Avaya Solution & Interoperability Test Lab

Application Notes for Configuring MTS Allstream SIP Trunking Service with Avaya Aura® Communication Manager Evolution Server Release 6.0.1 and Avaya Session Border Controller for Enterprise Release 4.0.5 – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between MTS Allstream SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager Evolution Server 6.0.1, Avaya Session Border Controller for Enterprise 4.0.5 and various Avaya endpoints. This documented solution does not extend to configurations without Avaya Session Border Controller for Enterprise.

MTS Allstream is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing is conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between MTS Allstream SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager 6.0.1 configured as an Evolution Server, Avaya SBC for Enterprise (Avaya SBCE) and various Avaya endpoints. This documented solution does not extend to configurations without Avaya SBCE.

Customers using this Avaya SIP-enabled enterprise solution with MTS Allstream SIP Trunking Service are able to place and receive PSTN call via a broadband connection. This converged network solution is an alternative to traditional PSTN trunk such as analog and/or ISDN-PRI.

2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

MTS Allstream is a member of the Avaya DevConnect Service Provider program. The general test approach is to connect a simulated enterprise to MTS Allstream SIP Trunking Service via the public Internet and exercise the features and functionality listed in Section 2.1. The simulated enterprise is comprised of Avaya Aura® Communication Manager; Avaya Session Border Controller for Enterprise and various Avaya endpoints.

2.1. Interoperability Compliance Testing

To verify MTS Allstream SIP Trunking Service interoperability, the following features and functionalities are covered during the interoperability compliance test:

- Response to SIP OPTIONS queries.
- Incoming PSTN call to various phone types. Phone type included H.323, digital and analog telephone at the enterprise. All inbound calls from PSTN are routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN call from various phone types. Phone type included H.323, digital and analog telephone at the enterprise. All outbound calls to PSTN are routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator (1XC) H.323 soft phone. Both the1XC Computer Mode (where 1XC is used for call control as well as audio path) and the 1XC Telecommuter Mode (where 1XC is used for call control and a separate telephone is used for audio path) are tested.
- Dialing plans including local, long distance, international, outbound toll-free, operator assisted calls, local directory assistance (411)… etc.
- Proper codec negotiation with G.729 and G.711MU codecs.
- DTMF tone transmissions as out-of-band RTP events as per RFC 2833.
- Caller ID presentation and Caller ID restriction.

- Response to incomplete call attempts and trunk errors.
- Voicemail navigation for inbound and outbound calls.
- Incoming and outgoing fax over IP with G.711MU codec.
- User features such as hold and resume, transfer and conference.
- Off-net call forwarding with SIP Diversion method.
- EC500 mobility (extension to cellular).
- Routing inbound vector call to call center agent queues.
- Network Call Redirection using reINVITE for transferring of inbound call back to PSTN.
- Session Timers implementation from both ends of the enterprise and the service provider.

Items are not supported or not tested including the following:

- Inbound toll-free and outbound emergency calls (911) are supported but are not tested as part of the compliance test because MTA Allstream does not provide the necessary configuration.
- T.38 fax is not supported.
- Off-net call forwarding using History-Info method is not supported.
- SIP phone and 1XC SIP soft phone are not tested. The SIP extension feature is comprised of Communication Manager and Session Manager functionality. The compliance test does not cover Session Manager; therefore the SIP extension is not tested.

2.2. Test Results

Interoperability testing of MTS Allstream SIP Trunking Service with the Avaya SIP-enabled enterprise solution is completed with successful results for all test cases with the exception of the observations/limitations described below.

1. **Off-net blind transfer is not successful with REFER method**. In an inbound call scenario, Communication Manager receives an initial INVITE with REFER method present in Allow header. Communication Manager transfers off-net an inbound call to PSTN with a REFER request. MTS Allstream SIP Trunking Service does not properly support REFER as it accepts the REFER with a SIP/202 message but the call transfer is not successful. Both call legs are still maintained after transfer is complete. This issue is corrected by disabling the **Network Call Redirection** flag on outgoing trunk group to make Communication Manager transfer the call with **reINVITE** request. The detail configuration is described in **Section 5.7**.
2. **Network Call Redirection is not successful with "302 Moved Temporarily"**. A vector DN is programmed to use "302 Moved Temporarily" to redirect an inbound call to PSTN before answering. MTS Allstream SIP Trunking Service does not properly support "302 Moved Temporarily" as it sends an ACK to the "302 Moved Temporarily" but does not redirect the call to PSTN party specified in the Contact header.
3. **The untrusted calling party name from Communication Manager is not examined**. In an outbound call scenario, PSTN displays the original untrusted calling party name from Communication Manager. MTS Allstream SIP Trunking Service does not examine the calling party name before sending to PSTN. This is a known issue on MTS Allstream SIP Trunking Service and there is no available resolution at this time.
4. **The calling party name for outbound call is not being displayed by PSTN**. In an outbound call scenario, Communication Manager sends both calling party name and

number to PSTN. But in some cases, PSTN phone displays the Calling Party Number only and no Calling Party Name. In other cases, PSTN phone displays both Calling Party Name and Number. The Calling Party Name may be overridden by MTS Allstream SIP Trunking Service or by intermediate service providers that route the call through PSTN. This issue has low user impact, and is listed here simply as an observation.

5. **Communication Manager extension holds an inbound call but the incoming audio traffic is still being sent by MTS Allstream SIP Trunking Service**. To hold an inbound call, Communication Manager sends a reINVITE with a=sendonly in SDP. MTS Allstream SIP Trunking Service responds with 200OK and a=recvonly in SDP which means that the incoming audio traffic on Communication Manager will be deactivated. But MTS Allstream SIP Trunking Service still sends audio when the call is in “recvonly” state. This issue has low user impact, it is listed here simply as an observation.
6. **In an inbound call scenario, MTS Allstream SIP Trunking Service does not refresh the Session Timer**. MTS Allstream SIP Trunking Service sends an initial INVITE with *Session-Expires: 3600; refresher: uac Min-SE: 600*. It means, as a user agent client, MTS Allstream SIP Trunking Service should refresh the Session Timer every 300 seconds by a reINVITE or UPDATE. In the compliance test, Communication Manager does not receive any Session Timer refresh signaling. This is a known issue on MTS Allstream SIP Trunking Service and there is no available resolution at this time.
7. **Mismatched Media Format setting on trunk group causes audio path being delayed**. Configure Direct IP-IP Early Media setting on Communication Manager SIP signaling group to MTS Allstream SIP Trunking Service to "n" and a setting of "y" for the general SIP signaling group used for internal enterprise, and make an outbound call to PSTN. The wireshark trace shows that the Media Format for audio from Communication Manager is different than the Media Format expected by MTS Allstream SIP Trunking Service even though it is configured to be the same. PSTN phone hears a series of tones and delayed audio from Communication Manager to PSTN. To avoid this problem, the Direct IP-IP Early Media setting on Communication Manager must be set identically on signaling group dedicated to MTS Allstream SIP Trunking Service and on the general signaling group used for the internal enterprise SIP endpoints.
8. **Fax over IP using G.711MU codec is successful**. Communication Manager does not officially support fax call using G.711MU codec. However, in the compliance test the incoming and outgoing fax calls appear to work with G.711MU codec by setting “fax=off” in codec profile. The fax document is transmitted successfully with acceptable quality. Communication Manager handles the fax call like a regular voice call using G.711MU codec and it only supports this fax call in best effort. Notes: the codec set should have G.711MU codec as a first choice otherwise the fax call would not be successfully established. The codec set is configured in **Section 5.4** and **Section 6.3.5**.
9. **Off-net call transfer, the calling party name and number is not updated to PSTN party**. A Communication Manager extension transfers an incoming call off-net to PSTN. Communication Manager sends UPDATE with true connected Calling Party Name and Number to both PTSN parties. The calling party information is in the URI-User and URI-Host portions of Contact header. However, the Calling Party Name and Number are not been updated; the calling and called PTSN parties still display Calling Party Number of Communication Manager extension. This is a known issue on MTS Allstream SIP Trunking Service. It is recommended that MTS Allstream SIP Trunking Service support

the calling party information update. This feature also needs to be supported by the service provider hosting the PSTN parties. This issue has low user impact, it is listed here simply as an observation.

10. **Outgoing SIP Trunk from Communication Manager is frequently out of service.** Configure an outgoing SIP trunk group on Communication Manager to send OPTIONS heartbeat to Avaya SBCE. This outgoing OPTIONS is forwarded by Avaya SBCE to MTS Allstream SIP Trunking Service. Due to WAN condition, the OPTIONS is frequently not being responded. This causes the SIP trunk group on Communication Manager to go out of service. To resolve this issue, configure **Enable Layer 3 Test?** to **n** on the trunk group configuration on Communication Manager as shown in **Section 5.6**. Also, the OPTIONS heartbeat is enabled on Avaya SBCE under Server Configuration for MTS Allstream SIP Trunking as shown in **Section 6.2.6.1**. This setting will let the Avaya SBCE keep the SIP Trunk up between the enterprise and the service provider by originating the outgoing OPTIONS heartbeat. Note: The Server Configuration for Communication Manager in **Section 6.2.6.2** has OPTIONS heartbeat disabled. This means that Avaya SBCE still forwards the incoming OPTIONS heartbeat from MTS Allstream SIP Trunking to Communication Manager but it does not originate OPTIONS heartbeat to Communication Manager.
11. **Local calls from a Communication Manager extension (Phone1) routed via MTS Allstream SIP Trunking Service to another Communication Manager extension (Phone2) results in no audio.** This problem is believed to have low user impact because all other local outgoing calls from the enterprise to PSTN (i.e. calls within the same area code) complete successfully with audio. Audio is only impacted when calling another Communication Manager via MTS Allstream SIP Trunking Service. In general, these calls would be routed within the enterprise which avoids the problem. This failure scenario is also related to shuffling because if shuffling is disabled on the service provider trunk then there is no audio issue. However, it is not recommended to disable shuffling and the failing scenario avoided by routing these types of calls within the enterprise.
12. **Performing an "Application Restart" or editing the SigMa script on Ayaya SBCE causes the SigMa script not working.** There is no resolution currently. If the SigMa script does not work after an "Application Restart" or editing, please contact Avaya SBCE support by telephone number 1-866-861-3113 or 1-214-269-2424.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

For technical support on MTS Allstream SIP Trunking Service, please contact MTS Allstream technical support at:

- Phone: 204-941-8557 or 1-800-542-8703
- Website: http://www.mts.ca/mts/personal/support

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution connected to the MTS Allstream SIP Trunking Service (Vendor Validation circuit) through a public Internet WAN connection.

For security purposes, the real public IP addresses and PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

The Avaya components used to create the simulated customer site included:

- Avaya S8800 Servers running Avaya Aura® Communication Manager
- Avaya G650 Media Gateway
- Avaya S8800 Server running Avaya Aura® Messaging
- Avaya Session Border Controller for Enterprise
- Avaya 9600-Series IP Telephones (H.323)
- Avaya one-X® Communicator soft phones (H.323)
- Avaya digital and analog telephones

Located at the edge of the enterprise network is Avaya SBCE. It has a public side that connects to MTS Allstream SIP Trunking Service via internet and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise network flows through the Avaya SBCE which can protect the enterprise against any outside SIP-based attacks. Avaya SBCE provides network address translation at both the IP and SIP layers. The transport protocol between the Avaya SBCE and MTS Allstream SIP Trunking Service across the public network is UDP; the transport protocol between the Avaya SBCE and Communication Manager across the enterprise network is TCP.

In the compliance testing, the Avaya CPE environment is configured with SIP domain "**avaya.com**" for the enterprise. Avaya SBCE is used to adapt the enterprise SIP domain to the IP address based URI-Host known to MTS Allstream SIP Trunking Service. **Figure 1** below illustrates the network diagram for the enterprise. All voice application elements are connected to internal trusted LAN.

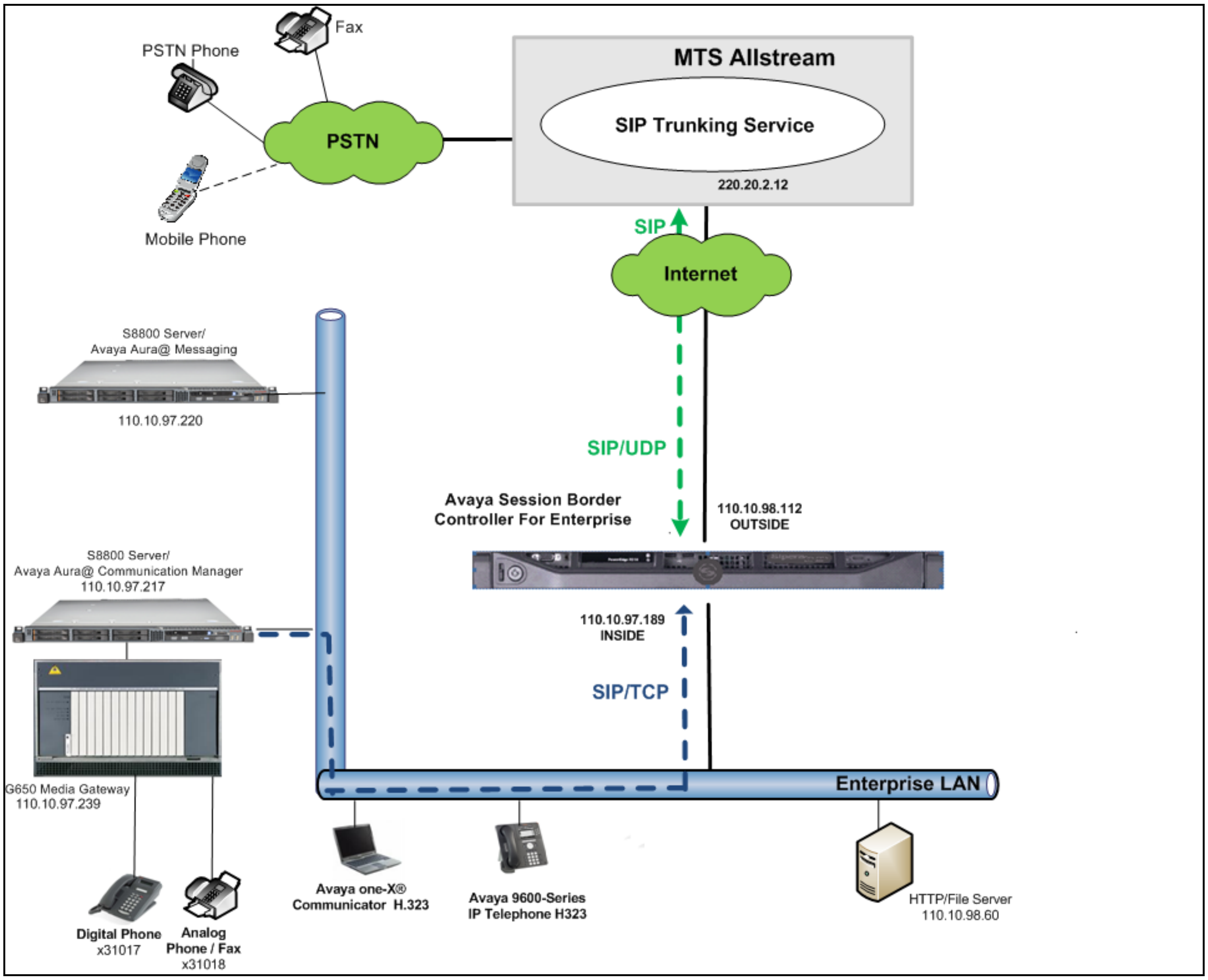


Figure 1: Avaya IP Telephony Network connecting to MTS Allstream SIP Trunking Service

4. Equipment and Software Validated

The following equipment and software are used for the sample configuration provided:

Avaya IP Telephony Solution Components	
Component	Release
Avaya Aura® Communication Manager running on an Avaya S8800 Server	6.0.1 (R016x.00.1.510.1-18621) (System Platform 6.0.3.0.3)
Avaya G650 Media Gateway IPSI TN2312BP CONTROL-LAN TN799DP MEDPRO TN2302 DIGITAL LINE TN2224 ANALOG LINE TN746B	 HW06 FW043 HW01 FW026 HW20 FW117 000006 000019
Avaya Aura® Messaging running on an Avaya S8800 Server	6.1-11.0
Avaya Session Border Controller for Enterprise	4.0.5 Q2
Avaya 9640 IP Telephone (H.323)	Avaya one-X® Deskphone Edition 6.0.1
Avaya one-X Communicator (H.323)	6.1.3.08-SP3-Patch2-35791
Avaya 1408 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
MTS Allstream SIP Trunking Service Components	
Component	Release
Genband S3	5.2.2.12
CS2K	CVM13

Table 1: Equipment and Software Tested

Note that this solution will be compatible with other Avaya Server and Media Gateway platforms running similar version of Communication Manager.

5. Configure Communication Manager

This section describes the procedure for configuring Communication Manager for inter-operating with the MTS Allstream SIP Trunking Service (MTS Allstream).

Two separate SIP trunk groups are created between Communication Manager and Avaya SBCE to carry traffic to and from service provider respectively. Inbound call flows from the MTS Allstream SIP Trunking Service to Avaya SBCE to Communication Manager. Once the call arrives at Communication Manager, further incoming call treatment, such as incoming digit translation and class of service restrictions may be performed. Outbound call to PSTN is first processed by Communication Manager for outbound feature treatment such as automatic route selection and class of service restrictions. Once Communication Manager selects the proper SIP trunk, the call is routed to Avaya SBCE for egress to the MTS Allstream SIP Trunking Service.

For the compliance test, Communication Manager sends 11 digits in the destination headers (e.g., Request-URI and To) and sends 10 digit in the source headers (e.g., From, Contact, and P-Asserted-Identity (PAI)). MTS Allstream sends 10 digits in destination headers and sent 11 digits in source headers.

It is assumed the general installation of the Communication Manager and the Avaya G650 Media Gateway has been previously completed and is not discussed here.

The Communication Manager configuration is performed using the System Access Terminal (SAT). Some screens in this section have been abridged for brevity and clarity in presentation.

5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to and from the service provider. The example shows that 24000 licenses are available and 222 are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sale representative to add the additional capacity or feature.

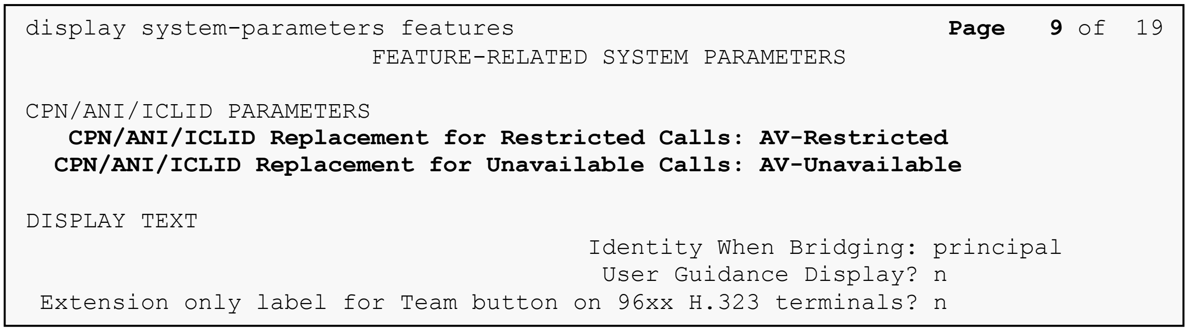
```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                  USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 5
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 2
                   Maximum Video Capable IP Softphones: 18000 3
                       Maximum Administered SIP Trunks: 24000 222
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

5.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to **all** to allow incoming call from PSTN to be transferred to another PSTN endpoint. If for security reasons, incoming call should not be allowed to transfer back to PSTN then leave the field set to **none**.

```
display system-parameters features                              Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                      Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                AAR/ARS Dial Tone Required? y
```

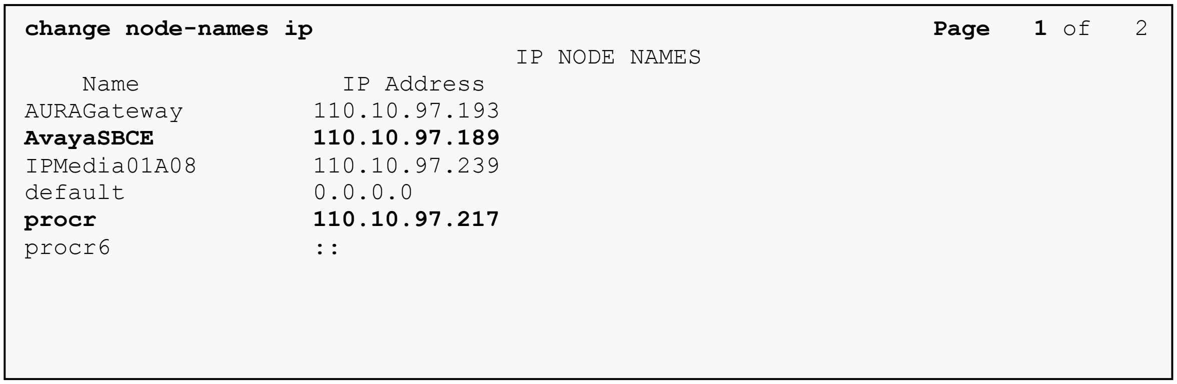
On **Page 9** verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. The compliance test used the value of **AV-Restricted** for restricted call and **AV-Unavailable** for unavailable call.



```
display system-parameters features                              Page   9 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
   CPN/ANI/ICLID Replacement for Restricted Calls: AV-Restricted
  CPN/ANI/ICLID Replacement for Unavailable Calls: AV-Unavailable

DISPLAY TEXT
                                       Identity When Bridging: principal
                                        User Guidance Display? n
 Extension only label for Team button on 96xx H.323 terminals? n
```

5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager **(procr**) and Avaya SBCE. These node names will be needed for defining the service provider signaling groups in **Section 5.6**.



```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AURAGateway         110.10.97.193
AvayaSBCE           110.10.97.189
IPMedia01A08        110.10.97.239
default             0.0.0.0
procr               110.10.97.217
procr6              ::
```

5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to be used for calls between the enterprise and the service provider. This compliance test used ip-codec-set 3. MTS Allstream supports G.729 and G.711MU. To use these codecs, enter G.711MU and G.729 in the **Audio Codec** column of the table in the order of preference. Default values can be used for all other fields.

The following screen shows the configuration for ip-codec-set 3. During testing, the codec set specifications are varied to test for individual codec support as well as codec negotiation between the enterprise and the network at call setup time.

```
change ip-codec-set 3                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 3

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n         2        20
 2: G.729            n         2        20
 3:
```

MTS Allstream only supports fax using G.711MU codec in the compliance test. The T.38 faxing is not currently supported. Communication Manager does not officially support fax call using G.711MU codec. However, incoming and outgoing fax call using G.711MU codec appear to work during testing when configuring fax = off. Communication Manager handles the call like a regular voice call and only supports fax call using G.711MU codec in best effort. **Note**: The codec profile should have G.711MU codec as a first choice otherwise the fax call would not be successfully established.

To use G.711MU codec for fax, set the **Fax Mode** to **off** on **Page 2**.

```
change ip-codec-set 3                                           Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? n



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

Note: To successfully send and receive fax using G.711MU codec, the voice call has to be established with G.711MU codec. The **Session Policy** configuration on Avaya SBCE in **Section 6.3.5** will make G.711MU the first choice in the preference codec.

5.5. IP Network Region

A separate IP network region for the service provider trunk groups is created. This allows separate codec or quality of service setting to be used (if necessary) for call between the enterprise and the service provider versus call within the enterprise or elsewhere. For the compliance test, **ip-network-region 3** is created by the **change ip-network-region 3** command with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In the compliance test, the domain name is **avaya.com** as assigned to the shared test environment in the Avaya test lab. This domain name appears in the "From" header of

SIP message originating from this IP region. Notes: Topology-Hiding configuration on Avaya SBCE in **Section 6.2.3** is used to convert this private domain name to the public IP Address of the Avaya SBCE for the URI-Host portion in From and PAI headers.

- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Set both **Intra-region** and **Inter-region IP-IP Direct Audio** to **yes.** This is the default setting. Shuffling can be further restricted at the trunk level under Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 3                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location: 1       Authoritative Domain: avaya.com
    Name: MTS Allstream
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              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
```

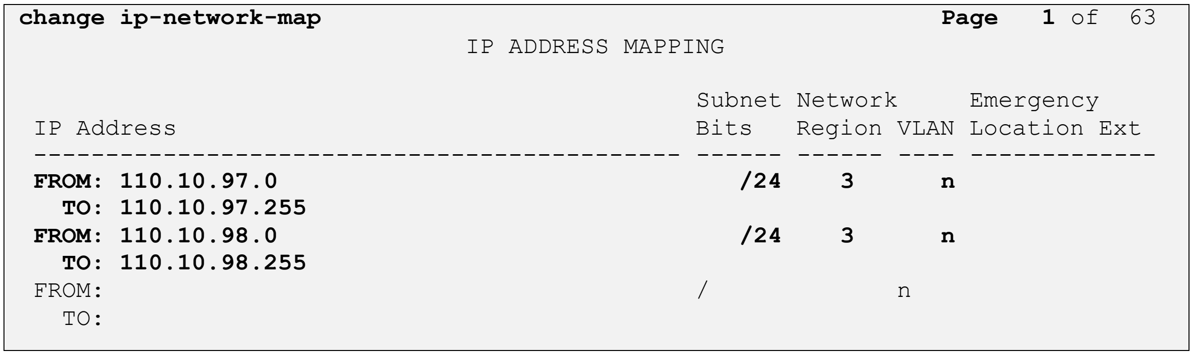
On **Page 4**, define the IP codec set to be used for traffic between region 3 and other regions. In the compliance test, Communication Manager, G650 Media Gateway, IP phones and Avaya SBCE are assigned to the same region 3. To configure IP codec set between regions, enter the desired IP codec set in the **codec set** column of the table with appropriate destination region (**dst rgn**). Default values may be used for all other fields. The example below shows codec set 3 will be used for call between region 3 and other regions.

```
change ip-network-region 1                                      Page   4 of  19

 Source Region: 3     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    3                                                                 all
 2    3     y    NoLimit                                             n       t
 3    3     y    NoLimit                                             n       t
 4
```

Non-IP telephones (e.g., analog, digital) derive network region from IP interface MEDPRO card of Avaya G650 Media Gateway to which the device is connected. IP telephones can be assigned a network region based on an IP address mapping.

For the compliance test, devices with IP addresses in the 110.10.97.0/24 subnet are assigned to network region 3. These include Communication Manager, MEDPRO card of Avaya G650 Media Gateway and Avaya SBCE that are set up for a shared test environment. IP telephones used for the compliance test, including both the Avaya 9600 IP Telephones and the Avaya one-X® Communicator soft phones, are assigned to network region 3 with IP address in the 110.10.98.0/24 subnet. In a production environment, different sites will typically be on different networks and ranges of IP addresses assigned by the DHCP scope serving the site. They can be entered as one entry in the network map to assign all telephones in a range to a specific network region.

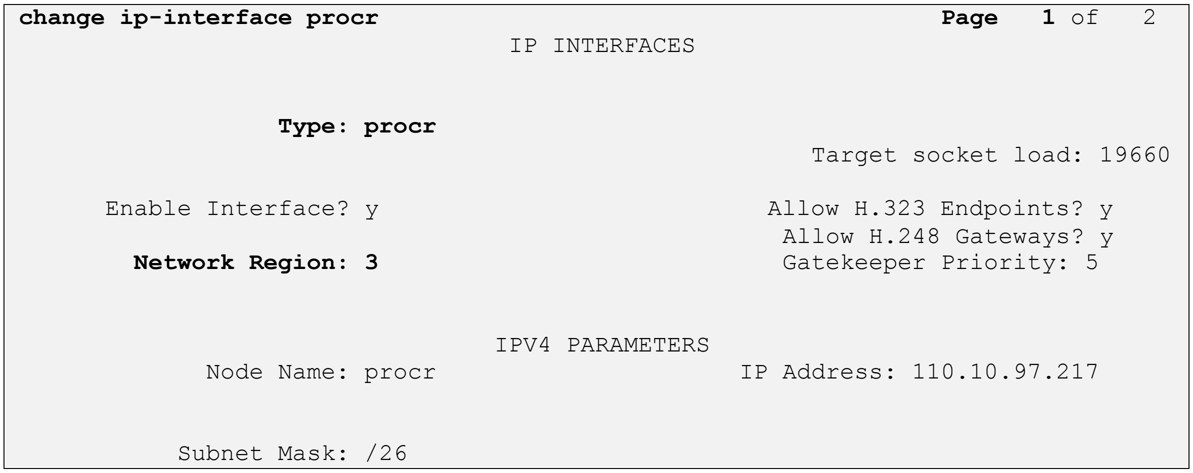
The following screen illustrates a subset of the IP network map configuration used to verify in these Application Notes.



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

                                                Subnet Network     Emergency
 IP Address                                     Bits   Region VLAN Location Ext
 ---------------------------------------------- ------ ------ ---- -------------
 FROM: 110.10.97.0                                 /24    3      n
   TO: 110.10.97.255
 FROM: 110.10.98.0                                 /24    3      n
   TO: 110.10.98.255
 FROM:                                          /             n
   TO:
```

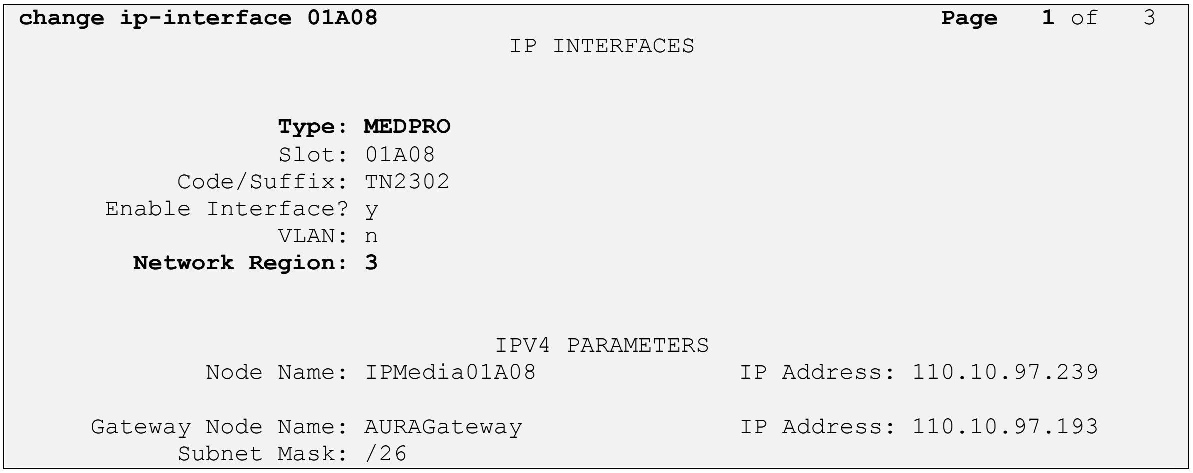
Next step is to allocate the resource on Communication Manager for IP network region 3. For SIP Trunk, the IP interface **procr** is used to process SIP signaling and MEDPRO card of Avaya G650 Media Gateway is used to process media. Both IP interfaces have to be assigned under the same region.

To defined network region 3 for ip interface **procr**, used **change ip-interface procr** command as shown in the following screen.



```
change ip-interface procr                                       Page   1 of   2
                                  IP INTERFACES


                  Type: procr
                                                       Target socket load: 19660

      Enable Interface? y                            Allow H.323 Endpoints? y
                                                     Allow H.248 Gateways? y
        Network Region: 3                            Gatekeeper Priority: 5


                                 IPV4 PARAMETERS
             Node Name: procr                     IP Address: 110.10.97.217


           Subnet Mask: /26
```

To defined network region 3 for ip interface of MEDPRO card, used **change ip-interface 01A08** command as shown in the following table. **Note**: 01A08 is the physical position of the MEDPRO card on the Avaya G650 Media Gateway; it is at cabinet 1, carrier A and slot 8.



```
change ip-interface 01A08                                       Page   1 of   3
                                  IP INTERFACES


                  Type: MEDPRO
                  Slot: 01A08
           Code/Suffix: TN2302
      Enable Interface? y
                  VLAN: n
        Network Region: 3


                                 IPV4 PARAMETERS
             Node Name: IPMedia01A08              IP Address: 110.10.97.239

     Gateway Node Name: AURAGateway               IP Address: 110.10.97.193
           Subnet Mask: /26
```

5.6. Signaling Group

Use the **add signaling-group** command to create two signaling groups between Communication Manager and Avaya SBCE, one for inbound calls from the service provider network and other for outgoing calls from the enterprise.

For the compliance test, signaling group 6 is created inbound calls from the service provider and is configured as follows:

- Set the **Group Type** field to **sip**.
- Set the **IMS Enabled** field to **n**.
- Set the **Transport Method** to the recommended default value of **tls** (Transport Layer Security). For ease of troubleshooting during testing, the compliance test is conducted with the **Transport Method** set to **tcp** The transport method specified here is used between the Communication Manager and Avaya SBCE.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061). The compliance test is conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to **5080** (the well-known port value for TCP is 5060).
- Set the **Peer Detection Enabled** field to **n**.
- Set the **Near-end Node Name** to **procr**. This node name maps to the IP interface of **procr** defined in **Section 5.3**.
- Set the **Far-end Node Name** to **AvayaSBCE**. This node name maps to the IP address of Avaya SBCE as defined in **Section 5.3**.
- Set the **Far-end Network Region** to the IP network region 3 defined for the service provider in **Section 5.5**.
- Set the **Far-end Domain** to blank.
- Set the **DTMF over IP field** to **rtp-payload**. This setting enables Communication Manager to send DTMF transmissions using RFC 2833.
- Set **Enable Layer 3 Test?** to **n**. This setting disables Communication Manager to send out OPTIONS heartbeat to Avaya SBCE on the incoming trunk group. Communication Manager will use ICMP ping as an alternative to check the status of the far end system. Please refer to **Section 2.2**, observation #10 for detail information.
- Set **Direct IP-IP Audio Connections** to **y**. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the SIP trunk and the enterprise endpoint. If this value is set to **n**, then Avaya Media Gateway will remain in the media path between the SIP trunk and the endpoint for the duration of the call. Depending on the number of media resources available in the Avaya Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- Verify that the **Direct IP-IP Early Media** is set to the same value of the signaling group used for the enterprise site. The default setting for this field is **n**. See the **Media Format** bullet item in **Section 2.2** observation #7 for more information.

- Change default setting of **6** for **Alternate Route Timer (sec)** to **12**. This allows more time for inbound PSTN calls to complete through MTS Allstream.
- Default values may be used for all other fields.

```
add signaling-group 6                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 6              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                             SIP Enabled LSP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? n  Peer Server: Others



   Near-end Node Name: procr             Far-end Node Name: AvayaSBCE
 Near-end Listen Port: 5080                Far-end Listen Port: 5080
                                        Far-end Network Region: 3

Far-end Domain:
                                             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? n                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 12
```

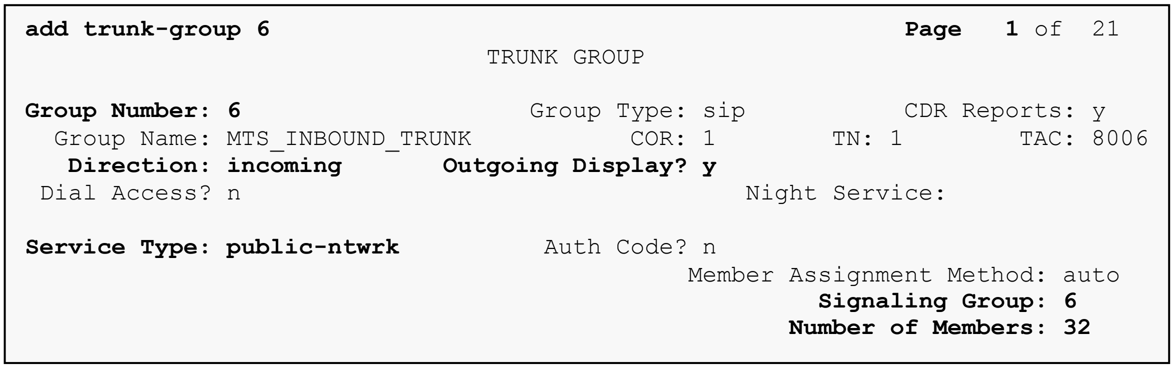
The signaling group for outbound calls from the enterprise to PSTN is similarly configured except that the Far-end Domain is set to **avaya.com**. This domain will be manipulated by Avaya SBCE to IP address in URI-Host that is known to MTS Allstream. For the compliance test, signaling group 7 is created and is shown below.

```
add signaling-group 7                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 7              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                             SIP Enabled LSP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? n  Peer Server: Others



   Near-end Node Name: procr             Far-end Node Name: AvayaSBCE
 Near-end Listen Port: 5080                Far-end Listen Port: 5080
                                        Far-end Network Region: 3

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? n                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 12
```

5.7. Trunk Group

Use the **add trunk-group** command to create trunk group for the two signaling groups created in **Section 5.6**. For the compliance test, trunk group 6 is configured for incoming calls and trunk group 7 is configured for outgoing calls as follows:

- Set the **Group Type** field to **sip**.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Direction** field to **incoming** for trunk group 6 and **outgoing** for trunk group 7.
- Set the **Outgoing Display** to **y** to enable name display on the trunk.
- Set the **Service Type** field to **public-ntwrk**.
- Set the **Signaling Group** to the signaling group shown in **Section 5.6**. The incoming trunk group 6 is set to incoming signaling group 6 and the outgoing trunk group 7 is set to outgoing signaling group 7.
- Set the **Number of Members** field to the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk.
- Default values are used for all other fields.



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

Group Number: 6                    Group Type: sip           CDR Reports: y
  Group Name: MTS_INBOUND_TRUNK           COR: 1        TN: 1        TAC: 8006
   Direction: incoming       Outgoing Display? y
 Dial Access? n                                   Night Service:

Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 6
                                                     Number of Members: 32
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval (sec)** is set to a value acceptable to service provider. This value defines the interval a re-INVITEs must be sent to keep an active session alive. For the compliance test, a default value of **600** seconds is used.

```
add trunk-group 6                                           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

 Disconnect Supervision - In? y
```

On **Page 3**, set the **Numbering Format** field to **private**. This field specifies the format of the calling party number (CPN) sent to the far-end. Beginning with Communication Manager 6.0, public numbers are automatically preceded with a + sign when passed in the From, Contact and P-Asserted Identity headers. The addition of the + sign impacted interoperability with service provider. Thus, the **Numbering Format** is set to **private** and the **Numbering Format** in the route pattern is set to **unk-unk** (see **Section 5.9**).

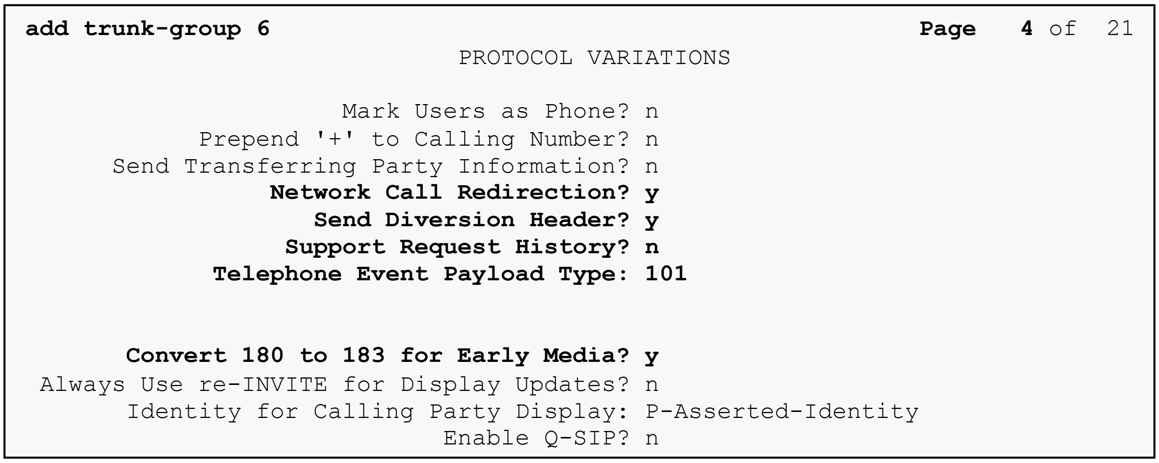
Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to **y**. This will allow the CPN displayed on local endpoint to be replaced with the value set in **Section 5.2**, if inbound call enabled CPN block. For outbound calls, these same settings request that CPN block be activated on the far-end destination if a local user requests CPN block on a particular call routed out from this trunk. Default values are used for all other fields.

```
add trunk-group 6                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y



                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y







 Show ANSWERED BY on Display? y
```

On **Page 4**, the **Network Call Redirection** field can be set to **y**. This setting the **Network Call Redirection** flag to **y** enables use of the SIP REFER message to transfer an incoming call to a vector number back to PSTN. For more information, please refer to **Section 2.2**, observation #1.

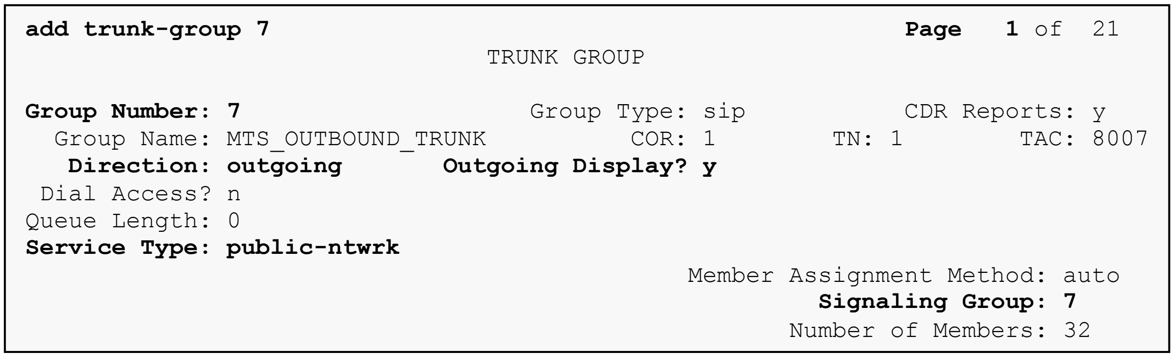
Note: The outgoing trunk group 7 in the later discussion will have **Network Call Redirection** set to **n**, this setting is to use reINVITE to off-net transfer an incoming call back to PSTN. For more information, please refer to **Section 2.2**, observation #1.

- Set the **Send Diversion Header** field to **y**. This field provides additional information to the network if the call has been re-directed. This is needed to support call forwarding of inbound call back to PSTN and Extension to Cellular (EC500) call scenario.
- Set the **Support Request History** field to **n**. This parameter determines if History-Info header will be excluded in the call-redirection INVITE from the enterprise.
- Set the **Telephone Event Payload Type** to **101**, the value is preferred by MTS Allstream.
- Set the **Convert 180 to 183 for Early Media** field to **y**.



```
add trunk-group 6                                               Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? y
                    Send Diversion Header? y
                  Support Request History? n
             Telephone Event Payload Type: 101


       Convert 180 to 183 for Early Media? y
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: P-Asserted-Identity
                             Enable Q-SIP? n
```

For outgoing trunk group configuration, the screen below shows **Page 1** of trunk group 7 for outgoing calls from the enterprise to MTS Allstream.



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

Group Number: 7                  Group Type: sip           CDR Reports: y
  Group Name: MTS_OUTBOUND_TRUNK        COR: 1        TN: 1        TAC: 8007
   Direction: outgoing       Outgoing Display? y
 Dial Access? n
Queue Length: 0
Service Type: public-ntwrk
                                                 Member Assignment Method: auto
                                                          Signaling Group: 7
                                                        Number of Members: 32
```

On **Page 4** of trunk group 7, the **Network Call Redirection** is set to “**n**”.

Note: **Network Call Redirection** is set to “**n**”. This setting is used to reINVITE an off-net transfer for an incoming call back to PSTN. For more information, please refer to **Section 2.2**, observation #1.

```
add trunk-group 7                                                 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? y
                     Support Request History? n
                Telephone Event Payload Type: 101
```

The configuration on other pages of trunk group 7 is identical to trunk group 6.

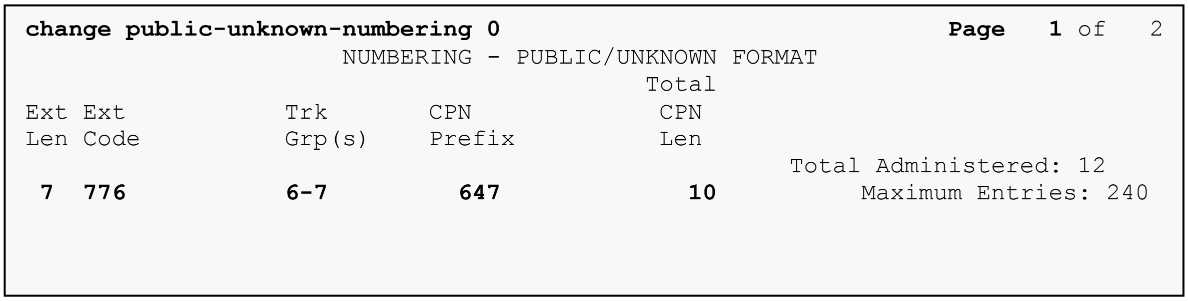
5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since private numbering is selected to define the format of this number (**Section 5.7**), use the **change private-numbering** command to create an entry for each extension which has a DID assigned. The DID numbers are provided by service provider. They are used to authenticate the caller.

Normally DID number is comprised of the local extension plus a prefix. A single private numbering entry can be applied for all extensions. In the example below, all stations with a 7-digit extension beginning with **776** when receiving or calling call on trunk group 6 or 7 will send the calling party number as the **Private Prefix** plus the extension number.

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

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 7  776              6-7          647              10    Total Administered: 4
                                                        Maximum Entries: 540
```

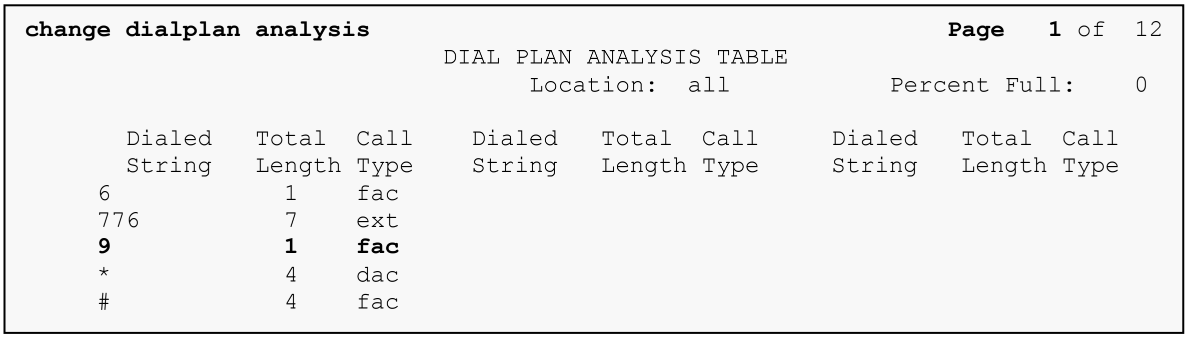
Even though private numbering is selected, currently the number used in the SIP Diversion header is derived from the public unknown numbering table and not the private numbering table. As a workaround for this, the entries in the private numbering table must be repeated in the public unknown numbering table.



```
change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                            Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                      Total Administered: 12
 7  776           6-7         647             10           Maximum Entries: 240
```

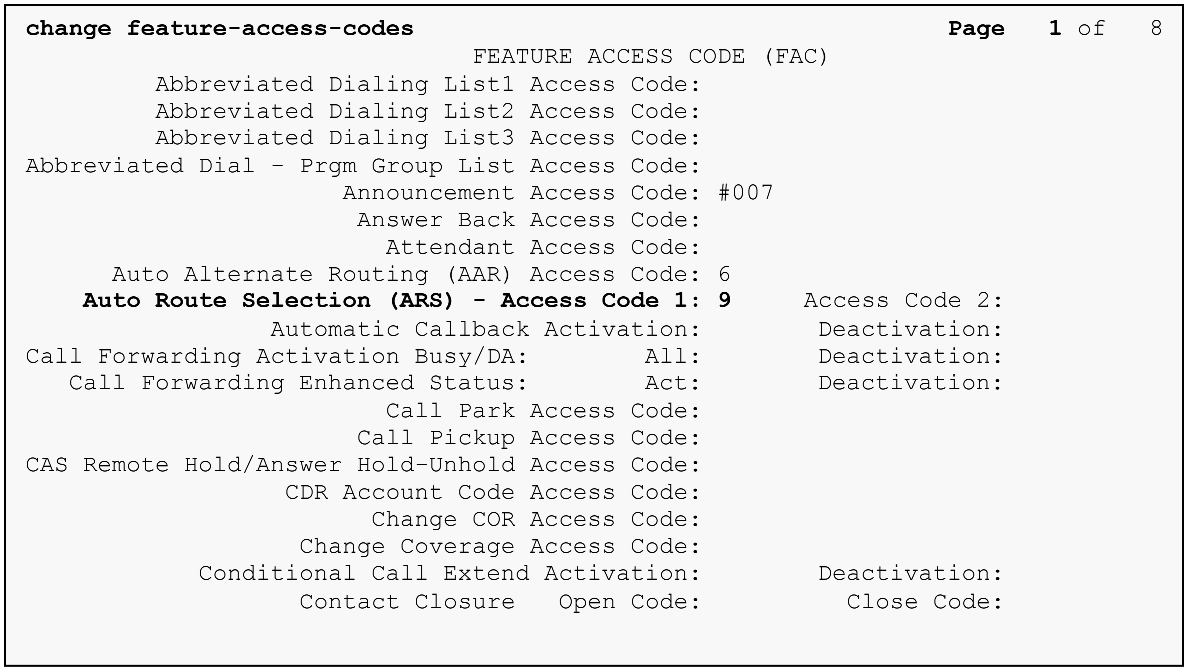
5.9. Outbound Routing

In these Application Notes, the **Automatic Route Selection** (ARS) feature is used to route outbound call via the SIP trunk to service provider. In the compliance test, a single digit 9 is used as the ARS access code. Enterprise caller will dial 9 to reach an outside line. To define feature access code (**fac**) 9, use the **change dialplan analysis** command as shown in the table below.



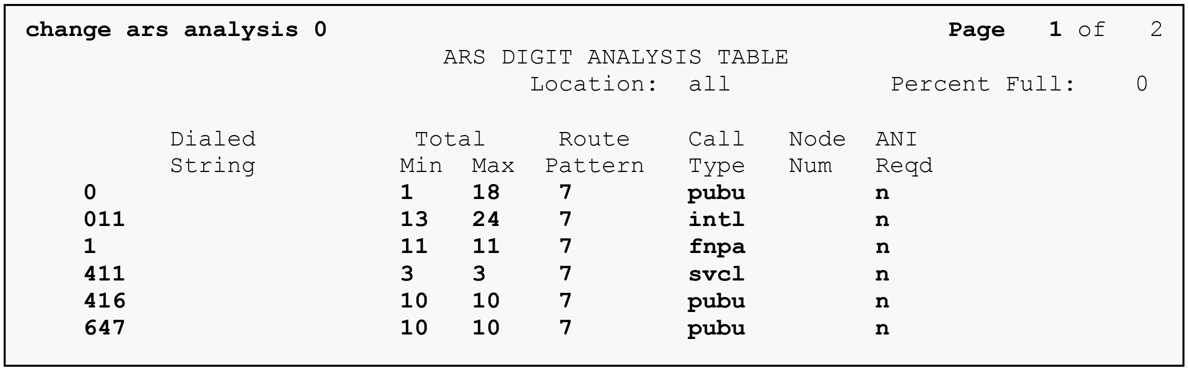
```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all         Percent Full:    0

       Dialed    Total  Call     Dialed    Total  Call      Dialed    Total  Call
       String    Length Type     String    Length Type      String    Length Type
     6             1    fac
     776           7    ext
     9             1    fac
     *             4    dac
     #             4    fac
```

Use the **change feature-access-codes** command to define 9 as the **Auto Route Selection (ARS) – Access Code 1**.



```
change feature-access-codes                                     Page   1 of   8
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: #007
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 6
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All:        Deactivation:
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

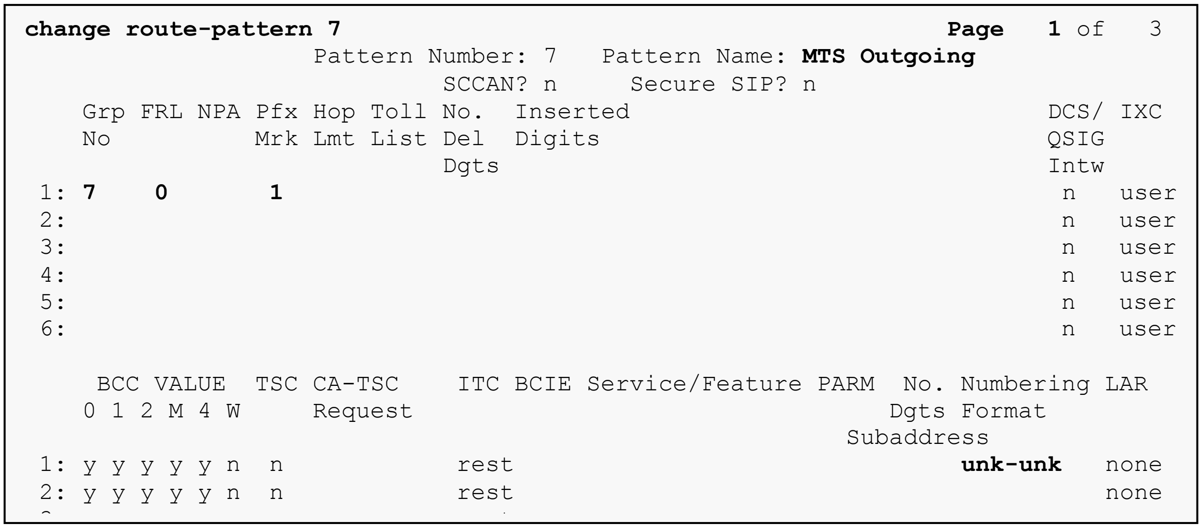
Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example below shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to route pattern 7 for outbound call which contains the SIP trunk to the service provider (as defined next).



```
change ars analysis 0                                           Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    0                     1    18    7        pubu         n
    011                   13   24    7        intl         n
    1                     11   11    7        fnpa         n
    411                   3    3     7        svcl         n
    416                   10   10    7        pubu         n
    647                   10   10    7        pubu         n
```

As being mentioned above, the route pattern defines which trunk group will be used for the outgoing calls and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for route pattern 7 in the following manner.

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the SIP service provider. For the compliance test, trunk group 7 is used.
- **FRL**: Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **Pfx Mrk**: The prefix mark (**Pfx Mrk**) of 1 will prefix any FNPA 10-digit number with a 1 and leave numbers of any other length unchanged. This will ensure 1 + 10 digits are sent to the service provider for the long distance North American Numbering Plan (NANP) numbers. All HNPA 10 digit numbers are left unchanged.
- **Numbering Format**: **unk-unk**. All calls using this route pattern will use the private numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.8**.



```
change route-pattern 7                                            Page   1 of   3
                    Pattern Number: 7   Pattern Name: MTS Outgoing
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 7    0       1                                                     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                               unk-unk   none
 2: y y y y y n  n            rest                                         none
```

5.10. Incoming Call Handling

When an incoming call arrives, Communication Manager applies incoming handling treatment on incoming trunk group 6 (created in **Section 5.7**). MTS Allstream sends 10 digits in Request-URI and To headers identically to the assigned DID number. The incoming call handling treatment will translate this DID number to an extension. In the compliance test, the DID numbers have prefix 647 which are deleted to normalize the incoming number to match 7 digits extension on Communication Manager.

Use the **inc-call-handling-trmt trunk-group** command to define an incoming handling for MTS Allstream. Following table shows the configuration in detail on incoming trunk group 6.

```
change inc-call-handling-trmt trunk-group 6                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 647776           3
```

5.11. Saving Communication Manager Configuration Changes

The command "**save translation all**" can be used to save the configuration changes made on Communication Manager.

6. Configure Avaya Session Border Controller for Enterprise

This section covers the configuration of Avaya Session Border Controller for Enterprise (Avaya SBCE). It is assumed that the software has already been installed. For additional information on these configuration tasks, see Reference [**9**] and [**10**].

The compliance test comprises of configuration for two major components, trunk server for service provider and call server for the enterprise. Each component consists of a set of Global Profiles, Domain Policies and Device Specific Settings. The configuration is performed using the Avaya SBCE web user interface as described in the following sections.

Trunk server configuration elements for service provider MTS Allstream:

- Global Profiles:
    - URI Groups
    - Routing
    - Topology Hiding
    - Server Interworking
    - Signaling Manipulation
    - Server Configuration
- Domain Policies:
    - Application Rules
    - Media Rules
    - Signaling Rules
    - Endpoint Policy Group
    - Session Policy
- Device Specific Settings:
    - Network Management
    - Media Interface
    - Signaling Interface
    - End Point Flows → Server Flows
    - Session Flows

Call server configuration elements at the enterprise for Communication Manager:

- Global Profiles:
    - URI Groups
    - Routing
    - Topology Hiding
    - Server Interworking
    - Server Configuration
- Domain Policies:
    - Application Rules
    - Media Rules
    - Signaling Rules

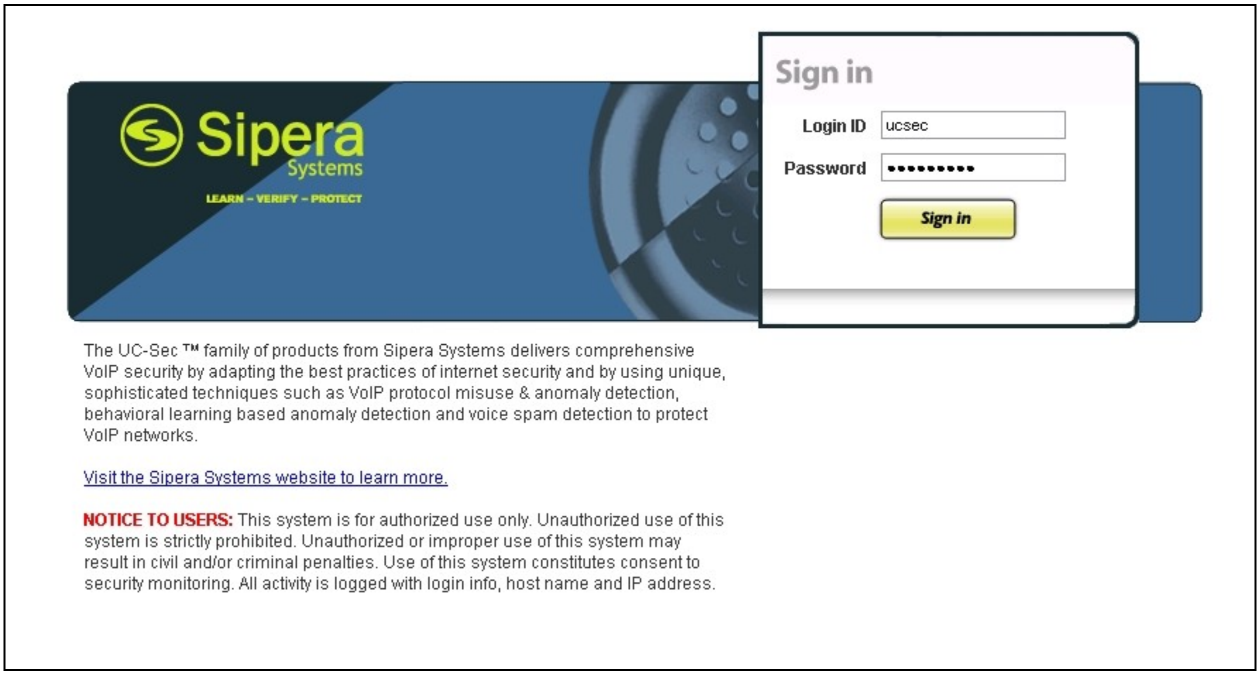
- Endpoint Policy Group
- Session Policy

- Device Specific Settings:
  - Network Management
  - Media Interface
  - Signaling Interface
  - End Point Flows → Server Flows
  - Session Flows

6.1. Avaya Session Border Controller for Enterprise Login

Use a Web browser to access the Unify Communication Security (UC-Sec) web interface, enter https://<ip-addr>/ucsec in the address field of the web browser (not shown), where <ip-addr> is the management LAN IP address of UC-Sec.

Enter appropriate credentials and click ***Sign In***.



Login ID: ucsec

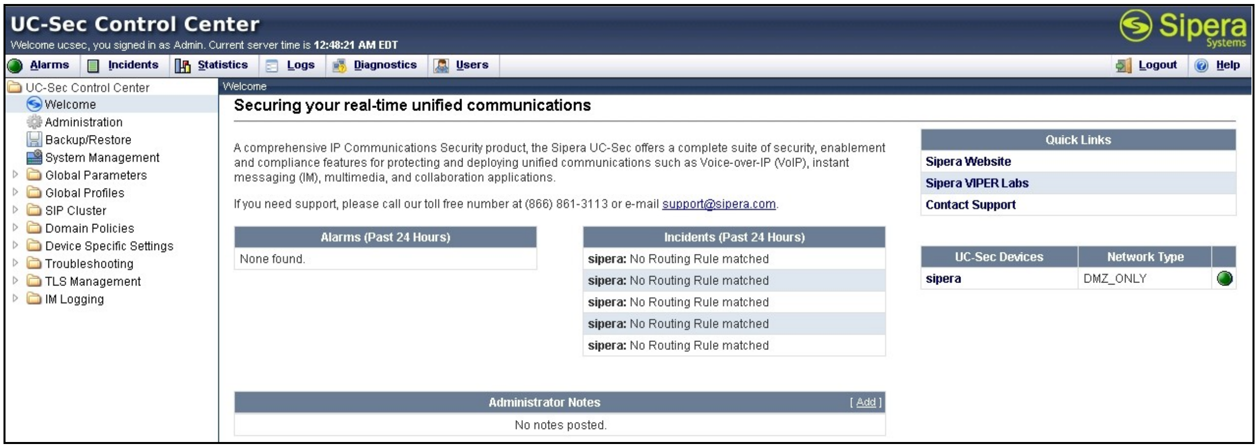
Password: •••••••••

The UC-Sec ™ family of products from Sipera Systems delivers comprehensive VoIP security by adapting the best practices of internet security and by using unique, sophisticated techniques such as VoIP protocol misuse & anomaly detection, behavioral learning based anomaly detection and voice spam detection to protect VoIP networks.

Visit the Sipera Systems website to learn more.

NOTICE TO USERS: This system is for authorized use only. Unauthorized use of this system is strictly prohibited. Unauthorized or improper use of this system may result in civil and/or criminal penalties. Use of this system constitutes consent to security monitoring. All activity is logged with login info, host name and IP address.

The main page of the **UC-Sec Control Center** will appear as shown below.



Welcome ucsec, you signed in as Admin. Current server time is 12:48:21 AM EDT

Securing your real-time unified communications

A comprehensive IP Communications Security product, the Sipera UC-Sec offers a complete suite of security, enablement and compliance features for protecting and deploying unified communications such as Voice-over-IP (VoIP), instant messaging (IM), multimedia, and collaboration applications.

If you need support, please call our toll free number at (866) 861-3113 or e-mail support@sipera.com.

Alarms (Past 24 Hours)
None found.

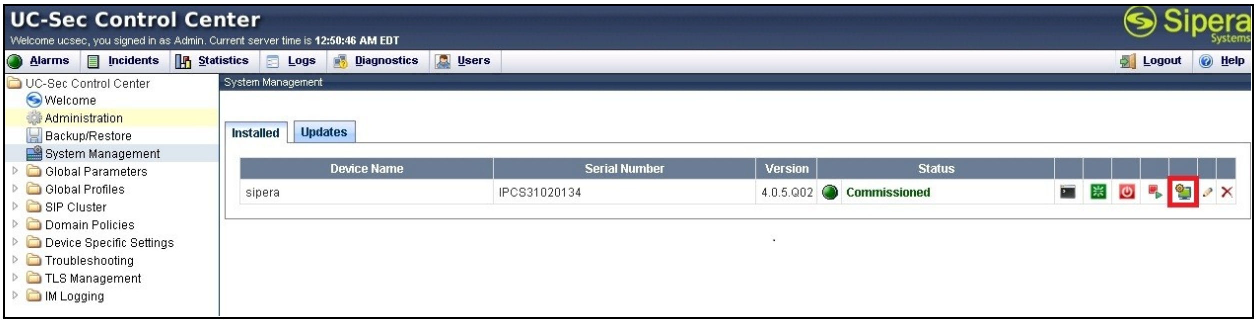
Incidents (Past 24 Hours)
sipera: No Routing Rule matched
sipera: No Routing Rule matched
sipera: No Routing Rule matched
sipera: No Routing Rule matched
sipera: No Routing Rule matched

Quick Links
Sipera Website
Sipera VIPER Labs
Contact Support

UC-Sec Devices	Network Type	
sipera	DMZ_ONLY	●

Administrator Notes [Add]
No notes posted.

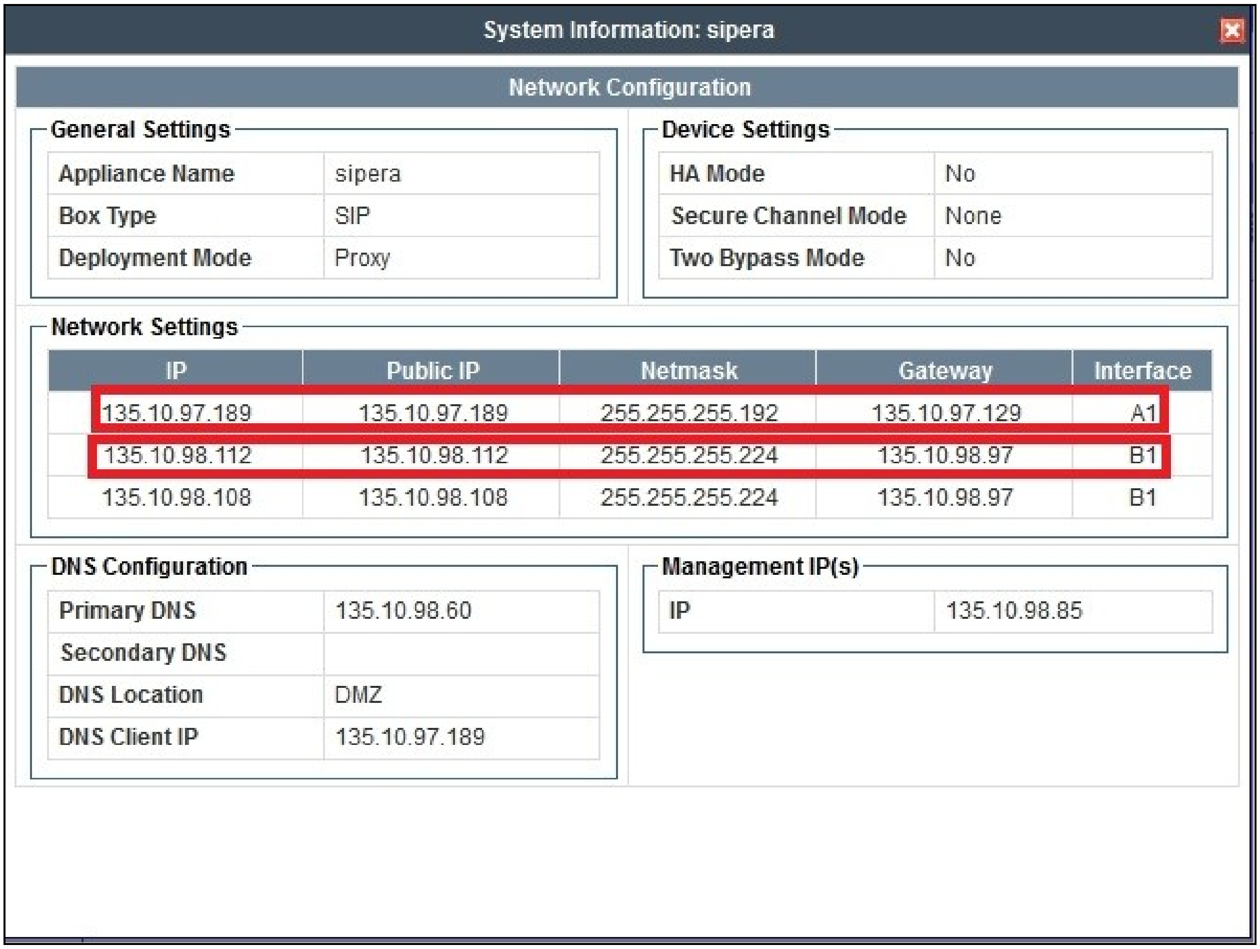
To view system information that has been configured during installation, navigate to **UC-Sec Control Center → System Management**. A list of installed devices is shown in the right pane. In the Compliance test, a single device named **sipera** is added. To view the configuration of this device, click the **View Config** icon (the third icon from the right) as shown below.



Welcome ucsec, you signed in as Admin. Current server time is 12:50:46 AM EDT

Device Name	Serial Number	Version	Status
sipera	IPCS31020134	4.0.5.Q02	● Commissioned

The **System Information** screen shows **Network Settings, DNS Configuration** and **Management IP** information provided during installation and corresponds to **Figure 1**. The **Box Type** is set to **SIP** and the **Deployment Mode** is set to **Proxy**. Default values are used for all other fields.



System Information: sipera

Network Configuration

General Settings

Appliance Name	sipera
Box Type	SIP
Deployment Mode	Proxy

Device Settings

HA Mode	No
Secure Channel Mode	None
Two Bypass Mode	No

Network Settings

IP	Public IP	Netmask	Gateway	Interface
135.10.97.189	135.10.97.189	255.255.255.192	135.10.97.129	A1
135.10.98.112	135.10.98.112	255.255.255.224	135.10.98.97	B1
135.10.98.108	135.10.98.108	255.255.255.224	135.10.98.97	B1

DNS Configuration

Primary DNS	135.10.98.60
Secondary DNS	
DNS Location	DMZ
DNS Client IP	135.10.97.189

Management IP(s)

IP	135.10.98.85

6.2. Global Profiles

Global Profiles allows for configuration of parameters across all UC-Sec appliances.

6.2.1. Uniform Resource Identifier (URI) Groups

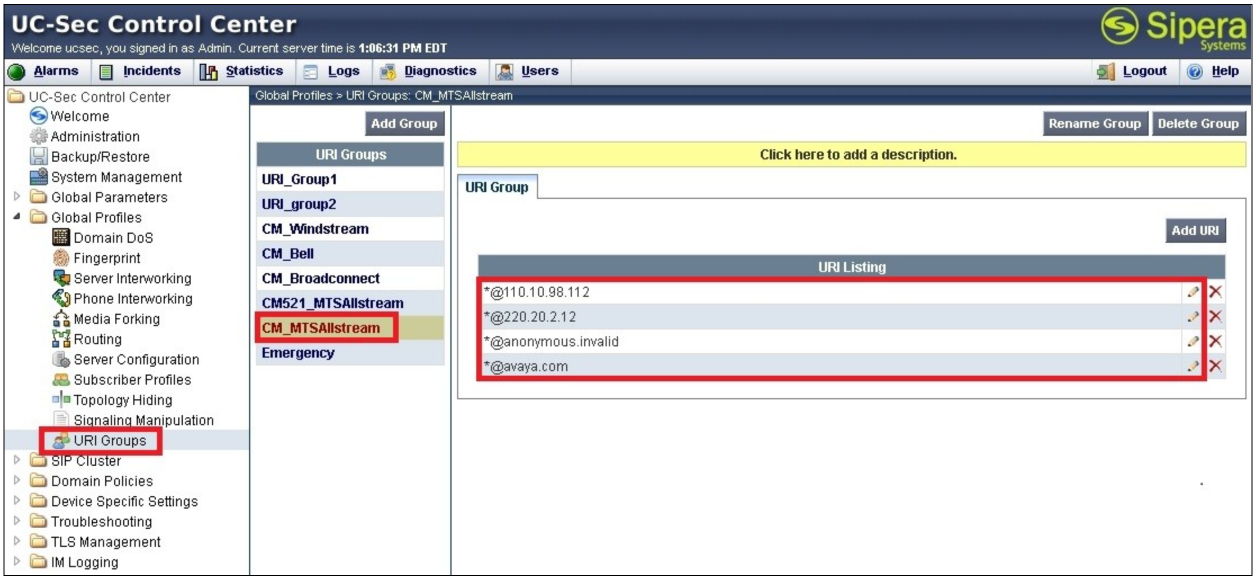
The **URI Group** feature allows user to create any number of logical URI groups that are comprised of individual SIP subscribers located in that particular domain or group. These groups are used by the various domain policies to determine which actions (Allow, Block, or Apply Policy) should be used for a given call flow.

To add an **URI Group**, select **UC-Sec Control Center → Global Profiles → URI Groups**. Click on **Add Group** (not shown).

In the compliance test, a URI Group named **CM_MTSAllstream** is added with URI type **Plain** (not shown) and consists of four domain *@anonymous.invalid, *@avaya.com, *@110.10.98.112 and *@220.20.2.12. Domain "anonymous.invalid" is defined for private calls received either from call server or trunk server had URI-Host masked by "anonymous.invalid". The enterprise domain name "avaya.com" is for SIP Trunk domain defined in **Section 5.5** between Communication Manager and Avaya SBCE. For the public SIP Trunk between Avaya SBCE and MTS Allstream, the Avaya SBCE public IP address 110.10.98.112 is set as URI-Host of From, PAI and Diversion headers while the public IP address of MTS Allstream 220.20.2.12 is set as URI-Host of Request-URI and To headers.

This URI-Group is used to match the From and To headers in a SIP call dialog received from both Communication Manager and MTS Allstream. If there is a match, the Avaya SBCE will apply the appropriate **Routing Profile** and **Server Flow** to route the inbound and outbound call to the right destination. The **Routing Profile** and **Server Flow** are configured in **Section 6.2.2** and **Section 6.4.4** appropriately.

The screenshot below illustrates the URI Listing for URI Group **CM_MTSAllstream**.



URI Listing
*@110.10.98.112
*@220.20.2.12
*@anonymous.invalid
*@avaya.com

6.2.2. Routing Profiles

Routing Profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by **Routing Profiles** include packet transport settings, name server addresses and resolution methods, next hop routing information and packet transport types.

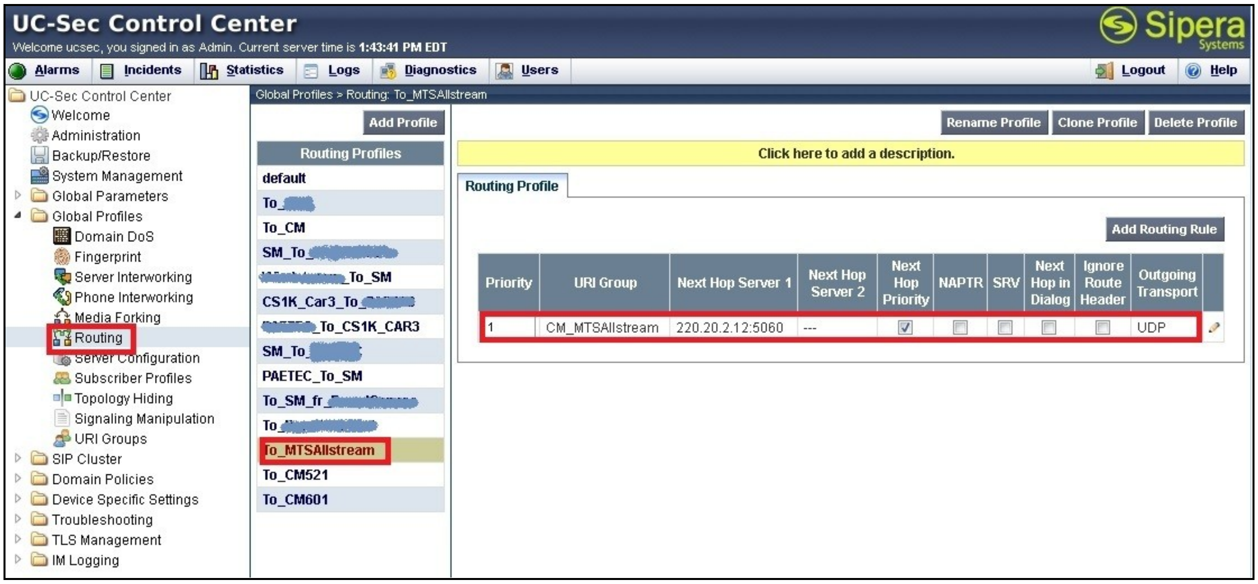
To create a **Routing Profile**, select **UC-Sec Control Center → Global Profiles → Routing**. Click on **Add Profile** (not shown).

In the compliance test, a **Routing Profile** named **To_MTSAllstream** is created to be used in conjunction with the server flow defined for Communication Manager. This entry is to route the outgoing call from the enterprise to MTS Allstream.

In the opposite direction, a **Routing Profile** named **To_CM601** is created to be used in conjunction with the server flow defined for MTS Allstream This entry is to route the incoming call from MTS Allstream to the enterprise.

6.2.2.1 Routing Profile for MTS Allstream

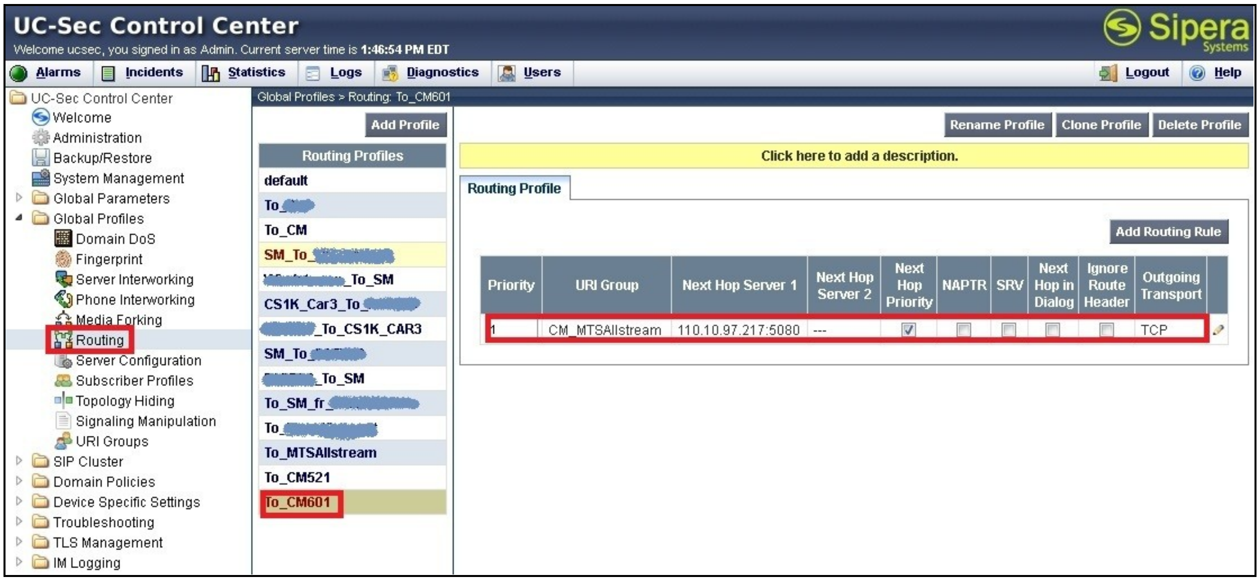
The screenshot below illustrate the **UC-Sec Control Center → Global Profiles → Routing**: **To_MTSAllstream**. As shown in **Figure 1**, MTS Allstream SIP Trunk is connected with transportation protocol **UDP**. If there is a match in the **To** header of the **CM_MTSAllstream** URI Group defined in **Section 6.2.1**, the call will be routed to the **Next Hop Server 1** which is the IP address of MTS Allstream SIP Trunk on port **5060**.



Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	CM_MTSAllstream	220.20.2.12:5060	---	☑	☐	☐	☐	☐	UDP

6.2.2.2 Routing Profile for Communication Manager

The routing profile **To_CM601** is defined to route call where the **To** header matches the URI-Group **CM_MTSAllstream** defined in **Section 6.2.1** to **Next Hop Server 1** which is the IP address of Communication Manager, on port **5080** as a destination. As shown in **Figure 1**, SIP Trunk between Communication Manager and Avaya SBCE is connected with transportation protocol **TCP**.



Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	CM_MTSAllstream	110.10.97.217:5080	---	☑	☐	☐	☐	☐	TCP

6.2.3. Topology Hiding

Topology Hiding is an Avaya SBCE security feature which allows changing certain key SIP message parameters to 'hide' or 'mask' how the enterprise network may appear to an unauthorized or malicious user.

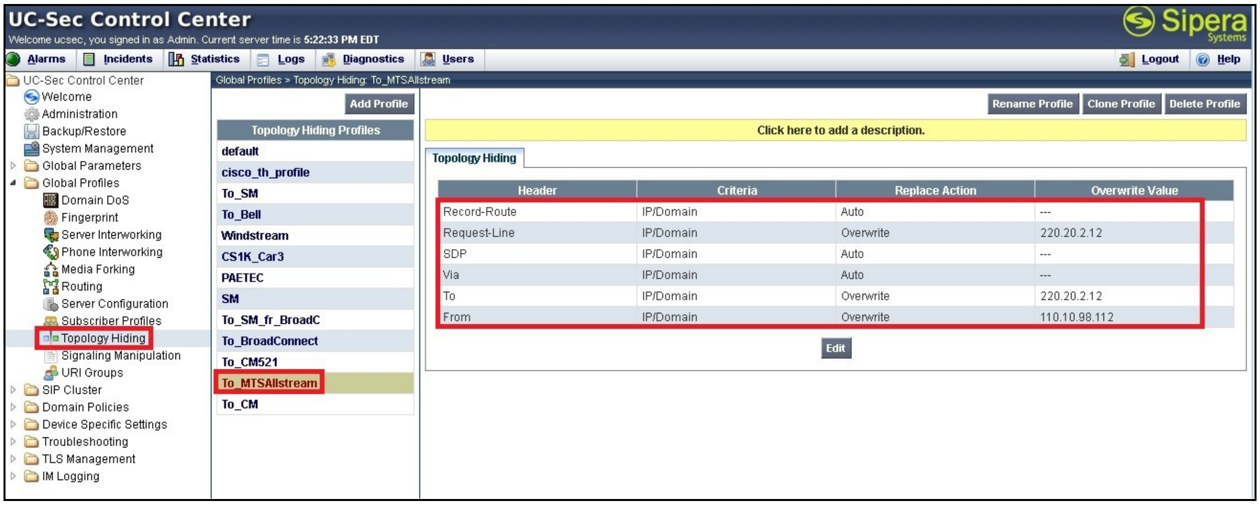
To create a **Topology Hiding** profile, select **UC-Sec Control Center → Global Profiles → Topology Hiding**. Click on **Add Profile** (not shown).

In the compliance test, two Topology Hiding profiles **To_MTSAllstream** and **To_CM** are created.

6.2.3.1 Topology Hiding Profile for MTS Allstream

Profile **To_MTSAllstream** is defined to mask the enterprise SIP domain **avaya.com** in Request-URI and To headers to IP **220.20.2.12** (the IP address MTS Allstream uses as URI-Host portion for Request-URI and To headers to meet the SIP specification requirement of MTS Allstream); mask the enterprise SIP domain **avaya.com** in From header to IP 110.10.98.112 (Avaya SBCE public IP address); and replace Record-Route, Via headers and SDP added by Communication Manager by external IP address known to MTS Allstream. It is to secure the enterprise network topology and also to meet the SIP requirement from service provider.

The screenshots below illustrate the **Topology Hiding** profile **To_MTSAllstream**.



Welcome ucsec, you signed in as Admin. Current server time is 5:22:33 PM EDT

Header	Criteria	Replace Action	Overwrite Value
Record-Route	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	220.20.2.12
SDP	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
To	IP/Domain	Overwrite	220.20.2.12
From	IP/Domain	Overwrite	110.10.98.112

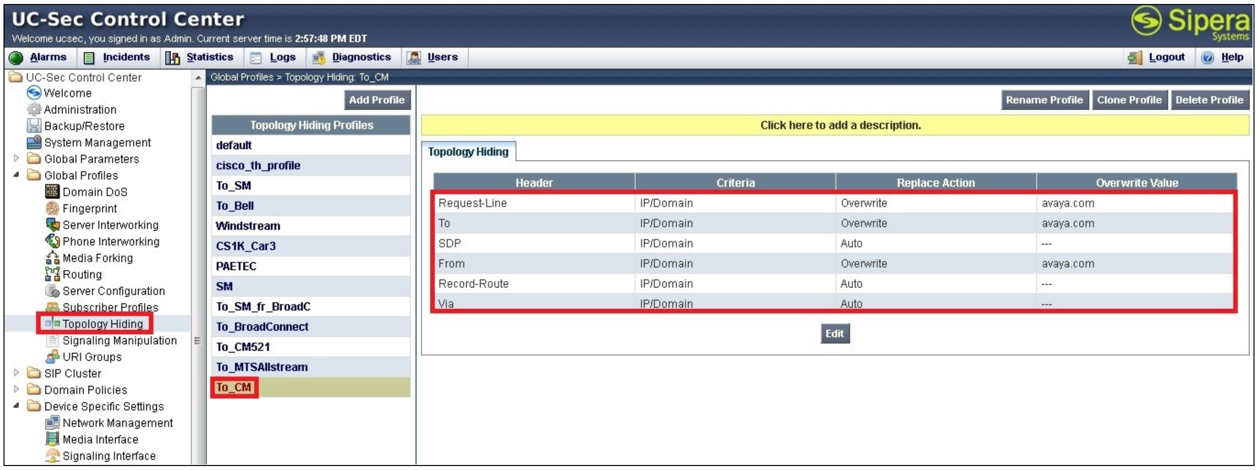
Notes:

- The **Criteria** should be selected as **IP/Domain** to give the Avaya SBCE the capability to mask both domain name and IP address present in SIP URI-Host.
- The masking applied on **From** header also applies to **Referred-By** and **P-Asserted-Identity** headers.
- The masking applied on **To** header also applies to **Refer-To** header.

6.2.3.2 Topology Hiding Profile for Communication Manager

Profile **To_CM** is also needed to mask MTS Allstream URI-Host in Request-URI, From, To headers to the enterprise SIP domain **avaya.com**; replace Record-Route, Via headers and SDP added by MTS Allstream by internal IP address known to Communication Manager.

The screenshots below illustrate the **Topology Hiding** profile **To_CM**.



UC-Sec Control Center

Welcome ucsec, you signed in as Admin. Current server time is **2:57:48 PM EDT**

Global Profiles > Topology Hiding: To_CM

Topology Hiding Profiles: default, cisco_th_profile, To_SM, To_Bell, Windstream, CS1K_Car3, PAETEC, SM, To_SM_fr_BroadC, To_BroadConnect, To_CM521, To_MTSAllstream, To_CM

Click here to add a description.

Topology Hiding

Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Overwrite	avaya.com
To	IP/Domain	Overwrite	avaya.com
SDP	IP/Domain	Auto	---
From	IP/Domain	Overwrite	avaya.com
Record-Route	IP/Domain	Auto	---
Via	IP/Domain	Auto	---

Notes:

- The **Criteria** should be **IP/Domain** to give the Avaya SBCE the capability to mask both domain name and IP address present in SIP URI-Host.
- The masking applied on **From** header also applies to **Referred-By** and **P-Asserted-Identity** headers.
- The masking applied on **To** header also applies to **Refer-To** header.

6.2.4. Server Interworking

Interworking Profile features are configured differently for Call and Trunk Servers.

To create a **Server Interworking** profile, select **UC-Sec Control Center → Global Profiles → Server Interworking**. Click on **Add Profile** (not shown).

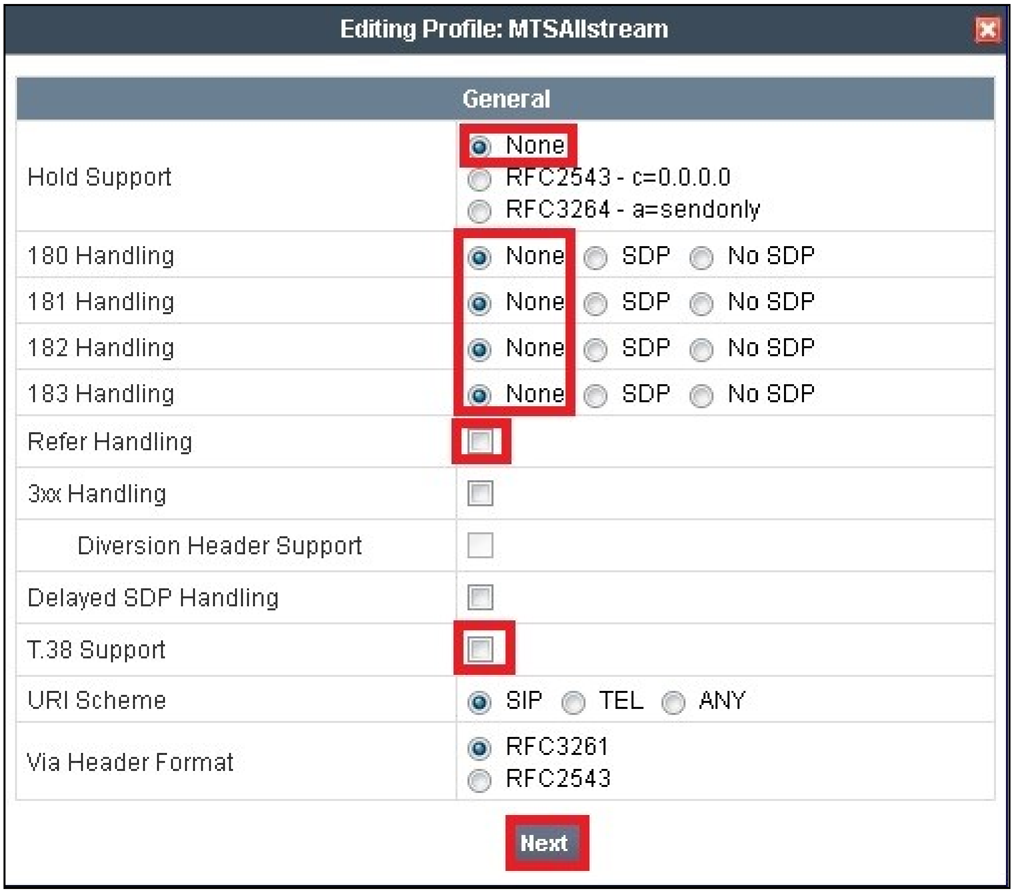
In the compliance testing, two profiles **MTSAllstream** and **CM** are created for MTS Allstream Trunk Server and Communication Manager Call Server.

6.2.4.1 Server Interworking profile for MTS Allstream

Profile **MTS Allstream** is defined to match the specification of MTS Allstream. The **General** settings are configured with following parameters while the other settings for **Timers**, **URI Manipulation**, **Header Manipulation** and **Advanced** are kept as default.

General settings:
- Hold Support = **None**. Avaya SBCE will not modify the hold/ resume signaling from Communication Manager to MTS Allstream.
- 18X Handling = **None**. Avaya SBCE will not handle 180X, it will keep the 18X messages from Communication Manager unchanged to MTS Allstream.
- Refer Handling = **unchecked**. Avaya SBCE will not handle Refer. It will keep the Refer message from Communication Manager unchanged to MTS Allstream.
- T.38 Support = **unchecked**. MTS Allstream does not support T.38 fax in the compliance testing.
- Privacy Enabled = **unchecked**. Avaya SBCE will not mask the From header with anonymous for outbound call to MTS Allstream. It depends on the Communication Manager to enable/ disable privacy on individual call basis.
- DTMF Support = **None**. Avaya SBCE will send original DTMF supported by Communication Manager to MTS Allstream.

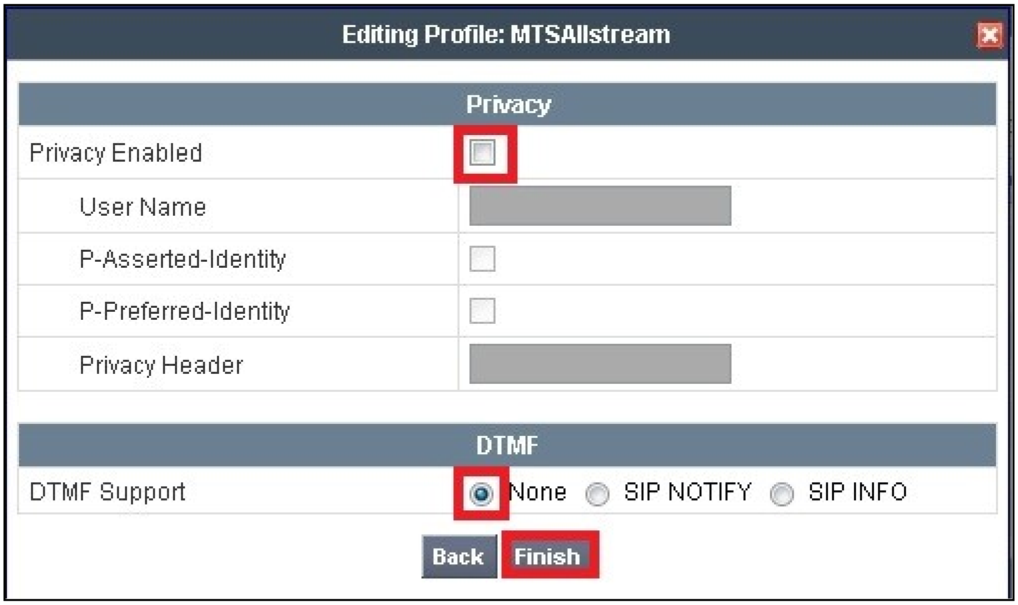
The screenshots below illustrate the **Server Interworking** profile **MTSAllstream**.



Editing Profile: MTSAllstream

General

Hold Support	● None ○ RFC2543 - c=0.0.0.0 ○ RFC3264 - a=sendonly
180 Handling	● None ○ SDP ○ No SDP
181 Handling	● None ○ SDP ○ No SDP
182 Handling	● None ○ SDP ○ No SDP
183 Handling	● None ○ SDP ○ No SDP
Refer Handling	□
3xx Handling	□
Diversion Header Support	□
Delayed SDP Handling	□
T.38 Support	□
URI Scheme	● SIP ○ TEL ○ ANY
Via Header Format	● RFC3261 ○ RFC2543



Editing Profile: MTSAllstream

Privacy

Privacy Enabled	□
User Name	
P-Asserted-Identity	□
P-Preferred-Identity	□
Privacy Header	

DTMF

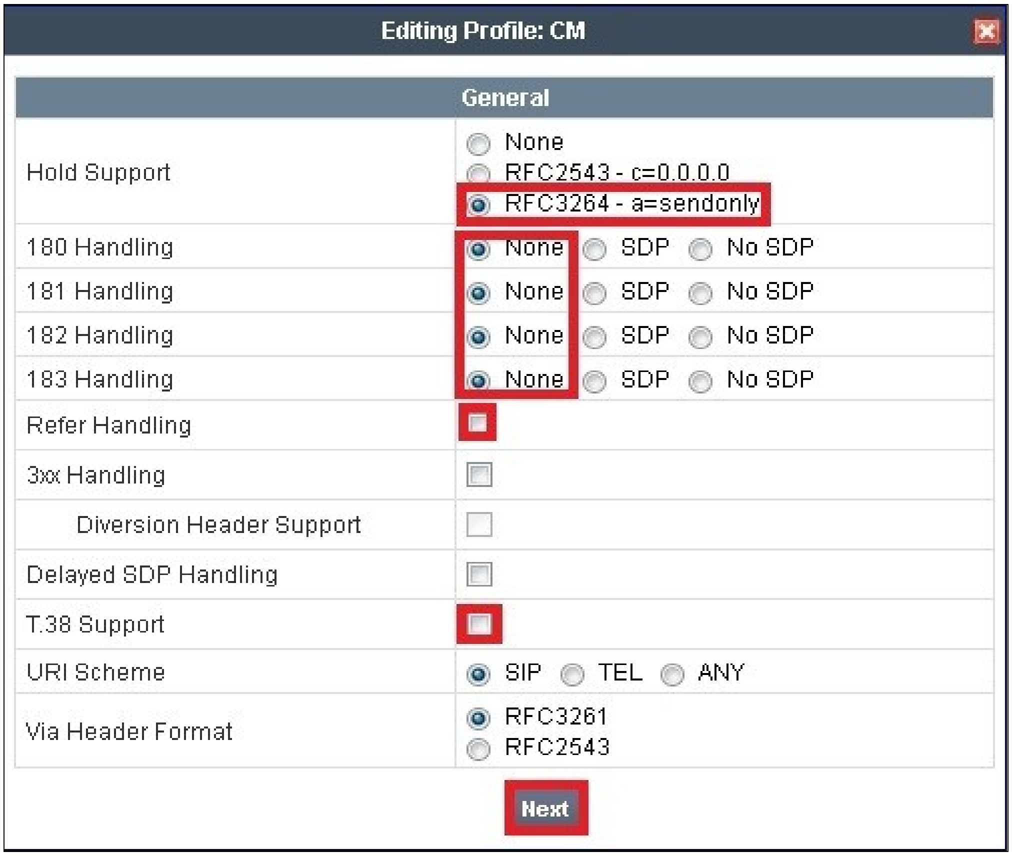
DTMF Support	● None ○ SIP NOTIFY ○ SIP INFO

6.2.4.2 Server Interworking profile for Communication Manager

Profile **CM** is defined to match the specification of Communication Manager. The **General** settings are configured with the following parameters while the other settings for **Timers**, **URI Manipulation**, **Header Manipulation** and **Advanced** are kept as default.

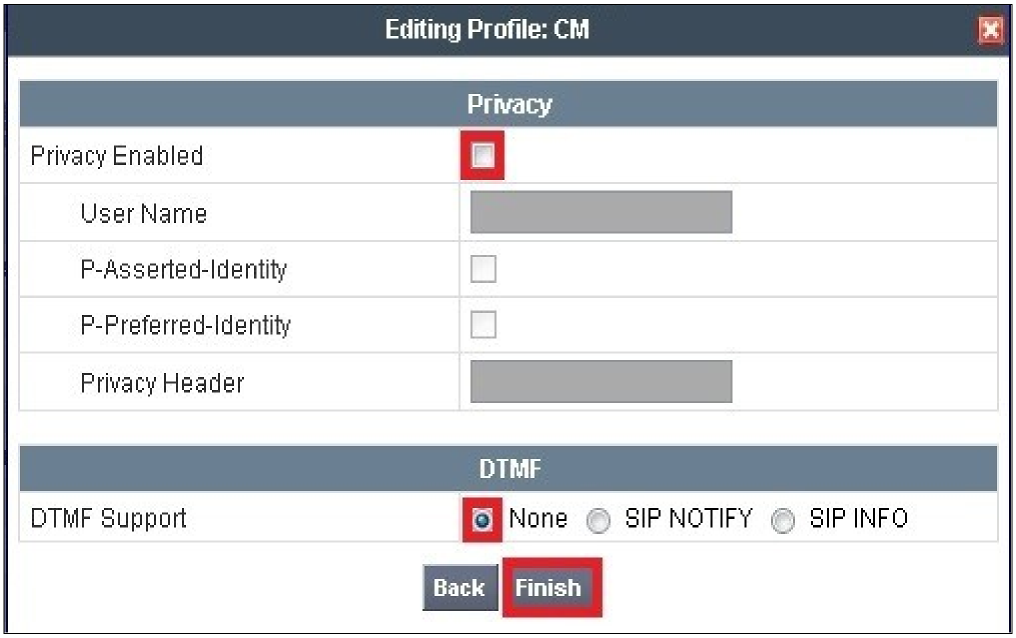
General settings:
- Hold Support = **RFC3264**. Communication Manager supports hold/ resume as per RFC3264.
- 18X Handling = **None**. Avaya SBCE will not handle 180X, it will keep the 18X messages from MTS Allstream to Communication Manager unchanged.
- Refer Handling = **unchecked**. Avaya SBCE will not handle Refer, it will keep the Refer messages from MTS Allstream to Communication unchanged.
- T.38 Support = **unchecked**. MTS Allstream does not support T.38 fax in the compliance testing.
- Privacy Enabled = **unchecked**. Avaya SBCE will not mask the **From** header with anonymous for inbound call from MTS Allstream. It depends on the MTS Allstream to enable/ disable privacy on individual call basis.
- DTMF Support = **None**. Avaya SBCE will send original DTMF supported by MTS Allstream to Communication Manager.

The screenshots below illustrate the **Server Interworking** profile **CM**.



Editing Profile: CM

General	
Hold Support	○ None ○ RFC2543 - c=0.0.0.0 ● RFC3264 - a=sendonly
180 Handling	● None ○ SDP ○ No SDP
181 Handling	● None ○ SDP ○ No SDP
182 Handling	● None ○ SDP ○ No SDP
183 Handling	● None ○ SDP ○ No SDP
Refer Handling	□
3xx Handling	□
Diversion Header Support	□
Delayed SDP Handling	□
T.38 Support	□
URI Scheme	● SIP ○ TEL ○ ANY
Via Header Format	● RFC3261 ○ RFC2543



Editing Profile: CM

Privacy	
Privacy Enabled	□
User Name	
P-Asserted-Identity	□
P-Preferred-Identity	□
Privacy Header	

DTMF	
DTMF Support	● None ○ SIP NOTIFY ○ SIP INFO

6.2.5. Signaling Manipulation

The **Signaling Manipulation** feature allows the ability to add, change and delete any of the headers in a SIP message. This feature will add the ability to configure such manipulation in a highly flexible manner using a proprietary scripting language called SigMa.

The SigMa scripting language is designed to express any of the SIP header manipulation operations to be done by the Avaya SBCE. Using this language, a script can be written and tied to a given **Server Configuration** which is configured in the next steps through the EMS GUI. The Avaya SBCE appliance then interprets this script at the given entry point or "hook point".

These Application Notes will not discuss the full feature of the Signaling Manipulation but will show an example of a script created during compliance testing to aid in **Topology Hiding**.

To create a **Signaling Manipulation** script, select **UC-Sec Control Center → Global Profiles → Signaling Manipulation**. Click on **Add Script** (not shown).

In the compliance testing, a SigMa script named **MTSAllstream** is created for Server Configuration MTS Allstream and described detail as following:

```
within session "ALL"
{
act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
{
%HEADERS["P-Asserted-Identity"][1].URI.HOST= "110.10.98.112";
}
act on message where %DIRECTION="INBOUND" and %ENTRY_POINT="AFTER_NETWORK"
                                         {
%HEADERS["Request_Line"][1].regex_replace("sip:110.10.98.112","sip:avaya.com");
%HEADERS["To"][1].regex_replace("sip:110.10.98.112","sip:ping@110.10.98.112");
%HEADERS["From"][1].regex_replace("sip:220.20.2.12","sip:ping@220.20.2.12");
}
}
```

The statement **`act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"`** is to specify the script will take effect on all type of SIP messages for outbound calls and the manipulation will be done after routing. The manipulation will be according to the rules contained in this statement. The **Topology-Hiding** profile **MTSAllstream** could mask the P-Asserted-Identity header properly. However, as a limitation, the P-Asserted-Identity header in "response" SIP message will still have the private enterprise SIP domain. Therefore, a SigMa rule is used to correct the URI-Host of P-Asserted-Identity header.

The statement **`act on message where %DIRECTION="INBOUND" and %ENTRY_POINT="AFTER_NETWORK"`** is to specify the script will take effect on all types of SIP messages for inbound calls and the manipulation will be done before routing. The manipulation will be according to the rules contained in this statement. The purpose of the SigMa script **MTAllstream** is to normalize the OPTIONS received from MTS Allstream. The header **From** and **To** need to be modified to have URI-User@URI-Host format, otherwise the OPTIONS will fail to match the URI-Group defined in **Section 6.2.1**. If unmatching happens, the **Routing**

Profile and **Server Flow** (discussed in **Section 6.4.4**) will not be applied to the call, and will result in dropped packets.

With the SigMa script in place, the OPTIONS heartbeat from MTS Allstream will be forwarded to Communication Manager. The 200OK response from Communication Manager will confirm the status of SIP Trunk as active. If there is no response, MTS Allstream will change the status of SIP Trunk to “out of service”.

Note: SigMa script for Communication Manager is unnecessary because in the compliance test Communication Manager is configured not to send OPTIONS heartbeat to MTS Allstream. The OPTIONS heartbeat is configured on Avaya SBCE under Server Configuration. Please see **Section 2.2**, observation #10 for more information.

6.2.6. Server Configuration

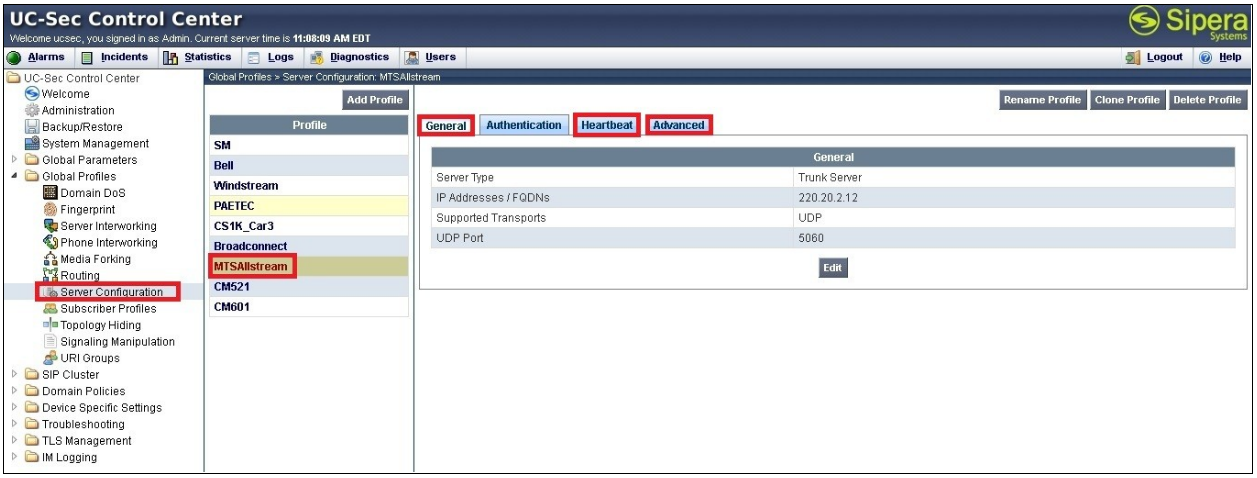
The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. These tabs are used to configure and manage various SIP Call Server specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics and trusted domains.

To create a Server Configuration entry, select **UC-Sec Control Center →Global Profiles →Server Configuration**. Click on **Add Profile** (not shown).

In the compliance testing, two separate Server Configurations are created, server entry **MTSAllsream** for MTS Allstream and server entry **CM601** for Communication Manager.

6.2.6.1 Server Configuration for MTS Allstream

The **Server Configuration** named **MTSAllstream** is added for MTS Allstream, it will be discussed in detail as below. **General**, **Heartbeat** and **Advanced** tabs are provisioned but no configuration is done for **Authentication tab** as MTS Alllstream does not implement Authentication on a SIP Trunk.

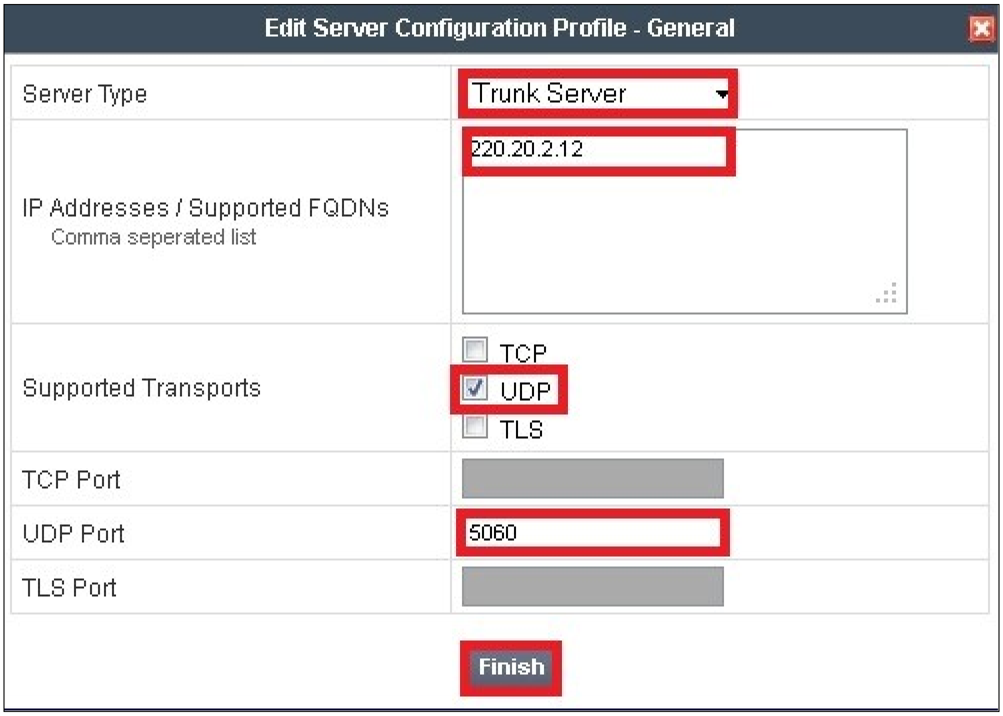


UC-Sec Control Center
Welcome ucsec, you signed in as Admin. Current server time is **11:08:09 AM EDT**

Global Profiles > Server Configuration: MTSAllstream

General	
Server Type	Trunk Server
IP Addresses / FQDNs	220.20.2.12
Supported Transports	UDP
UDP Port	5060

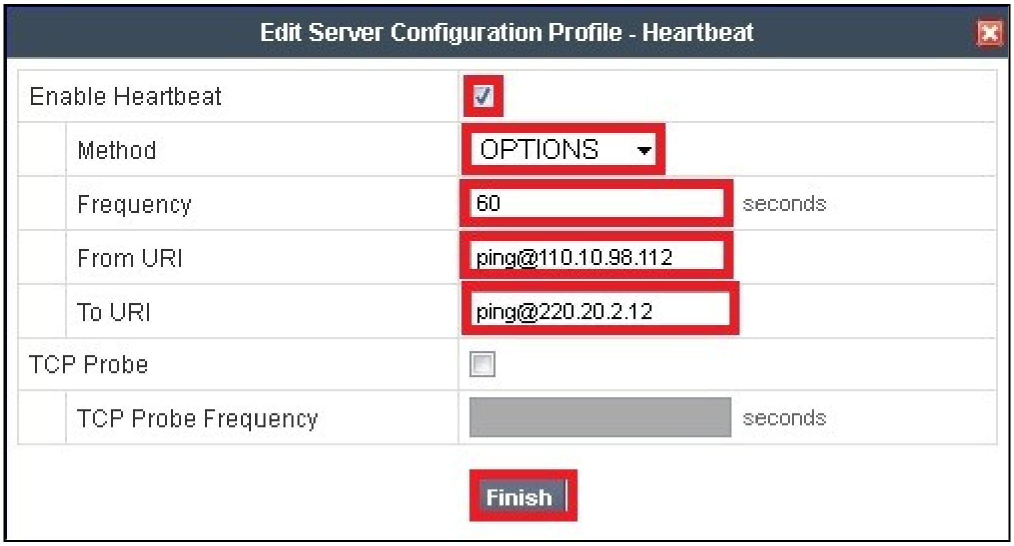
In the **General** tab, set **Server Type** for MTS Allstream to **Trunk Server**. In the compliance testing, MTS Allstream supports UDP and listens on port 5060.



Edit Server Configuration Profile - General

Server Type	Trunk Server
IP Addresses / Supported FQDNs Comma seperated list	220.20.2.12
Supported Transports	□ TCP ☑ UDP □ TLS
TCP Port	
UDP Port	5060
TLS Port	

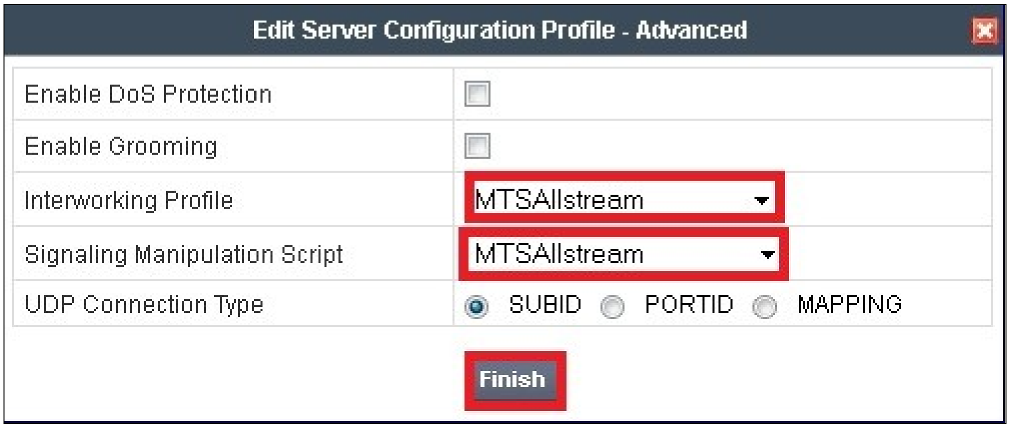
The OPTIONS heartbeat is implemented on Avaya SBCE under Server Configuration for MTS Allstream. Please see **Section 2.2**, observation #10 for more information. The following screenshot shows the configuration in Heartbeat tab in detail.



Edit Server Configuration Profile - Heartbeat

Enable Heartbeat	☑
Method	OPTIONS
Frequency	60 seconds
From URI	ping@110.10.98.112
To URI	ping@220.20.2.12
TCP Probe	□
TCP Probe Frequency	seconds

Under **Advanced** tab, for **Interworking Profile** drop down list, select **MTSAllstream** as defined in **Section 6.2.4** and for **Signaling Manipulation Script** drop down list select **MTSAllsteram** as defined in **Section 6.2.5**. These configurations are applied to the specific SIP profile and SigMa rules for the traffic from MTS Allstream. The other settings are kept as default.

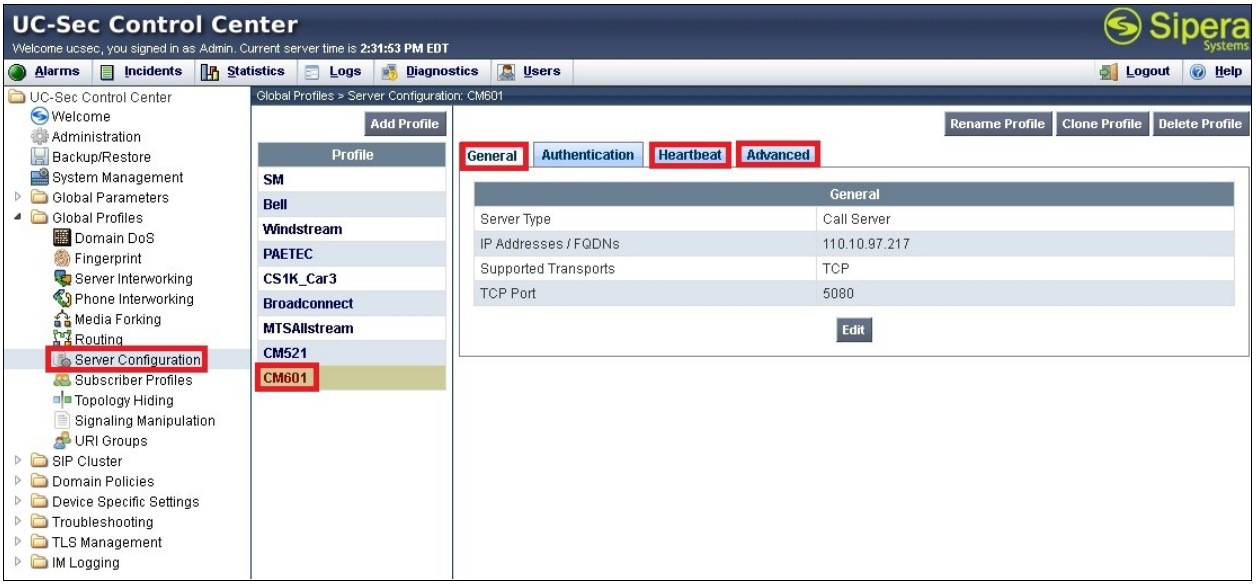


Edit Server Configuration Profile - Advanced

Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	MTSAllstream
Signaling Manipulation Script	MTSAllstream
UDP Connection Type	● SUBID ○ PORTID ○ MAPPING

6.2.6.2 Server Configuration for Communication Manager

The **Server Configuration** named **CM601** is added for Communication Manager is discussed in detail below. **General**, **Heartbeat** and **Advanced** tabs are provisioned but no configuration is done for **Authentication** tab.



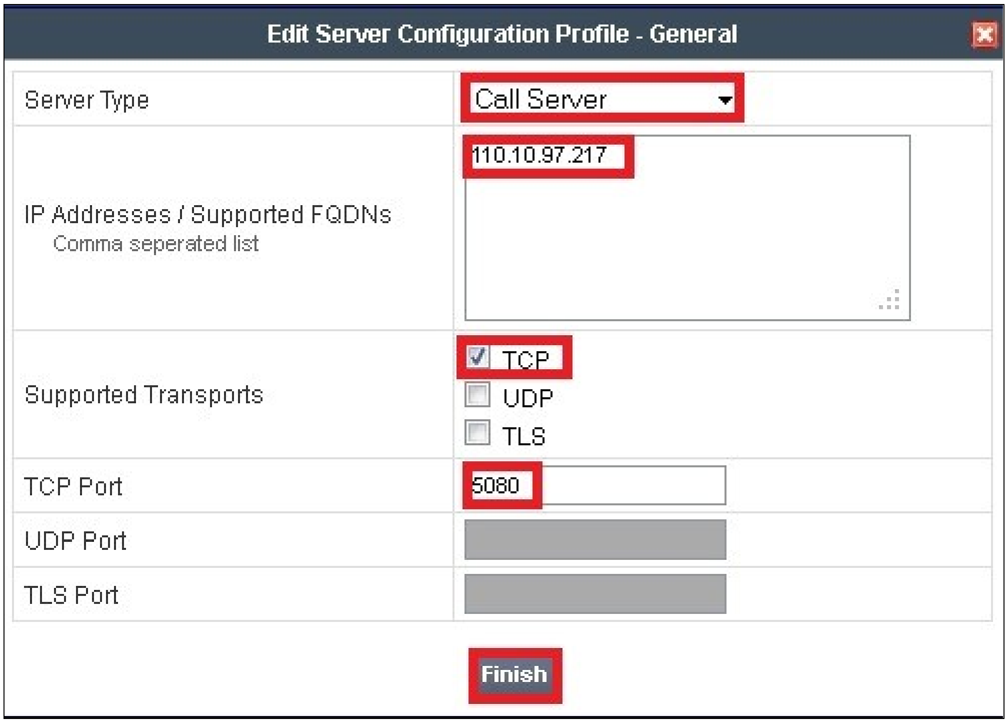
UC-Sec Control Center

Welcome ucsec, you signed in as Admin. Current server time is 2:31:53 PM EDT

Global Profiles > Server Configuration: CM601

General	
Server Type	Call Server
IP Addresses / FQDNs	110.10.97.217
Supported Transports	TCP
TCP Port	5080

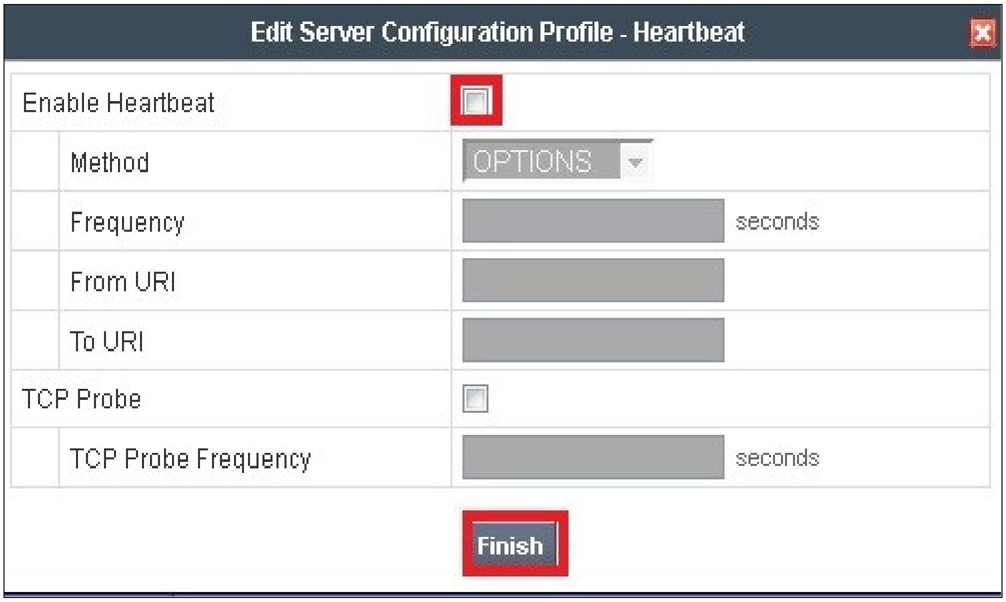
In the **General** tab, specify **Server Type** as **Call Server**. In the compliance testing, the link between Avaya SBCE and Communication Manager is TCP and Communication Manager listens on port 5080.



Edit Server Configuration Profile - General

Server Type	Call Server
IP Addresses / Supported FQDNs Comma seperated list	110.10.97.217
Supported Transports	☑ TCP ☐ UDP ☐ TLS
TCP Port	5080
UDP Port	
TLS Port	

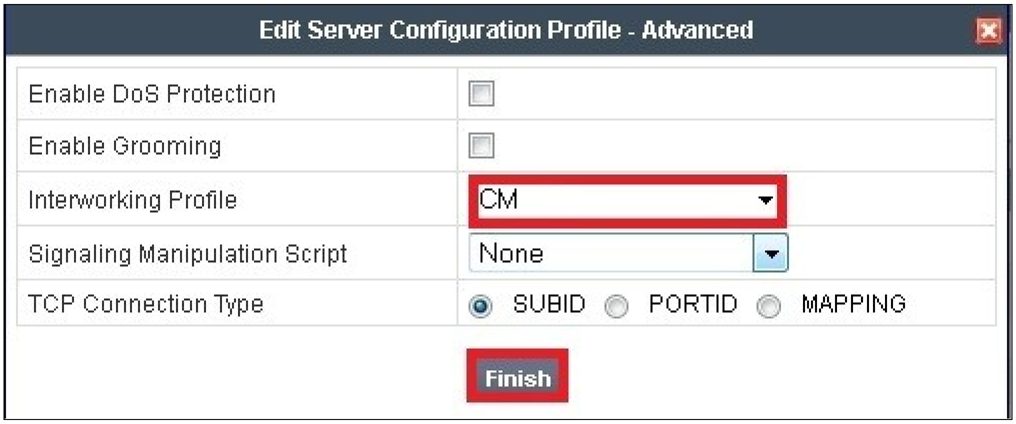
By default the Avaya SBCE forwards the OPTIONS from MTS Allstream to Communication Manager to check for SIP Trunk status. Thus, it is unnecessary to enable heartbeat on Avaya SBCE to send OPTIONS to Call Server. In the compliance testing, the Heartbeat is kept disabled as default. The following screenshot shows the configuration in Heartbeat tab in detail.



Edit Server Configuration Profile - Heartbeat

Enable Heartbeat	☐
Method	OPTIONS
Frequency	seconds
From URI	
To URI	
TCP Probe	☐
TCP Probe Frequency	seconds

Under **Advanced** tab, for **Interworking Profile** drop down list select **CM** as defined in **Section 6.2.4** and for **Signaling Manipulation Script** drop down list select **None**. The other settings are kept as default.



Edit Server Configuration Profile - Advanced

Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	CM
Signaling Manipulation Script	None
TCP Connection Type	● SUBID ○ PORTID ○ MAPPING

6.3. Domain Policies

The **Domain Policies** feature configures various rule sets (policies) to control unified communications based upon criteria of communication sessions originating from or terminating at the enterprise. This criterion can be used to trigger policies which, in turn, activate various security features of the UC-Sec security device to aggregate, monitor, control and normalize call flow. There are default policies available for use, or a custom domain policy can be created.

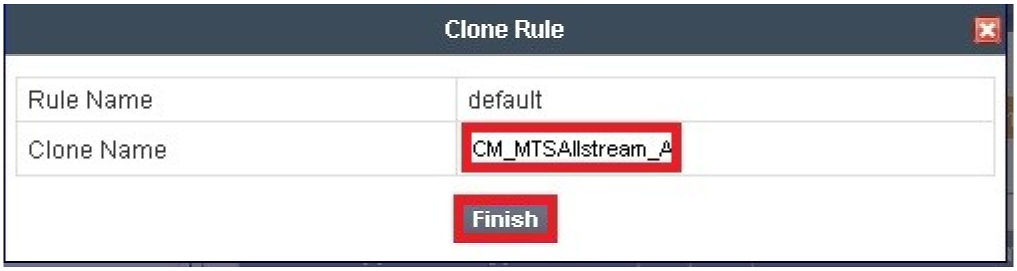
6.3.1. Application Rules

Application Rules define which types of SIP-based Unified Communications (UC) applications the UC-Sec security device will protect: voice, video, and/or Instant Messaging (IM). In addition, it is possible to configure the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion.

An **Application Rule** is created to set the number of concurrent voice traffic. The sample configuration cloned and modified the default application rule to increase the number of **Maximum Concurrent Session** and **Maximum Sessions Per Endpoint**.

To clone an application rule, navigate to **UC-Sec Control Center → Domain Policies → Application Rules**. With the default rule chosen, click on **Clone Rule** (not shown).

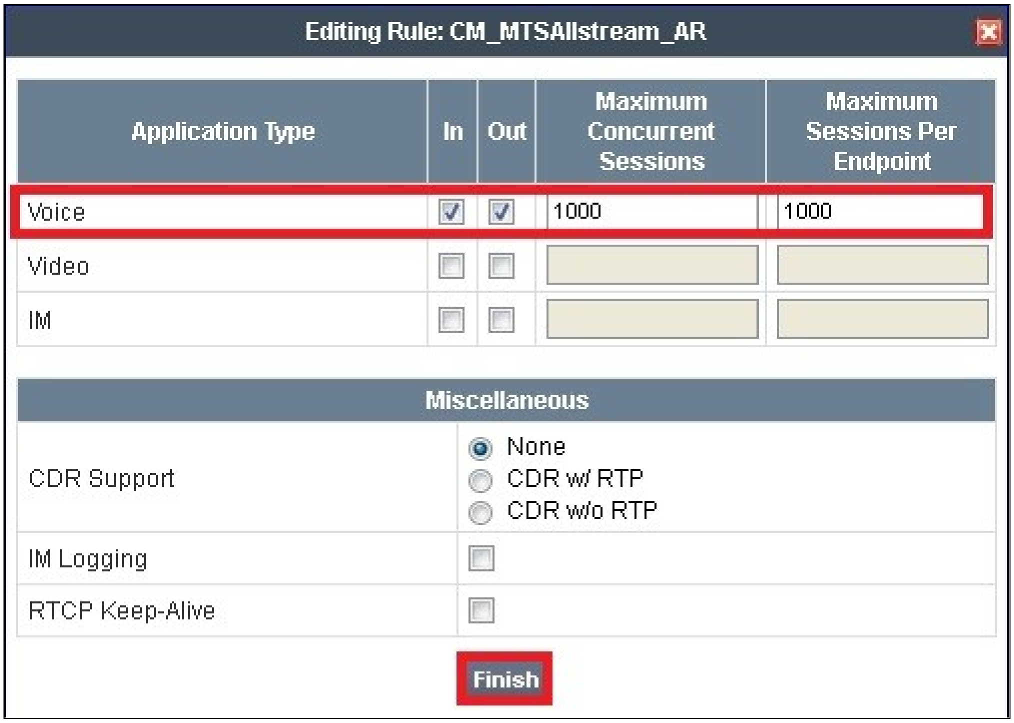
Enter a rule with a descriptive name **CM_MTSAllstream_AR** and click **Finish**.



Clone Rule

Rule Name	default
Clone Name	CM_MTSAllstream_A

Click **Edit** button (not shown) to modify the rule. Set the **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** for the **Voice** application to a value high enough for the amount of traffic the network is able process. The following screen shows the modified **Application Rule** with the **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** set to 1000. In the compliance test, Communication Manager is programmed to control the concurrent sessions by setting the number of members in the trunk group (**Section 5.7**) to the allotted number. Therefore, the values in the **Application Rule** named **CM_MTSAllstream_AR** are set high enough to be considered non-blocking.



Editing Rule: CM_MTSAllstream_AR

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Voice	☑	☑	1000	1000
Video	□	□		
IM	□	□		

Miscellaneous

CDR Support	● None ○ CDR w/ RTP ○ CDR w/o RTP
IM Logging	□
RTCP Keep-Alive	□

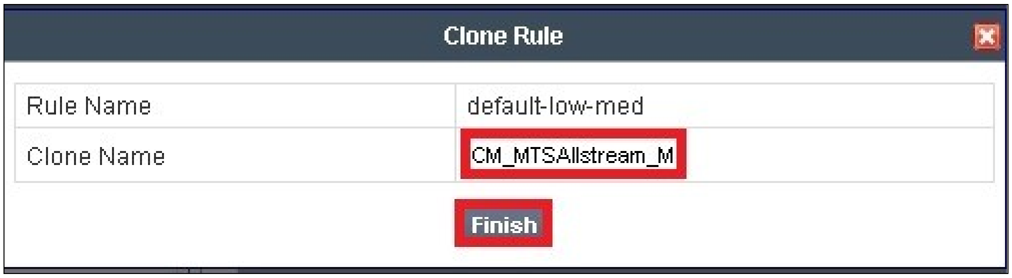
6.3.2. Media Rules

Media Rules define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packet matching the criteria will be handled by the UC-Sec security product.

A custom **Media Rule** is created to set the **Quality of Service** and **Media Anomaly Detection**. The sample configuration shows **Media Rule CM_MTSAllstream_MR** used for both the enterprise and MTS Allstream.

To create **Media Rule**, navigate to **UC-Sec Control Center → Domain Policies → Media Rules**. With **default-low-med** selected, click **Clone Rule** (not shown).

Enter a **Media Rule** with a descriptive name **CM_MTSAllstream_MR** and click **Finish**.

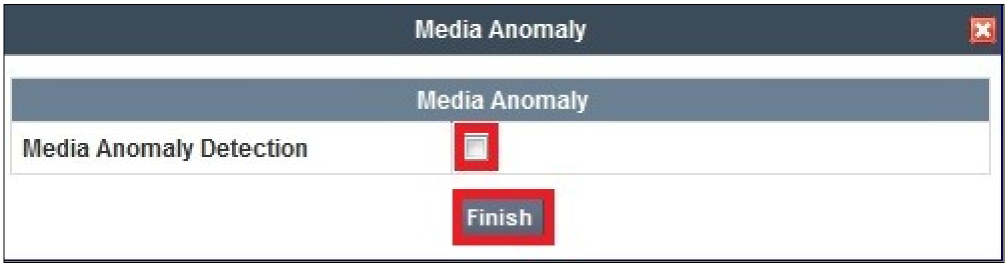


Clone Rule

Rule Name	default-low-med
Clone Name	CM_MTSAllstream_M

When the RTP packets of a call are shuffled from Communication Manager to an IP Phone, Avaya SBCE will interpret this as an anomaly and an alert will be created in the **Incidents Log**. Disabling **Media Anomaly Detection** prevents the **RTP Injection Attack** alerts from being created in the log during an audio shuffle.

To modify the rule, select the **Media Anomaly** tab (not shown) and click **Edit**, uncheck **Media Anomaly Detection** and click **Finish**.



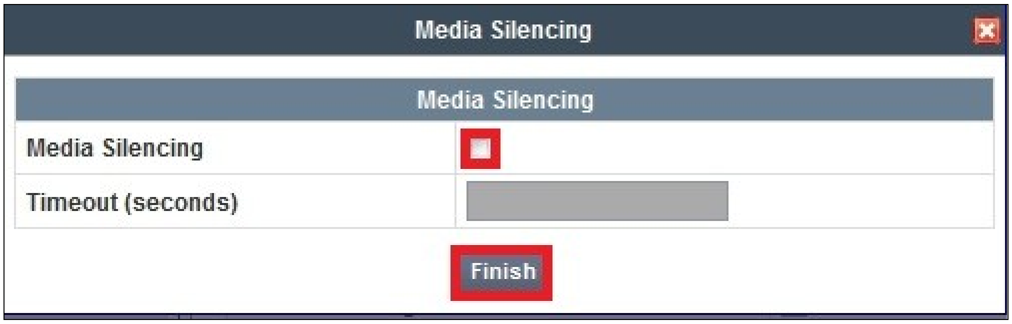
Media Anomaly

Media Anomaly

Media Anomaly Detection □

The **Media Silencing** feature detects the silence when the call is in progress. If the silence is detected and exceeds the allowed duration, Avaya SBCE generates alert in the **Incidents Log**. In the compliance test, the **Media Silencing** detection is disabled to prevent the call from unexpectedly disconnected due to a RTP packet lost on public WAN.

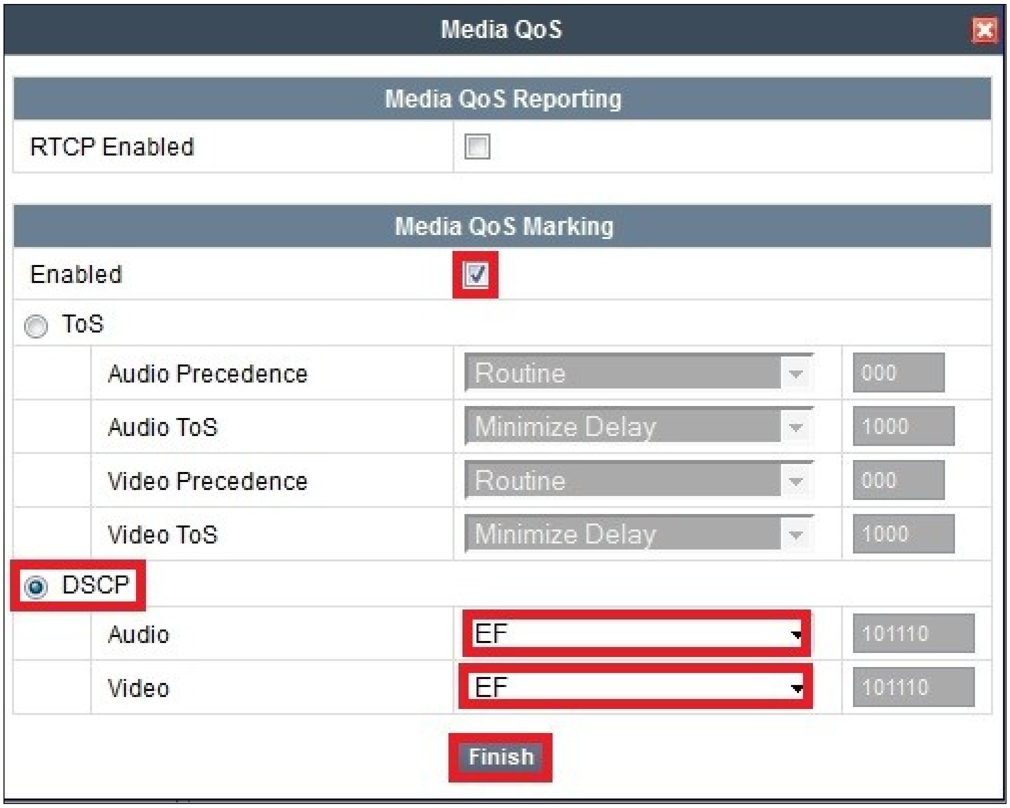
To modify the rule, select the **Media Silencing** tab and click **Edit**, uncheck **Media Silencing** and click **Finish**.



Media Silencing

Media Silencing	
Media Silencing	□
Timeout (seconds)	

Select the **Media QoS** tab and click **Edit** to configure the proper Quality of Service (QoS). Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for the media. The following screen shows the QoS values used for the compliance test.



Media QoS

Media QoS Reporting		
RTCP Enabled	□	
Media QoS Marking		
Enabled	☑	
○ ToS		
Audio Precedence	Routine	000
Audio ToS	Minimize Delay	1000
Video Precedence	Routine	000
Video ToS	Minimize Delay	1000
● DSCP		
Audio	EF	101110
Video	EF	101110

6.3.3. Signaling Rules

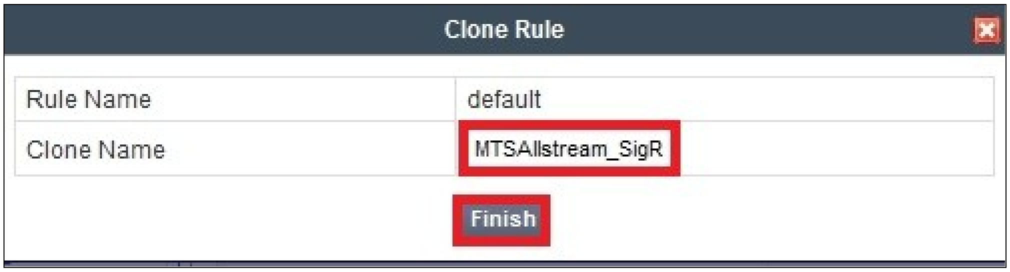
Signaling Rules define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by the UC-Sec, they are parsed and “pattern-matched” against the particular signaling criteria defined by these rules. Packets matching the criteria defined by the Signaling Rules are tagged for further policy matching.

To clone a signaling rule, navigate to **UC-Sec Control Center → Domain Policies → Signaling Rules**. With the **default** rule chosen, click on **Clone Rule** (not shown).

In the compliance testing, two **Signaling Rules** are created for MTS Allstream and Communication Manager.

6.3.3.1 Signaling Rule for MTS Allstream

Clone a **Signaling Rule** with a descriptive name **MTSAllstream_SigR** and click **Finish**.



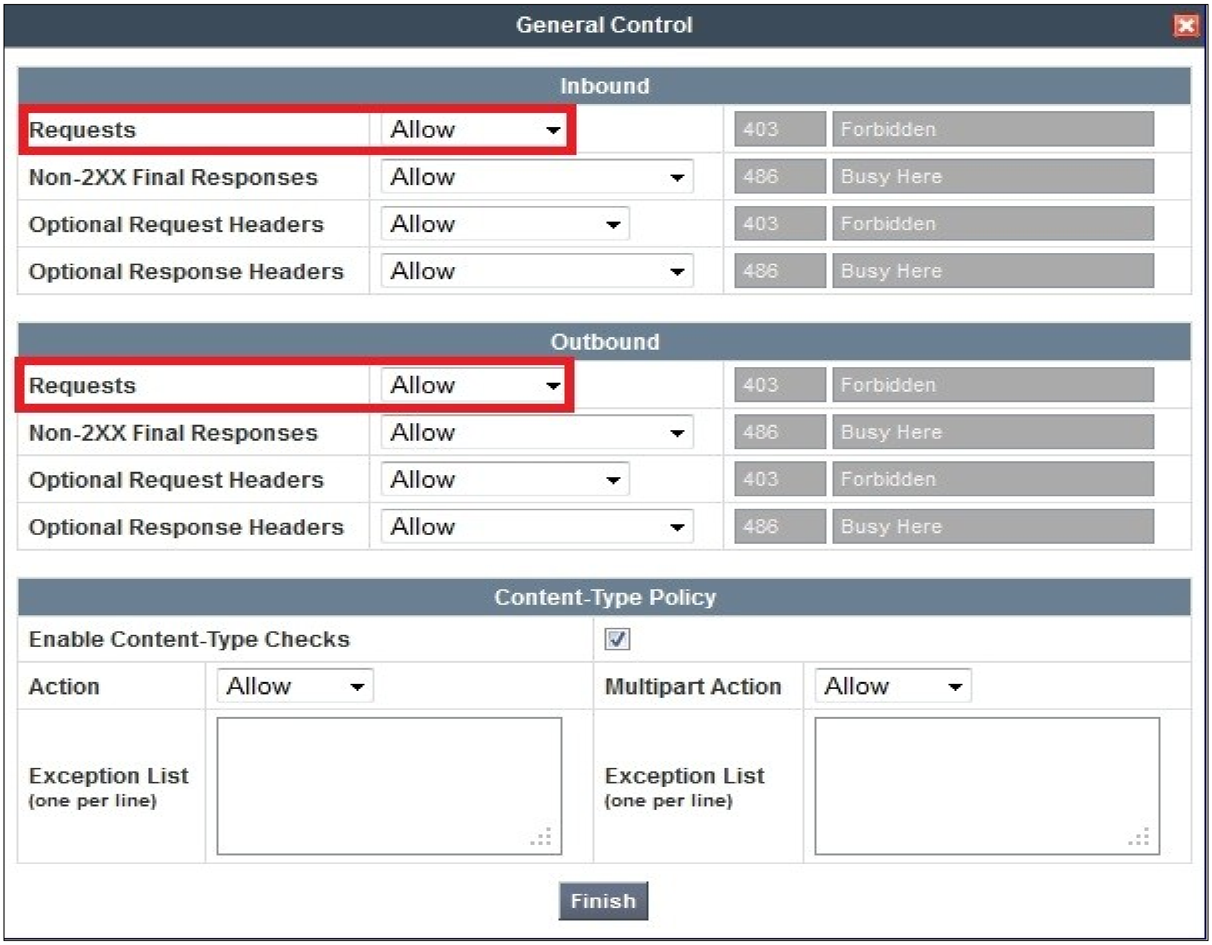
Clone Rule

Rule Name: default

Clone Name: MTSAllstream_SigR

The **MTSAllstream_SigR** is configured to allow MTS Allstream to accept inbound and outbound call requests. It also blocks Accept-Language, Alert-Info and P-Chanrging-Vector headers from Communication Manager because these headers are not required by MTS Allstream.

Being cloned from the **Signaling Rule default**, the **MTSAllstream_SigR** will block all requests with 403 Forbidden. To start accepting calls, go to **General** tab, click on **Edit**. Then change **Inbound** and **Outbound Request** to **Allow** as shown in following screenshot.



General Control

Inbound			
Requests	Allow	403	Forbidden
Non-2XX Final Responses	Allow	486	Busy Here
Optional Request Headers	Allow	403	Forbidden
Optional Response Headers	Allow	486	Busy Here

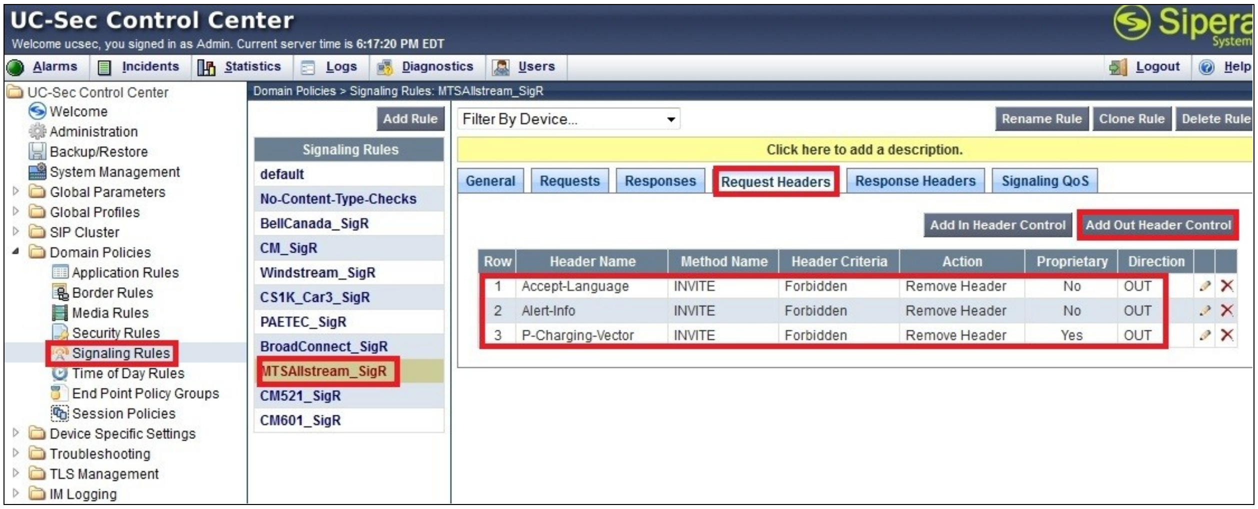
Outbound			
Requests	Allow	403	Forbidden
Non-2XX Final Responses	Allow	486	Busy Here
Optional Request Headers	Allow	403	Forbidden
Optional Response Headers	Allow	486	Busy Here

Content-Type Policy			
Enable Content-Type Checks	☑		
Action	Allow	Multipart Action	Allow
Exception List (one per line)		Exception List (one per line)	

Request Headers setting is to allow or block a header in particular direction for request method . The buttons “**Add In Header Control**” and “**Add Out Header Control**” are used to define the inbound and outbound Request header rules. The signaling rule **MTSAllstream_SigR** will be assigned to Server Configure for MTS Allstream as shown in **Section 6.2.6.1**.

The following screenshot shows three rules added to block the **Accept-Language**, **Alert-Info** and P-**Chanrging-Vector** headers.

- **Header Name**: Select the header to be manipulated.
- **Method Name**: Select **INVITE** in an outbound call request.
- **Header Criteria**: Click on **Forbidden** to block the header.
- **Action**: Select **Remove header** to delete the header.



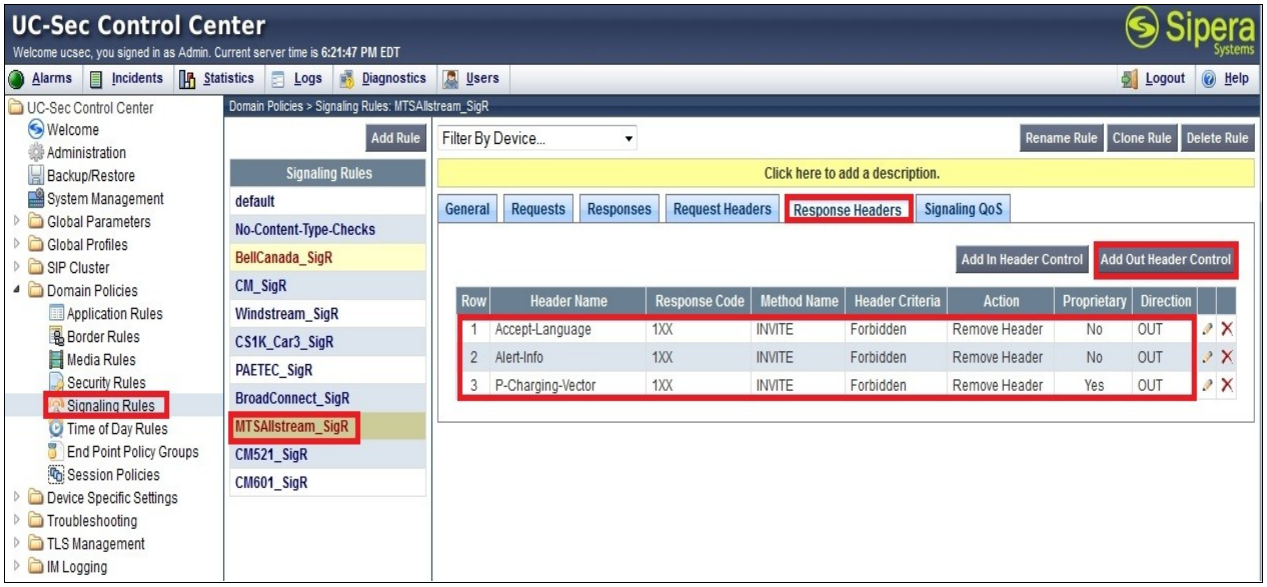
Row	Header Name	Method Name	Header Criteria	Action	Proprietary	Direction
1	Accept-Language	INVITE	Forbidden	Remove Header	No	OUT
2	Alert-Info	INVITE	Forbidden	Remove Header	No	OUT
3	P-Charging-Vector	INVITE	Forbidden	Remove Header	Yes	OUT

Notes: Pre-defined list does not have P-Charging-Vector header, but the Avaya SBCE provides an option to define this proprietary header.

Response Headers setting is to allow or block a header in particular direction for response method. The buttons “**Add In Header Control**” and “**Add Out Header Control**” are used to define inbound and outbound Response Header rules. The **Signaling Rule MTSAllstream_SigR** will be assigned to Server Configure for MTS Allstream as shown in **Section 6.2.6.1**.

The following screenshots show three rules added to block the **Accept-Language**, **Alert-Info** and **P-Chanrging-Vector** headers:

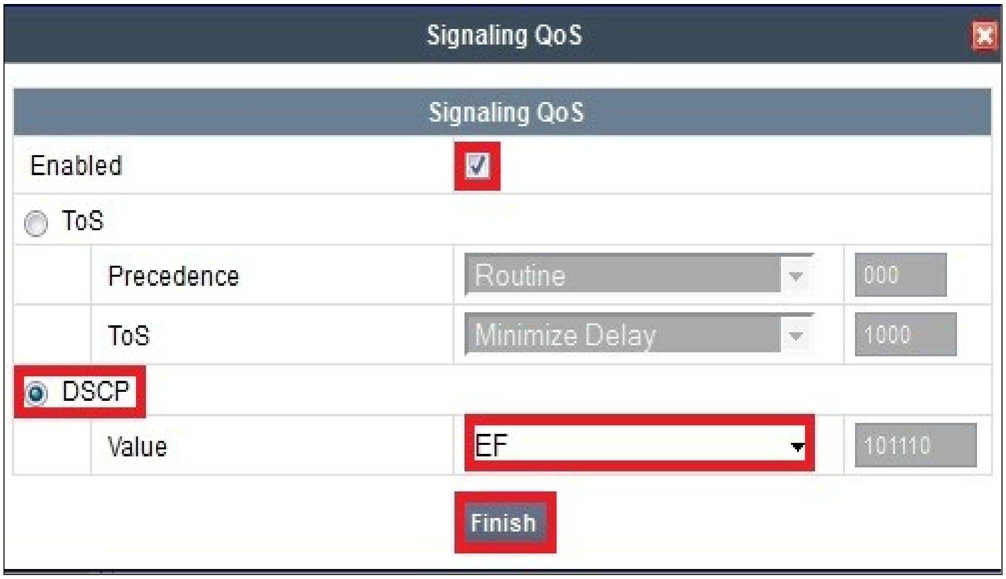
- **Header Name**: Select the header to be manipulated.
- **Method Name**: Select INVITE for an inbound call request.
- **Header Criteria**: Click on **Forbidden** to block the header.
- **Action**: Select **Remove header** to delete the header.



Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction
1	Accept-Language	1XX	INVITE	Forbidden	Remove Header	No	OUT
2	Alert-Info	1XX	INVITE	Forbidden	Remove Header	No	OUT
3	P-Charging-Vector	1XX	INVITE	Forbidden	Remove Header	Yes	OUT

Notes: Pre-defined list does not have **P-Charging-Vector** header, but the Avaya SBCE provides an option to define this proprietary header.

On the **Signaling QoS** tab, select the proper Quality of Service (QoS). Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for signaling. The following screen shows the QoS value used for the compliance test.



Enabled ☑

○ ToS

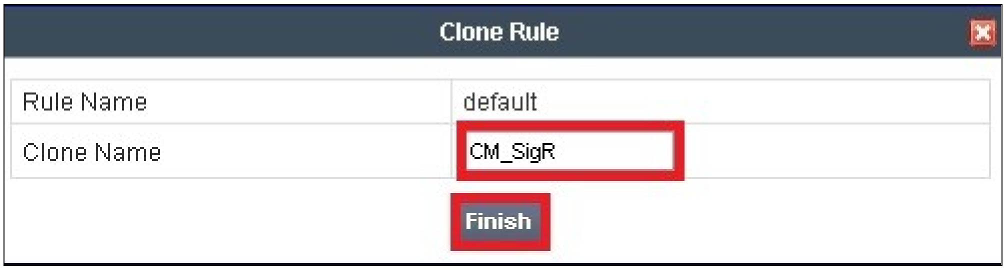
	Precedence	Routine	000
	ToS	Minimize Delay	1000

● DSCP

	Value	EF	101110

6.3.3.2 Signaling Rule for Communication Manager

Clone a **Signaling Rule** with a descriptive name **CM_SigR** and click **Finish**.

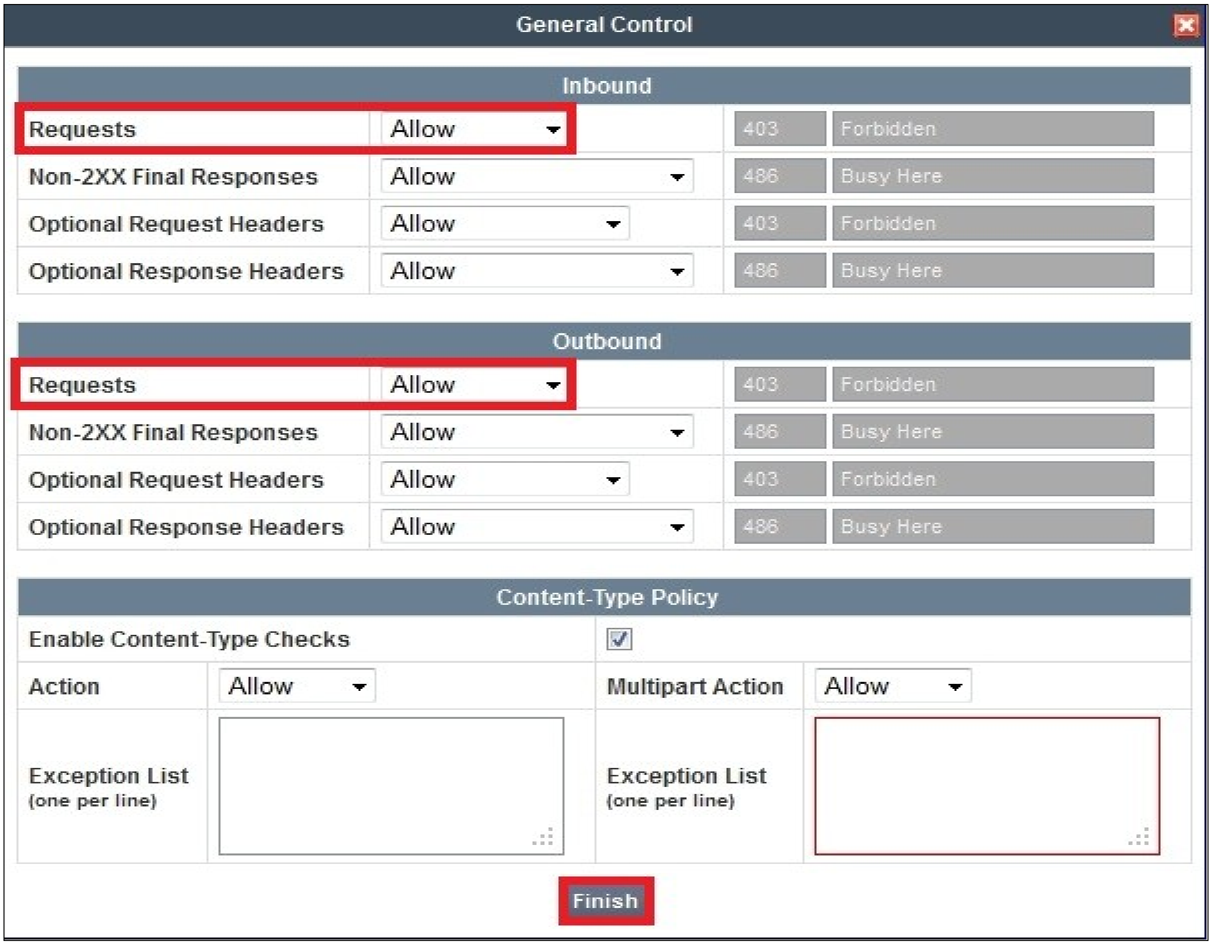


Clone Rule

Rule Name	default
Clone Name	CM_SigR

This **CM_SigR** is configured to allow Communication Manager to accept inbound and outbound call requests.

Being cloned from the **Signaling Rule default**, the **CM_SigR** will block all requests with 403 Forbidden. To start accepting calls, select CM_SigR then go to **General** tab, click on **Edit** (not shown). Then change **Inbound-Requests** and **Outbound-Requests** to **Allow** as shown in following screenshot and click **Finish**.



General Control

Inbound

Requests	Allow	403	Forbidden
Non-2XX Final Responses	Allow	486	Busy Here
Optional Request Headers	Allow	403	Forbidden
Optional Response Headers	Allow	486	Busy Here

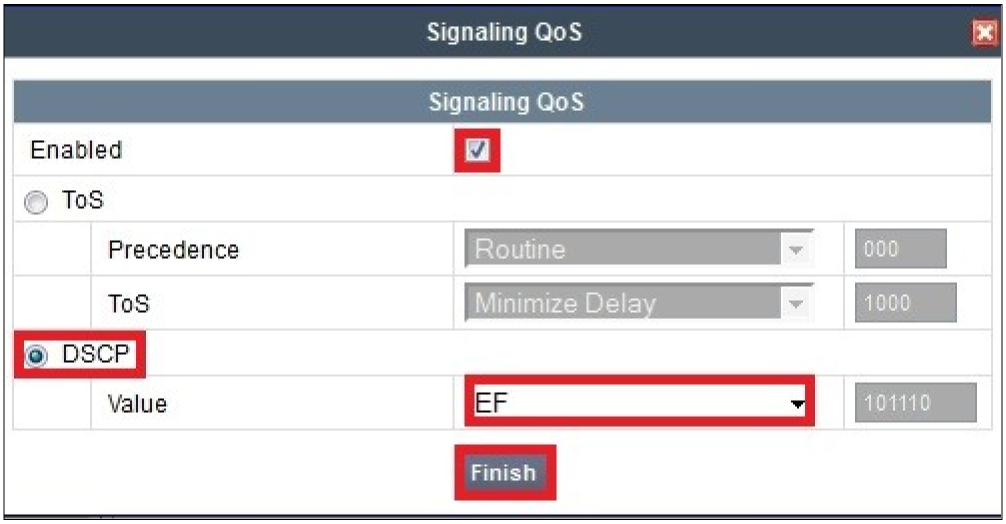
Outbound

Requests	Allow	403	Forbidden
Non-2XX Final Responses	Allow	486	Busy Here
Optional Request Headers	Allow	403	Forbidden
Optional Response Headers	Allow	486	Busy Here

Content-Type Policy

Enable Content-Type Checks	☑		
Action	Allow	Multipart Action	Allow
Exception List (one per line)		Exception List (one per line)	

On the **Signaling QoS** tab, select the proper Quality of Service (QoS). Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for signaling. The following screen shows the QoS value used for the compliance test.



Signaling QoS

Signaling QoS		
Enabled	☑	
○ ToS		
Precedence	Routine	000
ToS	Minimize Delay	1000
● DSCP		
Value	EF	101110

6.3.4. Endpoint Policy Groups

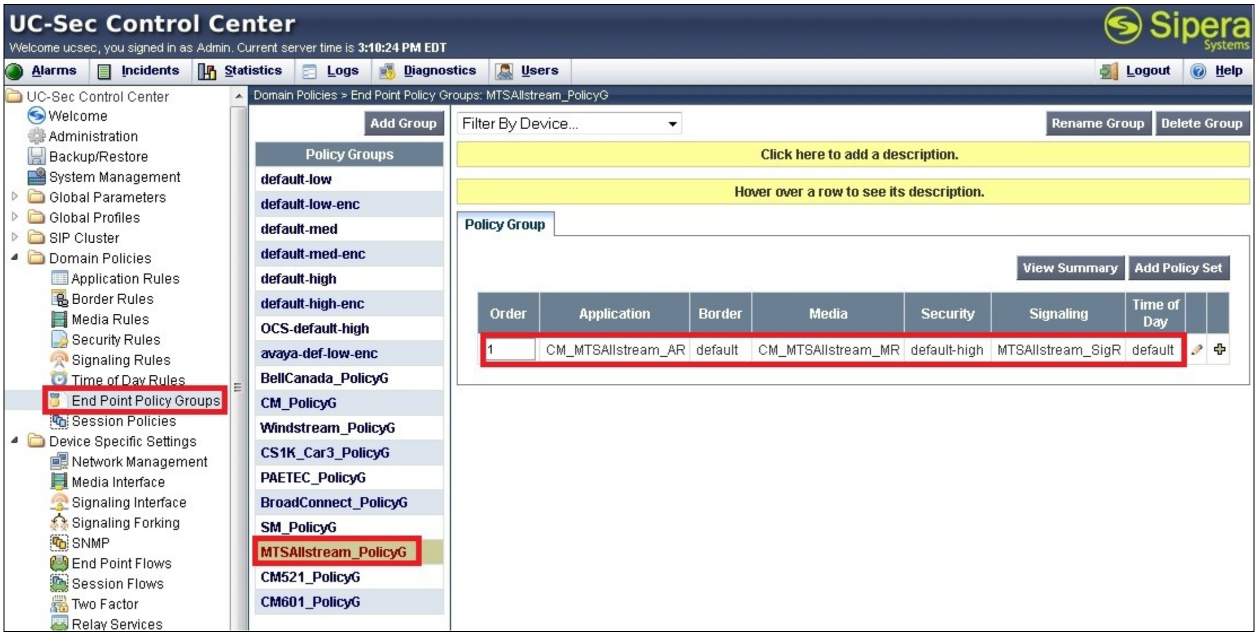
The rules created within the **Domain Policy** section are assigned to an **Endpoint Policy Group**. The **Endpoint Policy Group** is then applied to a **Server Flow** defined in the next section**.**

Endpoint Policy Groups are created for the Communication Manager and the MTS Allstream.

To create a new policy group, navigate to **UC-Sec Control Center → Domain Policies → Endpoint Policy Groups** and click on **Add Group** (not shown)**.**

6.3.4.1 Endpoint Policy Group for MTS Allstream

The following screen shows **MTSAllstream_PolicyG** created for MTS Allstream:
- Set **Application** and **Media** rules created in **Section 6.3.1** and **Section 6.3.2**.
- Set **Signaling** rule **CM_MTSAllstream_AR** created in **Section 6.3.3.1**.
- Set **Border** and **Time of Day** rules to **default**.
- Set **Security** rule to **default-high**.

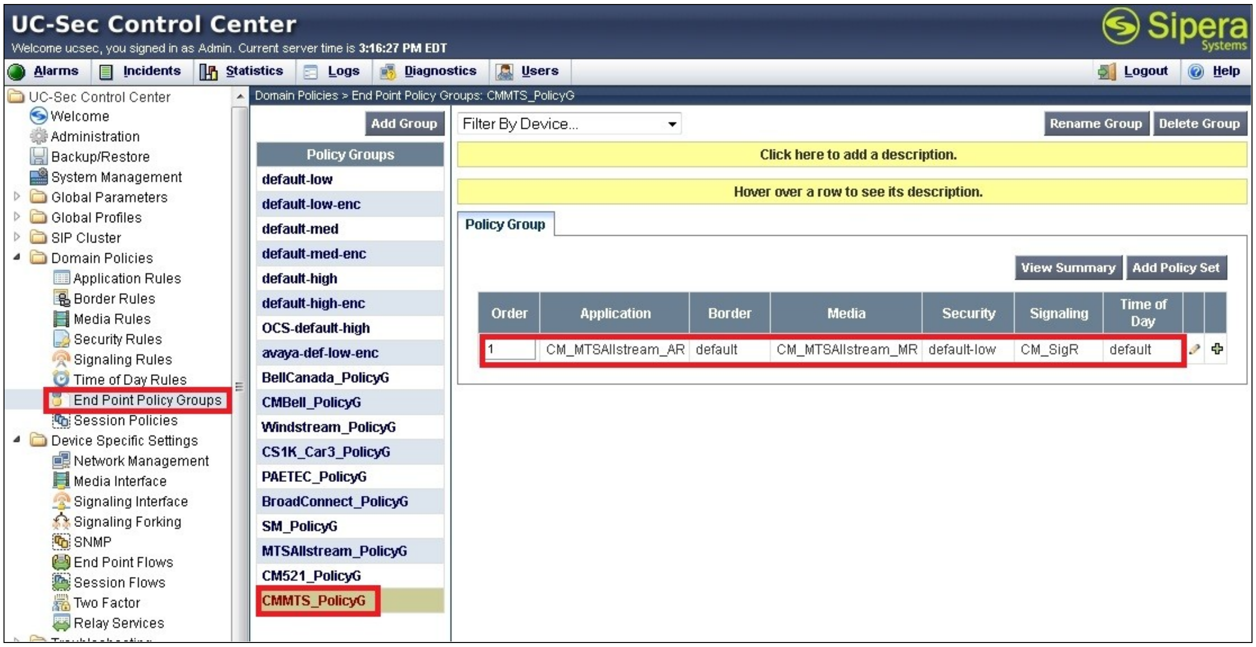


Domain Policies > End Point Policy Groups: MTSAllstream_PolicyG

Order	Application	Border	Media	Security	Signaling	Time of Day
1	CM_MTSAllstream_AR	default	CM_MTSAllstream_MR	default-high	MTSAllstream_SigR	default

6.3.4.2 Endpoint Policy Group for Communication Manager

The following screen shows **CM_PolicyG** created for **Communication Manager**:
- Set **Application** and **Media** rules created in **Section 6.3.1** and **Section 6.3.2**.
- Set **Signaling** rule **CM_SigR** created in **Section 6.3.3.2**.
- Set the **Border** and **Time of Day** rules to **default**.
- Set the **Security** rule to **default-low**.



Domain Policies > End Point Policy Groups: CMMTS_PolicyG

Order	Application	Border	Media	Security	Signaling	Time of Day
1	CM_MTSAllstream_AR	default	CM_MTSAllstream_MR	default-low	CM_SigR	default

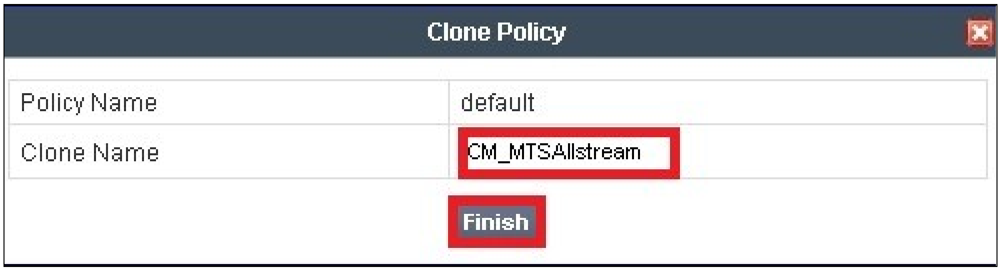
6.3.5. Session Policy

The **Session Policy** is applied based on the source and destination of a media session i.e., which codec is to be applied to the media session between its source and destination. The source and destination are defined in URI Group in **Section 6.2.1**.

In the compliance test, the **Session Policy** named **CM_MTSAllstream** is created to match the codec configuration on MTS Allstream. The policy also allows Avaya SBCE to anchor media in off-net call transfer scenarios.

To clone a **Session Policy**, navigate to **UC-Sec Control Center → Domain Policies e→ Session Policies**. With the **default** rule chosen, click on **Clone Rule** (not shown). It is applied to both Communication Manager and MTS Allstream.

Enter a descriptive name **CM_MTSAllstream** for the new policy and click **Finish**.

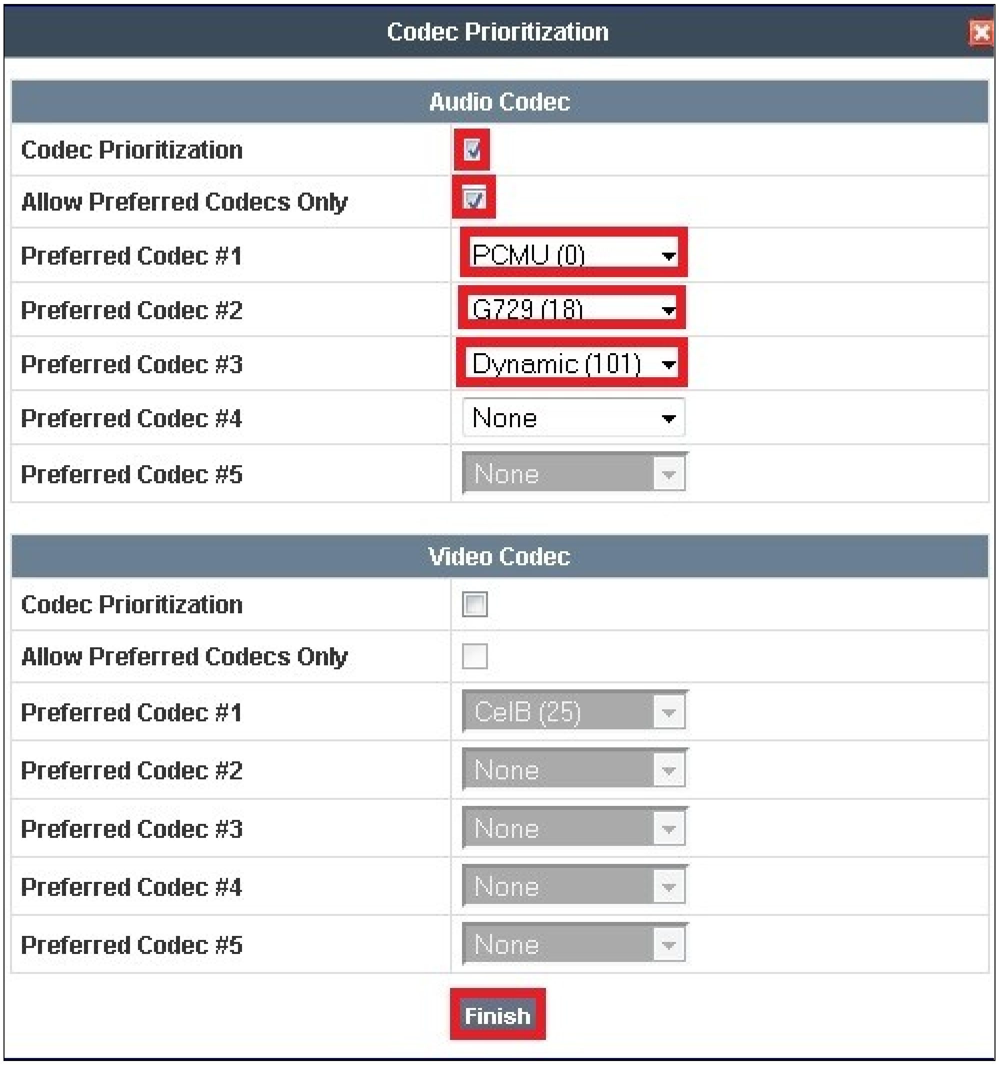


Policy Name	default
Clone Name	CM_MTSAllstream

MTS Allstream supports voice codec G.729 and G.711MU in prioritized order with payload 101 for RFC2833/ DTMF. To define **Codec Prioritization** for Audio Codec, select the profile **CM_MTSAllstream** created above, click on **Edit** (not shown). Select **Preferred Codec #1** as G.711MU, **Preferred Codec #2** as G.729 and **Preferred Codec #3** as Dynamic (101) for RFC2833/ DTMF. Check **Allow Preferred Codecs Only** to prevent the unsupported codec from being sent to both ends.

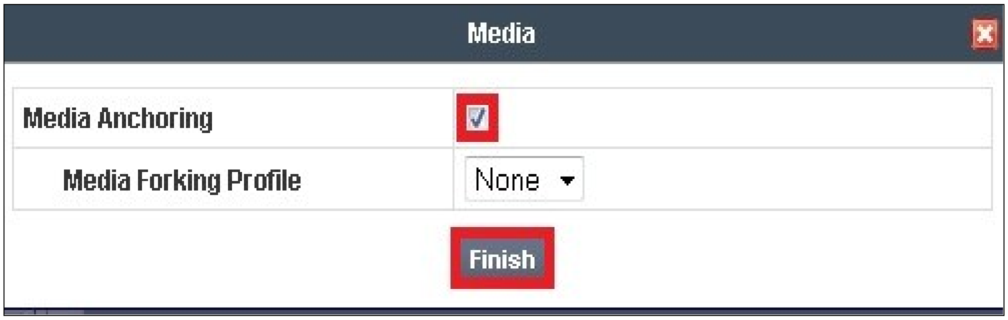
Notes:

- The T.38 fax is not yet supported by MTS Allstereeam SIP Trunking Service.
- This **Session Policy** prioritizes voice codec G.711MU to establish the voice call. It is mandatory for a G.711MU fax call to be successful because both Communication Manager and MTS Allstream cannot switch the voice call using different codec to G.711MU for fax.



Audio Codec	
Codec Prioritization	☑
Allow Preferred Codecs Only	☑
Preferred Codec #1	PCMU (0)
Preferred Codec #2	G729 (18)
Preferred Codec #3	Dynamic (101)
Preferred Codec #4	None
Preferred Codec #5	None
Video Codec	
Codec Prioritization	□
Allow Preferred Codecs Only	□
Preferred Codec #1	CelB (25)
Preferred Codec #2	None
Preferred Codec #3	None
Preferred Codec #4	None
Preferred Codec #5	None

To enable **Media Anchoring** on Avaya SBCE, select Session Policy C**M_MTSAllstream** created above then select tab **Media**, click **Edit** (not shown). Check on **Media Anchoring**.



Media Anchoring	☑
Media Forking Profile	None

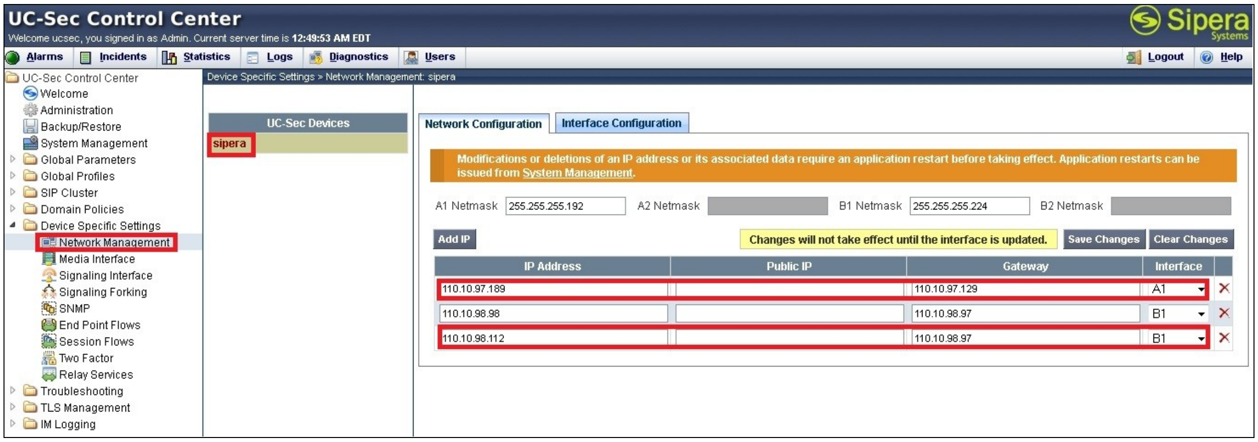
6.4. Device Specific Settings

The **Device Specific Settings** feature allows aggregate system information to be viewed and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network. Specifically, it gives the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality and protocol scrubber rules, end-point and session call flows, as well as the ability to manage system logs and control security features.

6.4.1. Network Management

The **Network Management** screen is where the network interface settings are configured and enabled. During the installation process of Avaya SBCE, certain network-specific information is defined such as device IP address (es), public IP address (es), netmask, gateway, etc. to interface the device to the network. This information populates the various **Network Management** tab displays, which can be edited as needed to optimize device performance and network efficiency.

Navigate to **UC-Sec Control Center → Device Specific Settings → Network Management** and under **Network Configuration** tab verify the IP addresses assigned to the interfaces and that the interfaces are enabled. The following screen shows the private interface is assigned to **A1** and the public interface is assigned to **B1**.



Welcome ucsec, you signed in as Admin. Current server time is **12:49:53 AM EDT**

Device Specific Settings > Network Management: sipera

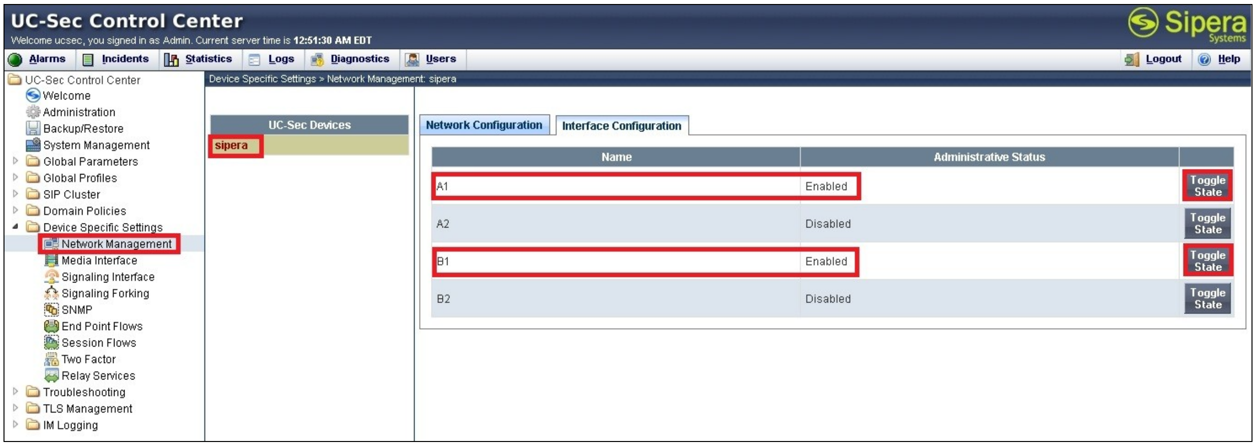
Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from System Management.

A1 Netmask 255.255.255.192 A2 Netmask B1 Netmask 255.255.255.224 B2 Netmask

Changes will not take effect until the interface is updated.

IP Address	Public IP	Gateway	Interface
110.10.97.189		110.10.97.129	A1
110.10.98.98		110.10.98.97	B1
110.10.98.112		110.10.98.97	B1

Enable the interfaces used to connect to the inside and outside networks on the **Interface Configuration** tab. The following screen shows interface **A1** and **B1** are **Enabled**. To enable an interface click it's **Toggle State** button.



Welcome ucsec, you signed in as Admin. Current server time is 12:51:30 AM EDT

Device Specific Settings > Network Management: sipera

Name	Administrative Status	
A1	Enabled	Toggle State
A2	Disabled	Toggle State
B1	Enabled	Toggle State
B2	Disabled	Toggle State

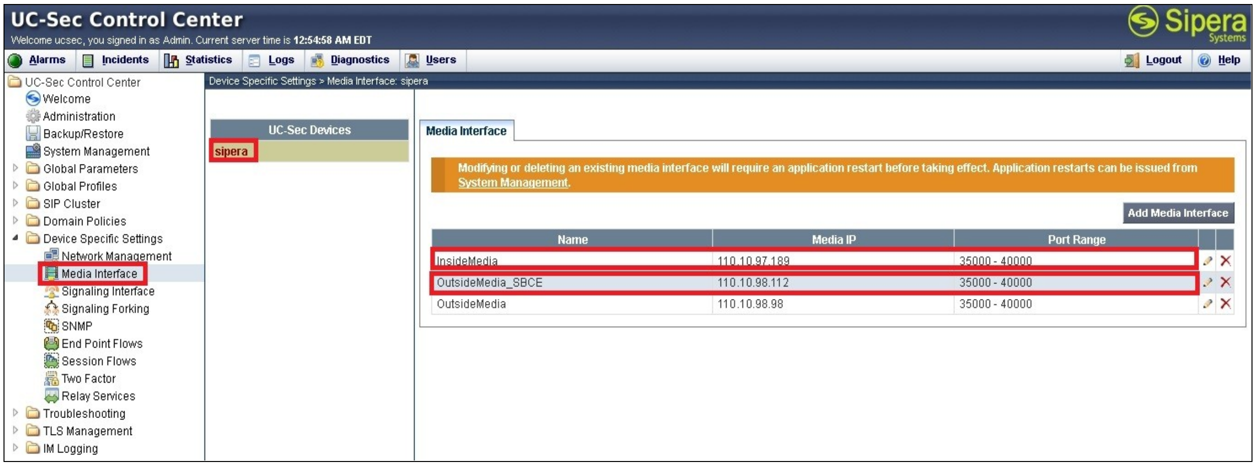
6.4.2. Media Interface

The **Media Interface** screen is where the media ports are defined. Avaya SBCE will open connection for RTP on the defined ports.

To create a new **Media Interface**, navigate to **UC-Sec Control Center → Device Specific Settings → Media Interface** and click **Add Media Interface** (not shown).

Media Interfaces are created for both the inside and outside interfaces. The following screen shows the **Media Interfaces** created in the compliance test.

Note: After the media interfaces are created, an application restart is necessary before the changes will take effect.



Welcome ucsec, you signed in as Admin. Current server time is 12:54:58 AM EDT

Device Specific Settings > Media Interface: sipera

Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from System Management.

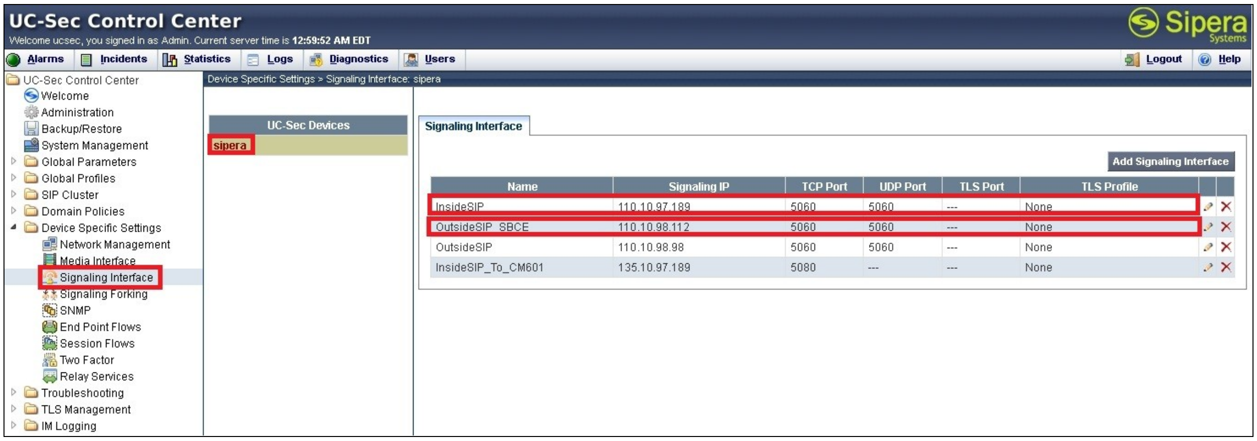
Name	Media IP	Port Range
InsideMedia	110.10.97.189	35000 - 40000
OutsideMedia_SBCE	110.10.98.112	35000 - 40000
OutsideMedia	110.10.98.98	35000 - 40000

6.4.3. Signaling Interface

The **Signaling Interface** screen is where the SIP signaling port is defined. Avaya SBCE will listen for SIP requests on the defined port.

To create a new **Signaling Interface**, navigate to **UC-Sec Control Center → Device Specific → Settings → Signaling Interface** and click **Add Signaling Interface** (not shown).

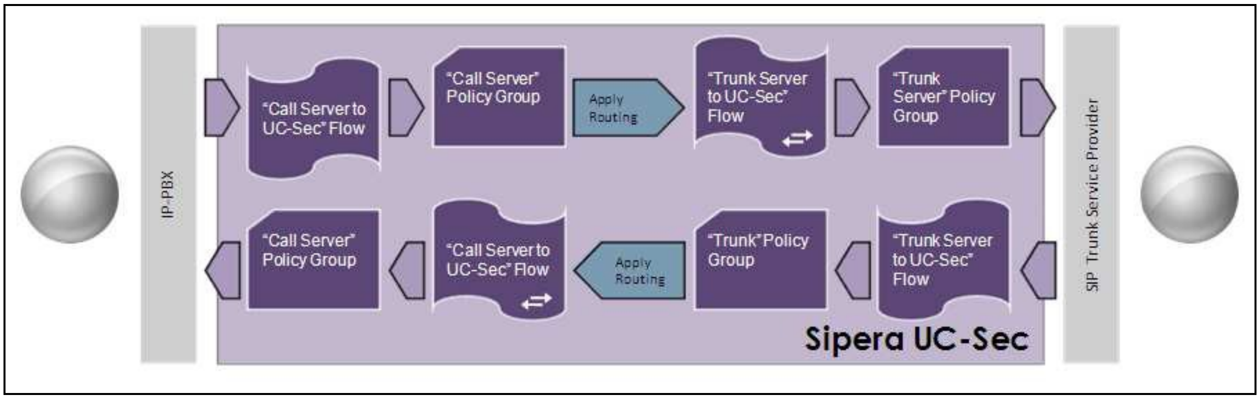
Signaling Interface is created for both inside and outside interfaces. The following screen shows the **Signaling Interfaces** created in the compliance test with TCP/5080 and UDP/5060 used respectively for the inside and outside IP interface.



Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile
InsideSIP	110.10.97.189	5060	5060	---	None
OutsideSIP_SBCE	110.10.98.112	5060	5060	---	None
OutsideSIP	110.10.98.98	5060	5060	---	None
InsideSIP_To_CM601	135.10.97.189	5080	---	---	None

6.4.4. End Point Flows - Server Flow

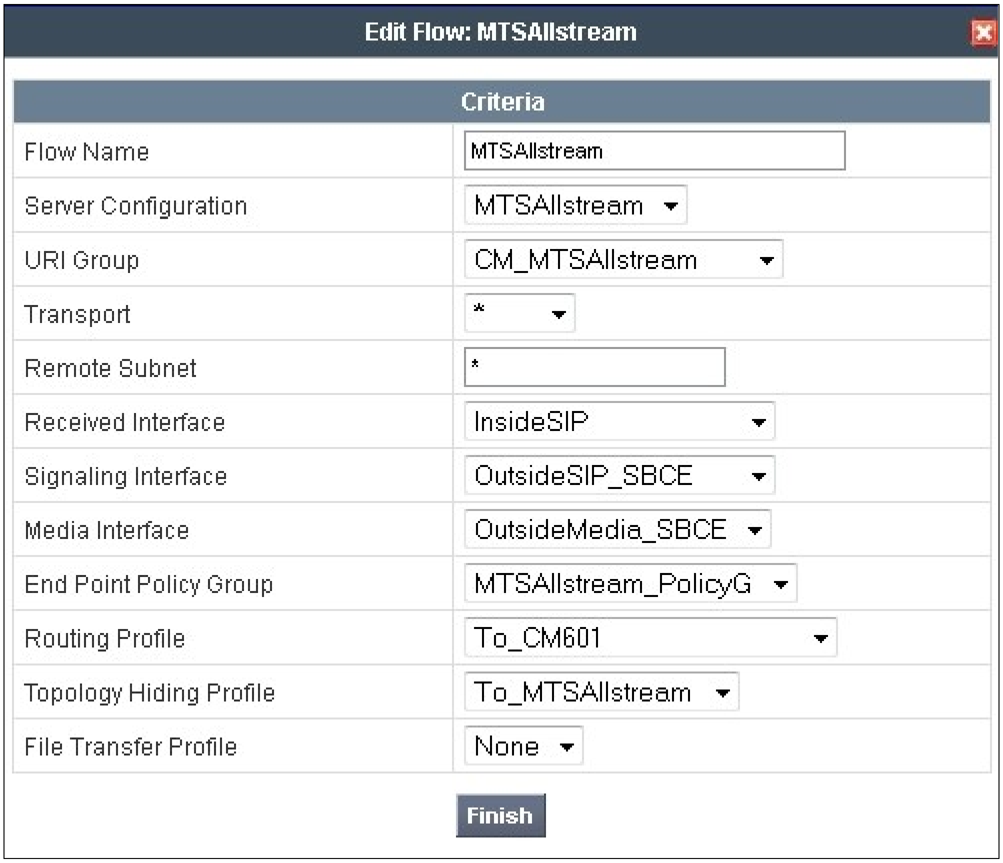
When a packet is received by UC-Sec, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screen illustrates the flow through Avaya SBCE to secure a SIP Trunk call.


IP-PBX
"Call Server to UC-Sec" Flow
"Call Server" Policy Group
Apply Routing
"Trunk Server to UC-Sec" Flow
"Trunk Server" Policy Group
"Call Server" Policy Group
"Call Server to UC-Sec" Flow
Apply Routing
"Trunk" Policy Group
"Trunk Server to UC-Sec" Flow
Sipera UC-Sec
SIP Trunk Service Provider

In the compliance test, separate **Server Flows** are created for MTS Allstream and Communication Manager. To create a **Server Flow**, navigate to **UC-Sec Control Center → Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add Flow** (not shown). In the new window that appears, enter the following values. The other fields are kept default.

- **Flow Name**: Enter a descriptive name.
- **Server Configuration**: Select a **Server Configuration** created in **Section 6.2.6** to assign to the Flow.
- **URI Group**: Select the URI Group created in **Section 6.2.1** to assign to the Flow.
- **Received Interface**: Select the Signaling Interface created in **Section 6.4.3** the Server Configuration is allowed to receive SIP messages from.
- **Signaling Interface**: Select the **Signaling Interface** used to communicate with the Server Configuration.
- **Media Interface**: Select the Media Interface created in **Section 6.4.2** used to communicate with the Server Configuration.
- **End Point Policy Group**: Select the End Point Policy Group created in **Section 6.3.4** assigned to the Server Configuration.
- **Routing Profile**: Select the Routing Profile created in **Section 6.2.2** the Server Configuration will use to route SIP messages to.
- **Topology Hiding Profile**: Select the Topology-Hiding profile created in **Section 6.2.3** to apply toward the Server Configuration.
- Click **Finish**.

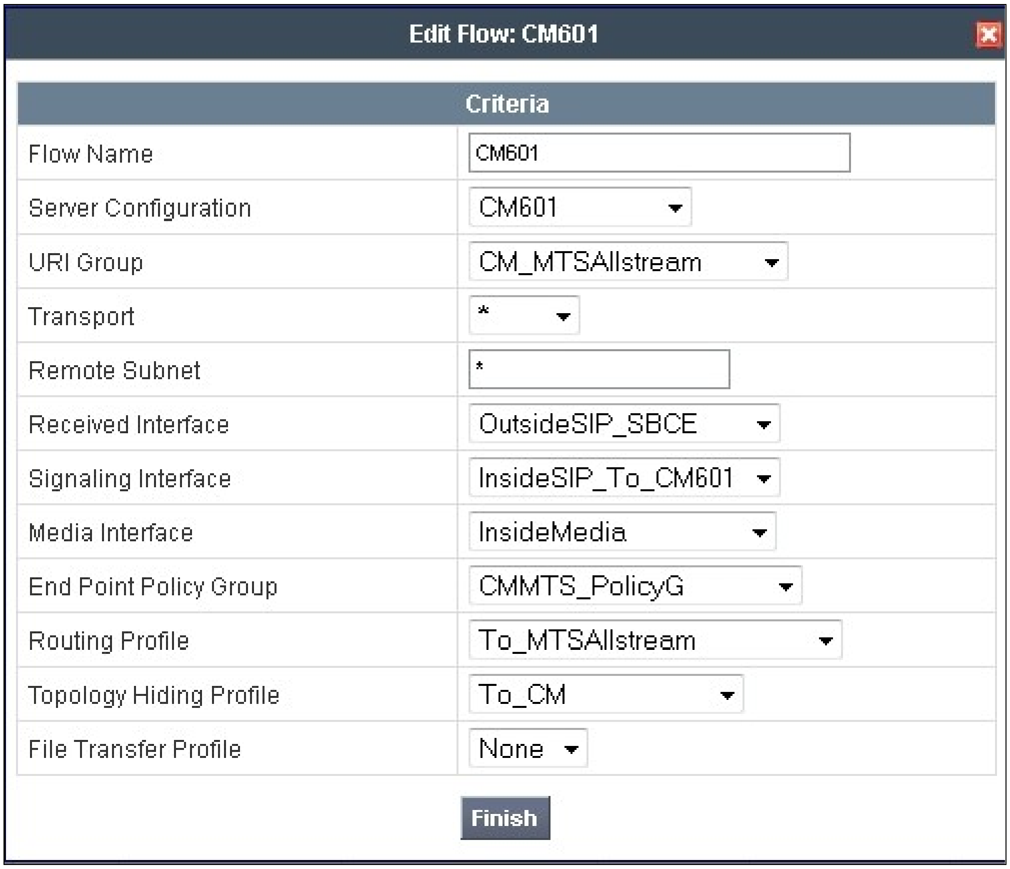
The following screen shows the Server **Flow Name (MTS Allstream)** configured for MTS Allstream.



Edit Flow: MTSAllstream

Criteria	
Flow Name	MTSAllstream
Server Configuration	MTSAllstream
URI Group	CM_MTSAllstream
Transport	*
Remote Subnet	*
Received Interface	InsideSIP
Signaling Interface	OutsideSIP_SBCE
Media Interface	OutsideMedia_SBCE
End Point Policy Group	MTSAllstream_PolicyG
Routing Profile	To_CM601
Topology Hiding Profile	To_MTSAllstream
File Transfer Profile	None

The following screen shows the Server **Flow Name** (**CM601**) configured for Communication Manager.



Criteria	
Flow Name	CM601
Server Configuration	CM601
URI Group	CM_MTSAllstream
Transport	*
Remote Subnet	*
Received Interface	OutsideSIP_SBCE
Signaling Interface	InsideSIP_To_CM601
Media Interface	InsideMedia
End Point Policy Group	CMMTS_PolicyG
Routing Profile	To_MTSAllstream
Topology Hiding Profile	To_CM
File Transfer Profile	None

6.4.5. Session Flows

The **Session Flows** feature allows to define certain parameters that pertain to the media portions of a call, whether it originates from the enterprise or outside the enterprise. This feature provides the complete and unparalleled flexibility to monitor, identify and control very specific types of calls based upon these user-definable parameters. **Session Flows** profiles SDP media parameters, to completely identify and characterize a call placed through the network.

To create a session flow, navigate to **UC-Sec Control Center → Device Specific Settings → Session Flows**. Click **Add Flow** (not shown).

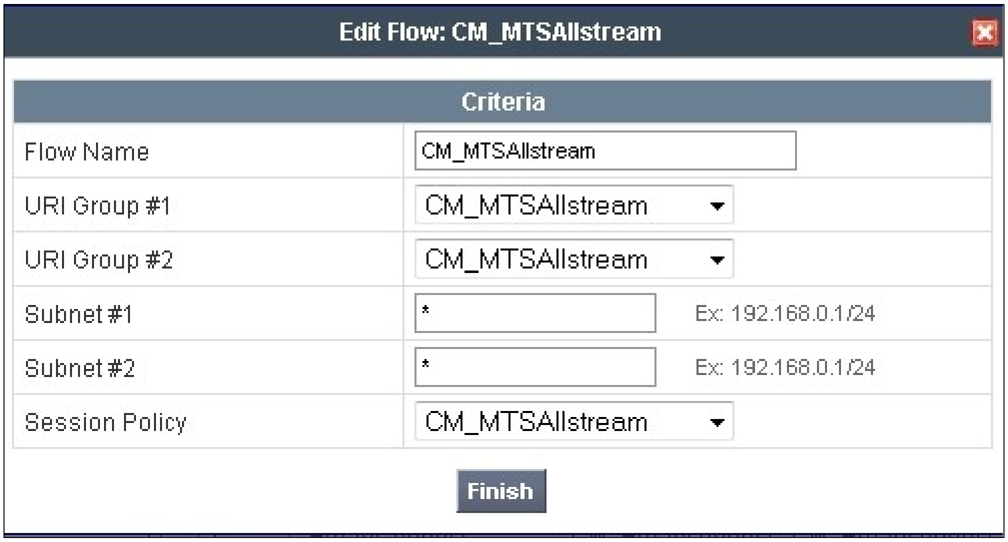
A common Session Flow is created for both Communication Manager and the MTS Allstream SIP Trunk. In the new window that appears, enter the following values. Use default values for the remaining fields:

- **Flow Name**: Enter a descriptive name.
- **URI Group #1**: Select the URI Group created in **Section 6.2.1** to assign to the Session Flow as the source URI Group.

- **URI Group #2**: Select the URI Group to assign to the Session Flow as the destination URI Group.
- **Session Policy**: Select the session policy created in **Section 6.3.5** to assign to the Session Flow.
- Click **Finish**.

Notes: A unique **URI Group** is used for source and destination, since it contains multiple URIs defined for the source as well as for the destination.

The following screen shows the **Session Flow** named **CM_MTSAllstream** is created.



Edit Flow: CM_MTSAllstream

Criteria		
Flow Name	CM_MTSAllstream	
URI Group #1	CM_MTSAllstream	
URI Group #2	CM_MTSAllstream	
Subnet #1	*	Ex: 192.168.0.1/24
Subnet #2	*	Ex: 192.168.0.1/24
Session Policy	CM_MTSAllstream	

7. MTS Allstream SIP Trunking Service Configuration

MTS Allstream is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya SBCE at enterprise side. MTS Allstream will provide the customer with the necessary information to configure the SIP connection from enterprise to the MTS Allstream. The information provided by MTS Allstream includes:

- IP address of the MTS Allstream Session Border Controller.
- MTS Allstream SIP domain. In the compliance test, MTS Allstream preferred to use IP address as a URI-Host.
- CPE SIP domain. In the compliance test, MTS Allstream preferred to use IP address of Avaya SBCE as a URI-Host.
- Supported codecs.
- DID numbers.
- IP addresses and port numbers used for signaling or media through any security devices.

The sample configuration between MTS Allstream and the Communication Manager for the compliance test is a static configuration. There is no registration on the SIP trunk implemented on either MTS Allstream or enterprise side.

8. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting commands.

Verification Steps:

1. Verify that endpoints at the enterprise site can place call to PSTN and that the call remains active for more than 35 seconds. This time period is included to satisfy SIP protocol timers.
2. Verify that endpoints at the enterprise site can receive call from PSTN and that the call can remain active for more than 35 seconds. This time period is included satisfy SIP protocol timers.
3. Verify that the user on PSTN can end an active call by hanging up.
4. Verify that an endpoint at the enterprise site can end an active call by hanging up.

Protocol Traces:
The following SIP headers are inspected using Wireshark traces:
- Request-URI: verify the request number and SIP domain
- From: verify the display name and display number
- To: verify the display name and display number
- P-Assert-Identity: verify the display name and display number
- Privacy: verify the "user, id" masking

The following attributes in SIP message body are inspected using Wireshark traces:
- Connection Information (c line): verify IP address of near end and far end endpoints
- Time Description (t line): verify session timeout value of near end and far end endpoints
- Media Description (m line): verify audio port, codec, DTMF event description
- Media Attribute (a line): verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes

Troubleshooting:

Avaya SBCE

Using a network sniffing tool (e.g., Wireshark) to monitor the SIP signaling messages between MTS Allstream and Avaya SBCE

Following is an example inbound call from MTS Allstream to Communication Manager.
- Inbound INVITE request from MTS Allstream:

```
INVITE sip:6477763571@110.10.98.112;user=phone SIP/2.0
Max-Forwards: 69
Session-Expires: 3600;refresher=uac
Min-SE: 600
Supported: timer, 100rel
To: <sip:6477763571@110.10.98.112;user=phone>
From: <sip:16139675258@220.20.2.12;user=phone>;tag=3541953942-353214
```

```
P-Asserted-Identity: <sip:16139675258@220.20.2.12;user=phone>
Call-ID: 88826-3541953942-353205@nextone-msw-lab-3.mtsallstream.com
CSeq: 1 INVITE
Allow: CANCEL, INVITE, BYE, OPTIONS, REGISTER, NOTIFY, INFO, REFER, SUBSCRIBE,
PRACK, UPDATE, MESSAGE, PUBLISH
Via: SIP/2.0/UDP 220.20.2.12:5060;branch=z9hG4bK4af7c3284dadb9cf037ccf2eb29f6d9f
Contact: <sip:16139675258@220.20.2.12:5060;tgrp=TOROONSBCIOT1>
Content-Type: application/sdp
Accept: application/sdp
Content-Length: 227

v=0
o=nextone-msw-lab-3 579335605 579335605 IN IP4 220.20.2.12
s=sip call
c=IN IP4 220.20.2.13
t=0 0
m=audio 20002 RTP/AVP 18 0 8 101
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

- 200OK/SDP response by Communication Manager:

```
SIP/2.0 200 OK
From: <sip:16139675258@220.20.2.12;user=phone>;tag=3541953942-353214
To: <sip:6477763571@110.10.98.112;user=phone>;tag=0784cf05e77e11d1b74f4990900
CSeq: 1 INVITE
Call-ID: 88826-3541953942-353205@nextone-msw-lab-3.mtsallstream.com
Contact: "MTS H323 3571" <sip:6477763571@110.10.98.112:5060;transport=udp>
Record-Route: <sip:110.10.98.112:5060;ipcs-line=33863;lr;transport=udp>
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,SUBSCRIBE,NOTIFY,REFER,INFO,PRACK,PUBLISH
Supported: 100rel,join,replaces,sdp-anat,timer
Via: SIP/2.0/UDP 220.20.2.12:5060;branch=z9hG4bK4af7c3284dadb9cf037ccf2eb29f6d9f
Accept-Language: en
Require: timer
Server: Avaya CM/R016x.00.1.510.1
P-Asserted-Identity: "MTS H323 3571" <sip:6477763571@avaya.com>
Session-Expires: 3600;refresher=uac
Content-Type: application/sdp
Content-Length: 198

v=0
o=- 1332965744 2 IN IP4 110.10.98.112
s=-
c=IN IP4 110.10.98.112
b=AS:64
t=0 0
m=audio 35232 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
```

Following is an example outbound call from Communication Manager to MTS Allstream.

- Outbound INVITE request from Communication Manager:

```
INVITE sip:16139675258@220.20.2.12 SIP/2.0
From: "MTS H323 3571" <sip:6477763571@220.20.2.12>;tag=05686a5f77e11dcb74f4990900
To: <sip:16139675258@220.20.2.12>
CSeq: 1 INVITE
Call-ID: 05686a5f77e11ddb74f4990900
Contact: "MTS H323 3571" <sip:6477763571@110.10.98.112:5060>
```

```
Record-Route: <sip:110.10.98.112:5060;ipcs-line=33874;lr;transport=udp>
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,SUBSCRIBE,NOTIFY,REFER,INFO,PRACK,PUBLISH
Supported: 100rel,join,replaces,sdp-anat,timer
User-Agent: Avaya CM/R016x.00.1.510.1
Max-Forwards: 70
Via: SIP/2.0/UDP 110.10.98.112:5060;branch=z9hG4bK-s1632-000876452326-1--s1632-
P-Asserted-Identity: "MTS H323 3571" <sip:6477763571@220.20.2.12>
Session-Expires: 1200;refresher=uac
Min-SE: 1200
Content-Type: application/sdp
Content-Length: 281

v=0
o=- 1332965790 1 IN IP4 110.10.98.112
s=-
c=IN IP4 110.10.98.112
b=AS:64
t=0 0
a=avf:avc=n prio=n
a=csup:avf-v0
m=audio 35234 RTP/AVP 0 8 18 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
```

- 200OK/SDP response by MTS Allstream:

```
SIP/2.0 200 OK
Session-Expires: 1200;refresher=uac
Require: timer
Via: SIP/2.0/UDP 110.10.98.112:5060;received=110.10.98.112;branch=z9hG4bK-s1632-
000876452326-1--s1632-
Record-Route: <sip:110.10.98.112:5060;ipcs-line=33874;lr;transport=udp>
To: <sip:16139675258@220.20.2.12>;tag=3541953990-700459
From: "MTS H323 3571" <sip:6477763571@220.20.2.12>;tag=05686a5f77e11dcb74f4990900
Call-ID: 05686a5f77e11ddb74f4990900
CSeq: 1 INVITE
Allow: CANCEL, INVITE, BYE, OPTIONS, REGISTER, NOTIFY, INFO, REFER, SUBSCRIBE,
PRACK, UPDATE, MESSAGE, PUBLISH
Contact: <sip:16139675258@220.20.2.12:5060>
Content-Type: application/sdp
Accept: application/sdp
Content-Length: 227

v=0
o=nextone-msw-lab-3 579823626 579823626 IN IP4 220.20.2.12
s=sip call
c=IN IP4 220.20.2.13
t=0 0
m=audio 20012 RTP/AVP 0 18 8 101
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Communication Manager Verification Steps

- **list trace station** <extension number> - Traces call to and from a specific station
- **list trace tac** <trunk access code number> - Trace call over a specific trunk group
- **status station** <extension number> - Displays signaling and media information for an active call on a specific station
- **status trunk** <trunk group number> - Displays trunk group information
- **status trunk** <trunk group number/channel number> - Displays signaling and media information for an active trunk channel

9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager 6.0.1 and Avaya Session Border Controller for Enterprise 4.0.5 to MTS Allstream SIP Trunking Service. MTS Allstream SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large the enterprises. MTS Allstream SIP Trunking Service provides a flexible, cost-saving alternative to traditional analog and ISDN-PRI trunks.

All of the test cases have been executed. Despite the number of observations seen during testing as noted in **Section 2.2**, the test results met the objectives outlined in **Section 2.1**. The MTS Allstream SIP Trunking Service is considered **compliant** with Avaya Aura® Communication Manager 6.0.1 and Avaya Session Border Controller for Enterprise 4.0.5.

10. References

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

[1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.03, February 2011.

[2] *Administering Avaya Aura® System Platform*, Release 6, June 2010.

[3] *Administering Avaya Aura® Communication Manager*, Release 6.0, June 2010, Document Number 03-300509.

[4] *Avaya Aura® Communication Manager Feature Description and Implementation,* Release 6.0, June 2010, Document Number 555-245-205.

[5] *Avaya one-X® Deskphone Edition for 9600 Series IP Telephones Administrator Guide,* Release 3.1, November 2009, Document Number 16-300698.

[6] *Avaya one-X® Deskphone SIP for 9600 Series IP Telephones Administrator Guide,* Release 2.6, June 2010, Document Number 16-601944.

[7] *Administering Avaya one-X® Communicator,* April 2011.

[8] *Using Avaya one-X® Communicator,* April 2011.

[9] *UC-Sec Install Guide (102-5224-400v1.01)*

[10] *UC-Sec Administration Guide (010-5423-400v106)*

[11] *RFC 3261 SIP: Session Initiation Protocol, http://www.ietf.org/*

[12] *RFC 3515, The Session Initiation Protocol (SIP) Refer Method, http://www.ietf.org/*

[13] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, http://www.ietf.org/*

[14] *RFC 4244, An Extension to the Session Initiation Protocol (SIP) for Request History Information, http://www.ietf.org/*

Product documentation for MTS Allstream SIP Trunking Service is available from MTS Allstream.

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.