** screen. The example below shows the dial plan after the entry for ***** (any value) has been added to the list. 

Step	Description
11.	**Number Pattern Properties** On the **Number Pattern Properties** screen, configure as follows: • **Number Pattern** – Set this field to the pattern to match. In this example, the value of ***** indicates any dialed number is acceptable. • **Peer** – Click the **Add** button. In the **Peer Properties** window that appears (not shown), enter the **Peer** IP Address and **Preference** value of ***1*** and click **OK**. In this example, only one peer is configured.  Lastly, click OK on the **Driver Properties** screen shown in **Step 10**, to accept the Driver Configuration.

Step	Description
12.	Once all the driver properties have been configured, go to **Start → Control Panel → Administrative Tools → Services** to stop and start the **XMFaxDriver** service to make the changes take effect. 

Step	Description
13.	**Configure Channels** On the main screen, navigate to **XMediusFAX → System Configuration → Hosts → WIN-8E644SJFMQO → Driver → Channels** in the left hand tree menu. Right-click on each channel in the right pane to set the **Mode** to ***Send***, ***Receive*** or ***Both***. During compliance testing, 9 channels were set to ***Send*** and 14 channels were set to ***Receive***.  s

8. Verification Steps

The following steps may be used to verify the configuration:

- Using System Manager, navigate to **Session Manager→System Status→SIP Entity Monitoring**, and click on the appropriate SIP Entities to verify that the Entity Link to Communication Manager is up.
- From the Communication Manager SAT, use the **status signaling-group *x*** command to verify that the SIP signaling group is in-service (where ***x*** is the signaling group number associated with the trunk between Communication Manager and Session Manager).
- From the Communication Manager SAT, use the **status trunk-group *y*** command to verify that the SIP trunk group is in-service (where ***y*** is the trunk group number for the trunk between Communication Manager and Session Manager).
- Verify that fax calls can be placed to/from the XMediusFAX fax server at each site.
- From the Avaya Communication Manager SAT, use the **list trace tac** command to verify that fax calls are routed over the expected trunks.

9. Conclusion

Sagemcom XMediusFAX passed compliance testing. These Application Notes describe the procedures required to configure Sagemcom XMediusFAX to interoperate with Session Manager and Communication Manager to support the network shown in **Figure 1**.

10. Additional References

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

[1] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Doc # 555-245-205, August 2010.
[2] *Administering Avaya Aura™ Communication Manager*, Doc # 03-300509, August 2010.
[3] *Administering Avaya Aura® Session Manager*, Doc # 03-603324, May 2011.
[4] *Installing and Configuring Avaya Aura® Session Manager*, Doc # 03-6034723, April 2011.

Product documentation for XMediusFAX 6.5.5 may be may be obtained from Sagemcom.

[5] *Sagemcom XMediusFAX Administrator Guide,* September 2010
[6] *Sagemcom XMediusFAX Installation and Maintenance Guide*, September 2010
[7] *Sagemcom XMediusFAX User Guide*, September 2010

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.