Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking using DeltaCom SIP Trunk Service with Avaya IP Telephony Solution - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Avaya Aura™ Communication Manager with Avaya Aura™ SIP Enablement Services co-resident on Communication Manager server. It includes the steps involved to configure Session Initiation Protocol trunking between DeltaCom's SIP Trunk Service and an Avaya IP Telephony Solution.

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

The objective of this compliance test is to verify that DeltaCom's Nortel CS2000 can interoperate with Avaya Aura™ Communication Manager 5.2 with Avaya Aura™ SIP Enablement Services co-resident in a SIP trunking environment. Communication Manager with SIP Enablement Services co-resident solution places an outbound call through a SIP trunk via a dedicated Internet connection terminating the call in DeltaCom's Nortel CS2000 with Acme Packet Session Border Controller SIP trunk solution. DeltaCom place an inbound call from the Nortel CS2000 through an Acme Packet Session Border Controller via a dedicated Internet connection terminating the call in Communication Manager with SIP Enablement Services SIP trunk solution.

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature testing. The feature testing evaluated the ability to make outbound/ inbound calls from SIP, H.323, Digital and Analog Phones including Fax Support in Communication Manager with SIP Enablement Services co-resident solution to phones in DeltaCom's Nortel CS2000 with Acme Packet Session Border Controller SIP Trunk Solution via a SIP trunk. Testing was carried out on codec support and negotiation supported by DeltaCom as well as supplementary features such as Call Hold, Forward, Transfer and Conference between the Service Provider and the Avaya IP Telephony Solution.

1.2. Support

Technical support from DeltaCom can be obtained through the following methods:

Phone:	1800 239 3000
E-mail:	support@deltacom.com
Web:	http://www.deltacom.com
Address:	1801 Hillyer Robinson Parkway Anniston AL 36207 USA

2. Reference Configuration

Figure 1 illustrates the network configuration used to verify DeltaCom's SIP Trunk solution. The main Site A was comprised of a S8300C Media Server with G450 Media Gateway, and had connections to the 9630 SIP Telephones, 1616 and 9630 IP Telephones, 2420 Digital Telephone, Analog Telephone and a standard Fax Machine, with a call originating from phones in Site A to Site B over the SIP trunk. A C363 T Ethernet Switch is used for local area connection within Site A. The branch Site B was comprised of Nortel Communication Server 2000 with Acme Packet Session Border Controller and it had connections to the following: Nortel IP Telephones, Nortel Digital Telephone, with a call originating from phones in Site B to Site A over a SIP Trunk. The firewall in Site A was programmed to allow access from DeltaCom public IP Address of the Acme Packet Session Border Controller and to allow traffic out from Communication Manager. Alternatively, the firewall in Site B was programmed to allow traffic out from the Acme Packet Session Border controller and allow access from Avaya S8300C Media Server with G450 Media Gateway in Avaya's public network

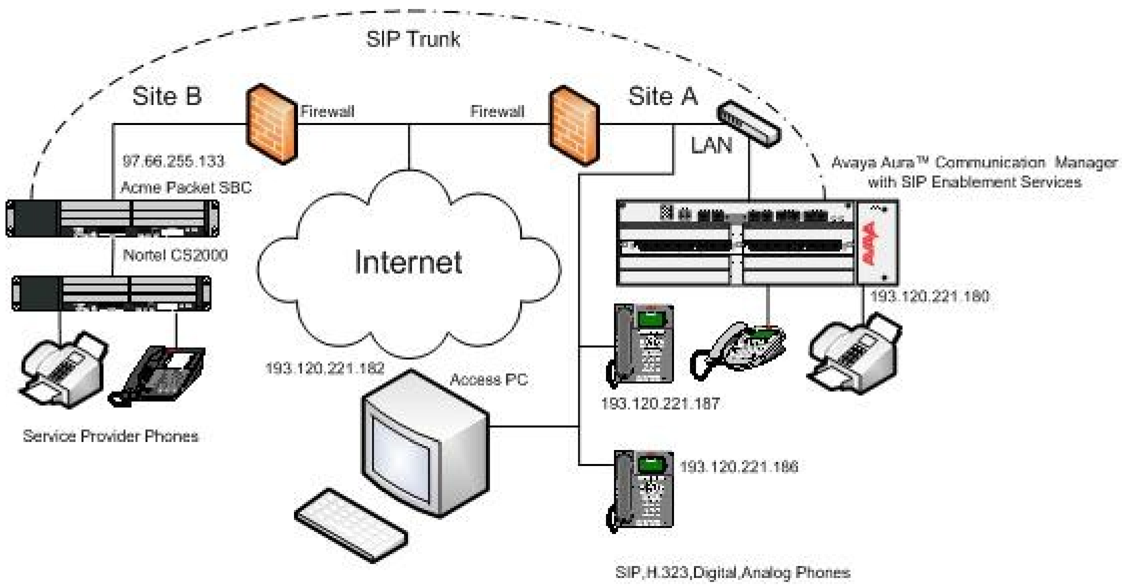

Figure 1: DeltaCom's Nortel CS2000 with Acme Packet SBC and Avaya Aura™ CM/SES

The Co-Residency of Communication Manager and SIP Enablement Services (SES) is a solution that helps reduce the cost of adding full, native support for SIP to your communications network by merging the hardware platforms of Release 5.2 of the Communication Manager software. The merged platform for co-residency is the Avaya S8300C Server, with compact flash replacing RAMDISK. With the co-resident hardware platform, SES and Communication Manager can operate more efficiently (for example, without firewall issues or the need to encrypt links between the two), and to share some of the same server resources and capabilities. Some of the pages of the Maintenance Web interface are also shared and reused, including the Web pages for system backup and restore capabilities. The system logging, process status and role-based web access controls also are the same for both. Both the SES software and Communication Manager are installed, patched and configured as usual with a separate license for each. Thus the transition to the new co-resident implementation is easier for existing administrators.

3. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided.

Avaya Aura™	Software
Avaya S8300C Server	Avaya Aura™ Communication Manager Release 5.2 R15x.02.0.947.3 Update: Service Pack 3
Avaya Media Gateway G450 Media Module 711 Analog Media Module 712 DCP	Vintage 30.10.4 MM FW 12 for HW 3 MM FW Vintage 9
Avaya Aura™ SIP Enablement Services co-resident	R5.2, SES05.2-02.0.947.3b Update: Service Pack 3
Avaya one-X®9600 Series IP Telephones (H.323)	Rel. 3.1 Service Pack 1
Avaya one-X® 9600 Series IP Telephones (SIP)	Rel. 2.5.x
Avaya 2420 Series Digital Telephone	Rel. 2420_R5.bin
Avaya one-X® 1616 IP Telephones	Rel. 1.2
Avaya 6520 Series Analog Telephone	-
Fax Machine (Standard)	Canon JX500
DeltaCom	**Software**
Nortel Communication Server 2000	CVM12
Acme Packet Session Border Controller	Net4250FWC6.0.0 MR-3 Patch 5 (WS Build 448)
Nortel Phones	-

4. Configure Avaya Aura™ Communication Manager

This section describes how to administer and configure SIP on a Communication Manager system so that Communication Manager (and SES, running on the same server) supports SIP endpoints. Administer and configure SIP trunking on the system with Communication Manager screens and fields, some of which are specific to SIP. Communication Manager must be functioning properly before you start SIP administration and SES implementation.

The sample configuration has been administered to use Trunk Group 2 to receive inbound calls in, and carry outbound calls to DeltaCom's SIP Trunk Solution. Communication Manager provides the capability to administer IP trunks as SIP trunks. These trunks are administered as a trunk type and associated with the appropriate type of signaling group on the server running Communication Manager. Then, the servers running SES enable these SIP trunks to be connected to SIP endpoints. The sample configuration uses SIP telephones, H.323 telephones, Digital telephones and Analog telephones to take in inbound calls and make outbound calls to DeltaCom SIP Trunk Solution in the testing environment. The following procedure describes how SIP Telephones are configured in both Communication Manager and SIP Enablement Server and uses Trunk Group 2 with Signaling Group 2.

- Administer System Parameter Customer-Options
- Administer Dial Plan Analysis
- Administer Media Gateway
- Administer IP Node Names
- Administer Trunk Group
- Administer Signaling Group
- Administer IP Network Region
- Administer IP Codec
- Administer ARS Analysis Table
- Administer Route Pattern
- Administer Incoming Call Handling Treatment
- Administer Public Numbering
- Administer Off-PBX Station Mapping
- Administer Station

4.1. Administer System Parameter Customer-Options

The **System Parameter Customer Options** screen shows what needs to be licensed for SIP Trunking on Communication Manager. As shown in **Figure 2** the **Maximum Off-PBX Telephone – OPS** field must be licensed. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to obtain additional capacity.

```
display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V15                                 Software Package: Standard
       Location: 1                                RFA System ID (SID): 1
       Platform: 22                               RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 900   18
                                      Maximum Stations: 450   8
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 400   0
                    Maximum Off-PBX Telephones -   OPS: 400   2
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

Figure 2: System Parameter Features Page 1

As shown in **Figure 3** there must also be licenses available for **Maximum Administered SIP Trunks.**

```
 display system-parameters customer-options                      Page   2 of  10
                               OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 0     0
           Maximum Concurrently Registered IP Stations: 450   0
             Maximum Administered Remote Office Trunks: 450   0
Maximum Concurrently Registered Remote Office Stations: 450   0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                       Maximum Video Capable Stations: 0     0
                  Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 200   10
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 0     0
                     Maximum Media Gateway VAL Sources: 2     1
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 3: System Parameter Features Page 2

4.2. Administer Dial Plan Analysis

This section describes the **Dial Plan Analysis** screen as shown in **Figure 4**. The Dial Plan Analysis Table is the system's guide to translating the digits dialed by users. This screen enables the administrator to determine the beginning digits and total length for each type of call that Communication Manager needs to interpret. In the example shown, the dialed string beginning with the number **1** and **four** digits in length will be used as our extension range for the SIP Telephones, H.323 Telephones, Digital Telephones and Analog Telephones. Therefore the extension range of the extensions used for test purposes is 1001 to 1007. The dialed string beginning with the number **2** and **three** digits in length will be used as a dial access code on the trunk. The dialed string beginning with the number **9** and **one** digit in length will be used as the feature access code to access the automatic alternate routing table in Communications Manager. The dialed string beginning with the characters **#** and ***** and three digits in length will also be used as feature access codes.

```
display dialplan analysis                                       Page   1 of  12
                            DIAL PLAN ANALYSIS TABLE
                                  Location:  all          Percent Full:    2

      Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
      String   Length Type    String   Length Type     String   Length Type
    1            4    ext
    2            3    dac
    4            6    ext
    9            1    fac
    *            3    fac
    #            3    fac
```

Figure 4: Dial Plan Analysis

4.3. Administer Media Gateway

This section describes the **Media Gateway** screen as shown in **Figure 5**. For the Media Gateway to register to Communication Manager it is crucial the correct Serial Number of the Media Gateway is entered in the **Serial No** field. For the SIP Enablement Server, the **Network Region** field must have the same value as the network region of the SIP authoritative domain. The Network Region indicates what is assigned to the media gateway. It is used by the primary server to allocate resources from the nearest Media Gateway. The Media Gateway has **Network Region 1** assigned to it.

```
 display media-gateway 1
                             MEDIA GATEWAY
         Number: 1                            Registered?  y
           Type: g450               FW Version/HW Vintage: 29 .22 .3  /1
           Name: DeltaCom                  MGP IP Address: 193.120.221.181
      Serial No: 09IS09228081       Controller IP Address: 193.120.221.180
   Encrypt Link? y                            MAC Address: 00:1b:4f:1e:5c:40
 Network Region: 1    Location: 1              Enable CF? n
                                               Site Data:
  Recovery Rule: none

Slot   Module Type            Name                   DSP Type  FW/HW version
 V1:   S8300                  ICC MM                 MP80      16   4
 V2:
 V3:
 V4:
 V5:   MM712                  DCP MM
 V6:   MM711                  ANA MM
 V7:   MM710                  DS1 MM
 V8:                                               Max Survivable IP Ext: 8
 V9:   gateway-announcements  ANN VMM
```

Figure 5: Media Gateway Page

4.4. Administer IP Node Names

This section describes the IP Node Name screen as shown in **Figure 6**. In Communication Manager the IP Address for the **procr** will be automatically populated from the installation. The IP Address used for the S8300C processor is **193.120.221.180**. Since Communication Manager is configured with SIP Enablement Services co-resident there is no need to configure an individual IP Address for SES.

```
 list node-names all

                       NODE NAMES

Type     Name              IP Address
IP       default           0.0.0.0
IP       procr             193.120.221.180
```

Figure 6: IP Node Names

4.5. Administer Trunk Group

This section describes each page of the **Trunk Group** screen as shown in **Figure 7**. Trunk Group 2 will be used to make and receive outbound and inbound calls to DeltaCom's SIP Trunk Testing Solution. In the Trunk Group screen, **Page 1**, type **sip** to specify the trunk group as SIP. The **Service Type** field indicates the service to which this trunk group is dedicated. Since Outbound calls and Inbound calls are originating from the public network, **public-ntwrk** is used for our **Service Type**. The trunk will be used for Outbound and Inbound calls, hence the **Direction** field is set to **two-way**. The **Signaling Group** number assigned to this trunk is **2**. The **Number of Members** assigned to this trunk group is **5**. All other fields on this page are left as default.

```
 display trunk-group 2                                           Page   1 of  21
                               TRUNK GROUP

Group Number: 2                    Group Type: sip          CDR Reports: y
  Group Name: Outbound/Inbound            COR: 1        TN: 1        TAC: 202
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                       Signaling Group: 2
                                                     Number of Members: 5
```

Figure 7: Trunk Group Page 1

All fields on **Page 2** of the trunk group are left as default.

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

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 600
```

Figure 8: Trunk Group Page 2

All fields on **Page 3** of trunk group 2 are left as default.

```
display trunk-group 2                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n







 Show ANSWERED BY on Display? y
```

Figure 9: Trunk Group Page 3

All fields on **Page 4** of trunk group 2 are left as default.

```
display trunk-group 2                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type:
```

Figure 10: Trunk Group Page 4

4.6. Administer Signaling Group

This section describes the **Signaling Group** screen as shown in **Figure 11**. The **Group Type** field describes the type of protocol to be used with the signaling group, **sip**. SIP is the protocol used. Verify the **Transport Method** field is set to **tls**. Since the system is to be configured with SES co-resident on CM the **Co-Resident SES** field is set to yes, **y**. The **Near-end Node Name** defaults to **procr** so that it can communicate with the SES application when the **Co-Resident SES** field is set to yes, **y**. The **Near-end Listen Port** automatically populates with port **6001** when **Co-Resident SES** field set to **y**, this is the same port number that the Co-Resident SES is using to communicate with Communication Manager. The **Far-end Node Name** is set to default **procr** for the Co-Resident SES signaling group. The **Far-end Listen Port** is set to **5061** for SIP over TLS. The **Far-end Domain** field is set to **97.66.255.133**. This is the IP Address of DeltaCom's Session Border Controller (SBC). The **Far-end Network Region** is set to **1**. All other values on the signaling group are set to default.

```
 display signaling-group 2
                               SIGNALING GROUP

 Group Number: 2             Group Type: sip
                       Transport Method: tls
  IMS Enabled? n          Co-Resident SES? y
     IP Video? n




   Near-end Node Name: procr                 Far-end Node Name: procr
 Near-end Listen Port: 6001                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
Far-end Domain: 97.66.255.133

                                             Bypass If IP Threshold Exceeded? 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? n                      Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

Figure 11: Signaling Group

4.7. Administer IP Network Region

The section describes the **IP Network Region** form as shown in **Figure 12** and **Figure 13**. The Authoritative Domain filed must be set to the same value as the SIP domain administered in the SIP Enablement Server. This is set in the Master Administration web interface in **Figure 23** in the SES system. Therefore the **Authoritative Domain** field was set to **sippri.com**. For the **Intra-region** and **Inter-region IP-IP Direct Audio** fields for the SIP installation set these two values to **yes**.

```
 display ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: sippri.com
    Name: Internal
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                  RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 12: IP Network Region Page 1

The IP Network Region screen was programmed to use IP Codec 1 from source to destination.

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

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

Figure 13: IP Network Region Page 3

4.8. Administer IP Codec

This section describes the **IP Codec Set** screen as shown in **Figure 14**. IP Codec **G.711MU**, **G.711A** and **G.729A** were used for testing purposes as these three codecs are only supported by DeltaCom's SIP Trunk Solution.

```
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.729A           n          2        20
3: G.711A           n          2        20
4:
```

Figure 14: IP Codec Set

4.9. Administer Automatic Route Selection

This section describes the **Automatic Route Selection** screen as shown in **Figure 15**. The user-dialed numbers are matched to the dialed string entry that most closely matches the dialed number. In order to make an Outbound call to DeltaCom test phone in their SIP Trunk Solution, a user dials **2562416070** the **10** digit dialed string will route to **Routing Pattern 1.** The **Call Type** field is set to **intl** since Route Index contains public network ISDN trunks that require international type of number encodings.

```
display ars analysis 256                                        Page   1 of   2
                            ARS DIGIT ANALYSIS TABLE
                                  Location:  all          Percent Full:    2

         Dialed           Total     Route     Call   Node   ANI
         String          Min  Max  Pattern    Type   Num    Reqd
   2562416070            10   10    1         intl          n
   2562416071            10   10    1         intl          n
   2562418000            10   10    1         intl          n
   2562419800            10   10    1         intl          n
   2562419898            10   10    1         intl          n
   2562419972            10   10    1         intl          n
   2564383340            10   10    1         intl          n
```

Figure 15: Automatic Route Selection

4.10. Administer Routing Pattern

This section describes the **Route Pattern** screen as shown in **Figure 16**. The route-pattern screen defines that the call made to 2562416070 will be sent out trunk group **2** with **FRL** set to **0**.

```
display route-pattern 1                                         Page   1 of   3
                   Pattern Number: 1   Pattern Name: Outbound
                            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: 2    0                                                              n   user
2:                                                                     n   user
3:                                                                     n   user
4:                                                                     n   user
```

Figure 16: Route Pattern

4.11. Administer Incoming Call Handling

This screen describes the **Incoming Call Handling Treatment** screen as shown in **Figure 17**. This Incoming Call Handling Treatment allows Communication Manager take in incoming dialed string from DeltaCom, strip the incoming dialed string away and insert a number of choice to allow the incoming dialed string to hit. The service feature is set to **public-ntwrk** as this was defined on trunk group 2. The Service Provider is sending a dialed string of **6** digits in length with digits **419830**. The 419830 was deleted on the trunk interface and extension **1001** was inserted so that when the DeltaCom dialed 419830 from their SIP trunk solution it would ring on extension 1001. The following is the case for dialed strings 419831 to 419834 originating in DeltaCom's SIP trunk solution.

```
display inc-call-handling-trmt trunk-group 2                    Page   1 of   3
                      INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    6  419830           6   1001
 public-ntwrk    6  419831           6   1005
 public-ntwrk    6  419832           6   1003
 public-ntwrk    6  419833           6   1006
 public-ntwrk    6  419834           6   1007
```

Figure 17: Incoming Call Handling

4.12. Administer Public Numbering

This screen describes the **public-unknown-numbering** page of Communication Manager as shown in **Figure 18**. This is where the Calling Party Number of choice can be inserted on an Outbound call to DeltaCom's SIP Trunk Solution. **Page 1** consists of parameters to enter the extension length, the extension code, the trunk group the call will be sent out on, the Calling Party Number (**CPN**) and the Calling Party Number Length. In the example the extension length of the extensions are **4** digits long. Extension code is as above 1001 to **1006**. The trunk group the Outbound call will be going out on is **Trunk Group 2**. The Calling Party Number DeltaCom wish to see when the call comes into their SIP Trunk Solution are listed above, Calling Party Number **2562419830** to 2562419834. The Calling Party Number length was **10** digits.

```
display public-unknown-numbering 1                          Page   1 of   2
                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext           Trk       CPN            CPN
Len Code          Grp(s)    Prefix         Len
                                                    Total Administered: 5
 4  1001          2         2562419832     10          Maximum Entries: 240
 4  1002          2         2562419834     10
 4  1003          2         2562419833     10
 4  1005          2         2562419831     10
 4  1006          2         2562419830     10
```

Figure 18: Public Numbering

4.13. Administer Off Station PBX Mapping

SIP phones that register to a SES server use **off-pbx-telephone station-mapping** to take and receive calls through Communication Manager. The extensions of both SIP Phones are **1003** and 1004 and also configured in the phone number field as shown in **Figure 19**. The SIP enabled phone application **OPS** is used in the configuration. Trunk 1 is the internal trunk that internal inbound and outbound calls will be made from the SIP Phones. The **Config Set** contains the desired call treatment for the stations which is set to **1**.

```
display off-pbx-telephone station-mapping 1003              Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 1003             OPS          -      1003             1          1
 1004             OPS          -      1004             1          1
```

Figure 19: Off-PBX-telephone Page 1

The **Call Limit** field is set to **2**. This is the maximum amount of simultaneous calls that can be active for each SIP Phone. The mapping mode field was set to both in this configuration setup. This is used to control the degree of integration between SIP Phones. The **Calls Allowed** field was set to **all**. This identifies the call filter type for a SIP Phone. The **Bridged Calls** field was set to **none** as it was not needed for testing purposes.

```
display off-pbx-telephone station-mapping 1003                  Page   2 of   3
                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station        Appl     Call         Mapping     Calls        Bridged       Location
Extension      Name     Limit        Mode        Allowed      Calls
1003           OPS       2           both         all          none
1004           OPS       8           both         all          none
```

Figure 20: Off PBX telephone Page 2

4.14. Administer Station Screen

This screen describes the **station** form setup for the SIP phone on Communication Manager as shown in **Figure 21**. The **Extension** used was **1003** with phone **Type 9630**. The **Name** of the phone was set to **SIP Phone** and all other values on **Page 1** of the station form were left as default.

```
 display station 1003                                                   Page   1 of   5
                                    STATION

Extension: 1003                        Lock Messages? n              BCC: 0
     Type: 9630                        Security Code:                 TN: 1
     Port: S00002                    Coverage Path 1:                COR: 1
     Name: SIP Phone                 Coverage Path 2:                COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                                         Time of Day Lock Table:
              Loss Group: 19       Personalized Ringing Pattern: 1
                                               Message Lamp Ext: 1003
            Speakerphone: 2-way             Mute Button Enabled? y
        Display Language: english                Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                         IP SoftPhone? n

                                                        IP Video? n


                                             Customizable Labels? y
```

Figure 21: Station Screen Page 1

5. Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of SIP Enablement Services (SES). SIP Enablement Services is configured via an Internet browser using the Administration web interface. It is assumed that SIP Enablement Services software and the license file have already been installed. The following procedures include:

- Access SIP Enablement Services
- Enable SIP Enablement Services
- Setup Master Admin Screen
- Edit System Properties Screen
- Add Host Screen
- Add Media Server
- Administer Address Maps to Communication Manager
- Administer Address Maps to Hosts
- Administer Trusted Host
- Administer SIP Phones on SES

5.1. Access SIP Enablement Services

Access the SES Administration web interface, by entering **http://<ip-addr>/admin** as the URL in an Internet browser, where *<ip-addr>* is the IP address of the SIP Enablement Services server. Log in with the appropriate credentials and then select the **Administration** link and then **SIP Enablement Services** from the main screen as shown in **Figure 22**.

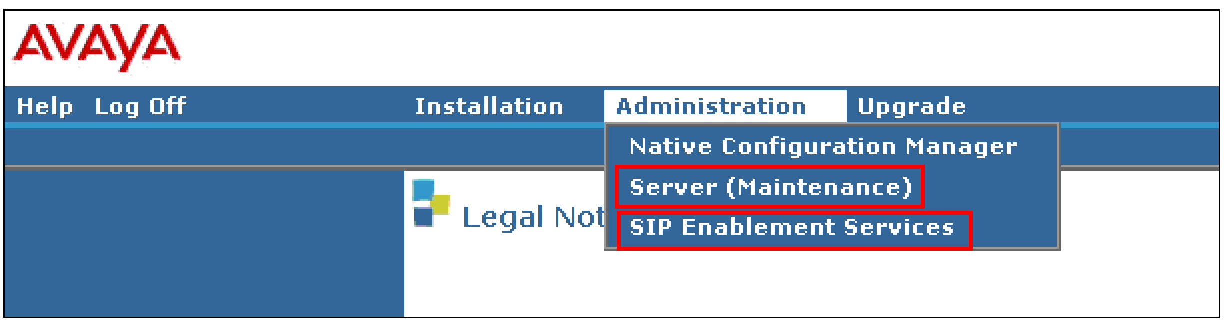

Figure 22: Access SIP Enablement Services

The Co-Residency of Communication Manager and SIP Enablement Services (SES) is a solution that merges the hardware platforms of Release of Communication Manager software as shown in **Figure 23**. **SES** is **enabled** on Communication Manager's System Management Interface by accessing **Server (Maintenance)** in **Figure 22** and then **SES Software** in **Figure 23.**

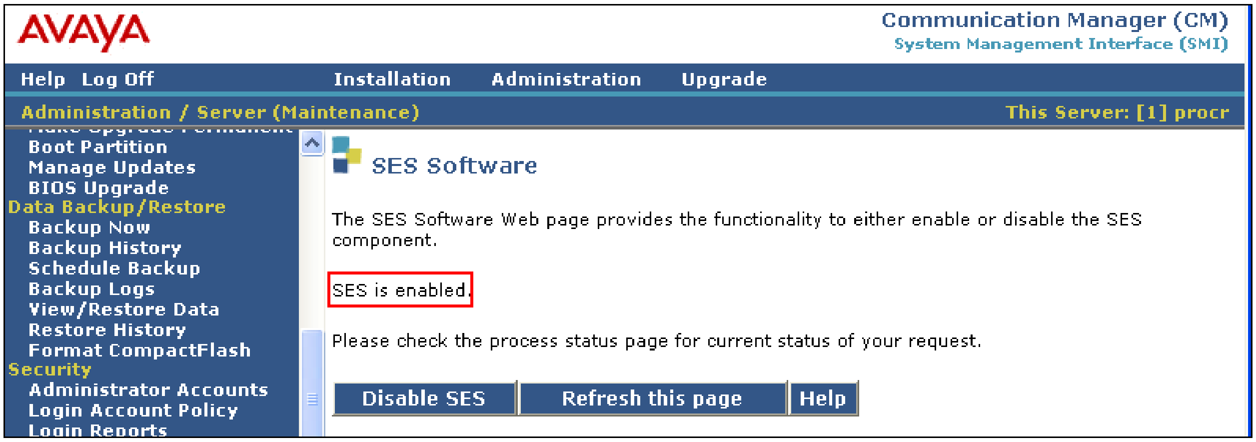

Figure 23: Enable SIP Enablement Services

5.2. Setup Master Admin Screen

The **Setup Master Administration** screen as shown in **Figure 24** allows the administrator to specify if the machine you are setting up is a SES edge server or/and SES home server. Select **This server is the SES Master Administration System for the SES Network** option.

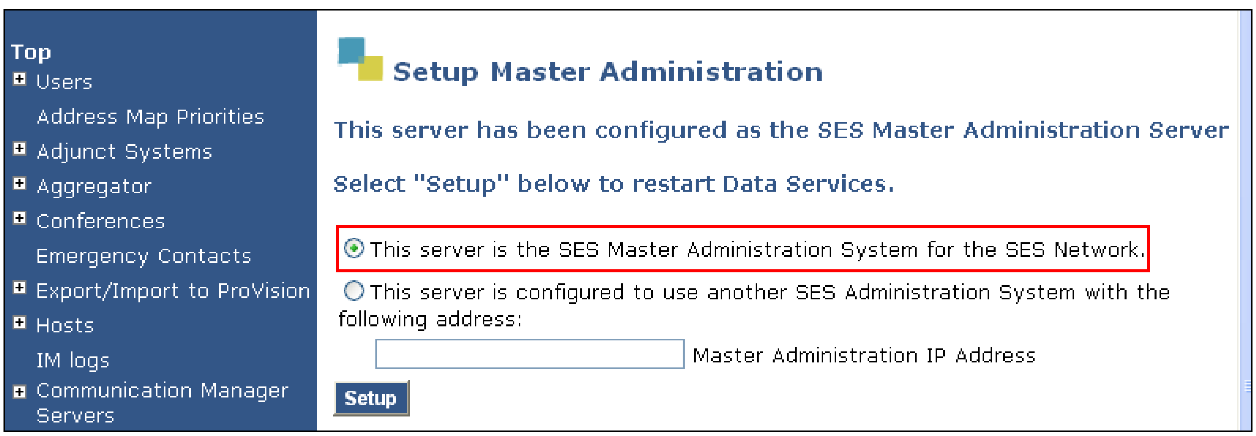

Figure 24: Setup Admin Screen

5.3. Edit System Properties Screen

The edit **System Properties** screen defines the server's type and domain as shown in **Figure 25**. This **SES Version** field displays the major and minor release number and the current load and build number of the Avaya software that is running on this SES server. The **System Configuration** field identifies the SES server as being a **Simplex** machine. This read-only field does not indicate the server's role of primary or backup. The **Host Type** field identifies the SES server as a home/edge type server. This read-only field does not indicate the server's role of primary or backup. The **SIP Domain** field indicates the domain name assigned to the SIP Enablement Services Configuration. This was set to **sippri.com**. The **SIP License Host** field requires the host name, fully qualified domain name or the IP address of the SES server that is running the WebLM application and has the associated license file installed. This entry shows the IP address of the SIP Enablement Server co-resident with Communication Manager software was entered as **193.120.221.180.**

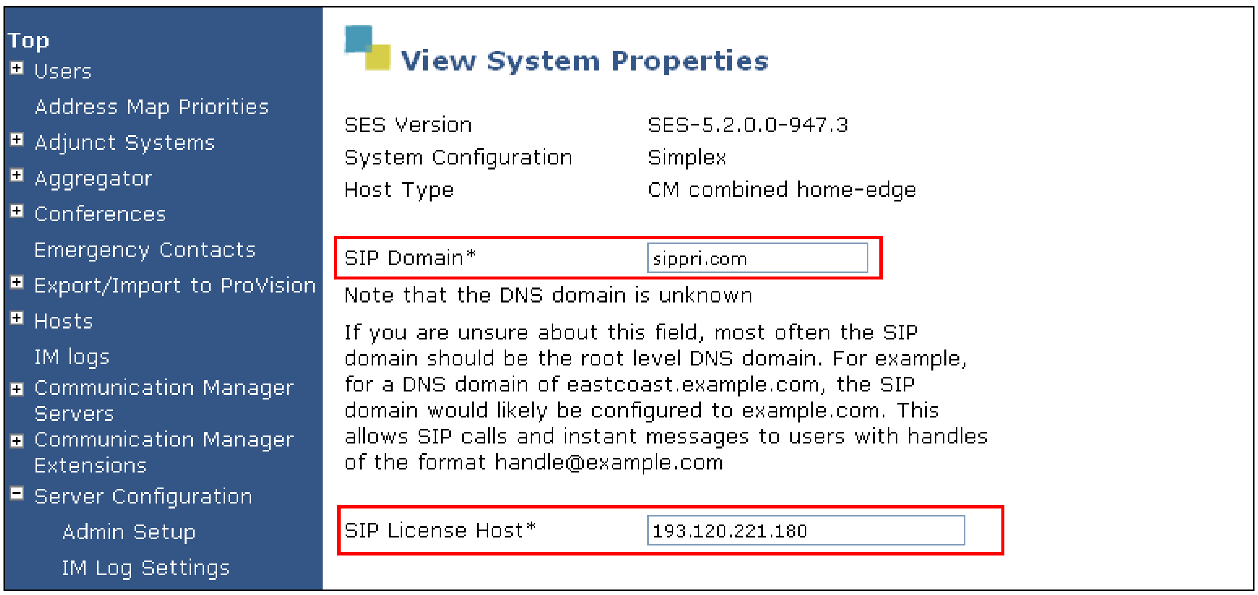

Figure 25: System Properties

5.4. Add Host Screen

The **Host IP Address** field contains the IP address for this combined home/edge server as shown in **Figure 26**. This was **193.120.221.180**. The **Profile Service Password** is for permissions between SES hosts. This is not used by the administrator; it is used by internal software components for secure communication between SES servers and the master administration system. Since the server is an S8300C running SES co-resident with Communication Manager software the **Host Type** functions as a **CM combined home-edge** server. In the **Listen Protocol** fields **UDP** and **TLS** were selected. The **Link Protocols** field refers to the trunk signaling between SIP Enablement Services and Communication Manager. Typically, the selection here matches the Signal Group value on Communication Manager. This was **TLS.** For third-party proxy servers you may select to link to SES with TLS, TCP or UDP.

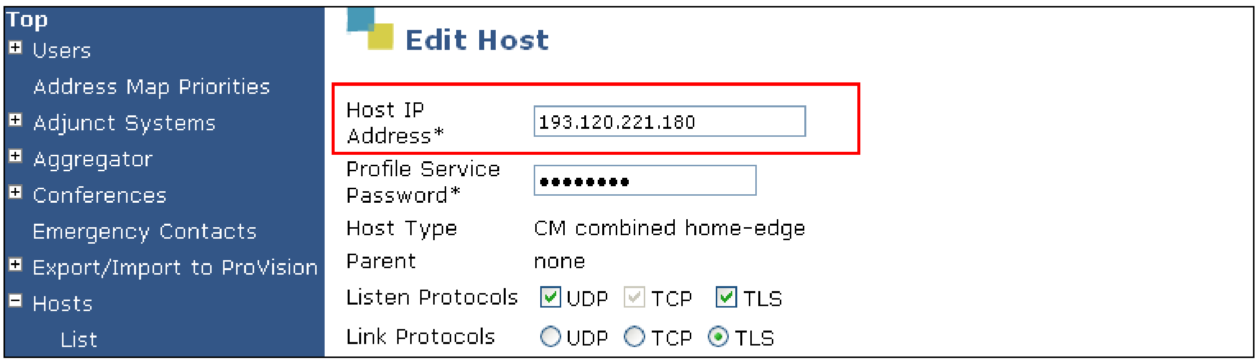

Figure 26: Host Screen

5.5. Add Media Server

The Communication Manager Server Interface screen shown in **Figure 27** assigns a home server to each interface for Communication Manager. Since the SES Edge server needs to obtain configuration information from co-resident Communication Manager running with an SES home, it will use the link to the specified Media Server Admin address. The two links may be the same IP address as shown **193.120.221.180**. The **SIP Trunk IP Address** field holds the IP address for the media server's processor Ethernet interface that terminates the SIP link form SES. This is **193.120.221.180**. The **SIP Trunk Port** field is for SES/Communication Manager co-resident installations only and allows configuration of SES with the same port number configured in Communication Manager signaling group field. When the signaling group in Communication Manager is configured, there is a co-resident check box, the selection of which will default the port to **6001.** The SIP Trunk port must match the near end listen port on the Signaling Group page in **Figure 11**.

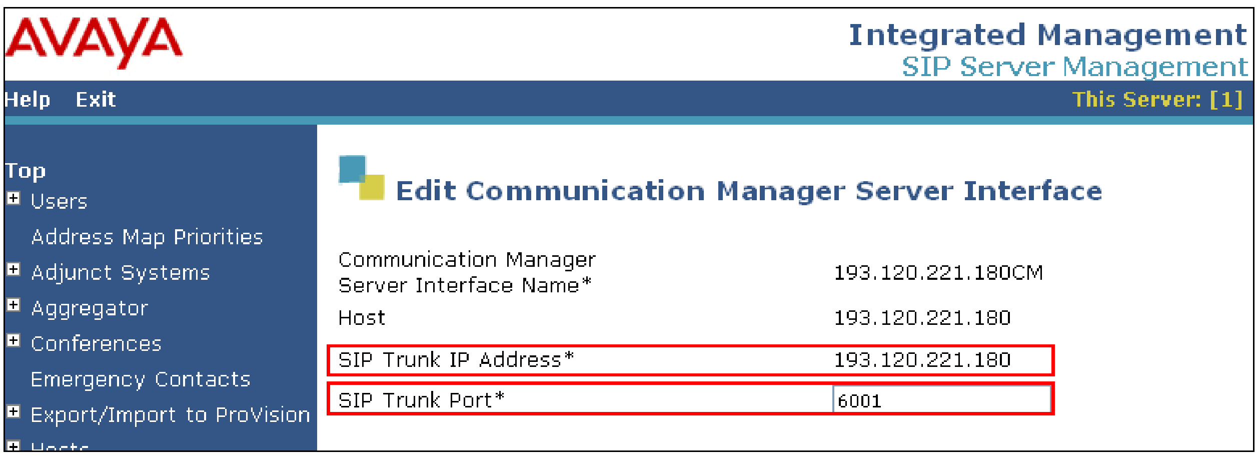

Figure 27: Communication Manager Server Interface

5.6. Administer Address Map to Communication Manager

The Communication Manager Map Entry Page shown in **Figure 28** describes how inbound number translations are taken in on the SES server from DeltaCom's SIP Trunk Solution. An address map identifies the relationship between the host server and Communication Manager server. These messages travel over the SIP trunk administered in Communication Manager. For example, an address map **Pattern** of **^sip:4198[0-9]*** matches the inbound number 419830 from DeltaCom. The pattern 4198[0-9]* will be the matched incoming call and will be redirected to a contact.

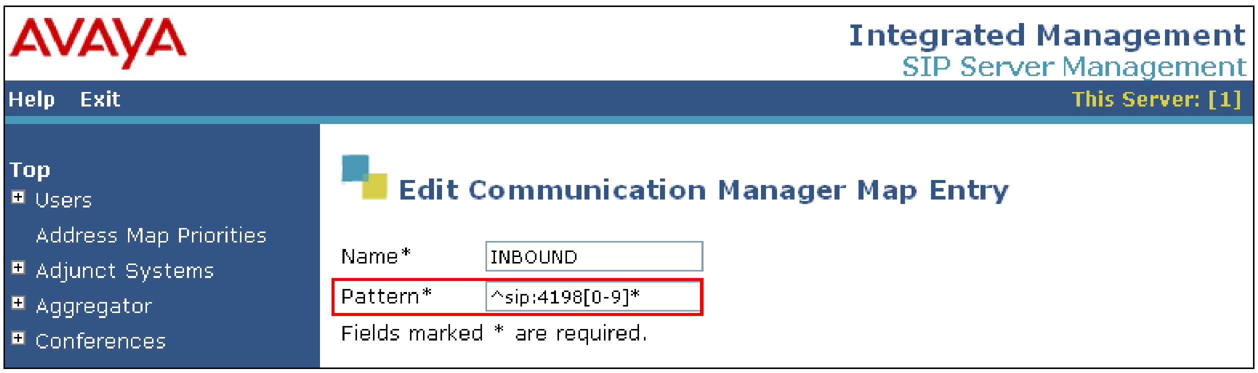

Figure 28: Address Map

Contact entries are constructed by the system shown in **Figure 29**. In our example the host has constructed a **Contact** dynamically by substituting **sip** as the protocol, **$(user)** to represent the user name or extension in the original request URI. The IP address of the Communication Manager server to try next **(193.120.221.180)**, the port number **(6001)** and name of the transport method to be used **(tls).**

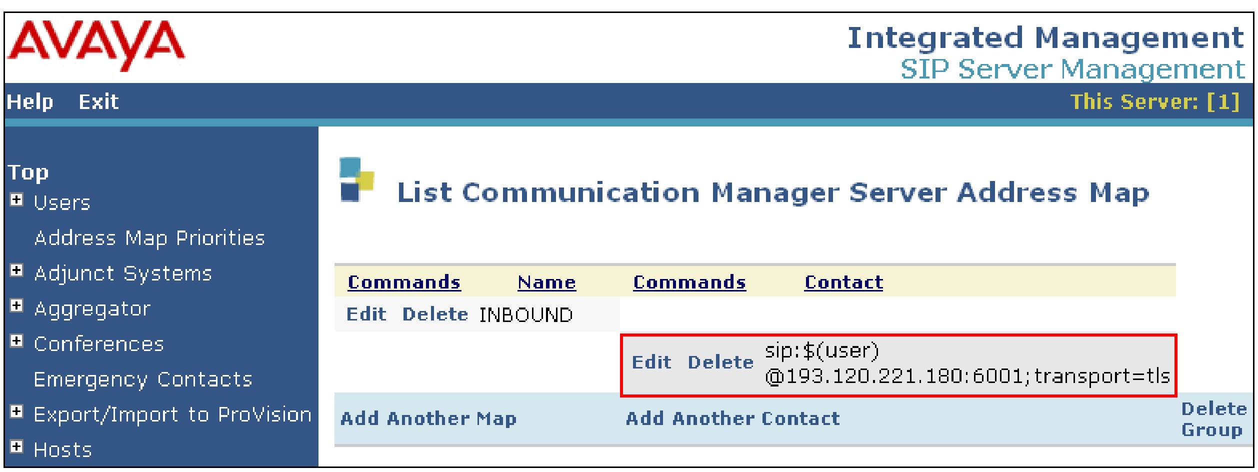

Figure 29: Communication Manager Server Address Map

5.7. Administer Address Map to Host

The Host Map Entry shown in **Figure 30** enables a SES server to redirect calls out of the current configuration to a specific server or endpoint. For the match pattern **^sip:25[0-9]]***, the dialed string 2562416070 will be redirected to the first contact associated with the group shown below.

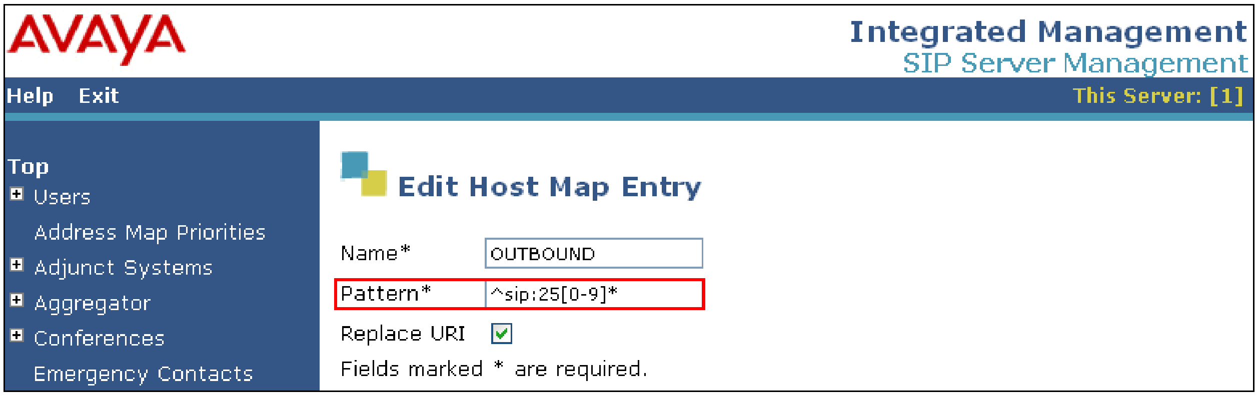

Figure 30: Host Map

The Host Contact address map shown in **Figure 31** is the IP address of another home server that you want to direct the calls to. The server evaluates the call for the regular expression provided in the pattern field. It finds a match and redirects the call to the first home computer named in the Contact field. The above example sends the contact address **sip:$(user)@97.66.255.133** or 2562416070@97.66.255.133 outbound from the SES to DeltaCom SIP Trunk Solution.

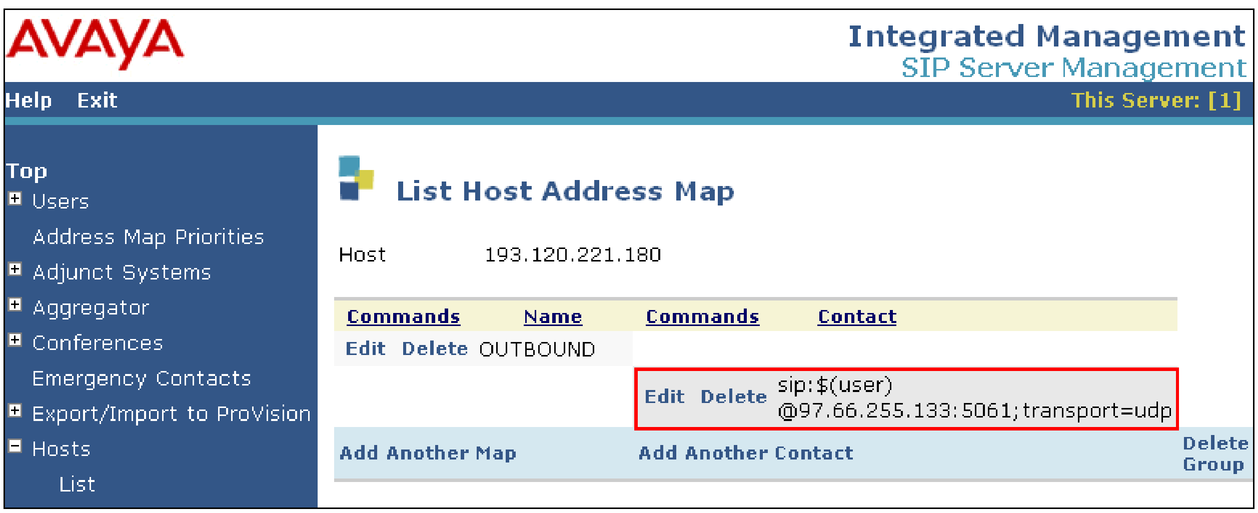

Figure 31: Host Address Map

5.8. Administer Trusted Host

This screen describes how to **Add Trusted Host** as shown in **Figure 32**. The IP Address of the far end domain of **DeltaCom** SIP Trunk Solution was added. This was **97.66.255.133**.

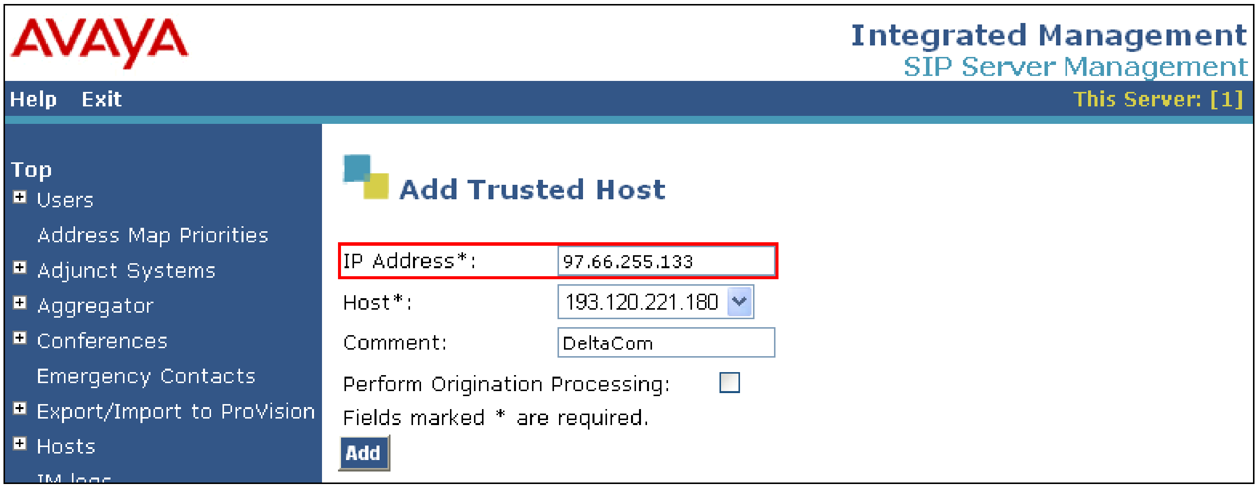

Figure 32: Add Trusted Host

5.9. Administer SIP Phones on SES

This screen allows SIP phone users to be added to the SES as shown in **Figure 33**. Users are added one at a time with this screen. A handle identifies the user on the SES system. Users **Primary Handle** must be the same as the **User ID**'s. In this example the **Primary Handle** and **User ID** is **1003**. The **Password** needs to be six characters long and was set to **123456**. This password is needed when the SIP phone registers to the SES server after the extension of the SIP phone is inputted. The **Host** IP address is populated automatically to **193.120.221.180**. Check the **Add Communication Manager Extension**. Press the **Add** button at the bottom of the screen. The SIP Phone extension 1003 must be added to Communication Manager also as shown in **Figure 34**.

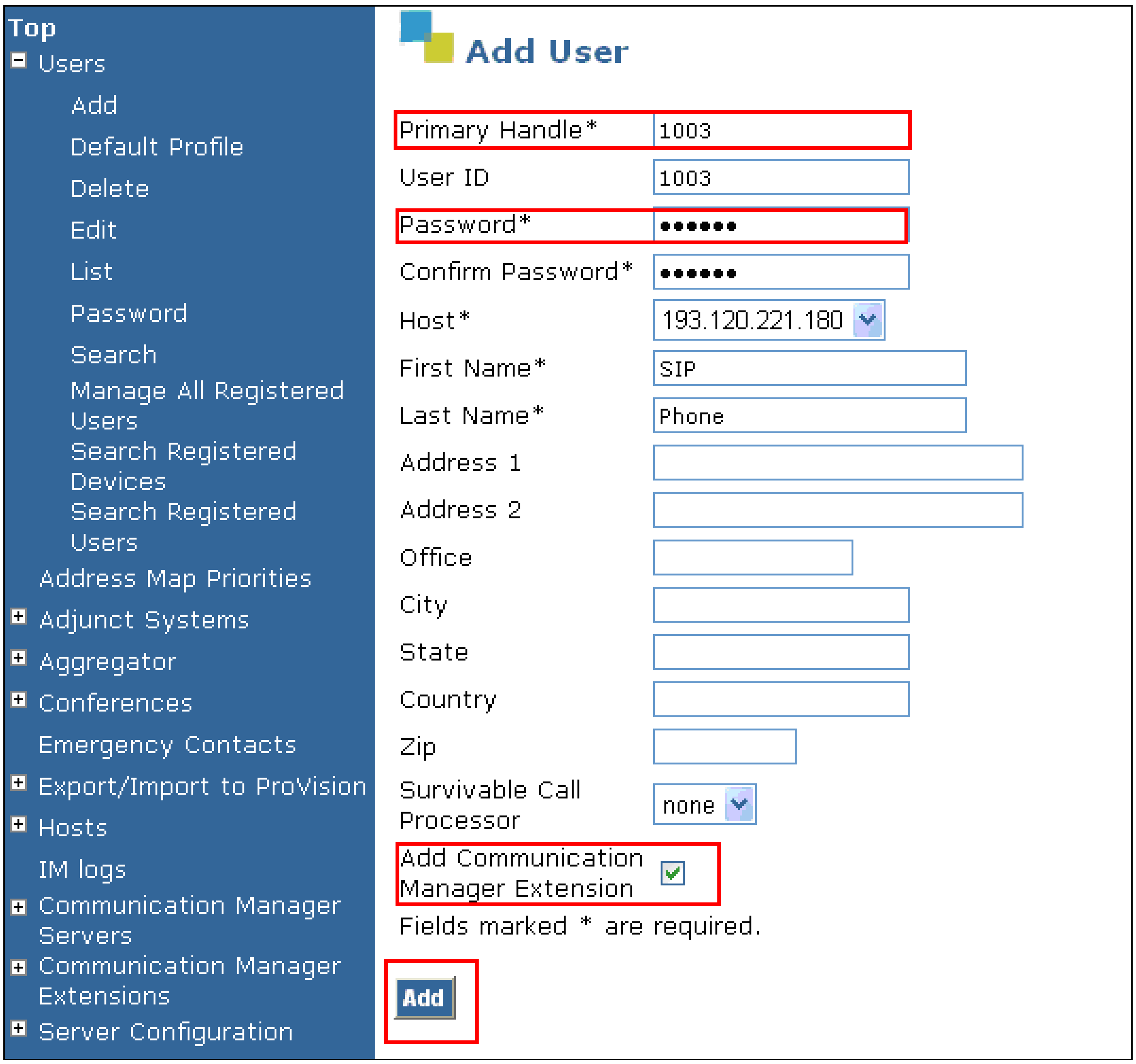

Figure 33: SIP Phone User

Once the **Add Communication Manager Extension** field is ticked the screen in **Figure 34** appears. Confirm that extension **1003** is **Communication Manager Extension** and press **Add**.

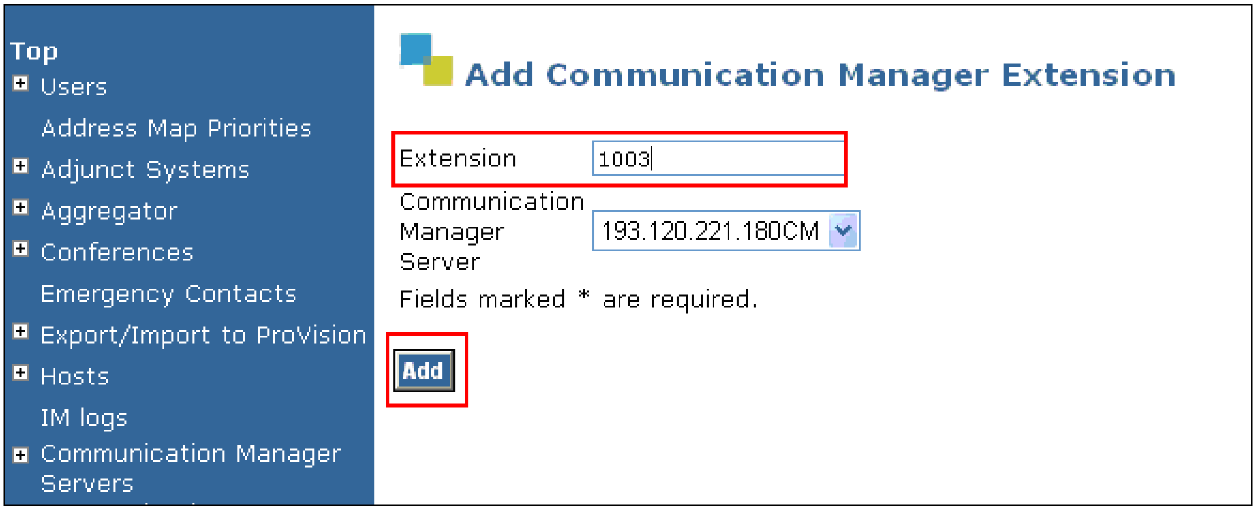

Figure 34: Communication Manager Extension

6. DeltaCom's SIP Trunk Solution

DeltaCom configuration is not mentioned or exposed in these application notes as requested by DeltaCom.

6.1. General Test Approach

A simulated enterprise site using an Avaya IP telephony solution was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use SIP trunks between the SIP Trunk Service provided by DeltaCom. The compliance test included the following:

- Incoming calls to Communication Manager with SIP Enablement Services made from DeltaCom's Nortel CS2000.
- Outgoing calls from Communication Manager with SIP Enablement Services made to DeltaCom's Nortel CS2000.
- Calls using G.729A, G.711MU, and G.711A coders.
- Fax calls to/from Communication Manager with SIP Enablement Services to DeltaCom's Nortel CS2000.
- DTMF transmission using RFC 2833 with successful Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction
- Direct IP-to-IP media with SIP and H.323 telephones.

6.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the DeltaCom's SIP Trunk Solution.

7. Conclusion

These Application Notes describe the configuration steps enabling customers using Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services co-resident to connect to DeltaCom's SIP Trunk Solution in a SIP Trunk environment.

8. References

This section references the Avaya documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1] *SIP Support in Avaya Communication Manager Running on Avaya Servers, May 2009* Document Number 555-245-206.

[2] *Administering Avaya Aura™ Communication Manager,* Document Number 03-300509

[3] *Avaya Aura™ SIP Enablement Services (SES) Implementation Guide*, May 2009, Document Number 16-300140

APPENDIX A:

Sample SIP INVITE Message from DeltaCom to Communication Manager with SIP Enablement Services co-resident.

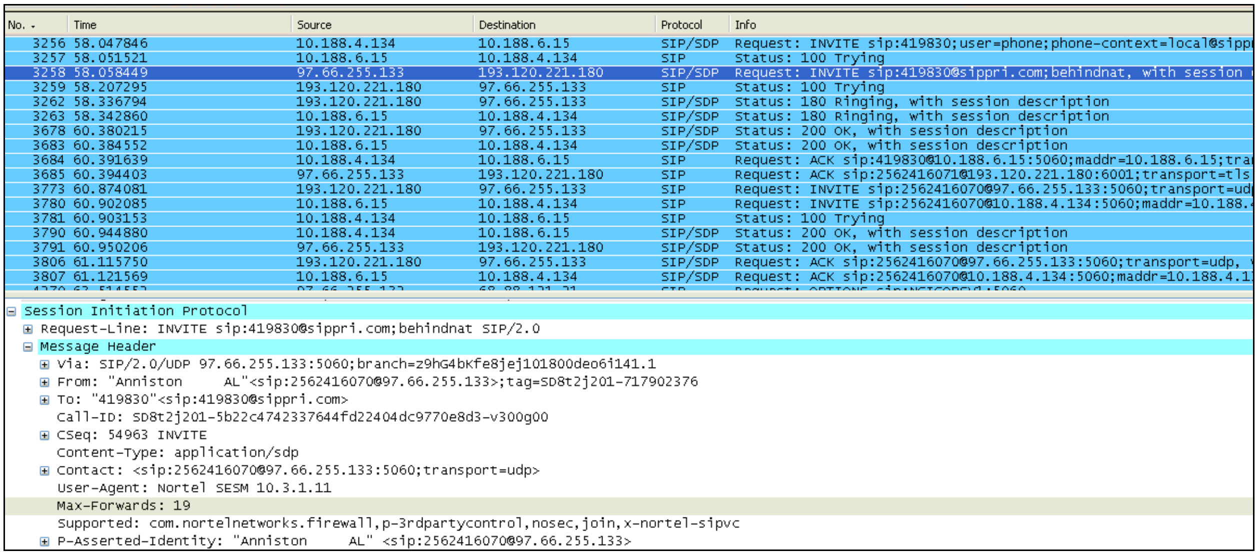

Figure 35: Wireshark Trace

Sample SIP INVITE Message from Communication Manager with SIP Enablement Services co-resident to DeltaCom.

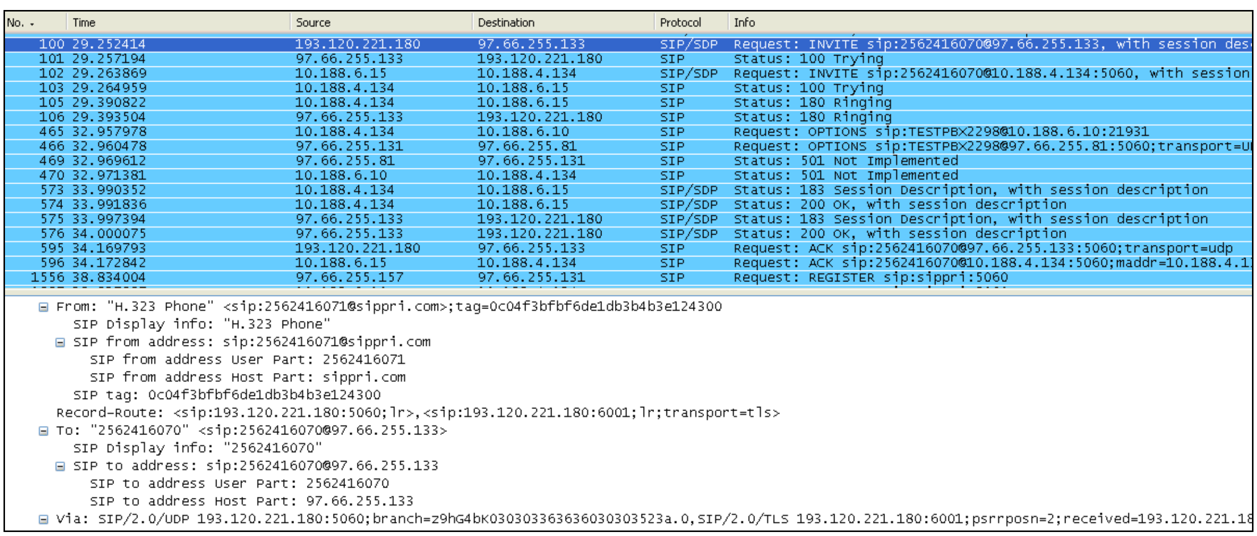

Figure 36: Wireshark Trace

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.