Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Avaya SIP Trunk Architecture with the Verizon Business IP Trunk service offer in a SIP Trunk Redundant (2-CPE) Environment – Issue 1.0

Abstract

These Application Notes describe the steps to configure the Avaya SIP trunk architecture with Verizon Business IP Trunk service offer in a SIP Trunk Redundant (2-CPE) environment. The SIP Trunk Redundant (2-CPE) architecture provides for redundant SIP trunking between the Verizon Business IP Trunk service offer and the Avaya SIP telephony solution consisting of Avaya Communication Manager (version 5.1.1), Avaya SIP Enablement Services (version 5.1.1), and various Avaya telephones.

The Verizon Business IP Trunk service offer referenced within these Application Notes is designed for business customers with an Avaya SIP telephony solution. The service provides local and/or long Distance PSTN calling via standards-based SIP trunks directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted by the Verizon Interoperability Test Lab with support from the Avaya Solution and Interoperability Test Lab.

Table of Contents

1. Introduction

These Application Notes describe the steps to configure the Avaya SIP trunk architecture with the Verizon Business IP Trunk service offer in a SIP Trunk Redundant (2-CPE) environment. The Verizon SIP Trunk Redundant (2-CPE) architecture provides for redundant SIP trunks between the Verizon Business IP Trunk service offer and an Avaya SIP telephony solution customer premises equipment (CPE) consisting of Avaya Communication Manager (version 5.1.1), Avaya SIP Enablement Services (version 5.1.1), and various Avaya SIP and H.323 telephones.

Note - The Verizon SIP Trunk Redundant (2-CPE) architecture is a service option and its use is not a requirement of the Verizon Business IP Trunk service offer.

The Verizon Business IP Trunk service offer described in these Application Notes is designed for business customers using Avaya Communication Manager and Avaya SIP Enablement Services. The service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks. These trunks connect to the Avaya systems directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

Verizon Business and Avaya developed the SIP Trunk Redundant (2-CPE) architecture to ensure that SIP trunk calls can be automatically rerouted to bypass SIP trunk failures due to network or component outages.

Note – The Avaya Communication Manager and Avaya SIP Enablement Services provisioning described in these Application Notes are also applicable on non SIP Trunk Redundant (2-CPE) architectures.

1.1. Verizon Business IP Trunk Service Offer Overview

The Verizon Business IP Trunk service offer described within these Application Notes is designed for business customers using Avaya Communication Manager and Avaya SIP Enablement Services. The service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks. These trunks connect to the Avaya systems directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

Single site users can choose combined local and long distance service packages that are available in 200+ metropolitan service areas or a long distance package only. Multi-site or campus configurations may only choose long distance service.

Outbound long distance voice and fax services include domestic direct-dial calling within the United States. They may also include integrated dedicated toll-free service, enabling customers to use a single, cost-effective connection for incoming toll-free calls and outbound VoIP traffic. International Europe, Middle East, and Africa (EMEA) dialing is also supported.

G.711 fax is supported over the Verizon Business IP Trunk service offer.

Note – An optional supported for G.711 fax configuration is the use of an Audio Codes MP-202 Gateway between Avaya Communication Manager and the fax device. See reference [10] for more information.

The Verizon Business IP Trunk service offer is delivered to the enterprise customer location using Verizon Business Internet Dedicated Access (IDA). Although through not described in these Application Notes, Verizon Business also supports service connections for their IP Trunk service offer using Private IP (PIP) service connections.

1.2. The SIP Trunk Redundant (2-CPE) Architecture Option

The SIP Trunk Redundant (2-CPE) architecture described in **Section 1** is based on a customer location having two Avaya SIP Enablement Services (SES) servers. One is designated as the Primary and one as the Secondary. Both Avaya SIP Enablement Services are provisioned identically except each will have a unique IP address. Avaya Communication Manager is provisioned with SIP trunks for “local” SIP calls, as well as SIP Trunks for inbound and outbound “off network” SIP calls. Local SIP calls are those between SIP, H.323, and legacy stations provisioned on Avaya Communication Manager. This local traffic also includes other SIP messages such as SIP phone registration, state updates, etc. Off network traffic includes all SIP messaging between the Avaya Communication Manager/ Avaya SIP Enablement Services environment and other destinations. These local and off network SIP trunks are provisioned between Avaya Communication Manager and both Avaya SIP Enablement Services servers (see **Figure 1**). Both of the Avaya SIP Enablement Services servers are provisioned with the ***same*** Fully Qualified Domain Name (FQDN).

Note - See **Section 1.5** for more information on Fully Qualified Domain Names (FQDN).

Voice and G.711 fax calls utilize their own inbound and outbound trunks. This allows specific voice and fax parameters to be provisioned (e.g. codec selection). See **Sections 3.1.6** and **3.2.7** for more information on SIP trunk provisioning.

For inbound calls (to CPE), the Verizon network may either send calls alternatively to the Primary and Secondary SIP Enablement Services servers or it may send calls to the Secondary SIP Enablement Services server only if calls cannot complete via the Primary. The method chosen is based on Verizon service provisioning for that particular customer. These SIP calls will be sent to Avaya Communication Manager from the Avaya SIP Enablement Services on one of the associated inbound SIP trunks.

For outbound calls (from CPE), Avaya Communication Manager is provisioned to select the Primary outbound SIP trunks first. The Secondary outbound SIP trunks are only used if outbound calls via the Primary trunks fail. In the reference configuration *Domain Based Routing* was used for outbound calls (see **Section 1.5** for more information on Domain based Routing). Domain

Based Routing relies on the Verizon network to provide call destination routing information to the Avaya SIP Enablement Services.

Local SIP trunks are used for calls between Avaya SIP telephones and Avaya H.323 and legacy telephones, as well as other Avaya SIP telephone/Avaya Communication Manager communications. See **Section 3.1.6** for a description of the SIP trunks used and their provisioning.

In the tested SIP Trunk Redundant (2-CPE) configuration, the Verizon network uses two Session Border Controllers (SBCs) between the Verizon Business IP trunk service offer network and the Avaya CPE. This adds an additional layer of redundancy.

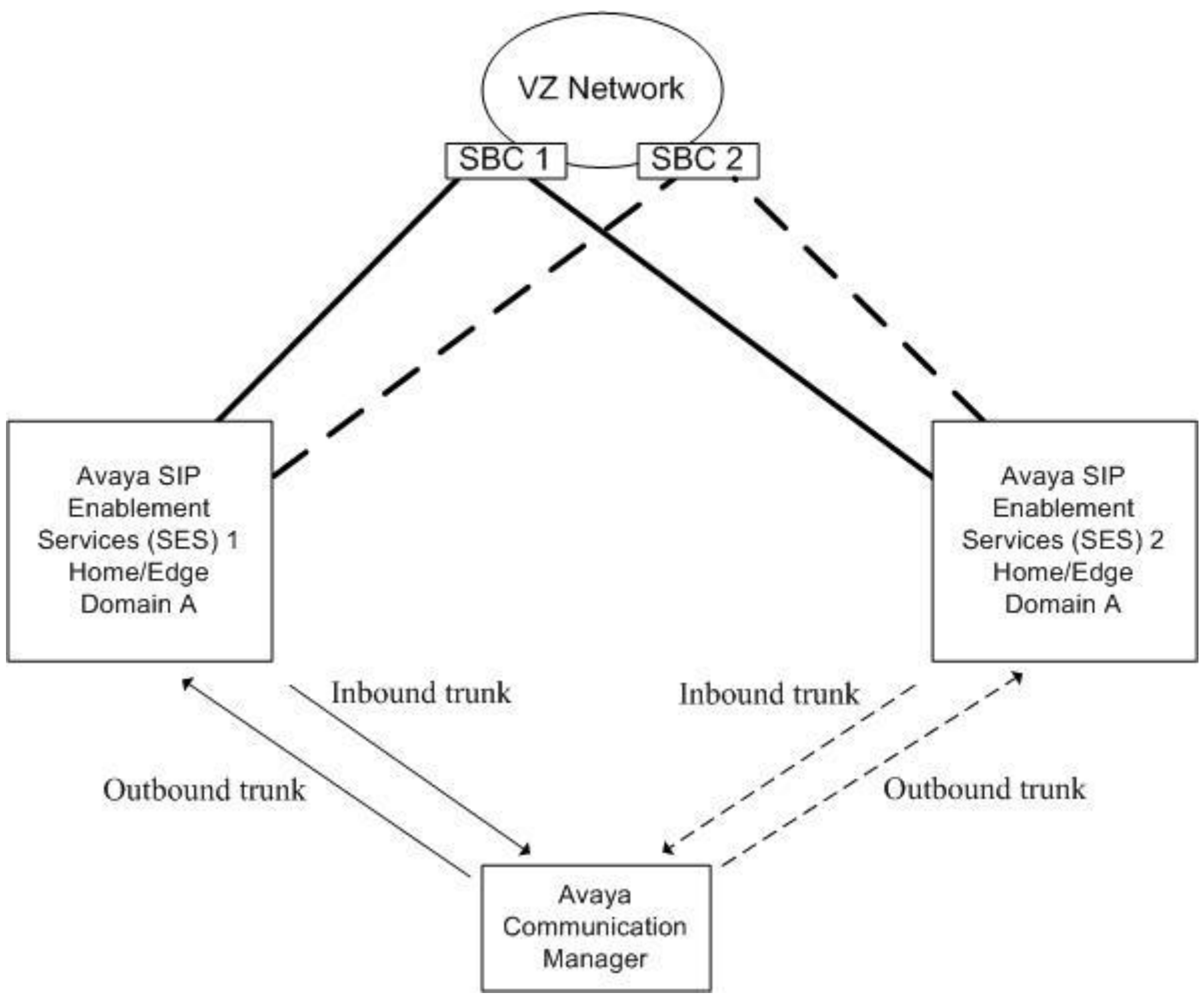

Figure 1: The SIP Trunk Redundant (2-CPE) Architecture

1.3. The SIP Trunk Redundant (2-CPE) Reference Configuration

Figures 2 and **3** illustrate two sample SIP Trunk Redundant (2-CPE) solutions. These are the configurations that were used for the DevConnect compliance testing. One configuration was used to test United States “domestic” dialing and the other was used to test Europe, Middle East, and Africa (EMEA) dialing.

1.3.1 Domestic and EMEA Common Architecture

The following architectural requirements were used in both the Domestic and EMEA test environments and are discussed in detail in subsequent sections.

- Primary and Secondary Avaya SIP Enablement Services servers.
  - Identical provisioning except for unique IP addresses.
  - Primary and Secondary Avaya SIP Enablement Services servers assigned the same Fully Qualified Domain Name (FQDN).
  - Communication Manager Server Maps for inbound calls.
  - No Host Maps required for outbound calls.
- Avaya Communication Manager.
  - Separate Primary and Secondary SIP trunks for Inbound Voice traffic.
    - Signaling Groups defined with <blank> Far-end Domain fields.
    - Signaling Groups defined with Near-end port 5061.
  - Separate Primary and Secondary SIP trunks for Outbound Voice traffic.
    - Signaling Groups defined with Far-end Domain field of the Verizon Fully Qualified Domain Name (FQDN).
    - Signaling Groups defined with Near-end port 5061.
  - Separate Primary and Secondary SIP trunks for Inbound G.711 fax traffic.
    - Signaling Groups defined with <blank> Far-end Domain field.
    - Signaling Groups defined with Near-end port 5062.
  - Separate Primary and Secondary SIP trunks for Outbound G.711 fax traffic.
    - Signaling Groups defined with Far-end Domain field of the Verizon Fully Qualified Domain Name (FQDN).
    - Signaling Groups defined with Near-end port 5062.
  - Separate Primary and Secondary SIP trunks for Local SIP telephone traffic.
    - Signaling Groups defined with Far-end Domain field of the Avaya SIP Enablement Services Fully Qualified Domain Name (FQDN).
    - Signaling Groups defined with Near-end port 5061.
  - Voice traffic uses G.729A and G.711Mu codecs.
  - Fax traffic uses G.711Mu and G.711A codecs.
  - Voice stations assigned a Class of Restriction (COR) with a higher Facility Restriction Level (FRL).
  - Fax stations assigned a Class of Restriction (COR) with a lower Facility Restriction Level (FRL).
  - Outbound Voice and G.711 fax calls utilize Look Ahead Routing (LAR) to select Primary and Secondary SIP Trunks based on the calling station COR/FRL.
  - Outbound Voice and G.711 fax calls will only select their associated Secondary trunks if the initial call via the Primary trunk fails.

1.3.2 Domestic Dialing – Avaya Components

The simulated domestic customer location included:

- An Avaya S8500 Media Server with an Avaya G650 Media Gateway. The S8500 served as the host processor for Avaya Communication Manager.
- Avaya 4600 Series IP telephones using the H.323 software bundle.
- Avaya 9600 Series IP telephones using the SIP software bundle.
- Two Avaya S8500 SIP Enablement Services servers.

1.3.3 Domestic Dialing Example

The following are examples of domestic outbound and inbound voice and G.711 fax calls. See **Sections 3.1** and **4.1** for Domestic Avaya Communication Manager and Avaya SIP Enablement Services provisioning.

Given:

- Voice station 1001
- Fax station 1004
- Voice Primary Inbound SIP trunk 12 and Secondary SIP trunk 13
- Voice Primary Outbound SIP trunk 14 and Secondary SIP trunk 15
- Fax Primary Inbound SIP trunk 20 and Secondary SIP trunk 21
- Fax Primary Outbound SIP trunk 18 and Secondary SIP trunk 19

Inbound

- Inbound voice calls to 1001 match the Avaya SIP Enablement Services Communication Manager voice Map that has the Contact with port 5061
  - Calls arrive on Inbound voice trunk 12 (Primary) or 13 (secondary) and connect to station 1001 using G729 or G711 codecs
- Inbound G.711 fax calls to 1004 match the Avaya SIP Enablement Services Communication Manager fax Map that has the Contact with port 5062
  - Calls arrive on Inbound fax trunk 20 (Primary) or 21 (Secondary) on port 5062 and connect to fax station 1004 using G711 codecs
- All inbound Communication Manager Signaling groups use a <blank> Far-end domain to accept calls from all foreign domains.

Outbound

- Communication Manager stations dial 9 and number 530352xxxx
  - ARS sends the call to Route Pattern 16
    - Route Pattern 16 is configured as follows:
      - Voice Trunk 14 (Primary) and FRL 2
      - Voice Trunk 15 (Secondary) and FRL 2
      - Fax Trunk 18 (Primary) and FRL 1
      - Fax Trunk 19 (Secondary) and FRL 1
        - Voice stations are set to COR 1 which has the higher FRL priority 2 set. These voice calls will select trunk 14 then 15 and go out on port 5061 using G729 or G711 codecs.
        - Fax stations are set to COR 2 which has the lower FRL priority 1 set. These G.711 fax calls will select trunk 18 then 19 and go out on port 5062 using G711 codecs.
  - All outbound Communication Manager Signaling groups send calls to the Far-end domain of schst1n0004.icpiptrunksit2.gsiv.com (Broadsoft Fully Qualified Domain Name (FQDN)).

- No SIP Enablement Services Host Maps (used for outbound calls) are required.

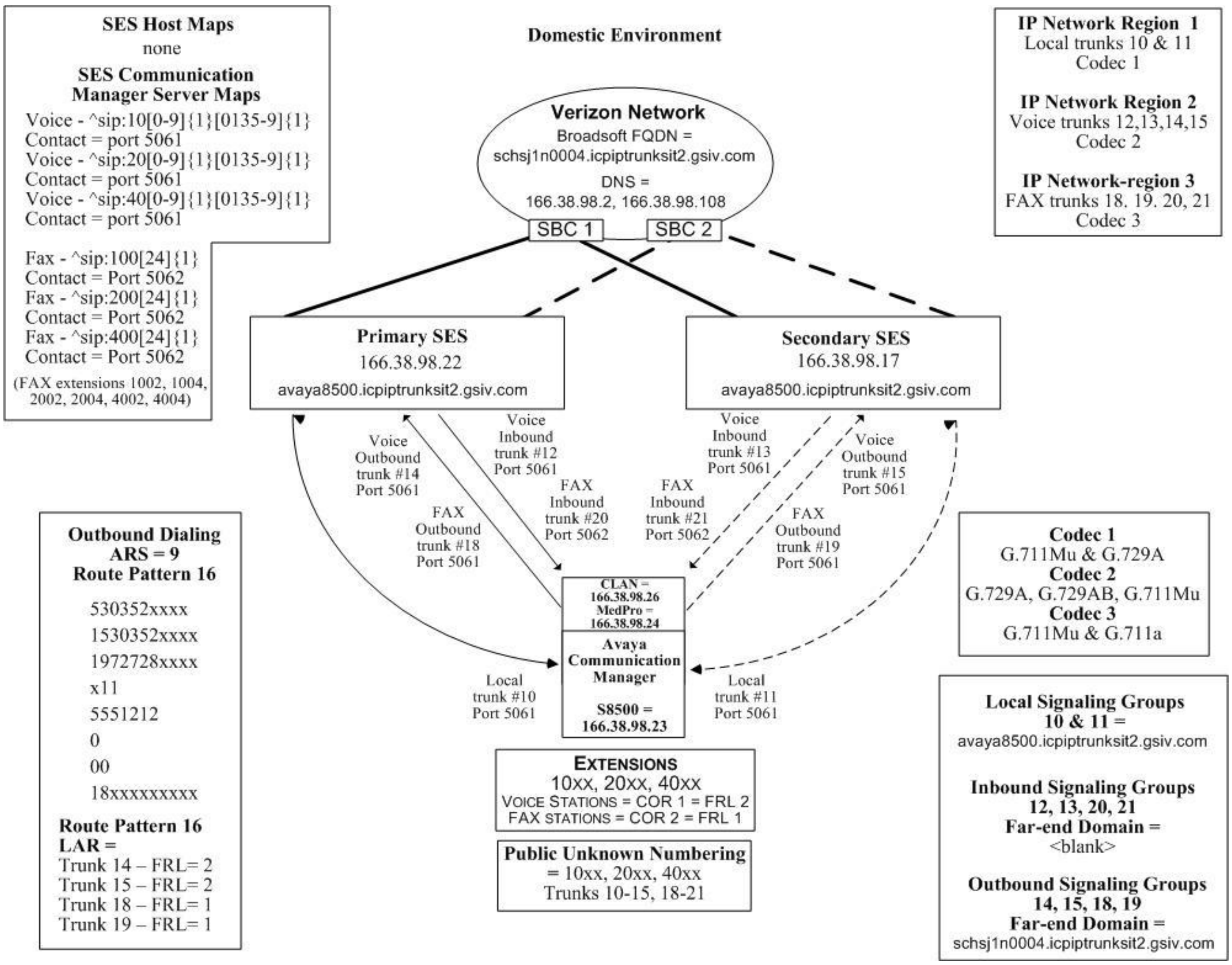

Figure 2: The SIP Trunk Redundant (2-CPE) Reference Configuration for Domestic Dialing

1.3.4 EMEA Dialing – Avaya Components

The Avaya SIP telephony solution used as a simulated domestic customer location included:

- An Avaya S8300 Media Server with an Avaya G350 Media Gateway. The S8300 served as the host processor for Avaya Communication Manager.
- Avaya 4600 Series IP telephones using the H.323 software bundle.
- Avaya 9600 Series IP telephones using the SIP software bundle.
- Two Avaya S8500 SIP Enablement Services servers.

1.3.5 EMEA Dialing Example

The following are examples of outbound and inbound voice and G.711 fax calls. See **Sections 3.2** and **4.2** for Avaya Communication Manager and Avaya SIP Enablement Services provisioning.

Given:

- Voice station 1031
- Fax station 1039
- Voice Primary Inbound SIP trunk 12 and Secondary SIP trunk 13
- Voice Primary Outbound SIP trunk 14 and Secondary SIP trunk 15
- Fax Primary Inbound SIP trunk 20 and Secondary SIP trunk 21
- Fax Primary Outbound SIP trunk 18 and Secondary SIP trunk 19

Inbound

- Inbound voice calls to 1031 match the Avaya SIP Enablement Services Communication Manager voice Map that has the Contact with port 5061
  - Calls arrive on Inbound voice trunk 12 (Primary) or 13 (secondary) and connect to station 1031 using G729 or G711 codecs
- Inbound G.711 fax calls to 1039 match the Avaya SIP Enablement Services Communication Manager fax Map that has the Contact with port 5062
  - Calls arrive on Inbound fax trunk 20 (Primary) or 21 (Secondary) on port 5062 and connect to fax station 1039 using G711 codecs
- All inbound Communication Manager Signaling groups use a <blank> Far-end domain to accept calls from all foreign domains.

Outbound

- Communication Manager stations dial 9 and number 00441189056849
  - ARS sends the call to Route Pattern 16
    - Route Pattern 16 is configured as follows:
      - Voice Trunk 14 (Primary) and FRL 2
      - Voice Trunk 15 (Secondary) and FRL 2
      - Fax Trunk 18 (Primary) and FRL 1
      - Fax Trunk 19 (Secondary) and FRL 1
        - Voice stations are set to COR 1 which has the higher FRL priority 2 set. These voice calls will select trunk 14 then 15 and go out on port 5061 using G729 or G711 codecs.
        - Fax stations are set to COR 2 which has the lower FRL priority 1 set. These G.711 fax calls will select trunk 18 then 19 and go out on port 5062 using G711 codecs.
  - All outbound Communication Manager Signaling groups send calls to the Far-end domain of schst1n0004.emeaiptrunksit2.gsiv.com (Broadsoft Fully Qualified Domain Name (FQDN)).
    - No Avaya SIP Enablement Services Host Maps (used for outbound calls) are required.

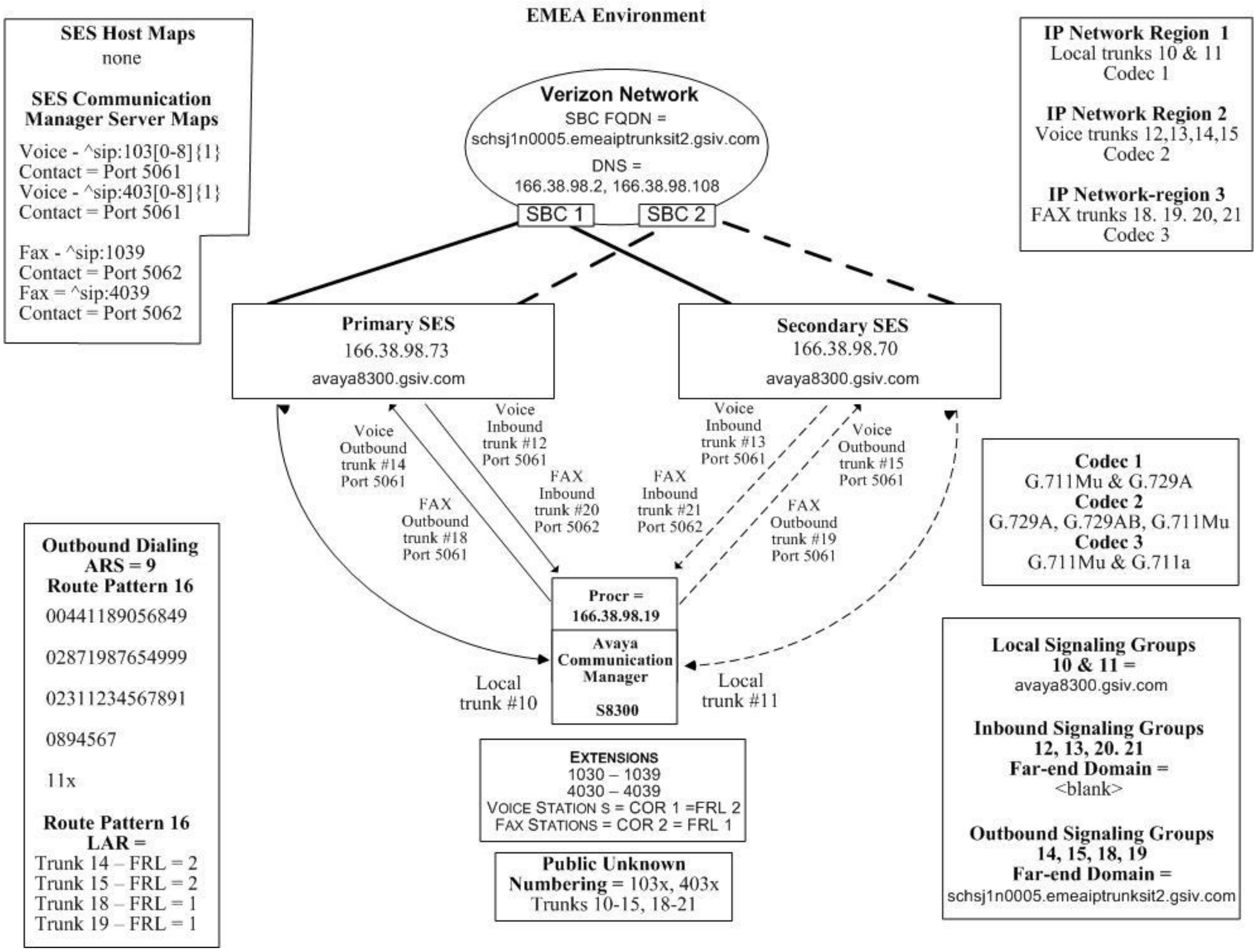

Figure 3: The SIP Trunk Redundant (2-CPE) Reference Configuration for EMEA Dialing

1.3.6 Reference Configuration - Verizon Interoperability Test Lab

The DevConnect compliance testing was conducted by the Verizon System Interoperability Test Lab in Richardson, Texas with support from the Avaya Solution and Interoperability Test Lab in Lincroft, New Jersey.

In the reference configuration, the Verizon Business solution included managed IP access service. Verizon Business is responsible for the configuration of this service and therefore those details are not included within these Application Notes.

Figure 4 shows the reference configuration used for the DevConnect compliance testing.

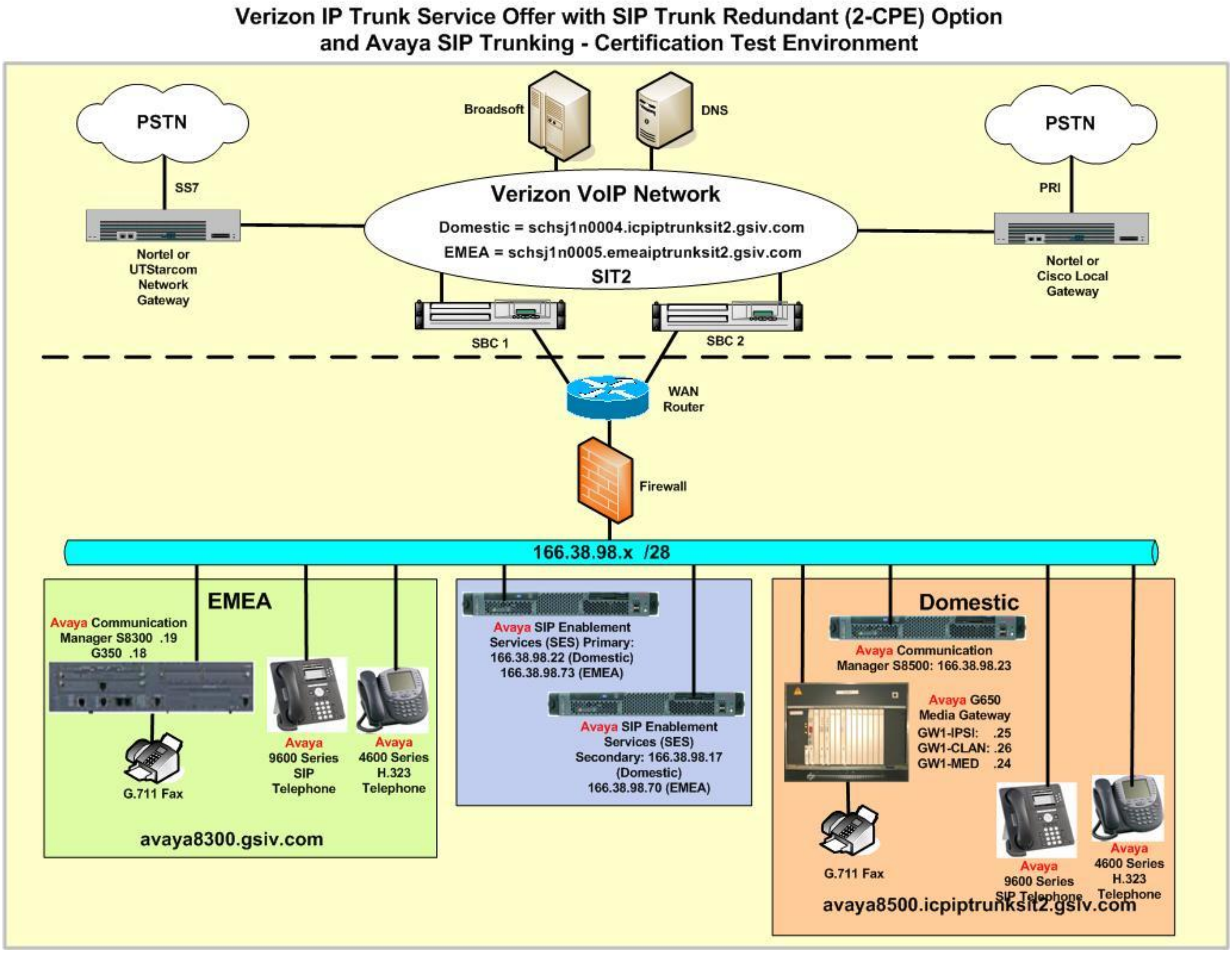

Figure 4: DevConnect Compliance Testing SIP Trunk Redundant (2-CPE) Reference Configuration

1.4. Avaya SIP Trunks

The public SIP trunks to the Verizon Business IP Trunk service offer works with any combination of Avaya analog, H.323 and SIP telephones. In the Avaya SIP trunk architecture, the Avaya SIP Enablement Services functions as a SIP proxy through which all incoming and outgoing SIP messages flow to the Verizon Business IP Trunk service offer. Avaya's SIP Enablement Services also supports protocol conversion between Verizon's SIP/UPD trunk termination and Avaya's Communication Manager's SIP/TLS (default) or SIP/TCP trunk interface. There is no direct SIP signaling path between the Verizon Business IP Trunk service offer and Avaya Communication Manager or Avaya SIP endpoints.

For incoming calls, the Avaya SIP Enablement Services uses Communication Manager Server Maps to direct the incoming SIP messages to Avaya Communication Manager. Once the message arrives at Avaya Communication Manager, further incoming call treatment such as incoming digit translations, class of service restrictions, etc., may be performed.

All outgoing calls to the PSTN are processed within Avaya Communication Manager and are subject to outbound features such as Automatic Route Selection, digit manipulation, class of service restrictions and Look Ahead Routing (see **Section 3.1.8**). Once Avaya Communication Manager selects a SIP trunk for the outbound call, the SIP signaling is routed to the appropriate Avaya SIP Enablement Services. Based on how the Avaya Communication Manager SIP trunking is defined, the Avaya SIP Enablement Services may use Host Maps (see **Appendix B**), an Outbound Proxy, or Domain Based Routing with DNS lookups (see **Section 1.5** below) to direct the outbound SIP calls. Domain Based Routing was used for outbound calls in the reference configuration.

Outbound Calling Method	Avaya Communication Manager Outbound SIP Trunk Far End Fully Qualified Domain Name (FQDN)	Avaya SIP Enablement Services Call Routing Method
Avaya SIP Enablement Services Host Maps	Avaya SIP Enablement Services SIP Fully Qualified Domain Name (FQDN)	Host Maps
Domain Based Routing	Foreign Fully Qualified Domain Name (FQDN)	Outbound Proxy or DNS lookup.

Table 1: SIP Trunk Outbound Call Routing Methods

1.5. Domain Based Routing for Outbound calls

In the reference configuration, the Avaya SIP Enablement Services uses DNS lookups to direct outbound calls to proper the Verizon Business Service destination. This method is referred to as *Domain Based Routing*. When Domain Based Routing is used, Avaya Communication Manager specifies a foreign Fully Qualified Domain Name (FQDN) in the outbound SIP trunk provisioning. A foreign Fully Qualified Domain Name (FQDN) is one that *does not* match the SIP Domain specified for the Avaya SIP Enablement Services (see **Section 4**). The Fully Qualified Domain Name (FQDN) used for the outbound SIP trunk was the Fully Qualified Domain Name (FQDN) of the Verizon Session Border Controller (SBC). This foreign domain is specified as the destination for the INVITE sent by Avaya Communications Manager to the Avaya SIP Enablement Services. The Avaya SIP Enablement Services then does a DNS query to the Verizon DNS server for this foreign domain and forwards the INVITE based on the DNS response. When a foreign Fully Qualified Domain Name (FQDN) is specified in conjunction with Domain Based Routing, no other outbound call route provisioning (e.g. Host Maps) is required for the Avaya SIP Enablement Services other than provisioning a DNS IP address(s). This provides the most flexible outbound route provisioning for the CPE as all the call route decisions take place in the Verizon network. In the reference configuration, the Verizon network used two SBCs. Both of these SBCs were specified in the DNS SRV responses to the Avaya SIP Enablement Services. This provided an additional layer of redundancy.

Note – There is an alternative method for configuring Domain Base Routing where Avaya Communication Manager that *does* use Avaya SIP Enablement Services Host Maps. This alternative method for configuring Domain Base Routing is described in reference [11].

1.5.1 Specifying Domains

The terms Fully Qualified Domain Name (FQDN) and Domain are sometimes used interchangeably even though they have different meanings. Typically *domain* means just that – a domain name such as *company.com*. An Fully Qualified Domain Name (FQDN) typically includes a *host name* with the domain – *big.company.com*. In the reference configuration, Verizon supplied Fully Qualified Domain Name (FQDN)s to be used by both EMEA and Domestic as well as supplying an Fully Qualified Domain Name (FQDN) for the Verizon service node for use in DNS queries.

1.5.2 Domain Based Routing Provisioning Example

The following steps describe how Domain Based Routing may be provisioned on Avaya Communication Manager and the Avaya SIP Enablement Services.

1. **Avaya Communication Manager**
   a. Avaya Communication Manager has the *Authoritative Domain* field of the *ip-network-region* form (see **Section 3.1.4**) provisioned with the Fully Qualified Domain Name (FQDN) of the Avaya SIP Enablement Services *SIP Domain* (see **Section 4.1**). This is the region associated with IP telephones or the region associated with the C-LAN/Procr for legacy phones.
   b. Avaya Communication Manager will use this value in the INVITE *From* header.
   c. The Avaya Communication Manager outbound trunk has a *foreign Fully Qualified Domain Name (FQDN)* specified in the Far-End Domain field of the associated Signaling Group form. The term Foreign means that the Fully Qualified Domain Name (FQDN) *does not* match the Fully Qualified Domain Name (FQDN) specified in the *SIP Domain* field of the Avaya SIP Enablement Services. Avaya Communication Manager will use this foreign Fully Qualified Domain Name (FQDN) in the *Request URI* of the INVITE.
   d. Avaya Communication Manager sends the INVITE to the Avaya SIP Enablement Services.

2. **Avaya SIP Enablement Services**
   a. The Avaya SIP Enablement Services compares the *Request URI* of the INVITE received from Avaya Communications Manager to its *SIP Domain* and sees that it is *not* authorative for the foreign Fully Qualified Domain Name (FQDN). Therefore, the Avaya SIP Enablement Services will *not* look at any provisioned *Host Maps*, making them unnecessary.
   b. The Avaya SIP Enablement Services will then do a *DNS SRV lookup* based on the *DNS server(s)* that has been provisioned in the Avaya SIP Enablement Services (see **Section 4.1**). The Avaya SIP Enablement Services will send the call to the destination specified in the SRV response.
   c. Alternatively the Avaya SIP Enablement Services can send the call to an *Outbound Proxy* if one has been provisioned on the Avaya SIP Enablement Services (see **Section 4.1**). If no DNS server or Outbound has been provisioned, the Avaya SIP Enablement Services will deny the call.

Note - Avaya's Best Practice recommendation is to use Port 5060 for all configurations, and by default the Avaya SIP Enablement Services will use port 5060 for outbound calls. If a different destination port is required, then Domain Based Routing *should not* be used and Avaya SIP Enablement Services Host Maps *must be used* to specify this non-standard port. See **Appendix B** for more information on provisioning Avaya Communication Manager and the Avaya SIP Enablement Services to utilize Host Maps for outbound calls.

1.6. Call Flows

To better understand how calls are routed between the PSTN and the enterprise sites using SIP trunks, two call flows are described in this section.

The first call scenario illustrated in **Figure 5** is an inbound call from PSTN to the enterprise site terminating on a telephone supported by Avaya Communication Manager.

1. A user on the PSTN dials a Verizon Business Service provided DID number that is assigned to an Avaya Communication Manager telephone at the enterprise site. The PSTN routes the call to the Verizon Business Service network (as the local service provider), which in turn routes the DID number to the assigned customer.

2. Based on the DID number, the Verizon Business Service offers the call to Avaya SIP Enablement Services using SIP signaling messages sent over the managed access facility. Note that the assignment of the DID number and the address of the Avaya SIP Enablement Services server was previously established during the ordering and provisioning of the service.

3. Avaya SIP Enablement Services routes the call to the Avaya Communication Manager over a SIP trunk.

4. Avaya Communication Manager terminates the call to the Avaya telephone. The same process occurs for calls to any Avaya analog, digital and H.323 IP telephone.

   - or –

4a. Inbound calls destined for a SIP extension at the enterprise are routed to Avaya Communication Manager. Avaya Communication Manager, acting as a SIP back-to-back user agent, then transmits the appropriate SIP signaling via Avaya SIP Enablement Services to the SIP telephone (as shown by the 4a arrow.)

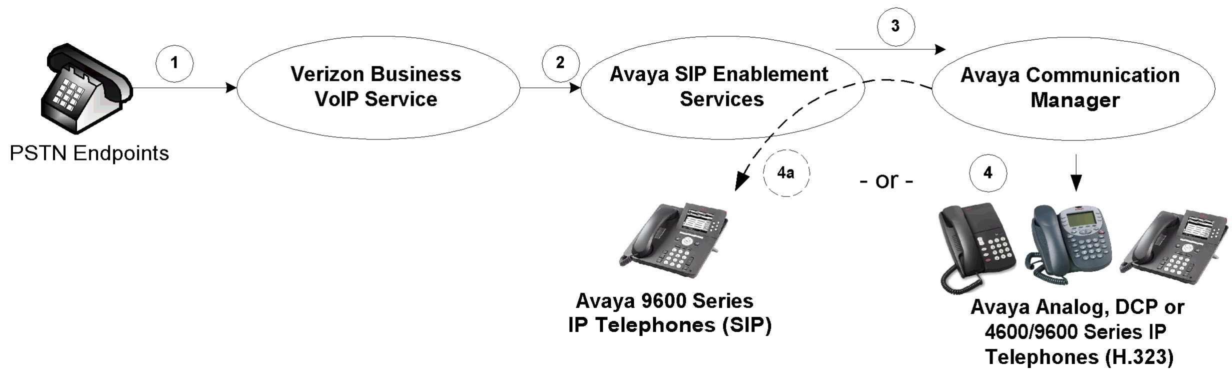

Figure 5: Incoming PSTN Calls to Avaya Communication Manager

The second call scenario illustrated in **Figure 6** is an outgoing call from an Avaya telephone at the enterprise site to the PSTN via the SIP trunk to the Verizon Business Service.

1. An Avaya H.323, analog or digital telephone served by Avaya Communication Manager originates a call to a user on the PSTN.

   - or-

1a. An Avaya SIP telephone originates a call that is routed via Avaya SIP Enablement Services (as shown by the 1a arrow) to Avaya Communication Manager.

2. Avaya Communication Manager handles the call request where origination treatment such as Class of Restrictions (COR) and Automatic Route Selection (ARS) is performed. Avaya Communication Manager selects the SIP trunk and sends the SIP signaling messages to Avaya SIP Enablement Services.

3. Avaya SIP Enablement Services routes the call to the Verizon Business Service.

4. The Verizon Business Service completes the call to the PSTN.

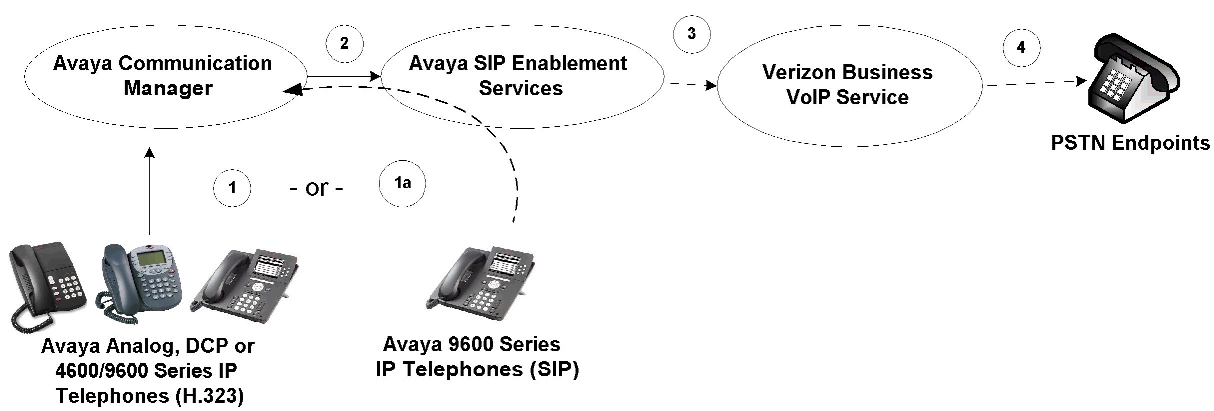

Figure 6: Outgoing Calls from Avaya Communication Manager to the PSTN

1.7. Low Bit Rate Voice and G.711 Fax

The Verizon Business Service supports both G.729A (low bit rate) and G.711MU (or G.711A) digital encoding methods used by PSTN services today. Lower bandwidth usage often makes G.729A a first choice for voice services over SIP trunks. In these Application Notes, the configuration is defined to support the use of G.729A as the first choice codec for voice calls.

The G.729 codec is often not suitable for fax; G.711 coding is required instead. Verizon Business does not support T.38 for fax or techniques to renegotiate the codec while a call is in progress. This means that calls may experience degraded performance unless special handing is performed.

Several strategies exist to provide special handling for G.711 fax calling:

- Connect G.711 fax directly to POTS lines from the Local Exchange Carrier.
- Use DID number assignment and routing techniques to route fax calls via TDM trunks (rather than using VoIP trunks from Verizon Business).
- Assign fax lines to dedicated Avaya Media Gateways separate from those used by voice lines. IP network region techniques can then be used to ensure only G.711 codecs are used for fax calls via the Verizon Business Service.
- Use DID number assignment and outbound routing techniques to force fax lines to use G.711 coding via the Verizon Business Service.

The use of a dedicated Avaya Media Gateway for fax lines is the most straightforward strategy for fax use via the Verizon Business Service. However, that strategy may result in high costs for a small number of fax lines and is not possible with smaller Avaya media gateway configurations. For this reason, the final strategy of using routing techniques will be discussed in the next section and demonstrated within these Application Notes.

1.8. Dial Plan and Routing Administration for Fax with the Verizon Business Service

The current Verizon Business Service recommends that fax calls be set up using a G.711 codec. This section outlines the overall administration strategy that will be implemented in the following sections of this document.

This strategy requires the following:

- Analog line ports must be specifically identified for fax.
- Fax must be assigned designated Verizon Business DID numbers. These numbers should be segregated into a contiguous block of numbers for ease of administration in the Avaya SIP Enablement Services.
- Two separate Avaya Communication Manager SIP trunk groups must be defined. One trunk group must use the G.729 codec as the first priority for normal voice calling. Another trunk group must be created for fax calls that only use the G.711 codec.
- The fax stations must be assigned a Class of Restriction different from voice stations. The Facility Restriction Level (FRL) within this COR must have less calling privileges than voice stations expected to use the G.729 codec.

- Outbound calls will use the Automatic Route Selection route patterns with FRL screening to prevent fax calls from using the G.729 trunk group.
- Incoming calls will use the Avaya SIP Enablement Services media server address maps to route fax designated DID numbers to the trunk group supporting the G.711 codec.

2. Equipment and Software Validated

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

Equipment	Firmware	Software
Avaya S8500 Servers	-	-
Avaya S8300 Server	-	-
Avaya G650 Media Gateway IPSI – TN2312BP C-LAN – TN799DP MedPro – TN2302AP	 HW12 FW43 HW1 FW26 HW20 FW118	- - -
Avaya G350 Media Gateway MM711AP Analog Module	28.18.0 HW27 FW 73	-
Avaya Communication Manager		R015x.01.1.415.1 (5.1.1) plus Service Pack 1 - 01.1.415.1-16402
Avaya SIP Enablement Services (SES)	-	ses-5.1.1-01.1.415.1
Avaya 4621SW IP Telephones	-	a10d01b2-8-3.bin (H.323)
Avaya 9630 IP Telephones	-	2.0.5.0 (SIP)
Verizon Business IP Trunk service offer	-	Integrated release 2008.4

Table 2: Equipment and Software Used in the Tested Configuration

3. Configure Avaya Communication Manager for SIP Trunking

This section describes the steps for configuring Avaya Communication Manager with the necessary signaling and media characteristics for the SIP trunk connection with the Verizon Business IP Trunk service offer.

Note - The initial installation, configuration, and provisioning of the Avaya servers for Avaya Communication Manager and Avaya SIP Enablement Services, Avaya Media Gateways and their associated boards, as well as Avaya telephones, are presumed to have been previously completed and are not discussed in these Application Notes.

As described in **Section 1**, two dialing environments were created for the reference architecture, Domestic and EMEA.

The EMEA site utilized Avaya Communication Manager on an S8300 server located in an Avaya G350 Media Gateway. This environment contained Avaya SIP, Avaya H.323, and analog fax endpoints.

The Domestic site utilized Avaya Communication Manager running on an Avaya S8500 server. Collocated with these servers is an Avaya G650 Media Gateway containing a C-LAN signaling processor card, a MedPro media processor card, and an IPSI controller card for communicating to the Avaya S8500 server. The Domestic site also contained Avaya SIP, Avaya H.323, and analog fax endpoints.

Note - As the Domestic/S8500 configuration is more complex, these Application Notes will focus on the Domestic provisioning. Unless otherwise noted, the configuration of the EMEA Avaya Communication Manager is similar. See **Section 3.2** for EMEA/S8300 specific provisioning of Avaya Communication Manager.

Note – The Avaya Communication Manager commands described in these Application Notes were administered using the System Access Terminal (SAT). SSH was used connect to SAT via the appropriate IP address, login and password.

Note – The Avaya Communication Manager and Avaya SIP Enablement Services provisioning described in these Application Notes are also applicable on non SIP Trunk Redundant (2-CPE) architectures.

3.1. Domestic (S8500 environment) Provisioning for Avaya Communication Manager

3.1.1 Verify System Capacity

The Avaya Communication Manager license file controls the customer capabilities. Contact an authorized Avaya representative for assistance if a required feature needs to be enabled.

1. On **Page 2** of the ***display system-parameters customer-options*** form, verify that the **Maximum Administered SIP Trunks** is sufficient for the combination of trunks to the Verizon Business IP Trunk service offer, trunks for SIP endpoints and any other SIP trunking applications. Be aware that for each call from a non-SIP endpoint to the Verizon Business IP Trunk service offer one SIP trunk is used for the duration of the call. However, due to Avaya SIP endpoint configuration, each Avaya SIP endpoint on a call with the Verizon Business IP Trunk service offer requires two SIP trunks for the duration of the call.

Note – If any changes are made to the **system-parameters customer-options** form, you must log out of SAT and log back in for the changes to take effect.

```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES
IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 800    4
           Maximum Concurrently Registered IP Stations: 2400   3
             Maximum Administered Remote Office Trunks: 800    0
Maximum Concurrently Registered Remote Office Stations: 2400   0
              Maximum Concurrently Registered IP eCons: 0      0
  Max Concur Registered Unauthenticated H.323 Stations: 0      0
                  Maximum Video Capable H.323 Stations: 0      0
                   Maximum Video Capable IP Softphones: 0      0
                       Maximum Administered SIP Trunks: 75     66
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
   Maximum Number of DS1 Boards with Echo Cancellation: 80     0
                             Maximum TN2501 VAL Boards: 10     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
```

Figure 7: System-Parameters Customer-Options Form – Page 2

2. On **Page 3** of the **System-Parameters Customer-Options** form, verify that the Automatic Route Selection (**ARS**) feature is enabled.

```
display system-parameters customer-options                    Page   3 of  10
                               OPTIONAL FEATURES
    Abbreviated Dialing Enhanced List? y           Audible Message Waiting? y
        Access Security Gateway (ASG)? n               Authorization Codes? y
        Analog Trunk Incoming Call ID? y                        CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                          CAS Main? n
Answer Supervision by Call Classifier? y                 Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? n
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                       DCS (Basic)? y
          ASAI Link Core Capabilities? n                 DCS Call Coverage? y
          ASAI Link Plus Capabilities? n                DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                           DS1 MSP? n
                                 ATMS? y             DS1 Echo Cancellation? y
                  Attendant Vectoring? y
```

Figure 8: System-Parameters Customer-Options Form – Page 3

3. On **Page 4** of the **System-Parameters Customer-Options** form, verify that the **IP Trunks**, **ISDN/SIP Network Call Redirection,** and **ISDN-PRI** features are enabled.

```
display system-parameters customer-options                    Page   4 of  10
                               OPTIONAL FEATURES
   Emergency Access to Attendant? y                              IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                        ISDN Feature Plus? y
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                 ISDN-PRI? y
              ESS Administration? n               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
     External Device Alarm Admin? y                 Media Encryption Over IP? y
  Five Port Networks Max Per MCC? n    Mode Code for Centralized Voice Mail? n
                Flexible Billing? y
   Forced Entry of Account Codes? y                 Multifrequency Signaling? y
      Global Call Classification? y         Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y               Multimedia IP SIP Trunking? n
                       IP Trunks? y
           IP Attendant Consoles? y
         (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 9: System-Parameters Customer-Options Form – Page 4

4. <u>*OPTIONAL*</u> - On **Page 6** of the **System-Parameters Customer-Options** form, the **ACD** and the indicated **Vectoring** features highlighted in **Figure 5** may be enabled.

```
display system-parameters customer-options                          Page   6 of  10
                         CALL CENTER OPTIONAL FEATURES
                          Call Center Release: 5.0
                                ACD? y                                Reason Codes? y
                        BCMS (Basic)? y                   Service Level Maximizer? y
          BCMS/VuStats Service Level? y                 Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? y     Service Observing (Remote/By FAC)? y
                   Business Advocate? n                  Service Observing (VDNs)? y
                     Call Work Codes? y                                Timed ACW? y
      DTMF Feedback Signals For VRU? y                        Vectoring (Basic)? y
                   Dynamic Advocate? n                    Vectoring (Prompting)? y
       Expert Agent Selection (EAS)? y                Vectoring (G3V4 Enhanced)? y
                            EAS-PHD? y                 Vectoring (3.0 Enhanced)? y
                   Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? y
               Least Occupied Agent? y     Vectoring (G3V4 Advanced Routing)? y
          Lookahead Interflow (LAI)? y                        Vectoring (CINFO)? y
Multiple Call Handling (On Request)? y      Vectoring (Best Service Routing)? y
    Multiple Call Handling (Forced)? y                     Vectoring (Holidays)? y
  PASTE (Display PBX Data on Phone)? y                    Vectoring (Variables)? y
         (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 10: System-Parameters Customer-Options Form – Page 6

3.1.2 Domestic Dial Plan

In the reference configuration the Domestic environment uses local extensions four digits in length beginning with 1, 2, 3, and 4. Trunk Access Codes (TAC) are 3 digits in length and begin with 6. The Feature Access Code (FAC) to access ARS is one digit in length (9).

The dial plan is modified with the ***change dialplan analysis*** command.

1. On **Page 1** of the form:
    - Local extensions:
        1. In the **Dialed String** field enter **1**
        2. In the **Total Length** field enter **4**
        3. In the **Call Type** field enter **ext**
        4. Repeat for **Dialed String 2, 3,** and **4**
    - TAC codes:
        1. In the **Dialed String** field enter **6**
        2. In the **Total Length** field enter **3**
        3. In the **Call Type** field enter **dac**
    - FAC code – ARS access:
        1. In the **Dialed String** field enter **9**
        2. In the **Total Length** field enter **1**
        3. In the **Call Type** field enter **fac**

```
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
     1            4    ext
     2            4    ext
     3            4    ext
     4            4    ext
     6            3    dac
     9            1    fac
```

Figure 11: Change Dialplan Analysis Form – Page 1

3.1.3 Domestic Node Names

In the **IP Node Names** form, verify (or assign) the node names to be used in this configuration using the ***change node-names ip*** command. As described in **Section 1**, Domestic uses an Avaya S8500 Server as the Avaya Communication Manager Platform. When the ***display node-names IP*** command is issued, the NIC IP address of the server is displayed as **procr**.

- **Procr** and **166.38.98.23** are the **Name** and **IP Address** of the Avaya Communication Manager S8500 server.
- **C-LAN-1a03** and **166.38.98.26** are used below as the **Name** and **IP Address** of the C-LAN signaling processor in the G650 Media Gateway.
- **Med-1a04** and **166.38.98.24** are used below as the **Name** and **IP Address** of the Media Processor in the G650 Media Gateway.
- **SBC** and **166.34.76.196** are used below as the **Name** and **IP Address** of the Verizon Session Border Controller (SBC). All traffic to and from the Verizon network passes through the SBC.
- **Primary-SES** and **166.38.98.22** are used below as the **Name** and **IP Address** of the Primary Avaya SIP Enablement Services server used in the SIP Trunk Redundant (2-CPE) configuration. **Note** – This IP address only applies to the Domestic environment. A different IP address is assigned for EMEA (see **Section 3.2**).
- **Secondary-SES** and **166.38.98.17** are used below as the **Name** and **IP Address** of the Secondary Avaya SIP Enablement Services server used in the SIP Trunk Redundant (2-CPE) configuration. **Note** – This IP address only applies to the Domestic environment. A different IP address is assigned for EMEA (see **Section 3.2**).

```
list node-names
                        NODE NAMES
Type     Name                 IP Address
IP       SBC                  166.34.76.196
IP       Primary-SES          166.38.98.22
IP       Secondary-SES        166.38.98.17
IP       clan-1a03            166.38.98.26
IP       default              0.0.0.0
IP       med-1a04             166.38.98.24
IP       procr                166.38.98.23
```

Figure 12: Domestic IP Node Names Form

3.1.4 Domestic IP-Network-Regions

Three network regions were defined in the reference configuration. Avaya equipment in the Domestic location is assigned to ip-network-region 1 as are the local SIP trunks. Voice trunks are assigned to ip-network-region 2. Fax trunks are assigned to ip-network-region 3.

Avaya Component	IP_Network-Region
C-LAN	1
MedPro	1
Local SIP Trunks 10 & 11	1
Voice SIP Trunks 12, 13, 14, & 15	2
Fax SIP Trunks 18, 19, 20, & 21	3

Table 3 – Domestic IP Network Regions

The SIP trunk ip-network-regions are defined in the SIP Signaling Group form Far-end region parameter (see **Section 3.1.6**).

Network region assignments for ip-interfaces may be verified with the ***list ip-interface all*** command.

```
list ip-interface all
                                IP INTERFACES
                                                                     Net
ON Type   Slot  Code Sfx Node Name/      Subnet Mask     Gateway Address Rgn VLAN
                         IP-Address
-- ----   ----  ---- --- --------------- --------------- --------------- --- ----
y C-LAN   01A03 TN799  D C-LAN-1a03       255.255.255.0   166.38.98.1      1   n
                         166.38.98.26
y MEDPRO  01A04 TN2302   med-1a04         255.255.255.0   166.38.98.1      1   n
                         166.38.98.24
```

Figure 13: IP-Interface IP-Network-Region Assignments – Domestic

The network-region for an ip-interface may be modified with the ***change ip-interface x*** command where **x** is the board location (the C-LAN interface is shown in the example below).

```
change ip-interface 01a03
                                  IP INTERFACES
                   Type: C-LAN
                   Slot: 01A03
            Code/Suffix: TN799  D
              Node Name: C-LAN-1a03
             IP Address: 166.38 .98 .26
            Subnet Mask: 255.255.255.0                            Link: 1
        Gateway Address: 166.38 .98 .1
  Enable Ethernet Port? y                      Allow H.323 Endpoints? y
         Network Region: 1                       Allow H.248 Gateways? y
                   VLAN: n                        Gatekeeper Priority: 5
 Target socket load and Warning level: 400
       Receive Buffer TCP Window Size: 8320
                                ETHERNET OPTIONS
                  Auto? n
                 Speed: 100Mbps
                Duplex: Full
```

Figure 14: IP-Interface IP-Network-Region Assignment.

The **IP-Network-Region** form specifies the parameters used by the Avaya Communication Manager components and how components defined to different regions interact with each other. The following ip-network-region assignments were used in the reference configuration. Other combinations are possible. In addition, specific codecs are used to communicate between these regions. See **Section 3.1.5** for the Codec form configurations.

Inter Region Communication	IP-Codec used
Region 1 to Region 1	Codec 1
Region 1 to Region 2	Codec 1
Region 1 to Region 3	Codec 2
Region 2 to Region 3	Codec 2

Table 4: Inter Region Codec Assignments

Note – Avaya IP telephones inherit the ip-network-region of the C-LAN (or procr for an Avaya S8300 based system) they register to. So if an IP phone registers to a C-LAN, that phone will become part of region 1. If an IP phone needs to be defined to a different region regardless of registration, this may be performed with the ***ip-network-map*** command.[1]

3.1.4.1 IP-Network-Region 1

As described in **Section 3.1.4**, ip-network-region 1 is defined for local Avaya IP interfaces as well as for local SIP trunks. The network regions are modified with the ***change ip-network-region x*** command, where x is the network region number (**Figure 15**).

1. On **Page 1** of the **IP Network Region** form:
   - Configure the **Authoritative Domain** field to match the **SIP Domain** name configured on the Avaya SIP Enablement Services **System Properties** field (see **Section 4.1**). In this configuration, the Domestic domain name is **avaya8500.icpiptrunksit2.gsiv.com** (provided by Verizon).

Note – The value placed in the **Authoritative Domain** field is used to create the SIP ***From*** header for outbound calls from stations in this region. In addition, inbound call Request URIs to stations in this region must match the Authoritative Domain.

- By default, **IP-IP Direct Audio** (media shuffling) is set to **yes** to allow audio traffic to be sent directly between SIP endpoints to reduce the use of media resources.
- Set the **Codec Set** to **1** for the corresponding calls within the IP Network Region.

```
change ip-network-region 1                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:        Authoritative Domain: avaya8500.icpiptrunksit2.gsiv.com
    Name: Local Region
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3029
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
```

Figure 15: IP Network Region 1 – Page 1

2. On **Page 3** of the **IP Network Region** form:
   - Define the **Codec Set** used for inter-region communications. **Codec Set *2*** is entered for communications with region **2**. **Codec Set *3*** is used for inter-region communication with region **3**.
   - Set the **direct WAN** field to **y**, indicating that devices in each region can directly communicate with each other.
   - Set the **WAN-BW-Limits** fields to **NoLimit** indicating that the Inter Network Region Connections are not constrained by bandwidth limits.
   - Set the **IGAR** (Inter-Gateway-Alternate-Routing) field to **n** because this field is not used in these Application Notes.

```
change ip-network-region 1                                    Page   3 of  19
                   Inter Network Region Connection Management
 src dst codec direct   WAN-BW-limits   Video      Intervening      Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC IGAR AGL
 1   1    1                                                                 all
 1   2    2     y    NoLimit                                             n
 1   3    3     y    NoLimit                                             n
```

Figure 16: IP Network Region 1 – Page 3

3.1.4.2 IP-Network-Region 2

As described in **Section 3.1.4**, ip-network-region 2 is defined for Voice SIP trunks. Provisioning is the same as for ip-network-region 1 except:

1. On **Page 1** of the **IP Network Region** form:
    - Set the **Codec Set** to **IP Codec Set 2** to be used for the corresponding calls within the IP Network Region.

```
change ip-network-region 2                                      Page   1 of  19
                               IP NETWORK REGION
Region: 2
Location:         Authoritative Domain: avaya8500.icpiptrunksit2.gsiv.com
    Name: Voice
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 17: IP Network Region 2 – Page 1

2. On **Page 3** of the **IP Network Region** form:
    - Define the **Codec Set** used for inter-region communications. **Codec Set 2** is entered for communications with region **1**. **Codec Set 3** is used for inter-region communication with region **3**.

```
change ip-network-region 2                                      Page   3 of  19
                   Inter Network Region Connection Management
 src dst codec direct   WAN-BW-limits   Video      Intervening      Dyn
 rgn rgn  set   WAN  Units     Total Norm  Prio Shr Regions          CAC IGAR AGL
 2   1    2     y    NoLimit                                              n
 2   2    2                                                                   all
 2   3    3     y    NoLimit                                              n
```

Figure 18: IP Network Region 2 – Page 3

3.1.4.3 IP-Network-Region 3

As described in **Section 3.1.4**, ip-network-region 3 is defined for fax SIP trunks. Provisioning is the same as for ip-network-region 1 except:

1. On **Page 1** of the **IP Network Region** form:

- Set the **Codec Set** to **IP Codec Set 2** to be used for the corresponding calls within the IP Network Region.

```
change ip-network-region 3                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location:          Authoritative Domain: avaya8500.icpiptrunksit2.gsiv.com
    Name: Fax
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? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 19: IP Network Region 3 – Page 1

2. On **Page 3** of the **IP Network Region** form:
   - Define the **Codec Set** used for inter-region communications. **Codec Set 3** is entered for communications with region **1**. **Codec Set *3*** is used for inter-region communication with region **2**.

```
display ip-network-region 3                                    Page   3 of  19
                   Inter Network Region Connection Management
 src dst codec direct   WAN-BW-limits   Video      Intervening      Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC IGAR
AGL
 3   1    3     y    NoLimit                                             n
 3   2    3     y    NoLimit                                             n
 3   3    3
all
```

Figure 20: IP Network Region 3 – Page 3

3.1.5 Domestic IP Codec Sets

Three codec sets are defined in the reference configuration. One for local intra customer location calls (ip-network-region 1), off network voice calls (ip-network-region 2), and off network G.711 fax calls (ip-network-region 3). **Table 5** shows the codecs defined to each of these codec sets.

IP-Codec Form	IP-Network-Region	Codecs Defined
Codec Form 1	1	G.711MU / G.729A
Codec Form 2	2	G.729AB / G.729A / G.711MU
Codec Form 3	3	G.729MU / G.711A

Table 5: Codec Form Codec Assignments

3.1.5.1 Intra Customer Location –IP-Codec-Set 1

G.711MU is typically used within the same location and is often specified first. G.729A is also specified as an option. Other codecs could be specified as well depending on local requirements. This codec set is associated with ip-network-region 1.

The **IP-Codec-Set** form is modified with the ***change ip-codec x*** command, where ***x*** is the codec form number.

1. On **Page 1** of the form:
   - Configure the **Audio Codec** field **1** to **G.711MU.**
   - Configure the **Audio Codec** field **1** to **G.729A.**

```
change ip-codec-set 1                                          Page   1 of   2
                          IP Codec Set
    Codec Set: 1
    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2: G.729A           n          2        20
```

Figure 21: IP Codec Set 1

2. On **Page 2** of the form:
   - Configure the **Fax** field to **off.**
   - Configure the **Fax Redundancy** field to **0.**
   - Let all other fields default.

```
change ip-codec-set 1                                        Page   2 of   2
                          IP Codec Set
                              Allow Direct-IP Multimedia? n
                    Mode                Redundancy
    Fax             off                   0
    Modem           off                   0
    TDD/TTY         off                   3
    Clear-channel   n                     0
```

Figure 22: IP Codec Set 1 – Page 2

3.1.5.2 Voice Calls – IP-Codec-Set 2

G.729AB was picked as the first option as it uses less bandwidth and is compatible with annexb (noise suppression). G.729A was used as the second choice, with G.711MU as the third choice. This codec set is associated with ip-network-region 2.

1. On **Page 1** of the form:
   - Configure the **Audio Codec** field **1** to **G.729AB.**
   - Configure the **Audio Codec** field **1** to **G.729A.**
   - Configure the **Audio Codec** field **2** to **G.711MU.**

```
display ip-codec-set 2                                            Page   1 of   2
                          IP Codec Set
    Codec Set: 2
    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729AB           n          2        20
 2: G.729A            n          2        20
 3: G.711MU           n          2        20
```

Figure 23: Voice Call IP Codec Set 2

2. On **Page 2** of the form set the values shown in **Figure 22** for codec set 1.

3.1.5.3 Fax Calls – IP-Codec-Set 3

G.711Mu was picked as the first option with G.711A as the third choice. This codec set is associated with ip-network-region 3.

1. On **Page 1** of the form:
   - Configure the **Audio Codec** field **1** to **G.711MU.**
   - Configure the **Audio Codec** field **2** to **G.711A.**

```
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.711A            n          2        20
```

Figure 24: Fax Call IP Codec Set 3

2. On **Page 2** of the form set the values shown in **Figure 22** for codec set 1.

3.1.6 Domestic SIP Trunk Groups

As described in **Section 1.2** SIP trunks are defined for local SIP communication as well as for, off network voice and G.711 fax calls. In addition, for each of these call types, SIP trunks are provisioned to the Primary and Secondary Avaya SIP Enablement Services servers. **Table 6** lists the SIP trunks used in the reference configuration. A SIP trunk is created in Avaya Communication Manager by provisioning a SIP Trunk Group as well as a SIP Signaling Group.

SIP Trunk Function	Avaya Communication Manager SIP Signaling Group/Trunk Group	Avaya Communication Manager SIP Signaling Group *Far-End Domain*	Avaya Communication Manager IP Network Region
Local - Primary	Trunk 10	Avaya SIP Enablement Services *SIP Domain* avaya8500.icpiptrunksit2.gsiv.com	1
Local - Secondary	Trunk 11	Avaya SIP Enablement Services *SIP Domain* avaya8500.icpiptrunksit2.gsiv.com	1
Inbound Voice - Primary	Trunk 12	<blank>	2
Inbound Voice - Secondary	Trunk 13	<blank>	2
Outbound Voice - Primary	Trunk 14	Verizon network Fully Qualified Domain Name (FQDN) schsj1n0005.icpiptrunksit2.gsiv.com	2
Outbound Voice - Secondary	Trunk 15	Verizon network Fully Qualified Domain Name (FQDN) schsj1n0005.icpiptrunksit2.gsiv.com	2
Inbound Fax - Primary	Trunk 20	<blank>	3
Inbound Fax - Secondary	Trunk 21	<blank>	3
Outbound Fax - Primary	Trunk 18	Verizon network Fully Qualified Domain Name (FQDN) schsj1n0005.icpiptrunksit2.gsiv.com	3
Outbound Fax - Secondary	Trunk 19	Verizon network Fully Qualified Domain Name (FQDN) schsj1n0005.icpiptrunksit2.gsiv.com	3

Table 6: Avaya SIP Trunk Configuration for SIP Trunk Redundant (2-CPE)

3.1.6.1 Configure Local SIP Trunks – 10 and 11

Create Primary and Secondary Signaling Groups and Trunk Groups 10 and 11 to support the local SIP phone registration and for voice calls between the local Avaya SIP stations and non-SIP phones (H.323/DCP/analog) defined on Avaya Communication Manager. Local SIP Trunk 10 is defined to the Primary Avaya SIP Enablement Services server and Local SIP Trunk 11 is defined to the Secondary Avaya SIP Enablement Services server.

1. Using the ***add signaling-group 10*** command, configure the **Primary** Local Signaling Group as follows:

   - Set the **Group Type** field to **sip**.
   - The **Transport Method** field will default to **tls** (Transport Layer Security). Note that this specifies the transport method used between Avaya Communication Manager and Avaya SIP Enablement Services, not the transport method used to the Verizon Business.
   - Specify the C-LAN used for SIP signaling (node name **clan-1a03**) and the Primary Avaya SIP Enablement Services server (node name **Primary-SES**) as the two ends of the signaling group in the **Near-end Node Name** and **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in **Section 3.1.3.**
   - Specify **5061** in the **Near-End** and **Far-end Listen Port** fields.

Note – See **<u>Section 3.1.6</u>** regarding Near-End port values for inbound voice and G.711 fax calls.

- Enter the value **1** into the **Far-end Network Region** field. This value is for the **IP Network Region** defined in **Section 3.1.5**.
- Enter the domestic Avaya SIP Enablement Services server SIP Domain of the Primary Avaya SIP Enablement Services server (see **<u>Section 4.1</u>**) in the **Far-end Domain** field. In the reference configuration, the Avaya SIP Enablement Services domestic domain name (for both the Primary and Secondary Avaya SIP Enablement Services servers) is **avaya8500.icpiptrunksit2.gsiv.com**.
- The **Direct IP-IP Audio Connections** field should be set to **y** to allow RTP voice paths to be established directly between IP telephones and the Verizon Business IP Trunk service offer. When compatible conditions exist, this allows the voice packets to follow a direct path between the telephones and the network edge, potentially reducing media processor resources and network usage.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Avaya Communication Manager to send DTMF tones using RFC 2833.
- The default values for the other fields may be used.

```
add signaling-group 10
                                SIGNALING GROUP
 Group Number: 10             Group Type: sip
                        Transport Method: tls
   Near-end Node Name: clan-1a03            Far-end Node Name: Primary-SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
      Far-end Domain: avaya8500.icpiptrunksit2.gsiv.com
                                            Bypass If IP Threshold Exceeded? n
        DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
        Enable Layer 3 Test? n                          IP Audio Hairpinning? n
 Session Establishment Timer(min): 3             Alternate Route Timer(sec): 6
```

Figure 25: Primary Signaling Group 10 – Primary Local SIP Trunk

2. Using the ***add signaling-group 11*** command, add the **Secondary** Local Signaling Group. Provisioning is the same as for Primary Local Signaling Group10 except:
   - Specify the Secondary Avaya SIP Enablement Services server (node name **Secondary-server**) as the **Far-end Node Name**. This field value is taken from the **IP Node Names** form shown in **Figure 12.**

```
add signaling-group 11
                                SIGNALING GROUP
 Group Number: 10            Group Type: sip
                        Transport Method: tls
   Near-end Node Name: clan-1a03              Far-end Node Name: Secondary-SES
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
       Far-end Domain: avaya8500.icpiptrunksit2.gsiv.com
                                              Bypass If IP Threshold Exceeded? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
         Enable Layer 3 Test? n                          IP Audio Hairpinning? n
 Session Establishment Timer(min): 3              Alternate Route Timer(sec): 6
```

Figure 26: Secondary Local Signaling Group 11

3. Using the ***add trunk-group 10*** command, add the **Primary** Local Trunk group as follows:
   a. On Page 1 of the Trunk Group form:
      - Set the **Group Type** field to **sip**.
      - Choose a descriptive **Group Name** such as **Primary Local.**
      - Specify an available trunk access code (**TAC**) such as **610**.
      - Set the **Service Type** field to **tie.**
      - Enter **10** as the **Signaling Group** number. This value was previously determined during the **Signaling Group** configuration specified in **Figure 25.**
      - Specify the **Number of Members** used by this SIP trunk group (e.g. **5**).

```
add trunk-group 10                                          Page   1 of  21
                                TRUNK GROUP
Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: Primary_Local          COR: 1        TN: 1        TAC: 610
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                       Signaling Group: 10
                                                     Number of Members: 5
```

Figure 27: Primary Local Trunk Group 10 – Page 1

   b. On Page 3 of the **Trunk Group** form:
      - Set the **Numbering Format** field to **public.** This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 10                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y
                     Numbering Format: public
                                                UUI Treatment: service-provider
                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
```

Figure 28: Primary Local Trunk Group 10 – Page 3

c. On Page 4 of the **Trunk Group** form:
   - Set the **Telephone Event Payload Type** to **101** to match the configuration on the Verizon Business IP Trunk service offer.

```
add trunk-group 10                                          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
             Telephone Event Payload Type: 101
```

Figure 29: Primary Local Trunk Group 10 – Page 4

4. Using the ***add trunk-group 11*** command, add the **Secondary** Local Trunk Group. Provisioning is the same as for Primary Local SIP Trunk 10 except on **Page 1** of the form:
   - Specify a new name (e.g. **Secondary_Local**).
   - Specify a new TAC code (e.g. **611**)
   - Specify **Signaling Group 11**.

```
add trunk-group 11                                          Page   1 of  21
                                TRUNK GROUP
Group Number: 11                   Group Type: sip          CDR Reports: y
  Group Name: Secondary_Local          COR: 1        TN: 1        TAC: 611
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                     Signaling Group: 11
                                                   Number of Members: 5
```

Figure 30: Secondary Local Trunk Group 11 – Page 1

3.1.6.2 Configure Inbound SIP Trunks

Create Primary and Secondary signaling groups and trunk groups to support inbound voice and G.711 fax calls with the Verizon Business IP Trunk service offer. Following the procedures described in **Section 3.1.6.1** create the Signaling and Trunk groups indicated below using the values specified in **Tables 6** through **9**. Note that the **Far-End Domain** field is left blank to accept incoming calls from any far end domain or IP address.

To discriminate between incoming voice and G.711 fax calls, the incoming voice calls will use port 5061 (default) and the G.711 fax call will use port 5062. These port values are set in the Avaya SIP Enablement Services (see **Section 4.1**).

All other values should match those shown in **Section 3.1.6.1**.

3.1.6.2.1 Inbound Voice Signaling Groups

Signaling Group Name	Group Number	Far-end Node Name	Near-end listen-port	Far-end Network Region	Far-end Domain
Primary_ Inbound_Voice	12	Primary-SES	5061	2	<blank>
Secondary_Inbound_Voice	13	Secondary-SES	5061	2	<blank>

Table 7: Inbound Voice Signaling Groups

3.1.6.2.2 Inbound Fax Signaling Groups

Signaling Group Name	Group Number	Far-end Node Name	Near-end listen-port	Far-end Network Region	Far-end Domain
Primary_ Inbound_Fax	20	Primary-SES	5062	3	<blank>
Secondary_Inbound_Fax	21	Secondary-SES	5062	3	<blank>

Table 8: Inbound Fax Signaling Groups

3.1.6.2.3 Inbound Voice Trunk Groups

Trunk Group Name	Group Number	TAC	Signaling Group
Primary_ Inbound_Voice	12	612	12
Secondary_Inbound_Voice	13	613	13

Table 9: Inbound Voice Signaling Groups

3.1.6.2.4 Inbound Fax Trunk Groups

Trunk Group Name	Group Number	TAC	Signaling Group
Primary_ Inbound_Fax	20	620	20
Secondary_Inbound_Fax	21	621	21

Table 10: Inbound Fax Signaling Groups

3.1.6.3 Configure Outbound Voice SIP Trunks

Create Primary and Secondary signaling groups and trunk groups to support outbound voice and G.711 fax calls with the Verizon Business IP Trunk service offer. Using the procedures described in **Section 3.1.6.1** create the following Signaling and Trunk groups using the values specified in **Tables 10** through **13** below. Note that the **Far-End Domain** field specifies the Fully Qualified Domain Name (FQDN) of the Verizon network. All other values should match those shown in **Section 3.1.6.1**.

3.1.6.3.1 Outbound Voice Signaling Groups

Signaling Group Name	Group Number	Far-end Node Name	Far-end Network Region	Far-end Domain
Primary_ Outbound_Voice	14	Primary-SES	2	schsj1n0005.icpiptrunksit2.gsiv.com
Secondary_Outbound_Voice	15	Secondary-SES	2	schsj1n0005.icpiptrunksit2.gsiv.com

Table 11: Outbound Voice Signaling Groups

3.1.6.3.2 Outbound Fax Signaling Groups

Signaling Group Name	Group Number	Far-end Node Name	Far-end Network Region	Far-end Domain
Primary_ Outbound_Fax	18	Primary-SES	3	schsj1n0005.icpiptrunksit2.gsiv.com
Secondary_Outbound_Fax	19	Secondary-SES	3	schsj1n0005.icpiptrunksit2.gsiv.com

Table 12: Outbound Fax Signaling Groups

3.1.6.3.3 Outbound Voice Trunk Groups

Trunk Group Name	Group Number	TAC	Signaling Group
Primary_ Outbound_Voice	14	614	14
Secondary_Outbound_Voice	15	615	15

Table 13: Outbound Voice Trunk Groups

3.1.6.3.4 Outbound Fax Trunk Groups

Trunk Group Name	Group Number	TAC	Signaling Group
Primary_ Outbound_Fax	18	618	18
Secondary_Outbound_Fax	19	619	19

Table 14: Outbound Fax Trunk Groups

3.1.7 Domestic Public Unknown Numbering

In the reference configuration, the extensions on Domestic Avaya Communication Manager use a 4 digit dialing plan using extensions 10xx, 20xx, and 40xx. The **Public-Unknown-Numbering** form allows Avaya Communication Manager to use these extensions as the calling party number for outbound calls. Otherwise *Anonymous* is displayed as the calling number. Each extension string is defined for the *outbound* trunk group that the extensions may use. These trunks may be defined individually or in contiguous ranges.

Use the ***change public-unknown-numbering x*** command, where *x* is the leading digit of the dial plan extensions (e.g. **1, 2, or 4**). For example to define the extensions beginning with 1, enter the following:

- Set the **Ext Len** field to **4**.
- Set the **Ext Code** field to **1**. This is the matching digit(s) of the extension. Setting this value to **10** would work as well.
- Set the **Trk Grp(s)** field to **10-11** (local), **12-15** (voice), and **18-21** (fax)**.**
- Set the **Total CPN Len** field to **4**. This is the total number of digits in the extension.
- Repeat these steps for extensions beginning with **2** and **4**.

All provisioned public-unknown-numbering entries can be displayed by entering the command ***display public-unknown-numbering 0*** as show in **Figure 31**.

```
display 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: 6
                                                    Maximum Entries: 9999
 4  1             10-15                     4
 4  1             18-21                     4
 4  2             10-15                     4
 4  2             18-21                     4
 4  4             10-15                     4
 4  4             18-21                     4
```

Figure 31: Public-unknown-numbering Form

3.1.8 Domestic Call Routing

3.1.8.1 Outbound Calls

As described in **Section 1.5**, Domain Based Routing is used for outbound calls. Avaya Communication Manager has outbound SIP trunks (14, 15, 18, and 19) that specify a foreign Far-End Domain on their Signaling Group forms. Since the Avaya SIP Enablement Services is not authoritative for this foreign domain, the Avaya SIP Enablement Services will do a DNS query on the foreign domain and route the call accordingly. Using this method, no Host Maps need to be defined in the Avaya SIP Enablement Services.

The following sections describe additional Avaya Communication Manager required for outbound dialing. These include Automatic Route Selection (ARS) and Look Ahead Routing (LAR). ARS determines how an outbound call is processed and LAR allows for SIP trunk fail-over from the Primary SIP trunks to the Secondary SIP trunks.

3.1.8.1.1 ARS

Automatic Route Selection feature is used to route calls via the SIP trunks to the Verizon Business IP Trunk service offer, which in turn completes the calls to the destination. In the reference configuration ARS is triggered by dialing a 9 (feature access code or FAC) and then dialing the called number. ARS matches on the called number and sends the call to a specified route pattern.

1. Verify that the appropriate extensions are defined in the **Public-Unknown-Numbering** form (see **Section 3.1.7**). This allows local extension to be used as the calling number.
2. Use the ***change dialplan analysis*** command to add **9** as a feature access code (**fac**).
   - Set **Dialed String** to **9**.
   - Set **Total Length** to **1**.
   - Set **Call Type** to **fac**.

```
change dialplan analysis                                        Page   1 of  12
                                DIAL PLAN ANALYSIS TABLE
                                     Location:  all          Percent Full:    1
      Dialed   Total  Call    Dialed   Total  Call      Dialed   Total  Call
      String   Length Type    String   Length Type      String   Length Type
        9        1    fac
```

Figure 32: Dialplan Analysis Form

3. Use the ***change feature-access-codes*** command to specify **9** as the access code for external dialing.
   - Set **Auto Route Selection (ARS) – Access Code 1:** to **9**.

```
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:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    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:
```

Figure 33: Feature-Access-Codes Form – Page 1

4. Use the ***change ars analysis*** command to configure the route pattern selection rule based upon the number dialed following the ARS access digit "9". In the reference configuration, domestic outbound calls are placed to the following numbers (*x* represents any digit):

   - 530352xxxx
   - 1530352xxxx
   - 1972728xxxx
   - 18xxxxxxxxx
   - 1xxx5551212
   - 011
   - 0

   For example, to specify the 530352xxxx called number, enter the command ***change ars analysis 530352*** and enter the following values:
   - Set the **Dialed String** field to **530352**.

- Set the **Total Min** field to **10**.
- Set the **Total Max** field to **10**.
- Set the **Route Pattern** field to **16**.
- Set the **Type** field to **fnpa**.

Using the same procedure, specify the other called number patterns in the ARS table. **Figure 34** shows the completed ARS table. Note that route pattern 16 was used for all outbound calls in the reference configuration.

```
display 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    1      16        op           n
    011                  11   20     16        intl         n
    1xxx5551212          11   11     16        fnpa         n
    1530352              11   11     16        fnpa         n
    18                   11   11     16        fnpa         n
    1972728              11   11     16        fnpa         n
    530352               10   10     16        fnpa         n
```

Figure 34: ARS Analysis Form

3.1.8.1.2 Route Patterns and LAR

Look Ahead Routing is a method for having a list of alternate trunks specified for the outbound calls. If the outbound call attempt fails for the first trunk on the list, the subsequent trunks listed will be tried in the order they appear on the list. LAR is specified on the route pattern form. Route pattern 16 was used in the reference configuration.

In addition to LAR, the reference configuration used Facility Restriction Level (FRL) to determine which outbound trunks could be used for voice and Verizon Business IP Trunk service offer G.711 fax calls. Each outbound trunk is assigned an FRL on the route pattern form. In addition, all voice extensions were provisioned with an FRL of 2 and all fax extensions were provisioned with an FRL of 1 (see **Section 1.3.3**). The higher the FRL, the higher the access level (e.g. an extension with an FRL of 1 cannot access a trunk specified with an FRL of 2. However an extension with an FRL of 2 can access trunks with FRLs of 2 and 1). In this manner G.711 fax calls are limited to trunks 18 and 19, while voice calls can use trunks 14, 15, 18, and 19.

Note - Route patterns may also be used to add or delete digits prior to sending them out the specified trunk(s). This feature was not used in the reference configuration.

1. Use the **change route-pattern** command to define the outbound SIP trunk groups included in the route pattern that ARS selects (see **Figure 35**).
   - **Primary voice trunk** - This trunk will always be selected first for outbound voice calls.
     - Set the first **Grp No** field to ***14***.

- Set the **FRL** field to ***2***.
- Set the **LAR** field to ***next***.
- **Secondary voice trunk** - This trunk will be selected for voice calls only if the voice call attempt fails over trunk 14.
  - Set the second **Grp No** field to ***15***.
  - Set the **FRL** field to ***2***.
  - Set the **LAR** field to ***next***.
- **Primary Fax trunk** - This trunk will always be selected first for outbound G.711 fax calls. It will also be used as the third choice for voice calls if attempts over trunks 14 and 15 fail.
  - Set the third **Grp No** field to ***18***.
  - Set the **FRL** field to ***1***.
  - Set the **LAR** field to ***next***.
- **Secondary Fax trunk** - This trunk will be selected for G.711 fax calls only if the Verizon Business IP Trunk service offer G.711 fax call attempt over trunk 18 fails. It will also be used as the fourth choice for voice calls if attempts over trunks 14, 15, and 18 fail.
  - Set the fourth **Grp No** field to ***19***.
  - Set the **FRL** field to ***1***.
  - Leave the **LAR** field at ***none*** (default).

```
change route-pattern 16                                            Page   1 of   3
                    Pattern Number: 16  Pattern Name: Outbound
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               QSIG
                             Dgts                                      Intw
 1: 14   2                                                              n   user
 2: 15   2                                                              n   user
 3: 18   1                                                              n   user
 4: 19   1                                                              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                                        next
 2: y y y y y n  n            rest                                        next
 3: y y y y y n  n            rest                                        next
 4: y y y y y n  n            rest                                        none
 5: y y y y y n  n            rest                                        none
 6: y y y y y n  n            rest                                        none
```

Figure 35: Route Pattern 16 – Outbound Calls with LAR

3.1.8.1.3 Locations Form

One of the parameters on the locations form specifies the default outbound route pattern that Avaya Communication Manager will use if no other route criteria are matched.

1. Use the **change locations** command to designate the default outbound SIP trunk route pattern (route pattern **16** below) in the **Proxy Sel Rte Pat** field.

```
change locations
                                   LOCATIONS
                ARS Prefix 1 Required For 10-Digit NANP Calls? y
 Loc  Name          Timezone Rule  NPA                                   Proxy Sel
 No                  Offset                                                Rte Pat
 1:  Main           + 00:00   0                                             16
```

Figure 36: Locations Form Administration

3.1.8.2 Incoming Calls

As described in **Section 3.1.6.2**, SIP trunks 12 and 13 are used for inbound voice calls and SIP trunks 20 and 21 are used for inbound G.711 fax calls.

In the reference configuration the Verizon network used the voice and fax extensions defined on Avaya Communication Manager as the called number for inbound calls. Therefore no incoming digit manipulation was required. However if an incoming called number needed to be changed to match a provisioned extension, this can be performed with the ***change inc-call-handling-trmt trunk-group x*** command, where **x** is the receiving trunk. For example if a called number 8001234567 need to be mapped to extension 1001, and the call came into trunk 10, then the following provisioning would be used:

Use the **change inc-call-handling-trmt trunk-group 10** command to administer this assignment.

- Leave ***public-ntwrk*** (default) in the **Service/Feature** field.
- Enter ***10*** into the **Called Len** field to match the length of the incoming digits.
- Enter ***8001234567*** into the **Called Number** field as the digit pattern to be matched.
- Enter ***10*** into the **Del** field as the number of digits that should be deleted from the end of the incoming digits.
- Enter ***1001*** into the **Insert** field.

```
change inc-call-handling-trmt trunk-group 10                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
 public-ntwrk    10 8001234567       10  1001
```

Figure 37: Example - Change Incoming Call Handling Treatment

3.1.9 Domestic Avaya Communication Manager Extensions

In the Domestic reference configuration 4 digit voice and fax extensions were provisioned. Voice extensions used the ranges 10xx, 20xx, and 40xx. Fax extensions were a subset of these ranges, 1002 and 1004, 2002 and 2004, as well as 4002 and 4004.

Figure 38 shows an example of a voice extension (Avaya IP phone) Note that the **COR** value is ***1*** (default) for the voice extension. Since the phone is an IP device, a virtual port **S00000** is assigned by the system. By default three call appearances are defined on page 4 of the form. See **Section 3.3** for Avaya SIP telephone provisioning.

```
display station 1001                                            Page   1 of   6
                                     STATION
Extension: 1001                          Lock Messages? n               BCC: 0
     Type: 4621                       Security Code:                   TN: 1
     Port: S00000                      Coverage Path 1:                 COR: 1
     Name: Avaya                       Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 1001
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n
                                              Customizable Labels? y


display station 1001                                            Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                              Headset? n
       Jack:                                              Speaker? n
      Cable:                                             Mounting: d
      Floor:                                          Cord Length: 0
   Building:                                            Set Color:
ABBREVIATED DIALING
     List1:                   List2:                    List3:
BUTTON ASSIGNMENTS
 1: call-appr                           5:
 2: call-appr                           6:
 3: call-appr                           7:
 4:                                     8:
    voice-mail Number:
```

Figure 38: Voice Extension – Avaya IP Phone

Figure 39 shows an example of a fax extension. Note that he **COR** value is **2** for the Verizon Business IP Trunk service offer fax extension. The Verizon Business IP Trunk service offer fax station is connected to an analog port on Avaya Communication Manager. In the reference configuration an analog board is located in slot 01a07 (the command ***list configuration all*** may be used to find board locations on Avaya Communication Manger). The fax station is connected to port 7 of the analog board. Therefore the port specified for this fax extension is ***01A0707***. By default one call appearance is define on page 4 of the form.

```
display station 1004                                            Page   1 of   4
                                     STATION
Extension: 1004                         Lock Messages? n              BCC: 0
     Type: 2500                         Security Code: *               TN: 1
     Port: 01A0707                    Coverage Path 1:                COR: 2
     Name: Fax-port                   Coverage Path 2:                COS: 1
                                      Hunt-to Station:              Tests? y
STATION OPTIONS
      XOIP Endpoint type: auto            Time of Day Lock Table:
              Loss Group: 1           Message Waiting Indicator: none
    Off Premises Station? n
           Survivable COR: internal
   Survivable Trunk Dest? y
                                             Remote Office Phone? n

display station 1004                                            Page   4 of   4
                                     STATION
 SITE DATA
      Room:                                            Headset? n
      Jack:                                            Speaker? n
     Cable:                                           Mounting: d
     Floor:                                       Cord Length: 0
  Building:                                          Set Color:
ABBREVIATED DIALING
     List1:                List2:                  List3:
HOT LINE DESTINATION
         Abbreviated Dialing List Number (From above 1, 2 or 3):
                                                      Dial Code:
     Line Appearance: call-appr
Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                        IP SoftPhone? y
                                             IP Video Softphone? n
                                             Customizable Labels? y
```

Figure 39: Fax Extension – Analog Port

As described in **Section 3.1.8**, outbound calls from voice and fax extensions were provisioned with different FRLs. These FRLs are associated with an extension via the **Class of Restriction** form. Voice extensions are assigned a COR of 1 (default) and fax extensions are defined with a COR of 2 (**Figure 40**).

For the voice COR 1 use the **change cor 1** command and enter the following:

- Enter ***Voice Calls*** in the **cor description** field.
- Enter ***2*** into the **FRL** field.

```
change cor 1                                                   Page   1 of  23
                              CLASS OF RESTRICTION
               COR Number: 1
          COR Description: Voice calls
                      FRL: 2                                 APLT? y
  Can Be Service Observed? n          Calling Party Restriction: none
Can Be A Service Observer? n           Called Party Restriction: none
        Time of Day Chart: 1      Forced Entry of Account Codes? n
         Priority Queuing? n               Direct Agent Calling? n
     Restriction Override: all       Facility Access Trunk Test? n
     Restricted Call List? n                Can Change Coverage? y
            Access to MCT? y           Fully Restricted Service? n
Group II Category For MFC: 7           Hear VDN of Origin Annc.? n
         Send ANI for MFE? n            Add/Remove Agent Skills? n
            MF ANI Prefix:             Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                        Can Be Picked Up By Directed Call Pickup? n
                                    Can Use Directed Call Pickup? y
                                    Group Controlled Restriction: inactive
```

Figure 40: COR 1 – Voice Extensions

For the fax extension COR 2 (**Figure 41**) use the **change cor 2** command and enter the following:

- Enter ***Fax Calls*** in the **cor description** field.
- Enter ***1*** into the **FRL** field.

```
change cor 2                                                   Page   1 of  23
                              CLASS OF RESTRICTION
               COR Number: 2
          COR Description: Fax calls
                      FRL: 1                                 APLT? y
  Can Be Service Observed? n          Calling Party Restriction: none
Can Be A Service Observer? n           Called Party Restriction: none
        Time of Day Chart: 1      Forced Entry of Account Codes? n
         Priority Queuing? n               Direct Agent Calling? n
     Restriction Override: none      Facility Access Trunk Test? n
     Restricted Call List? n                Can Change Coverage? n
            Access to MCT? y           Fully Restricted Service? n
Group II Category For MFC: 7           Hear VDN of Origin Annc.? n
         Send ANI for MFE? n            Add/Remove Agent Skills? n
            MF ANI Prefix:             Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                        Can Be Picked Up By Directed Call Pickup? n
                                    Can Use Directed Call Pickup? n
                                    Group Controlled Restriction: inactive
```

Figure 41: COR 2 – Fax Extensions

3.1.10 Save Domestic Avaya Communication Manager Provisioning

Enter the ***save translation*** command to make the changes permanent.

3.2. EMEA (S8300 environment) Specific Provisioning For Avaya Communication Manager

The following Avaya Communication Manager provisioning is required for the EMEA S8300 environment. All other provisioning is common with the Domestic environment.

3.2.1 Verify System Capacity

On Page 5 of the ***display system-parameters customer-options*** form (**Figure 42**), verify that the **Processor Ethernet parameter** is set to ***Y*.**

Note - The Avaya Communication Manager license file controls the customer capabilities. Contact an authorized Avaya representative for assistance if a required feature needs to be enabled.

```
display system-parameters customer-options                      Page   5 of  10
                                OPTIONAL FEATURES

               Multinational Locations? n            Station and Trunk MSP? n
 Multiple Level Precedence & Preemption? n      Station as Virtual Extension? n
                    Multiple Locations? n
                                             System Management Data Transfer? n
          Personal Station Access (PSA)? n              Tenant Partitioning? n
                       PNC Duplication? n      Terminal Trans. Init. (TTI)? n
                  Port Network Support? n              Time of Day Routing? n
                       Posted Messages? n      TN2501 VAL Maximum Capacity? y
                                                       Uniform Dialing Plan? y
                    Private Networking? y     Usage Allocation Enhancements? y
               Processor and System MSP? n
                    Processor Ethernet? y               Wideband Switching? n
                                                                  Wireless? n
                         Remote Office? n
          Restrict Call Forward Off Net? y
                 Secondary Data Module? y
        (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 42: System-Parameters Customer-Options Form – Page 5

3.2.2 EMEA Fully Qualified Domain Name (FQDN)S

The EMEA environment used different local and foreign Fully Qualified Domain Name (FQDN)s than the Domestic environment.

- EMEA local Fully Qualified Domain Name (FQDN) – **avaya8300.gsiv.com**
- EMEA Verizon Fully Qualified Domain Name (FQDN) – **schsj1n0005.emeaiptrunksit2.gsiv.com**

3.2.3 EMEA Dial Plan

In the reference configuration the EMEA environment uses local extensions four digits in length beginning with 1 and 4. Trunk Access Codes (TAC) are 3 digits in length and begin with 6. The Feature Access Code (FAC) to access ARS is one digit in length (9).

The dial plan is modified with the ***change dialplan analysis*** command (**Figure 43**).

1. On **Page 1** of the form:
   - Local extensions:
     1. In the **Dialed String** field enter **1**
     2. In the **Total Length** field enter **4**
     3. In the **Call Type** field enter **ext**
     4. Repeat for **Dialed String 4**
   - TAC codes:
     1. In the **Dialed String** field enter **6**
     2. In the **Total Length** field enter **3**
     3. In the **Call Type** field enter **dac**
   - FAC code – ARS access:
     1. In the **Dialed String** field enter **9**
     2. In the **Total Length** field enter **1**
     3. In the **Call Type** field enter **fac**

```
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
    1            4    ext
    4            4    ext
    6            3    dac
    9            1    fac
```

Figure 43: EMEA Dialplan Analysis Form – Page 1

3.2.4 EMEA Environment Node Names

The EMEA environment uses an S8300 platform for running Avaya Communication Manager. This platform does not use C-LAN or MedPro boards for call signaling or media processing. Instead the S8300 Procr is used for signaling and an Avaya G350 Media Gateway is used for media. In addition the EMEA environment uses different Primary and Secondary Avaya SIP Enablement Services servers.

- **Procr** and **166.38.98.19** are the **Name** and **IP Address** of the Avaya Communiction Manager S8300 server.
- **Primary-SES** and **166.38.98.73** are used below as the **Name** and **IP Address** of the Primary Avaya SIP Enablement Services server used in the EMEA SIP Trunk Redundant (2-CPE) configuration.
- **Secondary-SES** and **166.38.98.70** are used below as the **Name** and **IP Address** of the Secondary Avaya SIP Enablement Services server used in the EMEA SIP Trunk Redundant (2-CPE) configuration.
- All other values match the Domestic configuration.

```
list node-names
                        NODE NAMES
Type     Name              IP Address
IP       Primary-SES       166.38.98.73
IP       Secondary-SES     166.38.98.70
IP       default           0.0.0.0
IP       procr             166.38.98.19
```

Figure 44: EMEA IP Node Names Form

3.2.5 EMEA IP-Network-Regions

The EMEA environment uses the same network-region assignments as the Domestic environment. Refer to the commands shown in **Section 3.1.4** for provisioning.

Avaya Component	IP_Network-Region
Procr	1
G350 Media Gateway	1
Local SIP Trunks 10 & 11	1
Voice SIP Trunks 12, 13, 14, & 15	2
Fax SIP Trunks 18, 19, 20, & 21	3

Table 15 – EMEA IP Network Regions

The same codec definitions are used in the EMEA environment as in the Domestic environment.

Inter Region Communication	IP-Codec used
Region 1 to Region 1	Codec 1
Region 1 to Region 2	Codec 1
Region 1 to Region 3	Codec 2
Region 2 to Region 3	Codec 2

Table 16: Inter Region Codec Assignments

Note - All the EMEA network regions use **avaya8300.gsiv.com** as their authorative domain.

3.2.6 EMEA IP Codec Sets

The EMEA environment uses the same Codec set provisioning as the Domestic environment (see **Section 3.1.6**). **Table 17** shows the codecs defined to each of these EMEA codec sets. Local calls use codec 1, voice calls use codec 2, and G.711 fax calls use codec 3.

IP-Codec Form	IP-Network-Region	Codecs Defined
Codec Form 1	1	G.711MU / G.729A
Codec Form 2	2	G.729AB / G.729A / G.711MU
Codec Form 3	3	G.729MU / G.711A

Table 17: EMEA Codec Form Codec Assignments

3.2.7 EMEA SIP Trunk Groups

The EMEA environment uses the same SIP trunk configuration as the Domestic environment. The only difference is that EMAE uses Verizon Fully Qualified Domain Name (FQDN) **schsj1n0005.emeaiptrunksit2.gsiv.com** as the far-end domain in outbound Signaling Groups 14, 15, 18, and 19.

SIP Trunk Function	Avaya Communication Manager SIP Signaling Group/Trunk Group	Avaya Communication Manager SIP Signaling Group *Far-End Domain*	Avaya Communication Manager IP Network Region
Local - Primary	Trunk 10	Avaya SIP Enablement Services *SIP Domain* avaya8300.gsiv.com	1
Local - Secondary	Trunk 11	Avaya SIP Enablement Services *SIP Domain* avaya8300.gsiv.com	1
Inbound Voice - Primary	Trunk 12	<blank>	2
Inbound Voice - Secondary	Trunk 13	<blank>	2
Outbound Voice - Primary	Trunk 14	Verizon network Fully Qualified Domain Name (FQDN) schsj1n0005.emeaiptrunksit2.gsiv.com	2
Outbound Voice - Secondary	Trunk 15	Verizon network Fully Qualified Domain Name (FQDN) schsj1n0005.emeaiptrunksit2.gsiv.com	2
Inbound Fax - Primary	Trunk 20	<blank>	3
Inbound Fax - Secondary	Trunk 21	<blank>	3
Outbound Fax - Primary	Trunk 18	Verizon network Fully Qualified Domain Name (FQDN) schsj1n0005.emeaiptrunksit2.gsiv.com	3
Outbound Fax - Secondary	Trunk 19	Verizon network Fully Qualified Domain Name (FQDN) schsj1n0005.emeaiptrunksit2.gsiv.com	3

Table 18: EMEA SIP Trunk Configuration

3.2.8 EMEA Public Unknown Numbering

The EMEA environment uses the same public unknown numbering format as the Domestic environment. See **Section 3.1.7**.

3.2.9 EMEA Call Routing

3.2.9.1 Outbound Calls

The EMEA environment also uses Domain Based Routing is used for outbound calls.

3.2.9.1.1 ARS

Like the Domestic environment, the EMEA environment is provisioned to use ARS for outbound dialing. However the ARS analysis for EMEA is provisioned to support international dialing.

1. Use the ***change ars analysis*** command to configure the route pattern selection rule based upon the number dialed following the ARS access digit "9". In the reference configuration, EMEA outbound calls are placed to the following numbers (*x* represents any digit):

   - 0044xxxxxxxxxx
   - 0287198xxxxxxx
   - 02311234xxxxxxx
   - 0894xxxx
   - 11x

   Figure 45 shows the completed ARS table for EMEA. Note that route pattern **16** was used for all outbound calls in the reference configuration and the **Call Type** specified is ***intl***.

```
display ars analysis 00                                          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
    0044                  14   14     16        intl         n
    0287198               14   14     16        intl         n
    02311234              15   15     16        intl         n
    11                      3   3     16        intl         n
```

Figure 45: EMEA ARS Analysis Form

3.2.9.1.2 Route Patterns and LAR

EMEA uses the same route pattern and LAR provisioning as was used for Domestic (see **Section 3.1.8**).

3.2.9.1.3 Locations Form

EMEA uses the same locations provisioning as was used for Domestic (see **Section 3.1.8.1.3**).

3.2.9.2 Incoming Calls

EMEA uses the same incoming call provisioning as was used for Domestic (see **Section 3.1.8.2**).

3.2.10 EMEA Avaya Communication Manager Extensions

In the EMEA reference configuration 4 digit voice and fax extensions were provisioned. Voice extensions used the ranges 10xx and 40xx. Fax extensions were a subset of these ranges, 1039 and 4049. Stations were provisioned the same as in the Domestic environment (see **Section 3.1.9**).

As described in **Section 3.1.8.1** for Domestic, outbound calls from EMEA voice and fax extensions were provisioned with different FRL values. These FRL values are associated with an extension via the **Class of Restriction** form. Voice extensions are assigned a COR of 1 (default) and fax extensions are defined with a COR of 2.

3.2.11 Save EMEA Avaya Communication Manager Provisioning

Enter the ***save translation*** command to make the changes permanent.

3.3. Avaya SIP Endpoint Configuration

This section describes the administration of Avaya SIP telephones on Avaya Communication Manager (see <u>**Section 4.1.1.8**</u> for SIP endpoint provisioning on the Avaya SIP Enablement Services). SIP telephones are optional and not required for the Verizon Business IP Trunkk service offer. Domestic is used in the following examples. EMEA provisioning is similar.

3.3.1 Add a Station for the SIP Endpoint

The first step in adding a station for Avaya SIP telephones registered with Avaya SIP Enablement Services is to assign a station as shown in **Figure 46**. In the following example an Avaya 96xx SIP telephone is provisioned.

Using the **add station** command from the SAT:

- Set the station **Type** field to **9600SIP**.
- Enter a **Name** for the station (e.g. **9630 SIP**).
- The **Security Code** is left blank for SIP stations. (Note: SIP phone registration passwords are administered within the Avaya SIP Enablement Services **Add User** screen. See <u>**Section 4.1.1.8**</u>.
- Let all other fields default.

```
add station 1002                                              Page   1 of   6
                                     STATION
Extension: 1002                          Lock Messages? n               BCC: 0
     Type: 9600SIP                          Security Code:               TN: 1
     Port: S00003                      Coverage Path 1: 1               COR: 1
     Name: 9630 SIP                    Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 1002
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                  Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n
                                              Customizable Labels? y
```

Figure 46: Avaya 9600 SIP Station Administration – Page 1

By default 3 call appearances (**call-appr**) are provided on page 4 of the **Station** form for the phone type 9600SIP (**Figure 47**).

```
add station 1002                                              Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                             Headset? n
       Jack:                                             Speaker? n
      Cable:                                            Mounting: d
      Floor:                                         Cord Length: 0
   Building:                                           Set Color:
ABBREVIATED DIALING
     List1:                List2:                     List3:
BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: call-appr                          6:
 3: call-appr                          7:
 4:                                    8:
```

Figure 47: Avaya 9600 SIP Station Administration – Page 4

The parameters to administer call appearances and other settings are described in [1].

3.3.2 Configure Off-PBX Telephone Station Mapping

The second step of administering an Avaya SIP station is to configure the **Off-PBX-Telephone Station-Mapping** form. This form directs incoming calls to the extension number created earlier (**Section 3.3.1**) to be routed via a SIP trunk group to the intended SIP telephone.

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 48**, enter the following:

- In the **Station Extension** field, enter the extension number from the station defined in **Section 3.3.1** (e.g. **1002**).
- Set the **Application** field to **OPS**.
- Set the **Phone Number** field to the digits to be sent over the SIP trunk. In these Application Notes, this is the **Primary Handle** value that will be entered in the Avaya SIP Enablement Services and matches the station extension (e.g. **1002**).
- Set the **Trunk Selection** field to **10**, which is the number assigned to the SIP trunk group used for the SIP stations (see **Section 3.1.6**).
- Let all other fields default.

```
change off-pbx-telephone station-mapping 1002                    Page   1 of   2

                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
  1002              OPS       -          1002               10         1
```

Figure 48: Off-PBX Telephone Station Mapping for 9640 SIP Telephone – Page 1

On page 2 of the form (**Figure 49**):

1. Set the **Call Limit** field to the maximum number of calls that may be active simultaneously at the station. In this example, the **Call Limit** is set to **3**, which corresponds to the number of call appearances configured on the station form (See **Figure 47**).
2. Accept the default values for the other fields.

```
change off-pbx-telephone station-mapping 1002                    Page   2 of   2

                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged       Location
 Extension     Limit      Mode         Allowed     Calls
 1002            3        both           all        none
```

Figure 49: Off-PBX Telephone Station Mapping for 9640 SIP Telephone – Page 2

3.3.3 Save EMEA Avaya Communication Manager Changes

Enter the ***save translation*** command to make the changes permanent.

4. Configure Avaya SIP Enablement Services (SES)

This section covers the administration of Avaya SIP Enablement Services. **Section 4.1** describes the provisioning of the Domestic Primary and Secondary Avaya SIP Enablement Services servers. EMEA Avaya SIP Enablement Services provisioning is similar and is shown in **Section 4.2**.

Avaya SIP Enablement Services is configured via an Internet browser using SIP Server Management screens. It is assumed that Avaya SIP Enablement Services software together with the Avaya Communication Manager and the Avaya SIP Enablement Services license files have already been installed. For additional information on these installation tasks, refer to [4].

As described in **Section 1**, the Domestic and EMEA environments use a standalone Avaya SIP Enablement Services provisioned as an Edge/Home. A Primary and a Secondary Avaya SIP Enablement Services server are provisioned for the SIP Trunk Redundant (2-CPE) architecture.

4.1. Domestic Avaya SIP Enablement Services Servers

Using the Domestic SIP Trunk Redundant (2-CPE) configuration shown in **Figure 2** as a guide, provision the Domestic Primary and Secondary Avaya SIP Enablement Services servers.

4.1.1 Primary Avaya SIP Enablement Services Server

4.1.1.1 Log in to Avaya SIP Enablement Services

Access the **Server Management** pages by entering ***http://<ip-addr>/admin*** as the URL in an Internet browser, where **<ip-addr>** is the IP address of the Avaya SIP Enablement Services server defined during installation.

Log in with the appropriate credentials and the screen shown in **Figure 50** is displayed.

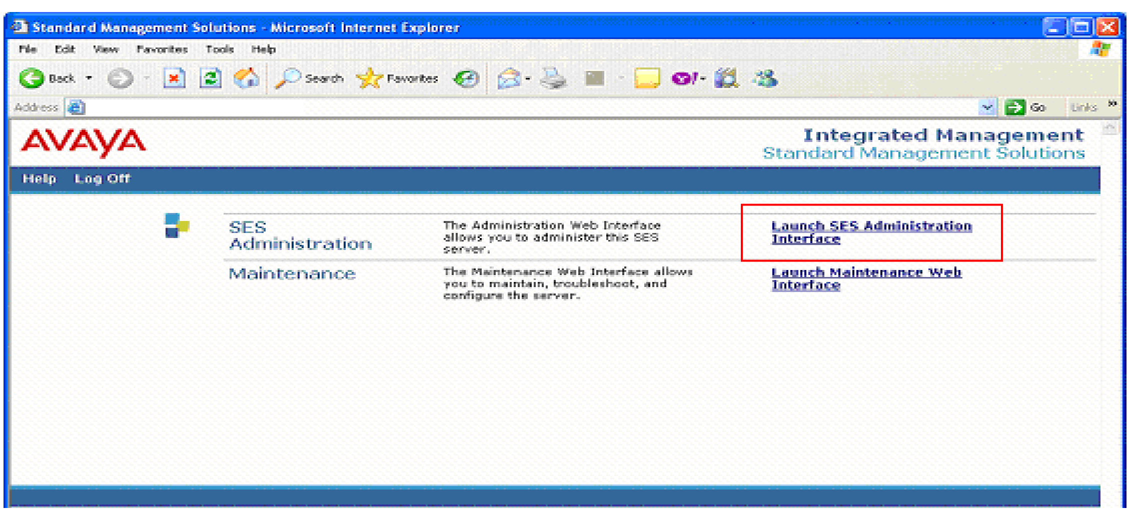

Figure 50: Avaya SIP Enablement Services Management Main Page

Click on **Launch SES Administration Interface** for the administration home page and the **Top SIP Server Management** page shown in **Figure 51** will open.

4.1.1.2 Verify System Properties

From the left pane of the **Top SIP Server Management** page, select the **Server Configuration** option and then select **System Properties**.

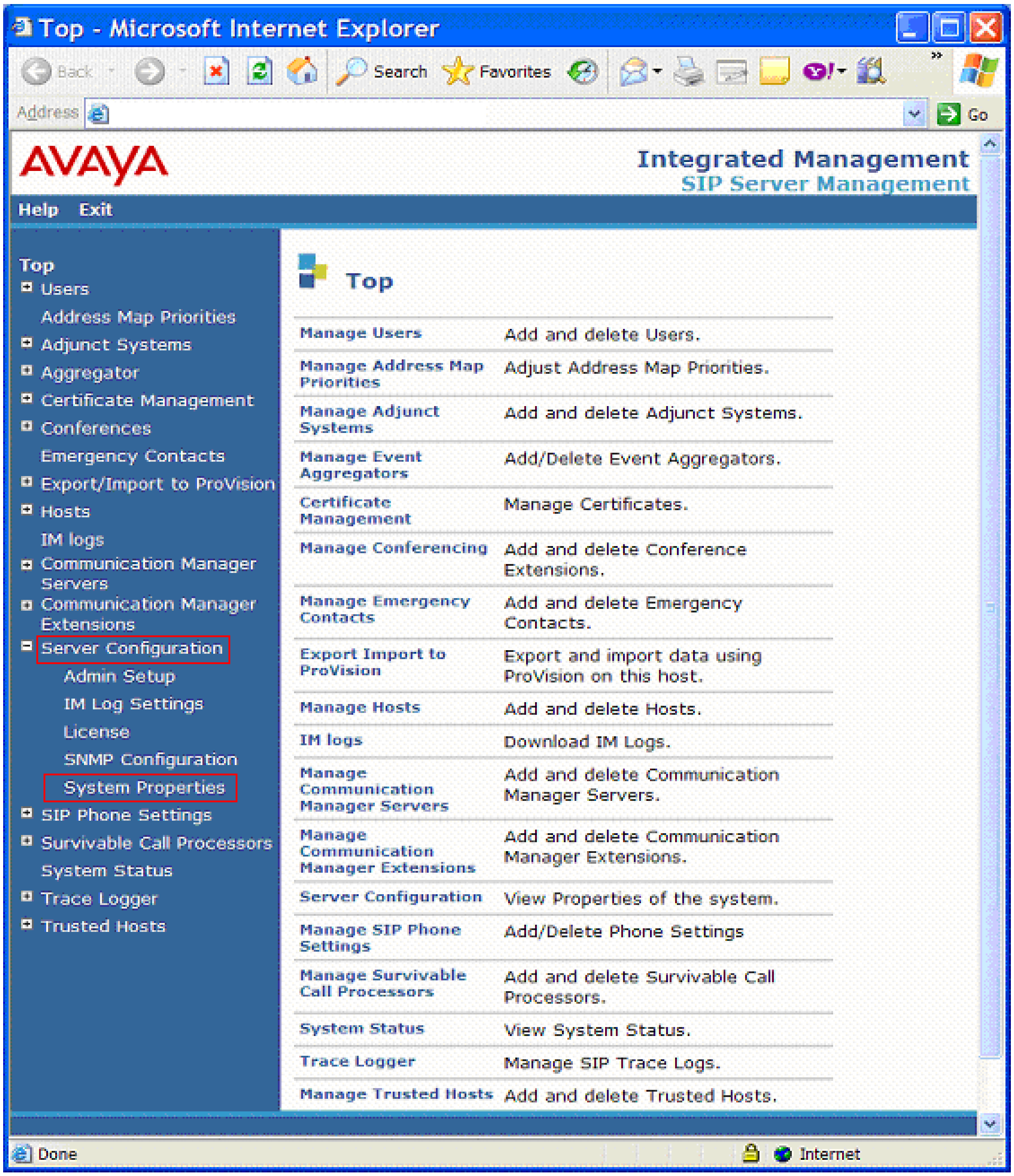

Figure 51: Avaya SIP Enablement Services Administration Top Page

The **View System Properties** page (**Figure 52**) displays the **Avaya SES Version** and the **Network Properties** entered during the installation process.

In the **View System Properties** page:

1. Verify the Avaya SIP Enablement Services Host information using the **Edit Host** page. In these Application Notes the Avaya SIP Enablement Services **Host Type** is **SES combined home/edge** (defined during installation)**.**
2. Enter the **SIP Domain**. The Domestic SIP Domain ***avaya8500.gsiv.com*** is used in the reference configuration (see **Section 3.1.6**).
3. Select **Update**.

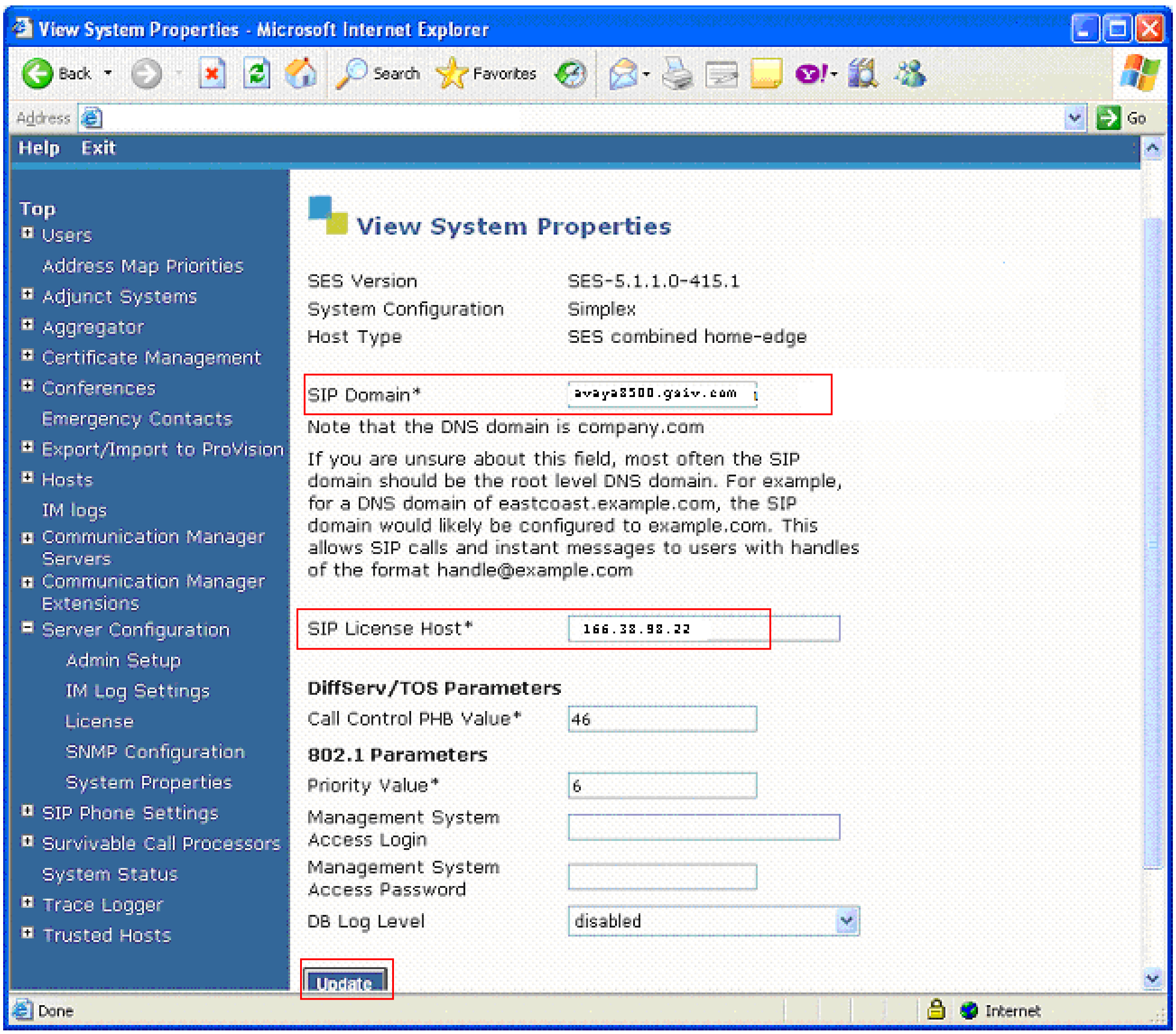

Figure 52: Domestic Primary Avaya SIP Enablement Services Server System Properties

4.1.1.3 Verify the Avaya SIP Enablement Services Host Information

Display the **Edit Host** page by following the **Hosts** link in the left navigation pane and then clicking on the **Edit** option under the **Commands** section of the **List Hosts** screen (**Figure 53**).

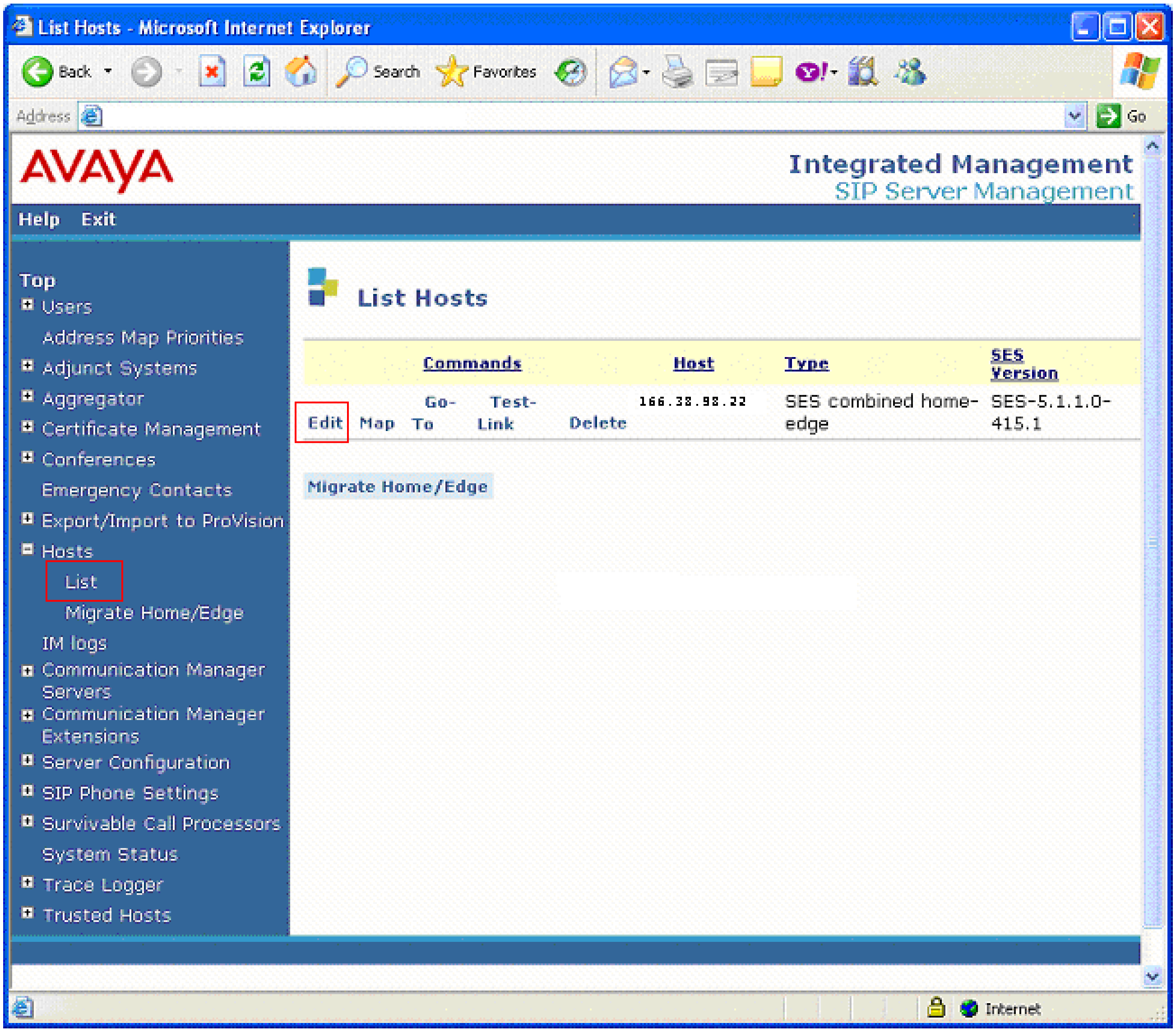

Figure 53: Domestic Primary Avaya SIP Enablement Services Server List Host

On the **Edit Host** screen (**Figure 54**):

- Verify that the IP address of this combined Avaya SIP Enablement Services Home/Edge server is in the **Host IP Address** field (e.g. **166.38.98.22**).
- Do not modify the **Profile Service Password** fields.
- Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols** (TCP is enabled by default).
- Verify that **TLS** is selected as the **Link Protocol**.
- Verify that the **Outbound Routing Allowed** options **Internal** and **External** are checked.

- Leave the **Outbound Proxy** and **Outbound Direct Domains** fields blank. This will cause the Avaya SIP Enablement Services to do a DNS query for the foreign Fully Qualified Domain Name (FQDN).
- Default values for the remaining fields may be used.
- Click the **Update** button only if changes are necessary. Otherwise exit the **Edit Host** page by selecting the **Top** link on the left navigation bar.

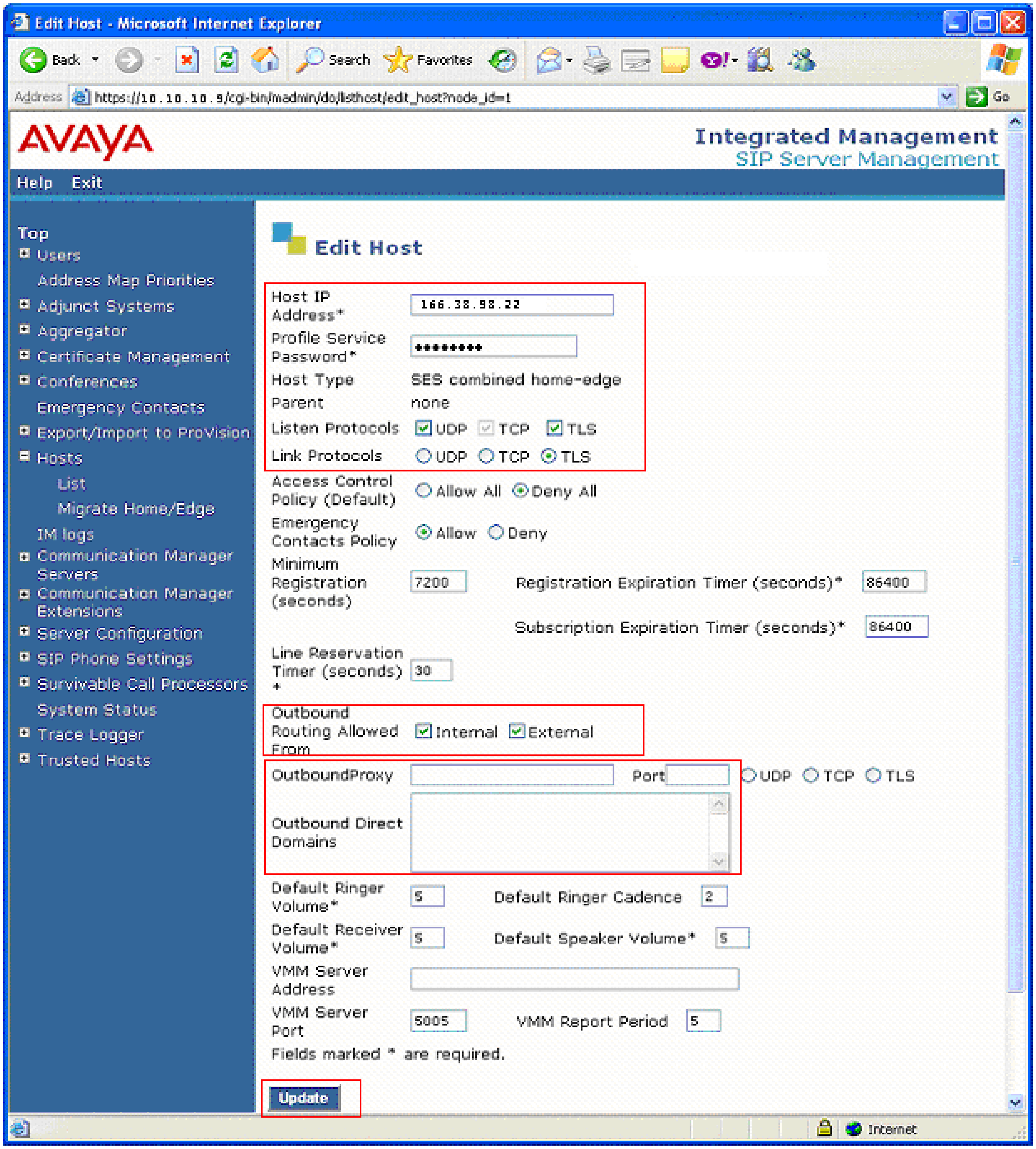

Figure 54: Domestic Primary Avaya SIP Enablement Services Server Edit Host

4.1.1.4 Configure Avaya Communication Manager Server Interfaces

Expand the **Communication Manager Servers** option within any Avaya SIP Enablement Services **SIP Server Management** page, and select **Add** to add a new **Communication Manager Server** or **List** to display or edit an existing **Communication Manager Server** configuration (**Figure 55**).

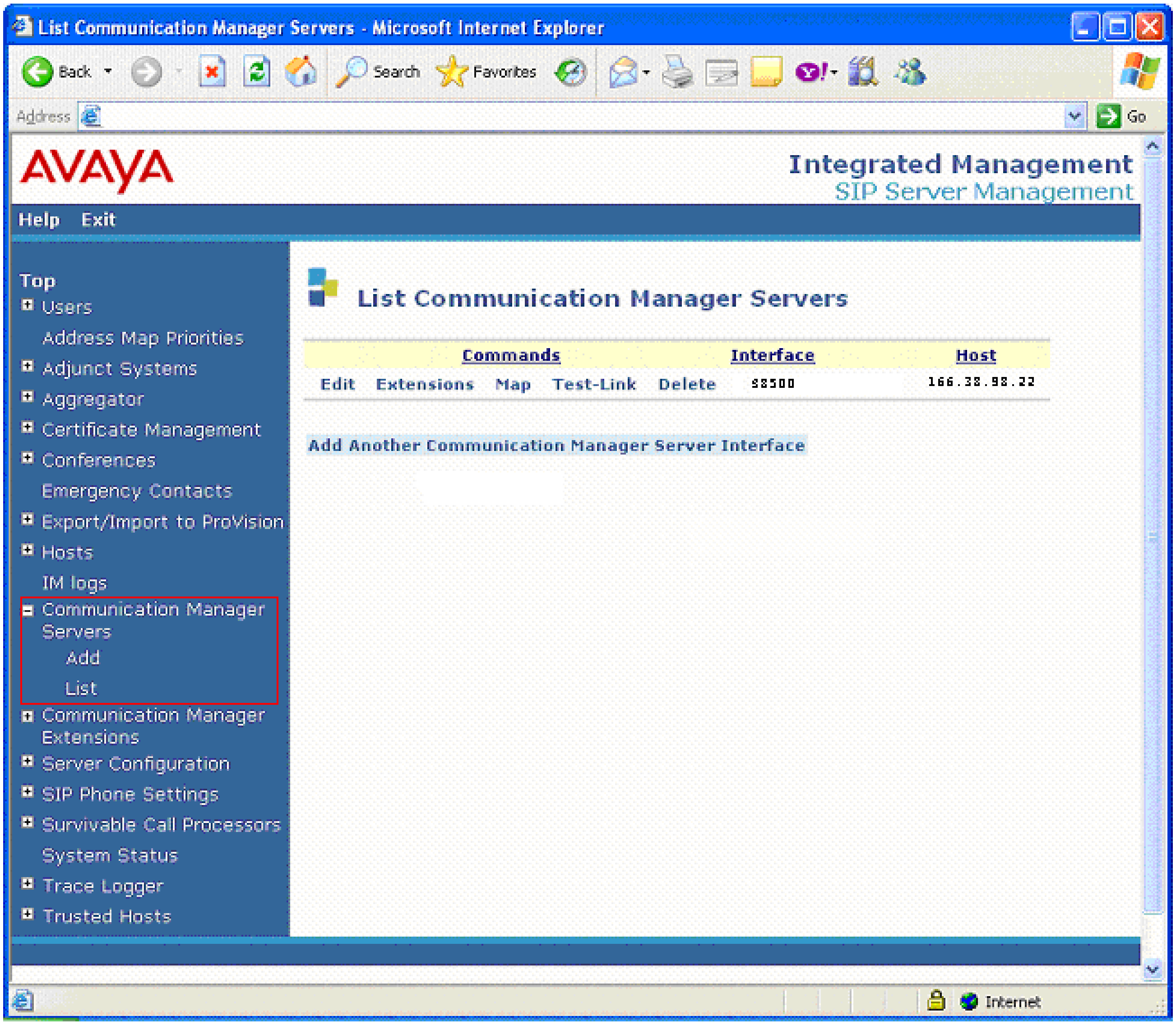

Figure 55: Domestic Primary Avaya SIP Enablement Services Server List Media Server

Figure 56 shows an existing configuration, which matches Signaling Group 10 configuration (SIP Telephones trunk) on Avaya Communication Manager (see **Section 3.1.6**).

On the **Edit Communication Manager Server Interface** screen:

- Specify a **Communication Manager Server Interface Name** (e.g. **S8500**).
- Verify that **TLS** is selected as the **Link Protocol**.

- Specify the IP address of the Avaya Communication Manager SIP trunk interface. This is the IP address of the **Near-end Node Name** specified in the Avaya Communication Manager Signaling Group form for the associated SIP Trunk (e.g. **166.38.98.26**).
- Specify the IP address of Avaya Communication Manager in the **Communication Manager Server** Admin Address field (e.g. **166.38.98.23**).
- Specify ***5022*** (SSH) for the **Communication Manager Server** Admin Port field.
- Specify the proper credentials for the **Communication Manager Server** Admin Login and Password fields.
- Verify that **SSH** is selected for **SMS Connection Type**.
- Click the **Update** button only if changes are necessary. Otherwise exit the **Edit Host** page by selecting the **Top** link on the left navigation bar.

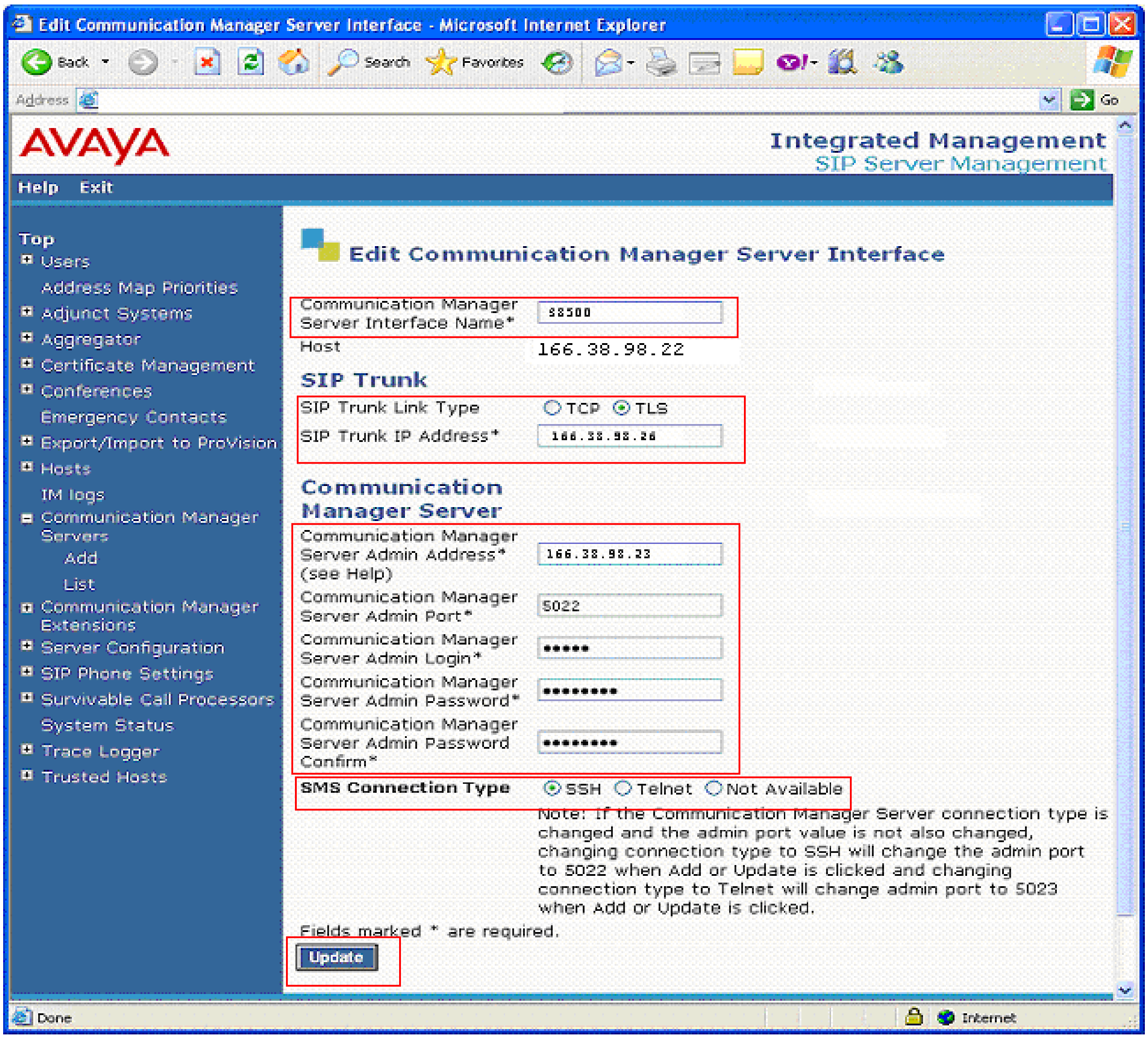

Figure 56: Domestic Primary Avaya SIP Enablement Services Server Edit Communication Manager Server Interface

4.1.1.5 Configure Trusted Hosts

Avaya SIP Enablement Services will deny inbound calls from unknown foreign nodes. As described in **Section 1.3**, the Verizon network used two SBCs in the SIP Trunk Redundant (2-CPE) architecture. Therefore both Verizon SBCs must be specified as Trusted Hosts in Avaya SIP Enablement Services.

1. From the Avaya SIP Enablement Services "**Top**" web page, select **Trusted Hosts**.
2. Expand the **Trusted Hosts** option and select **Add** to add a new **Trusted Host** or select **List** to display or edit an existing **Trusted Host** configuration (**Figure 57**).

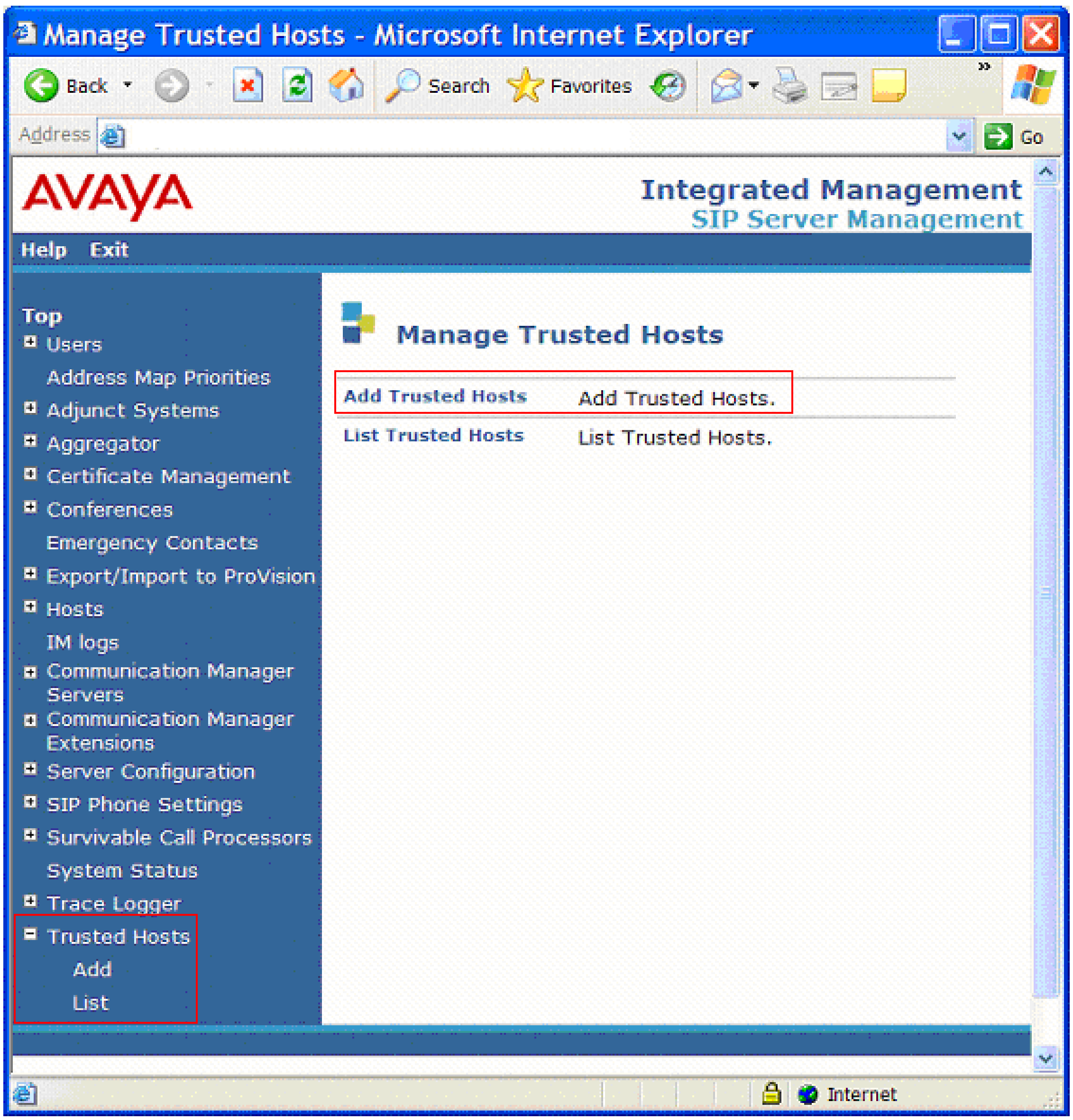

Figure 57: Domestic Primary Avaya SIP Enablement Services Server Trusted Hosts

3. On the **Edit Host** screen (**Figure 58**):
   - Specify the IP address of the first Verizon SBC (e.g. **166.34.93.140**).
   - Specify a description of the Trusted Host entry (e.g. **VZ SBC 1**).
   - Select **Update** to save changes.

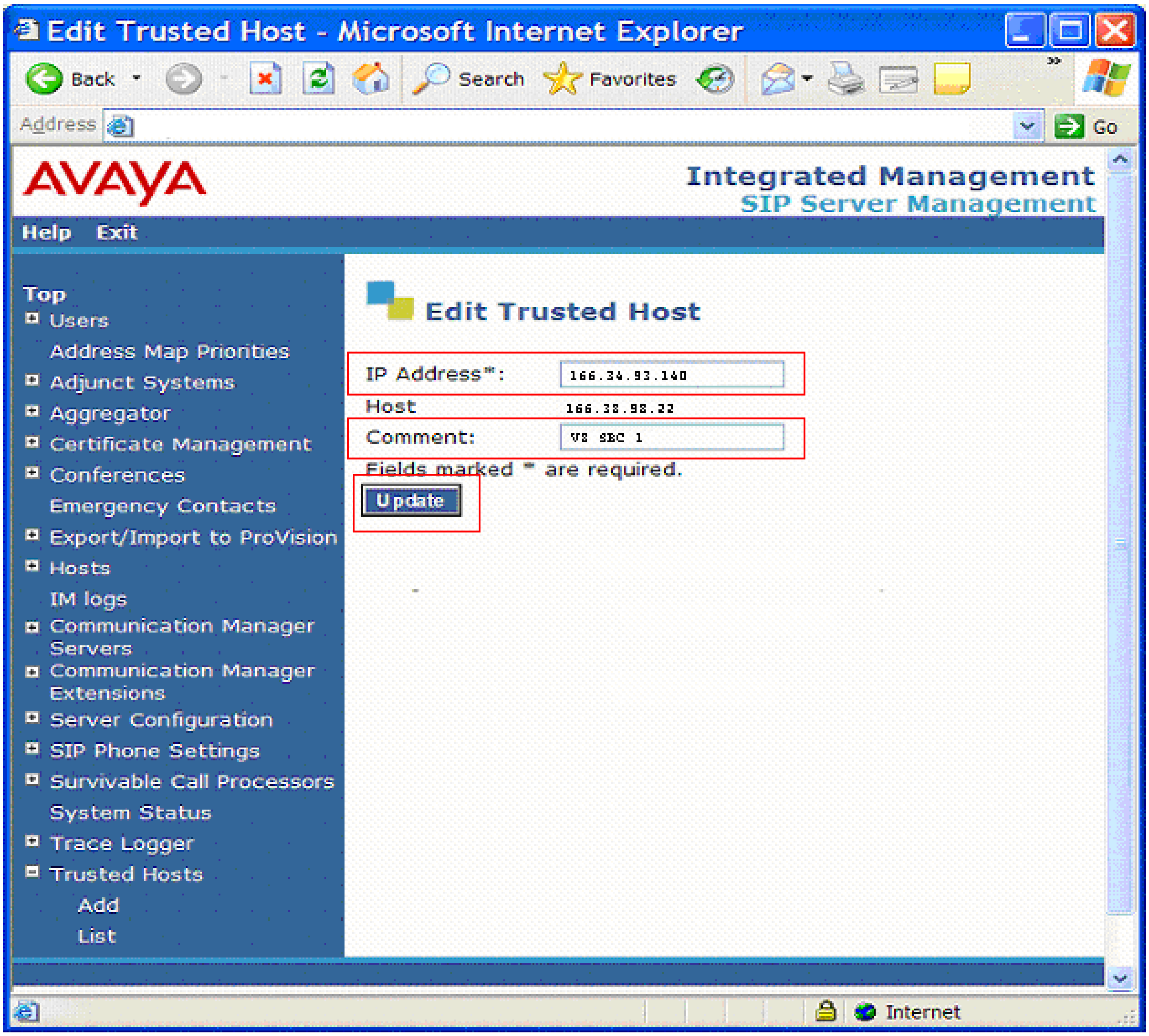

Figure 58: Domestic Primary Avaya SIP Enablement Services Server Edit Trusted Hosts

4. Repeat steps two and three to add the IP address of the second Verizon SBC (e.g. **166.34.93.141**, **VZ SBC 2**).

4.1.1.6 Configure Domestic Call Routing

The Avaya SIP Enablement Services functions as a SIP proxy server for the SIP trunking with the Verizon Business IP Trunk service offer. In this role, the Avaya SIP Enablement Services must direct outbound SIP calls originating from Avaya Communication Manager to the Verizon SBCs. In a similar manner for incoming calls, the Avaya SIP Enablement Services must route messages received from the Verizon SIP network to Avaya Communication Manager.

4.1.1.6.1 Domain Based Routing – Outbound Calls

Note – Domain Based Routing requires that a DNS server IP address(s) as well as a destination Fully Qualified Domain Name (FQDN) is supplied by the Service Provider. The DNS IP addresses may be specified during the Avaya SIP Enablement Services installation or by the procedures described in this section.

As described in **Section 1.5**, Domain Based Routing is used for outbound calls from Avaya Communication Manager to the Avaya SIP Enablement Services.

In the Domestic reference configuration Avaya Communication Manager specifies the Verizon Domestic Fully Qualified Domain Name (FQDN) ***schsj1n0004.icpiptrunksit2.gsiv.com*** in the Far-End Domain field of the outbound SIP Trunk Signaling Groups (14, 15, 18, and 19, see **Section 3.1.6**). The Avaya SIP Enablement Services receives an INVITE from Avaya Communication Manager with the destination URL of ***<callednumber>@ schsj1n0004.icpiptrunksit2.gsiv.com,*** and with no Outbound Proxy specified in the Avaya SIP Enablement Services (see **Section 4.1.1.3**), the Avaya SIP Enablement Services will issue a DNS SRV record query for the foreign domain. The Verizon network will then determine the proper destination and respond with the appropriate destination IP address. The Avaya SIP Enablement Services will then send the INVITE to the IP address provided in the SRV response.

This method of call routing provides the maximum flexibility since network changes within the Verizon service do not require any modification to Avaya provisioning.

Note – By default the Avaya SIP Enablement Services will use port 5060 for outbound calls. If a different destination port is required, then Domain Based Routing *should not* be used and Avaya SIP Enablement Services Host Maps *must be used* to specify this non-standard port. See **Appendix B** for more information on provisioning Avaya Communication Manager and the Avaya SIP Enablement Services to utilize Host Maps for outbound calls.

4.1.1.6.2 Inbound Call Routing

The SIP message routing for inbound calls uses **Communication Manager Server Address Maps** that examine some or all of the *called number* (matching on a defined pattern) and route to a specific predetermined destination (called a **Contact**).

The *called number* is contained within the *user* part of the Uniform Resource Identifier (URI) of an incoming SIP INVITE message. The URI usually takes the form of *sip:user@destination*, where *destination* can be the CPE (Avaya SIP Enablement Services) Fully Qualified Domain Name (FQDN) or IP address. The *user* part for SIP trunking in these Application Notes will only contain digits[1].

The Communication Manager Server Address Map patterns are specified using Linux regular expression syntax. Patterns are generally designed to match a collection of *called numbers* that require identical SIP message routing. However, each pattern must also be specific enough to direct each unique *called number* to the proper signaling Contact**.** The Communication Manager Server Address Map patterns must also be mutually exclusive (non-overlapping) from all other Communication Manager Server Address Map patterns used in the Avaya SIP Enablement Services to ensure proper operation.

Appendix A provides a detailed description of the Linux regular expression syntax used within the address map patterns.

Note – Communication Manager Server Address Map provisioning is very flexible, with possibilities for either broad or narrow routing constraints. The following provisioning should be viewed as reference examples.

As described in **Section 1.2**, inbound voice and G.711 fax calls are processed differently by Avaya Communication Manager (codecs, FRLs, etc). Therefore voice and G.711 fax calls must be identified uniquely. This is done by having the Avaya SIP Enablement Services assign port 5061 (default) for inbound voice calls and port 5062 for inbound G.711 fax calls. These port assignments are defined via the Communication Manager Server Address Maps.

4.1.1.6.3 Inbound Calls

To configure the Communication Manager Server Address Maps for inbound calls:

- Expand the **Communication Manager Servers** link in the left navigation menu of any **SIP Server Management** page.
- Select **List** to display the **List Communication Manager Servers** page as shown in **Figure 59.**
- Click on the **Map** link to display the **List Communication Manager Server Address Map** (shown in **Figure 60**).

[1] SIP does permit mnemonic addressing such as "sip:john.doe@customer.com". However, this convention is not used in these Application Notes for SIP Trunking. Further discussion of this topic is beyond the scope of this document.

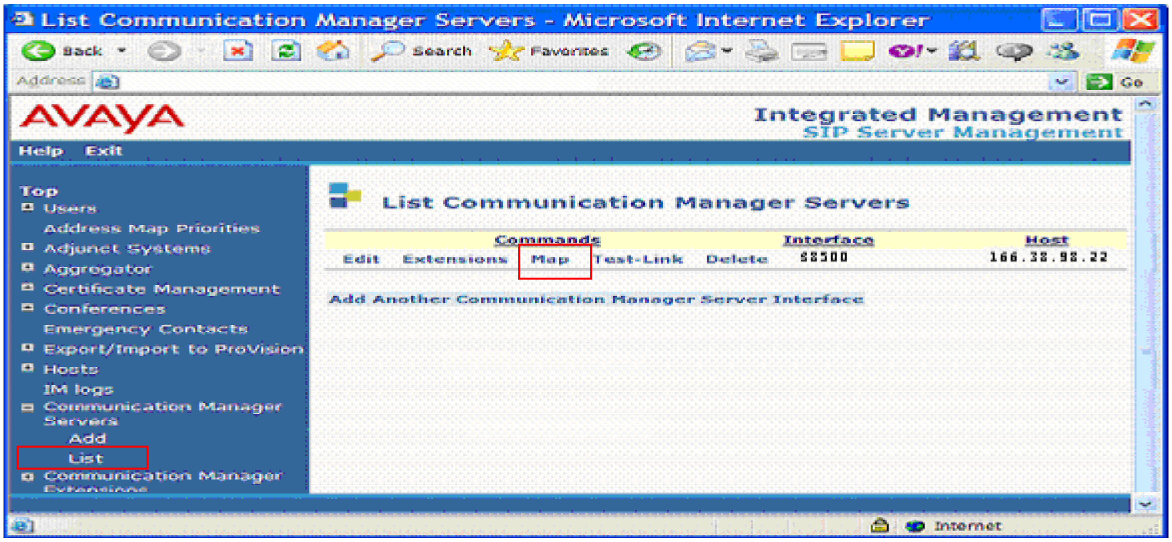

Figure 59: Domestic Primary Avaya SIP Enablement Services Server - List Communication Manager Servers

1. **Add Voice Call Maps.**
   Three voice maps need to be added to include all the voice extensions (10xx, 20xx, and 40xx) and to exclude the fax extensions (1002, 1004, 2002, 2004, 4002, and 4004)
   - Click on the ***Add Map In New Group*** button. The **Add Communication Manager Address Map** window will open (shown in **Figure 61**).

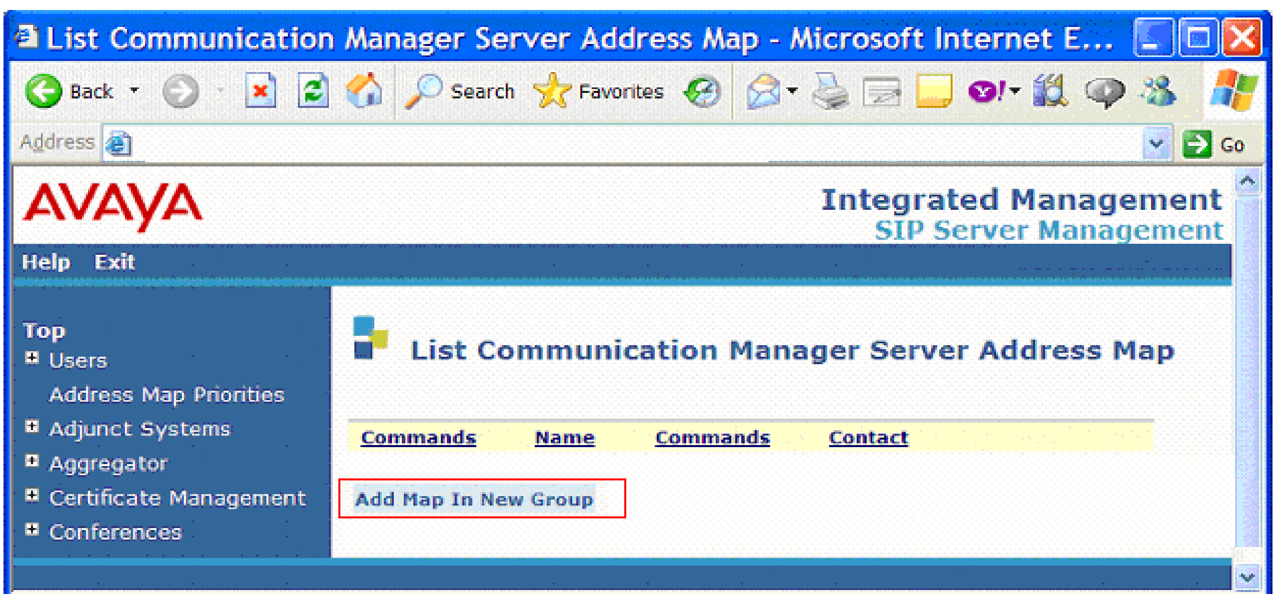

Figure 60: Domestic Primary Avaya SIP Enablement Services Server - List Communication Manager Server Address Maps

2. Enter a map for the 10xx voice extension range (**Figure 61**).
    - Enter a name for the map. (e.g. ***10xx***)
    - Enter the **Address Map Pattern** for incoming 10xx voice calls (excluding fax extensions 1002 and 1004) into the **Pattern** field. See **Appendix A** for more information on defining map patterns.
        1. ^***sip:10[0-9]{1}[0135-9]{1}*** This string means match any string where the first two digits are 10, the third digit is any number 0 through 9, and the fourth digit is 0, 1, 3, or 5 through 9. This will exclude 1002 and 1004.
    - Verify the ***Replace URI*** option is checked.
    - Click the **Add** button once the form is completed.

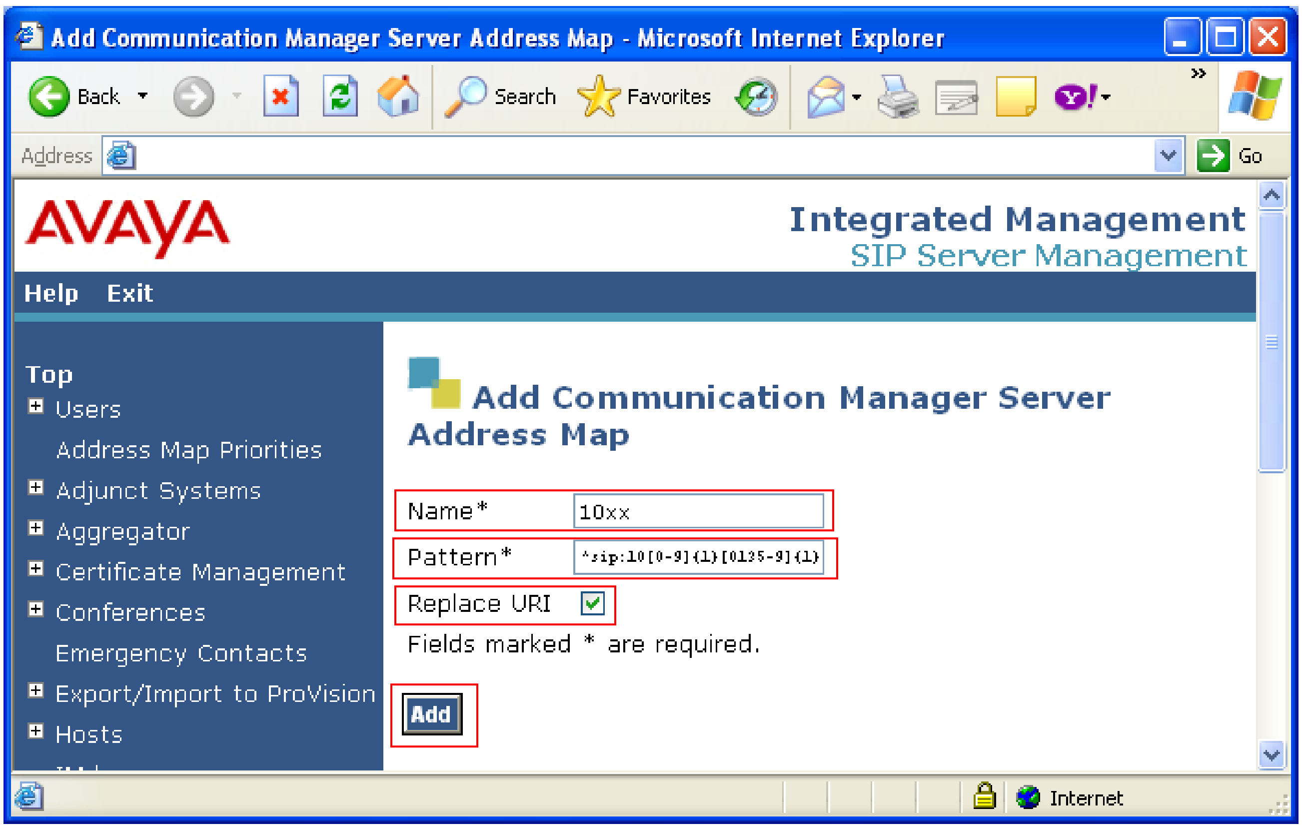

Figure 61: Domestic Primary Avaya SIP Enablement Services Server - Voice Address Map – 10xx

3. Enter a map for the 20xx voice extension range.
    a. Click on **Add Another Map**.
    b. Enter a name for the map. (e.g. ***20xx***)
    c. Enter the **Address Map Pattern** for incoming 20xx voice calls (excluding fax extensions 2002 and 2004) into the **Pattern** field.
        i. ^***sip:20[0-9]{1}[0135-9]{1}***
    d. Verify the ***Replace URI*** option is checked.
    e. Click the **Add** button once the form is completed.

4. Enter a map for the 40xx voice extension range.
    a. Click on **Add Another Map**.
    b. Enter a name for the map. (e.g. ***40xx***)
    c. Enter the **Address Map Pattern** for incoming 40xx voice calls (excluding fax extensions 4002 and 4004) into the **Pattern** field.
        i. ***^sip:40[0-9]{1}[0135-9]{1}***
    d. Verify the ***Replace URI*** option is checked.
    e. Click the **Add** button once the form is completed.

5. **Voice Call Contact**.

After the first **Communication Manager Address Map** is created, a corresponding media server **Contact** entry is created automatically.

sip:$(user)@166.38.98.26:5061;transport=tls

This **Contact** entry contains the IP address of the Domestic Avaya Communication Manager (the address of the C-LAN card), the port (***5061*** is the port for TLS) and the transport protocol (***tls***) to be used. The incoming digits sent in the ***user*** part of the original request URI will replace the $(user) string when the message is sent to the **Contact**.

If necessary the Contact information can be modified by clicking on the ***Edit*** button (see **Step 7**).

6. Add Fax Call Map.

Three fax maps need to be added to include only the fax extensions (1002, 1004, 2002, 2004, 4002, and 4004). Repeat steps 1 through 5 using the following information:

- Click on the ***Add Map In New Group*** button and enter a map for the 1002 and 1004 fax extensions.
    1. Enter a name for the map. (e.g. ***Fax 100x***)
    2. Enter the **Address Map Pattern** for incoming 1002 and 1004 fax station calls into the **Pattern** field.
        - ***^sip:100[24]{1}*** This string means match any string where the first three digits are 100, and the fourth digit is 2 or 4.
    3. Verify the ***Replace URI*** option is checked.
    4. Click the **Add** button once the form is completed.
    5. Click on **Add Another Map**.
    6. Enter a name for the map. (e.g. ***Fax 200x***)
    7. Enter the **Address Map Pattern** for incoming 2002 and 2004 fax station calls into the **Pattern** field.
        - ***^sip:200[24]{1}***
    8. Verify the ***Replace URI*** option is checked.
    9. Click the **Add** button once the form is completed.

10. Click on **Add Another Map**.
11. Enter a name for the map. (e.g. ***Fax 400x***)
12. Enter the **Address Map Pattern** for incoming 4002 and 4004 fax station calls into the **Pattern** field.
    - *^sip:400[24]{1}*
13. Verify the ***Replace URI*** option is checked.
14. Click the **Add** button once the form is completed.

7. Modify Fax Call Contact.

After the first fax **Communication Manager Address Map** is created in the new group, a corresponding media server **Contact** entry is created automatically. However this contact will specify the default value of 5061. As described in **Section 3.1.6**, inbound G.711 fax calls to Avaya Communication Manager are identified by using port 5062. Therefore the fax Contact must be changed to specify port 5062.

- Click on the Contact ***Edit*** button and the Edit Communication Manager Contact window will open (**Figure 62**).

1. Modify the **Contact** to specify port 5062.
    - ***sip:$(user)@166.38.98.26:5062;transport=tls***
2. Click on the **Submit** button.

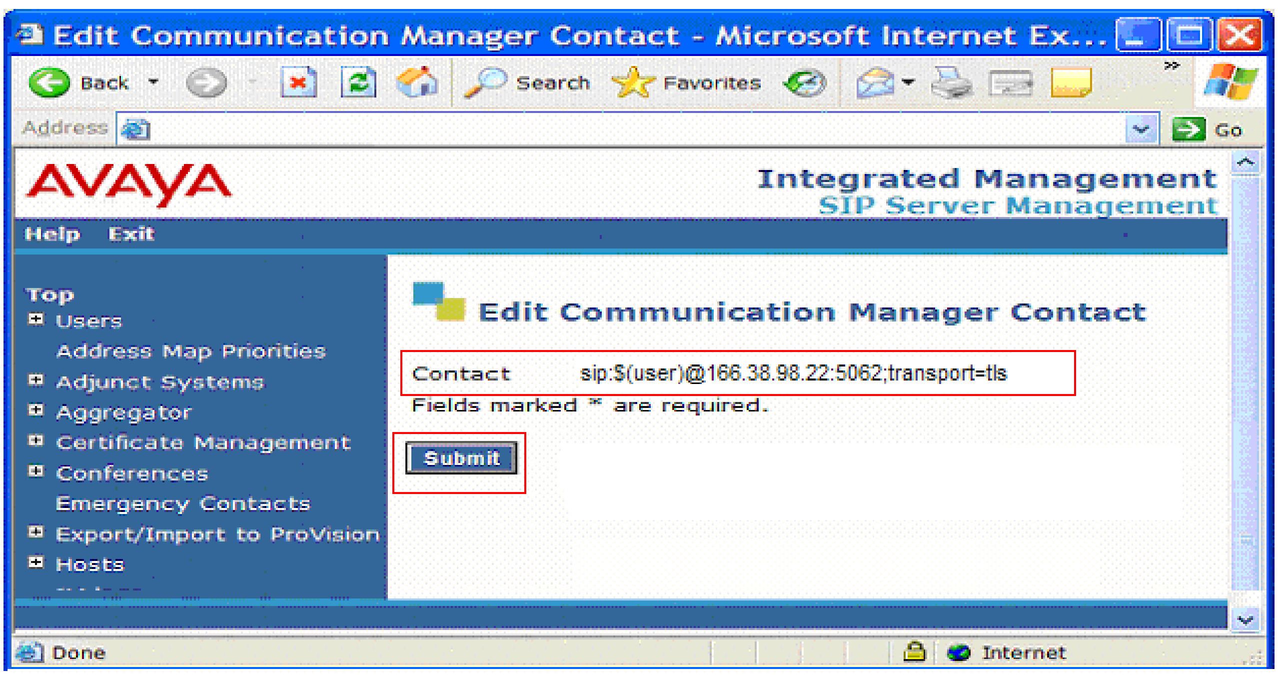

Figure 62: Domestic Primary Avaya SIP Enablement Services Server – Edit Fax Contact

4.1.1.7 Verifying/Configuring Avaya SIP Enablement Services DNS Information

Warning – The ***initial_setup*** procedure described in this section for modifying/adding DNS information could result in the ***re-initialization of the Avaya SIP Enablement Services data base and all existing provisioning would be lost***. It is **strongly** recommended that the Avaya SIP Enablement Services be backed up prior to this procedure.

The IP address of the DNS server used by the Avaya SIP Enablement Services is typically specified during the Avaya SIP Enablement Services installation process. The DNS address can be verified by logging on to the Avaya SIP Enablement Services Linux shell (using SSH and the appropriate credentials). Go to the **/etc** directory and enter the ***cat resolv.conf*** command. The current DNS entry is displayed (**Figure 63**).

craft@Avaya_SIL_SES> **cat resolv.conf**
resolv.conf generated by CCS Installer
nameserver 99.88.77.1
nameserver 99.88.77.2
search company.com

Figure 63: Avaya SIP Enablement Services Display Current DNS Configuration

If the DNS entry needs to be changed, connect to the Avaya SIP Enablement Services Linux shell (any directory) and enter the ***initial_setup*** command (the setup process may execute several background functions that may take a few minutes). **Figure 64** will be displayed.

Using the *Tab* key, move the cursor to the **Primary DNS IP Address** line and enter the new DNS address. If there is a secondary DNS address, enter it in the next line. Press the *Tab* key until the ***OK*** button is highlighted. Then press the *Enter* key. You can press the ***Exit*** button at any time to abort the process.

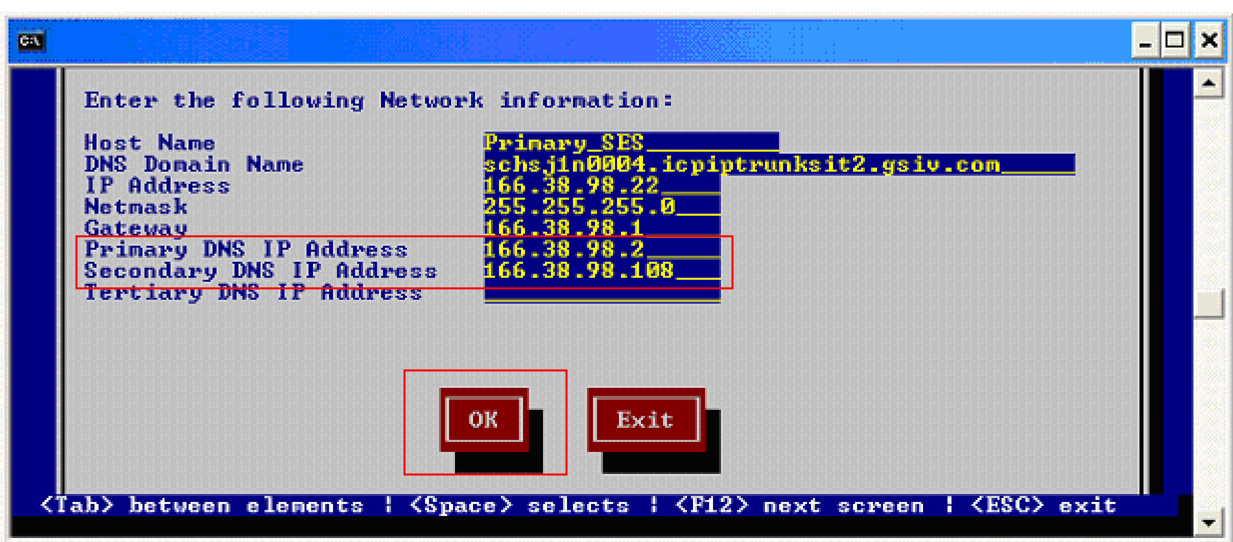

Figure 64: Domestic Primary Avaya SIP Enablement Services Server –

Changing the DNS IP Address

After the ***OK*** button is pressed (**Figure 64**), the screen showed in **Figure 65** will be displayed. Leaving the default value selected (e.g. **Simplex**), press the *Tab* key until the ***OK*** button is highlighted. Press the *Enter* key.

Figure 65: Domestic Primary Avaya SIP Enablement Services Server – Default Redundancy Configuration

After the ***OK*** button is pressed (**Figure 65**), the screen shown in **Figure 66** will open. Verify the ***Finish*** button is highlighted (use the *Tab* key to change the selection) and press the ***Enter*** key.

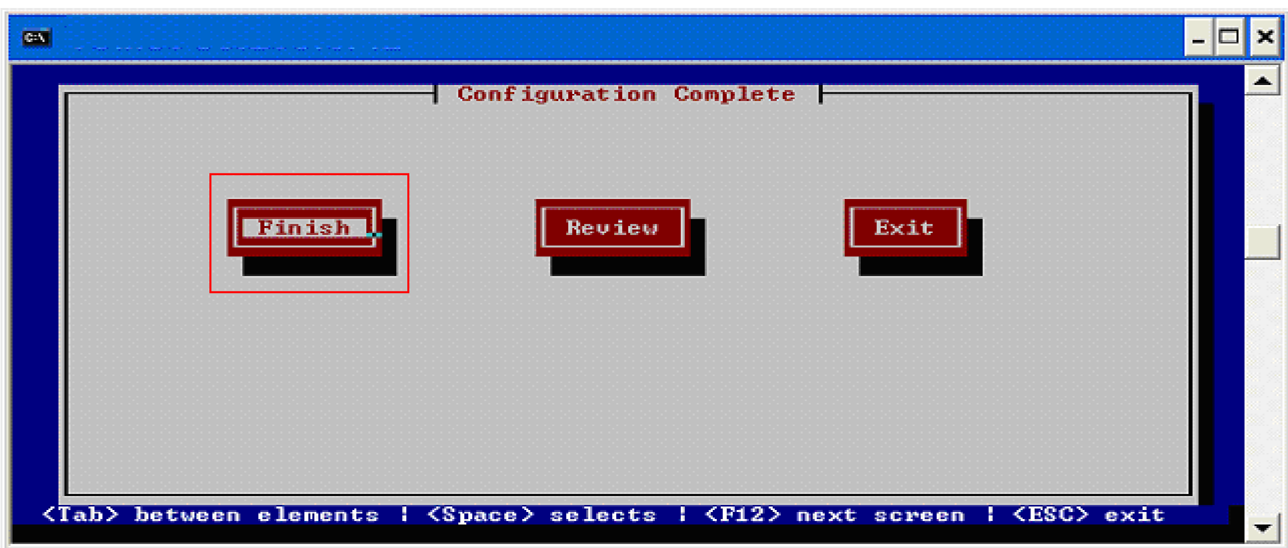

Figure 66: Domestic Primary Avaya SIP Enablement Services Server – Finish Changing the DNS IP Address

The screen shown in **Figure 67** will open asking to stop the Avaya SIP Enablement Services processes. Verify that the option **[y]** is offered. If not, enter ***y***. Press the ***Enter*** key to accept the default value **[y]**.

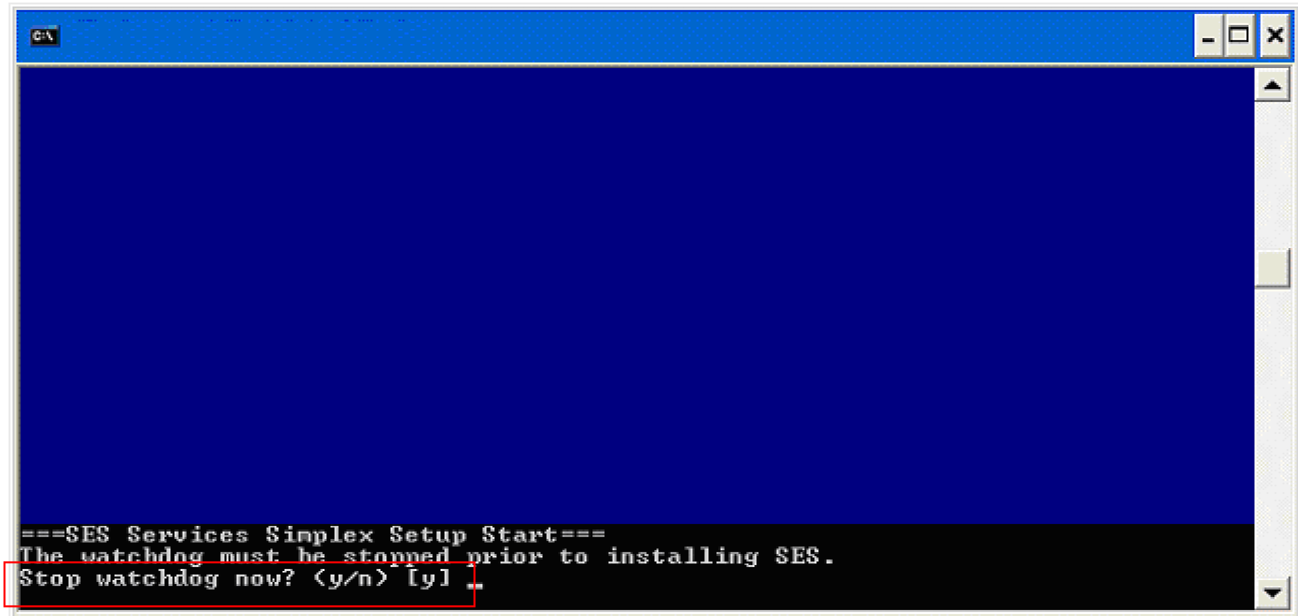

Figure 67: Changing the DNS IP Address –
Stopping the Avaya SIP Enablement Services Processes

Next the screen shown in **Figure 68** will open regarding the Avaya SIP Enablement Services database. Verify that the option **[n]** is offered. If not, enter ***n***. Press the ***Enter*** key to accept the default value **[n]**.

Note – <u>This step must be performed with caution</u>. If "y" is entered the database will be initialized and all existing Avaya SIP Enablement Services provisioning will be lost!

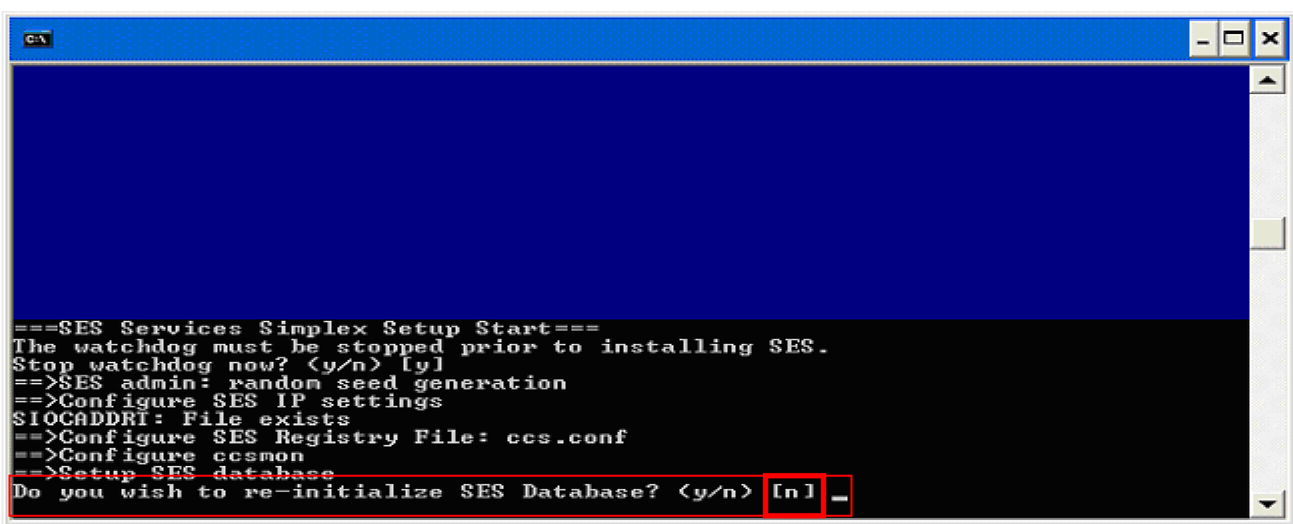

Figure 68: <u>Do Not</u> Initialize the Avaya SIP Enablement Services Database

Next the screen shown in **Figure 69** will open asking to restart the Avaya SIP Enablement Services processes. Verify that the option **[y]** is offered. If not, enter ***y***. Press the ***Enter*** key to accept the default value **[y]**.

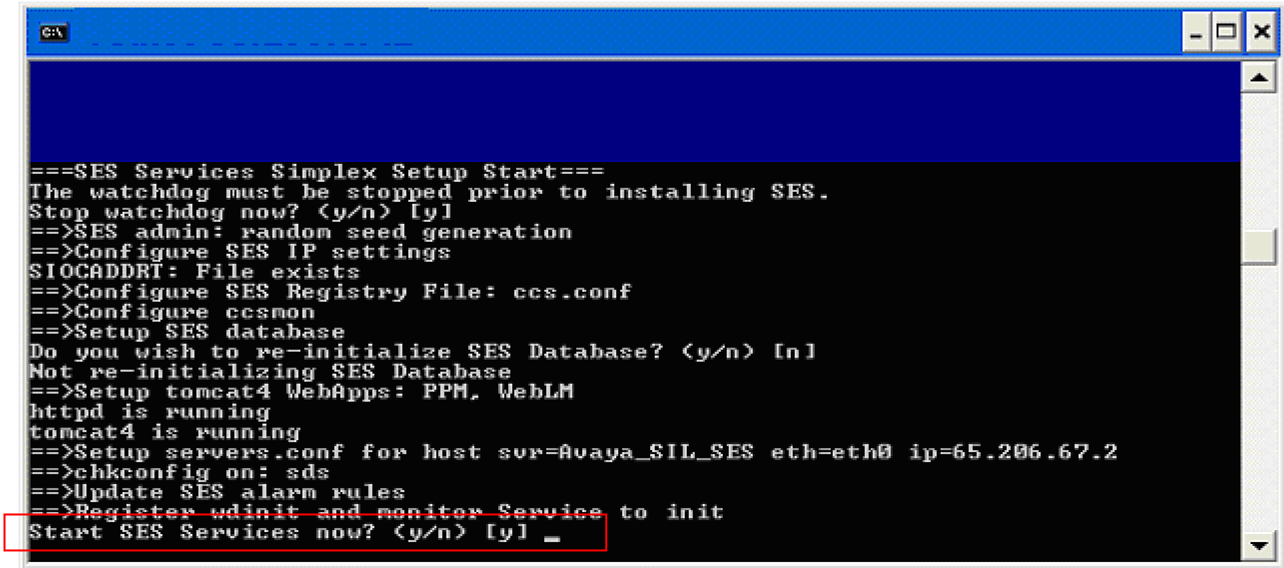

Figure 69: Changing the DNS IP Address – Restarting Avaya SIP Enablement Services

The Avaya SIP Enablement Services will then restart with the new DNS parameters.

4.1.1.8 Add an Avaya SIP Station (User)

Note - An Avaya SIP station must also be provisioned in Avaya Communication Manager (see **Section 3.3.1**).

Add a SIP User to the Avaya SIP Enablement Services as follows:

1. From the **Top** page of the SIP Enablement Services **SIP Server Management** web interface (**Figure 52**), expand the **Users** link in the left side blue navigation bar and click on the **Add User** link. The Add User window will open (**Figure 70**).

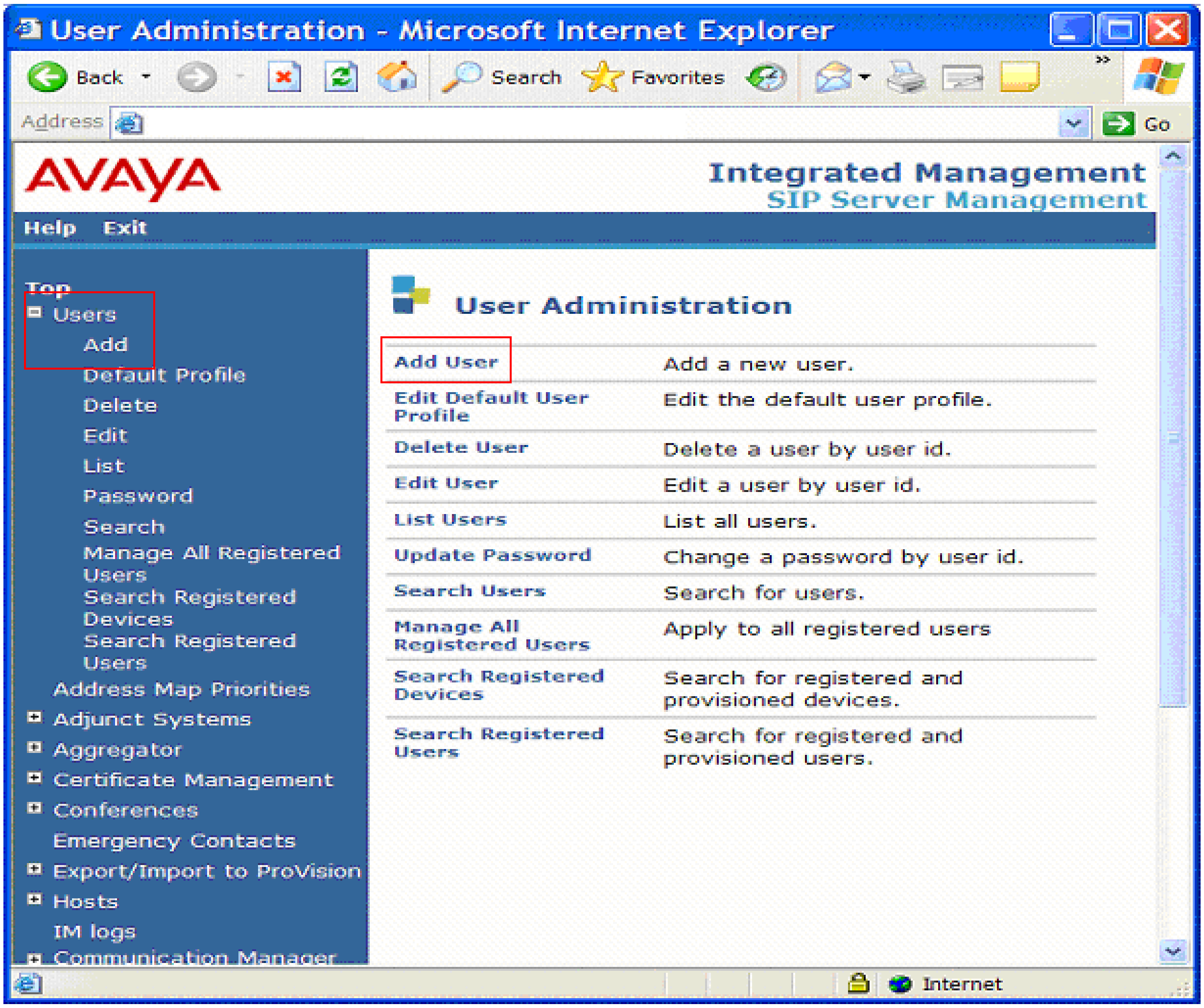

Figure 70: Domestic Primary Avaya SIP Enablement Services Server - User Administration

2. In the **Add User** page (**Figure 71**), enter the extension number for the SIP telephone in the **Primary Handle** and the **User ID** fields.
3. Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
4. Enter the **First Name** and **Last Name** of the user.
5. Select the **Add Media Server Extension** checkbox. This associates an Avaya Communication Manager extension with this SIP User. Calls from this user will be provided features and routing via Avaya Communication Manager.
6. In the **Host** field drop down menu, select the Avaya SIP Enablement Services server hosting the domain (**166.38.98.22**) for this user (see **Section 4.1.2** regarding User provisioning on the Secondary Avaya SIP Enablement Services) server.

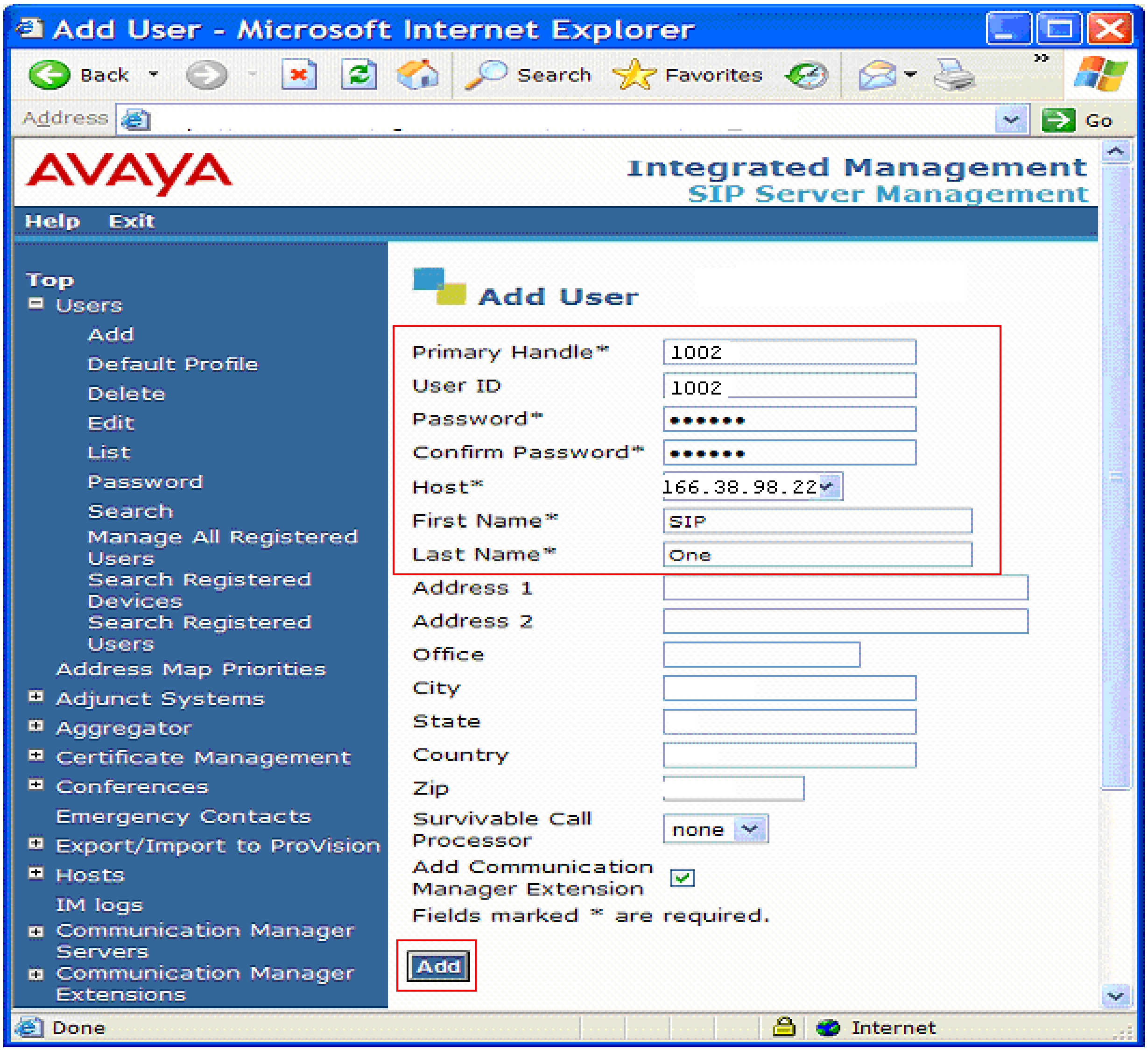

Figure 71: Domestic Primary Avaya SIP Enablement Services Server - Add User

7. Click the **Add** button. This will cause a confirmation screen to appear.
8. Click **Continue** on the confirmation screen. The **Add Communication Manager Extension** window will open (**Figure 72**).

Figure 72: Domestic Primary Avaya SIP Enablement Services Server - Add Communication Manager Extension

The SIP phone handle must now be associated with the corresponding extension on Avaya Communication Manager (see **Section 3.3.1**).

9. In the **Add Communication Manager Extension** page, enter the extension number (e.g., ***1002***) corresponding to the extension previously configured on Avaya Communication Manager. The **Communication Manager Extension** and the **Primary Handle** are usually the same, but are not required to be.
10. From the drop down menu, select the **Communication Manager Server** interface assigned to this extension.
11. Click the **Add** button.

4.1.2 Domestic Secondary Avaya SIP Enablement Services Server

The Domestic Secondary Avaya SIP Enablement Services server ***must*** be provisioned identically to the Primary Avaya SIP Enablement Services server. Repeat all steps shown in **Section 4.1.1**. There is only one exception to this. The Secondary Avaya SIP Enablement Services server has a unique IP address and name. These are usually specified at installation.

Care must be taken that any provisioning performed on either Avaya SIP Enablement Services is repeated on the other. This includes any Users (SIP stations) that are defined on the Primary Avaya SIP Enablement Services server.

> **Note** – Currently multi-registration on Avaya SIP telephones is not supported. If the Primary Avaya SIP Enablement Services server fails, the Avaya SIP telephones will loose registration. They must be manually provisioned to register to the Secondary Avaya SIP Enablement Services server. Alternatively a second **46xxsettings.txt** file may be created that specifies the Secondary Avaya SIP Enablement Services server, and used when the Primary Avaya SIP Enablement Services server is not available. See [9] for more information on the 46xxsettings.txt file.

4.1.2.1 Changing Avaya SIP Enablement Services IP Address/Name

If the IP address and or name of the Avaya SIP Enablement Services needs to be changed after installation, the steps described in **Section 4.1.1.7** to modify the DNS IP address(s) can also be used to modify these fields. **Figure 73** shows the fields to be changed if this becomes necessary.

In the reference configuration, the Secondary Avaya SIP Enablement Services server name is **Secondary_SES** and the IP address is **166.38.98.17**.

> **Note** – If the IP address is changed, this change must be reflected in the Avaya Communication Manager Node-Names form (see **Section 3.1.3**).

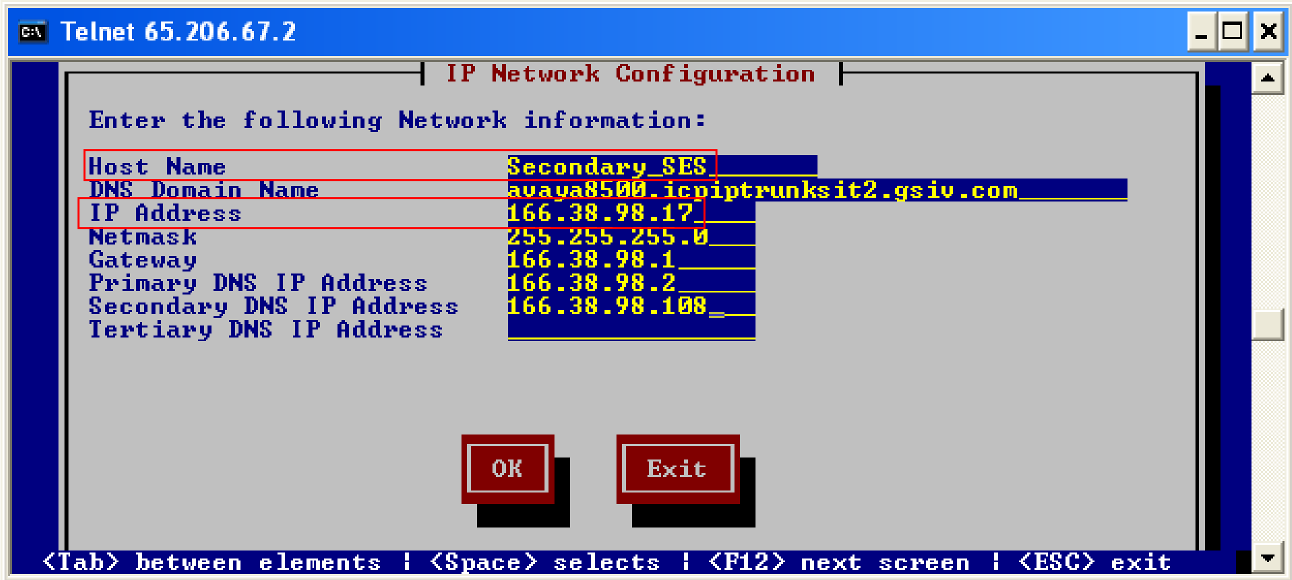

Figure 73: Modifying an Avaya SIP Enablement Services IP address

If the IP address of the Avaya SIP Enablement Services is changed, insure that the following Avaya SIP Enablement Services forms are also updated:

- **System Properties** (see **Figure 52**) – License Host field.
- **Host Form** (see **Figure 53**) – Host IP Address

4.2. EMEA Avaya SIP Enablement Services Specific Provisioning

Using the EMEA configuration shown in **Figure 3** as a guide, provision the EMEA Primary and Secondary Avaya SIP Enablement Services servers. The Avaya SIP Enablement Services provisioning in the EMEA environment is similar to the Domestic Avaya SIP Enablement Services provisioning described in **Section 4.1** with the exceptions:

4.2.1 Primary Avaya SIP Enablement Services Server

4.2.1.1 Verify System Properties – SIP Domain

- In the **View System Properties** page (see **Figure 52**):
  a. Enter the **SIP Domain**. The EMEA SIP Domain ***avaya8300.gsiv.com*** is used in the reference configuration (see **Section 1.3**).

4.2.1.2 Verify the Host Information

On the **Edit Host** screen (see **Figure 54**):

- Verify that the IP address of the EMEA Primary Avaya SIP Enablement Services server **Host IP Address** field is **166.38.98.73**.

4.2.1.3 Configure Avaya Communication Manager Server Interfaces

On the **Edit Communication Manager Server Interface** screen **(see Figure 56)**:

- Specify a **Communication Manager Server Interface Name** (e.g. **S8300**).

- Specify the IP address of the EMEA Avaya Communication Manager SIP trunk interface, **166.38.98.19**. This is the IP address of the **Near-end Node Name** specified in the Avaya Communication Manager Signaling Group form for the associated SIP Trunk (see **Section 3.2**).
- Specify the IP address of Avaya Communication Manager in the **Communication Manager Server** Admin Address field (e.g. **166.38.98.19**).

4.2.1.4 Configure Trusted Hosts

In the reference configuration, the EMEA Avaya SIP Enablement Services servers used the same Trusted Hosts as the Domestic configuration (see **Figure 57**).

4.2.1.5 Configure EMEA Call Routing

The EMEA environment also uses Domain Based Routing for outbound calls.

4.2.1.5.1 Domain Based Routing – Outbound Calls

In the EMEA reference configuration Avaya Communication Manager specifies the Verizon EMEA Fully Qualified Domain Name (FQDN) ***schsj1n0005.emeaiptrunksit2.gsiv.com*** in the Far-End Domain field of the outbound SIP Trunk Signaling Groups (14, 15, 18, and 19, see **Section 3.2**). The Avaya SIP Enablement Services receives an INVITE from Avaya Communication Manager with the destination URL of ***<callednumber>@ schsj1n0005.emeaiptrunksit2.gsiv.com***.

4.2.1.5.2 Inbound Call Routing

As described in **Section 3.1.6.2** for the Domestic environment, the EMEA environment also uses port 5061 (default) for inbound voice calls and port 5062 for inbound G.711 fax calls.

4.2.1.5.3 Inbound Calls

1. Add Voice Call Maps.
   Two voice maps need to be added to include all the voice extensions (103x and 403x) and to exclude the fax extensions (1039 and 4039)
   - Click on the ***Add Map In New Group*** button. The **Add Communication Manager Address Map** window will open (see **Figure 61**).
2. Enter a map for the 103x voice extension range.
   - Enter a name for the map. (e.g. ***103x***)
   - Enter the **Address Map Pattern** for incoming 10xx voice calls (excluding fax extension 1039) into the **Pattern** field. See **Appendix A** for more information on defining map patterns.
     1. ***^sip:103[0-8]{1}*** This string means match any string where the first three digits are 103, and the fourth digit is 0 through 8. This will exclude 1039.
   - Verify the ***Replace URI*** option is checked.
   - Click the **Add** button once the form is completed.
3. Enter a map for the 403x voice extension range.
   a. Click on **Add Another Map**.
   b. Enter a name for the map. (e.g. ***403x***)

c. Enter the **Address Map Pattern** for incoming 40xx voice calls (excluding fax extension 4039) into the **Pattern** field.
    i. ***^sip:403[0-8]{1}***
d. Verify the ***Replace URI*** option is checked.
e. Click the **Add** button once the form is completed.

4. Voice Call Contact.

After the first **Communication Manager Address Map** is created, a corresponding media server **Contact** entry is created automatically.

sip:$(user)@166.38.98.73:5061;transport=tls

Note - This **Contact** entry contains the IP address of the EMEA Avaya Communication Manager (the address of the **Procr**), the port (***5061*** is the port for TLS) and the transport protocol (***tls***) to be used. The incoming digits sent in the ***user*** part of the original request URI will replace the $(user) string when the message is sent to the **Contact**.

5. Add fax Call Map.

Two fax maps need to be added to include only the fax extensions (1039 and 4039). Repeat steps 1 through 4 using the following information:

- Click on the ***Add Map In New Group*** button and enter a map for the 1039 fax extensions.
    1. Enter a name for the map. (e.g. ***Fax 1039***)
    2. Enter the **Address Map Pattern** for incoming 1039 station fax calls into the **Pattern** field.
        - ***^sip:1039***
    3. Verify the ***Replace URI*** option is checked.
    4. Click the **Add** button once the form is completed.
    5. Click on **Add Another Map**.
    6. Enter a name for the map. (e.g. ***Fax 4039***)
    7. Enter the **Address Map Pattern** for incoming 4039 station fax calls into the **Pattern** field.
        - ***^sip:4039***
    8. Verify the ***Replace URI*** option is checked.
    9. Click the **Add** button once the form is completed.

6. Modify fax Call Contact.
   After the first fax **Communication Manager Address Map** is created in the new group, a corresponding media server **Contact** entry is created automatically. However this contact will specify the default value of 5061. As described in **Section 3.1.6.2**, inbound G.711 fax calls to Avaya Communication Manager are identified by using port 5062. Therefore the fax Contact must be changed to specify port 5062.

- Click on the Contact ***Edit*** button and the Edit Communication Manager Contact window will open (see **Figure 62**).
- Modify the **Contact** to specify port 5062.

sip:$(user)@166.38.98.19:5062;transport=tls

- Click on the **Submit** button.

4.2.1.6 Verifying/Configuring Avaya SIP Enablement Services DNS Information

The same procedures apply for modifying the EMEA DNS information as described for the Domestic environment (see **Section 4.1.1.7**)

4.2.1.7 Add an Avaya SIP Station (User)

The procedures for adding an Avaya SIP station are the same as described for the Domestic environment in **Section 4.1.1.8**.

4.2.2 EMEA Secondary Avaya SIP Enablement Services Server

As described for the Domestic environment (**Section 4.1.2**) the EMEA Secondary Avaya SIP Enablement Services server ***must*** be provisioned identically to the EMEA Primary Avaya SIP Enablement Services server. However the Secondary Avaya SIP Enablement Services server has a unique IP address and name from the EMEA Primary Avaya SIP Enablement Services server:

- Name = **E_Secondary_SES**
- IP Address = **166.38.98.70**

Otherwise repeat all steps shown in **Section 4.2.1**.

5. Verizon Business IP Trunk Service Offer Configuration

Information regarding Verizon Business IP Trunk service offer can be found at http://www.verizonbusiness.com/us/products/voip/trunking/ or by contacting a Verizon Business sales representative.

As described in **Section 1**, the reference configuration described in these Application Notes was constructed at the Verizon Interoperability Test Lab in Richardson, Texas. The Verizon staff performed all the LAN/WAN network provisioning as well as the provisioning of the Verizon IP Trunk service offer environment (referred to as "SIT2" in the reference configuration). In addition, Verizon provisioned their network to send inbound calls via the Primary or Secondary Avaya SES servers based on network conditions. Verizon provided the following information necessary to complete the Avaya Communication Manager and Avaya SIP Enablement Services administration discussed in these Application Notes.

5.1. Domestic and EMEA Inbound (to CPE) and Outbound numbers

The following numbers were used for inbound and outbound dialing in the Domestic and EMEA environments.

<table>
<tr><th></th><th>Inbound Numbers</th><th>Outbound Numbers</th></tr>
<tr><td>Domestic Voice</td><td>10xx, 20xx, 40xx</td><td rowspan="2">530352xxxx
1530352xxxx
1972728xxxx
18xxxxxxxxx
1xxx5551212
011
0</td></tr>
<tr><td>Domestic Fax</td><td>1002, 1004
2002, 2004
4002, 4004</td></tr>
</table>

Table 19: Domestic numbers

<table>
<tr><th></th><th>Inbound Numbers</th><th>Outbound Numbers</th></tr>
<tr><td>EMEA Voice</td><td>103x, 403x</td><td rowspan="2">00441189056849
02871987654999
02311234567891
0894567
11x</td></tr>
<tr><td>EMEA Fax</td><td>1039, 4039</td></tr>
</table>

Table 20: EMEA Numbers

5.2. Fully Qualified Domain Name (FQDN)s

The following Fully Qualified Domain Name (FQDN)s were provided by Verizon for the Domestic and EMEA environments in the reference configuration.

	CPE (Avaya)	Verizon Network
Domestic	avaya8500.icpiptrunksit2.gsiv.com	schsj1n0004.icpiptrunksit2.gsiv.com
EMEA	avaya8300.gsiv.com	schsj1n0005.emeaiptrunksit2.gsiv.com

Table 21: Fully Qualified Domain Name (FQDN)s

5.3. Verizon DNS Information

The following DNS information was provided by Verizon for the Domestic and EMEA environments in the reference configuration.

DNS	
IP address	166.38.98.2, 166.38.98.108

Table 22: DNS Information

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Verizon Business IP Trunk service offer and an Avaya SIP telephony solution using public SIP trunking. This section covers the general test approach and the test results.

6.1. General Test Approach

A simulated enterprise site consisting of an Avaya Communication Manager and Avaya SIP Enablement Services IP telephony solution supporting public SIP trunking was connected to a laboratory version of the Verizon Business IP Trunk service offer as shown in **Section 1**. The enterprise site was configured as if using the generally available service provided by Verizon Business.

The features and functionality covered during the SIP trunking interoperability compliance test are referenced in the Verizon Business Product Integration test requirements [12].

6.2. Test Results

Interoperability testing of the Verizon SIP Trunk Redundant (2-CPE) option reference configuration was completed with successful results.

The testing successfully demonstrated the ability for Avaya Communication Manager and Avaya SIP Enablement Services, working in conjunction with Verizon's SIP Trunk Redundant (2-CPE) network architecture, could support failover of the primary service connection to a redundant (secondary) service connection supported by both a secondary Avaya SIP Enablement Services server in the enterprise and a secondary SBC in Verizon's network (see **Figure 1** in **Section 1**).

The following issues shown in **Table 23** were observed during the testing.

Reported Issue	Solution
Fax failure rate exceeded test exit criteria.	**12/9** – Avaya provided resolution using **Audio Codes MP-202** gateway (see [10]).
Call Forwarding with Diversion Header not supported.	SIP Diversion Header supported in a future release of Avaya Communication Manager.
High availability does not automatically occur with Avaya SIP phones.	Feature available in a future release of the Avaya 96xx SIP telephones.
The Avaya SIP Enablement Services sends a DNS A record lookup prior to SRV record lookup.	The Avaya SIP Enablement Services operating as designed.
A malformed 2nd DNS A record lookup is sent by the Avaya SIP Enablement Services if the initial lookup did not get a response or receive a negative response.	The issue is under investigation by Avaya SIP Enablement Services development.
The 96xx SIP phone does not follow RFC2543 (c=0.0.0.0) or RFC 3264 (SendOnly) hold mechanisms.	Avaya notified Verizon that the 96xx SIP phone does use RFC3264 but since TLS is used by the 96xx, this transaction is not visible to typical protocol analyzers.
DTMF digits of less than 50 ms can be produced.	Verizon mentioned that it is possible to generate DTMF tones of durations less than 50 ms. Avaya Communication Manager development notified.
The Avaya Communication Manager 5.1 does not support the Update method.	Avaya Communication Manager development reports that Avaya Communication Manager will send an UPDATE but not receive it.
SIP 180 message is sent with SDP.	Avaya Communication Manager operating as designed. This issue is under discussion between Avaya and Verizon.
The Privacy header is lacking in calls invoking privacy.	Fix will be applied to a future release of Avaya Communication Manager.

Reported Issue	Solution
Avaya Communication Manager 5.1 sends an INVITE with no SDP within an existing conversation.	Avaya Communication Manager operating as designed.
Avaya Communication Manager 5.1 SIP: If the Avaya cannot complete a call it sends a 500 Internal Server Error instead of a 606	The Avaya SIP Enablement Services sends 500 per design.

Table 23: Summary of Issues Identified During Certification Testing

7. Verification Steps

This section provides the verification steps that may be performed to verify basic operation of the Avaya SIP telephony solution with the Verizon Business IP Trunk service offer.

7.1. Call scenarios

7.1.1 Domestic

1. Primary Avaya SIP Enablement Services server
   - Using one of the tracing methods described in **Section 8.2**, verify that the Domestic Primary Avaya SIP Enablement Services server is used for the following calls.
     - Dial a Domestic Avaya Communication Manager station number from the Verizon network and verify the incoming voice calls are received successfully on the Domestic Primary Avaya SIP Enablement Services, received on the associated Avaya Communication Manager station, there is 2-way audio, and that the call remains connected for at least 3 minutes.
     - Dial a Verizon network number from a Domestic Avaya Communication Manager station and verify the call is processed by the Domestic Primary Avaya SIP Enablement Services server. Verify there is 2-way audio, and that the call remains connected for at least 3 minutes.
     - Verify that calls are properly disconnected when either end disconnects.
2. Secondary Avaya SIP Enablement Services server
   - Disconnect the Ethernet connection to the Domestic Primary Avaya SIP Enablement Services server. Using one of the tracing methods described in **Section 8.2**, verify that the Domestic Secondary Avaya SIP Enablement Services server is used for the following calls.
     - Dial a Domestic Avaya Communication Manager station number from the Verizon network and verify the incoming voice calls are received successfully on the Domestic Secondary Avaya SIP Enablement Services server, received on the associated Avaya Communication Manager station, there is 2-way audio, and that the call remains connected for at least 3 minutes.
     - Dial a Verizon network number from a Domestic Avaya Communication Manager station and verify the call is processed by the Domestic Secondary Avaya SIP

Enablement Services server. Verify there is 2-way audio, and that the call remains connected for at least 3 minutes.
    - Verify that calls are properly disconnected when either end disconnects.
- Using Avaya Communication Manager SAT command ***busy trunk x***, (where *x* is the trunk number), busy out the Primary outbound SIP trunks 14 (voice), and 18 (fax). Using one of the tracing methods described in **Section 8**, verify that the Domestic Secondary Avaya SIP Enablement Services server is used for the following calls.
    - Dial a Verizon network number from a Domestic Avaya Communication Manager station and verify the call is processed by the Domestic Secondary Avaya SIP Enablement Services server. Verify there is 2-way audio, and that the call remains connected for at least 3 minutes.
    - Verify that calls are properly disconnected when either end disconnects.

7.1.2 EMEA

Repeat the steps in **Section 7.1.1** for the EMEA environment.

8. Troubleshooting Tools and Techniques

A protocol analyzer such as Wireshark may be used to monitor and verify SIP protocol exchanges.

The following Avaya Communication Manager and Avaya SIP Enablement Services commands may be used to troubleshoot call completion issues.

8.1. Avaya Communication Manager

- Use the SAT ***list trace station xxx*** command, where *xxx* is a station extension, to monitor station call progress.
- Use the SAT ***status station xxx*** command, where *xxx* is a station extension, to view station call states.
- Use the SAT ***list trace tac xxx*** command, where *xxx* is the tac code defined on the trunk group form, (e.g. **Section 3.1.6.1**) to monitor trunk activity.

For additional commands see [1].

8.2. Avaya SIP Enablement Services

The link between the Avaya SIP Enablement Services and Avaya Communication Manager uses TLS protocol. Most protocol analyzers cannot decode this. Therefore the Avaya SIP Enablement Services Trace Logger may be used to trace internal SIP protocol messages between the Avaya SIP Enablement Services and Avaya Communication Manager. The trace logger is accessed via the Avaya SIP Enablement Services web GUI.

1. Connect to the Avaya SIP Enablement Services web GUI using a web browser and appropriate credentials. The screen shown in **Figure 74** will open.
2. Select ***Launch SES Administration Interface***

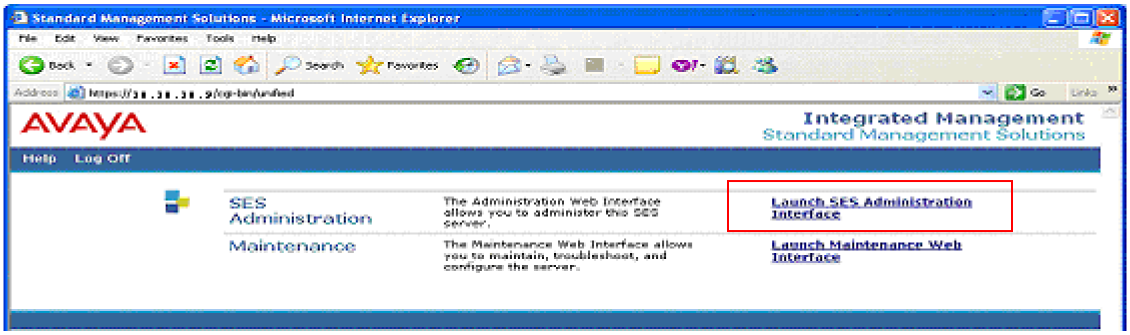

Figure 74: Avaya SIP Enablement Services Administration/Maintenance Web Interface

3. The "Top" window will open (**Figure 75**). Select ***Trace Logger*** and ***Configure Filters***.

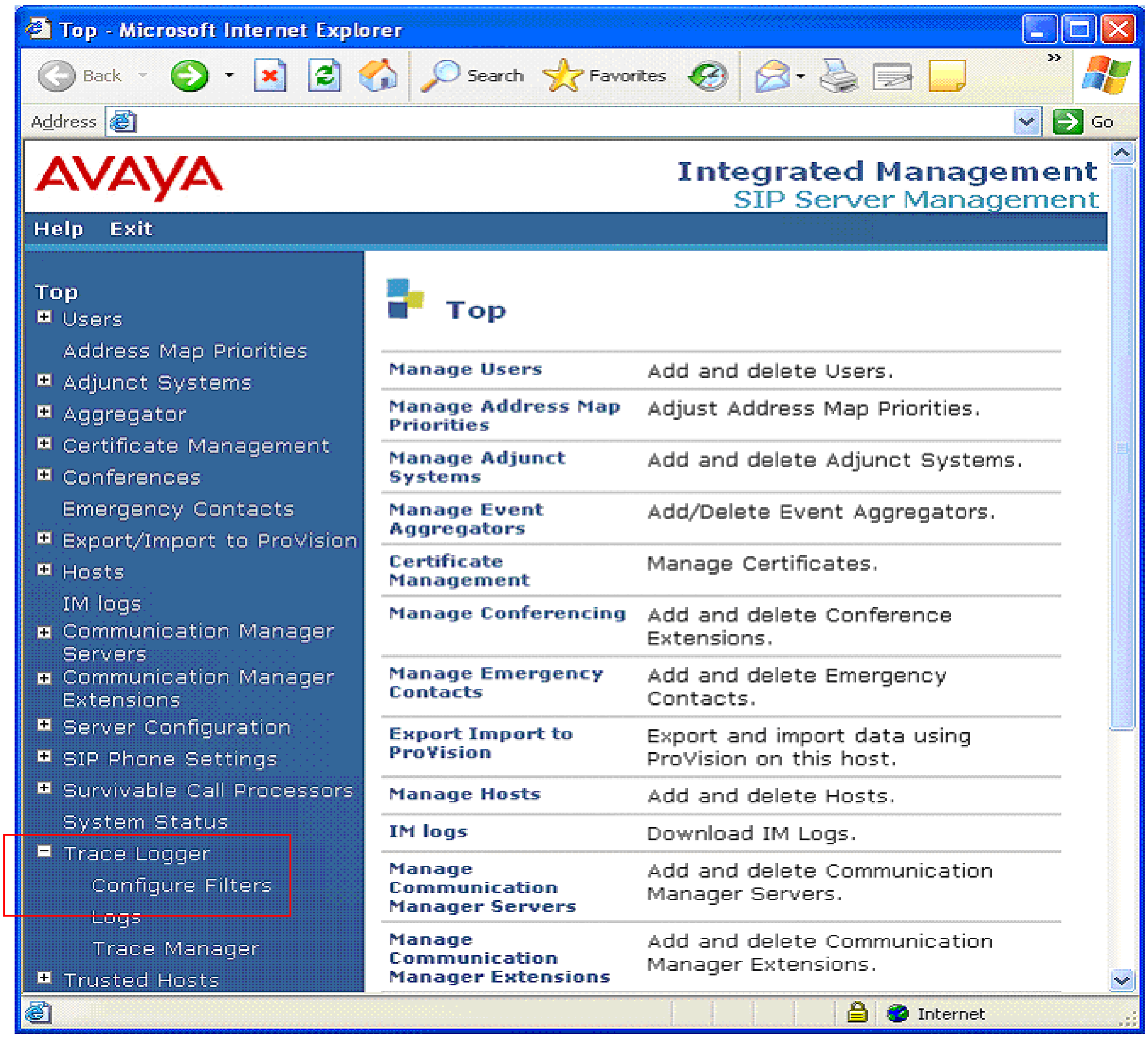

Figure 75: Avaya SIP Enablement Services "Top" Screen

4. The Filter Configuration window will open (**Figure 76)**. Select ***Add New Rule To Filter***.

Figure 76: Create Trace Filter

5. Create a filter to capture all traffic as shown in **Figure 77**.
   - **Filter Label** → Enetr a name for the filter.
   - **Message Type** → Enter ***Any***
   - **From** → Enter **.***
   - Let all other fields default.
   - Select **Update** to save the filter. The select **Continue** when prompted.

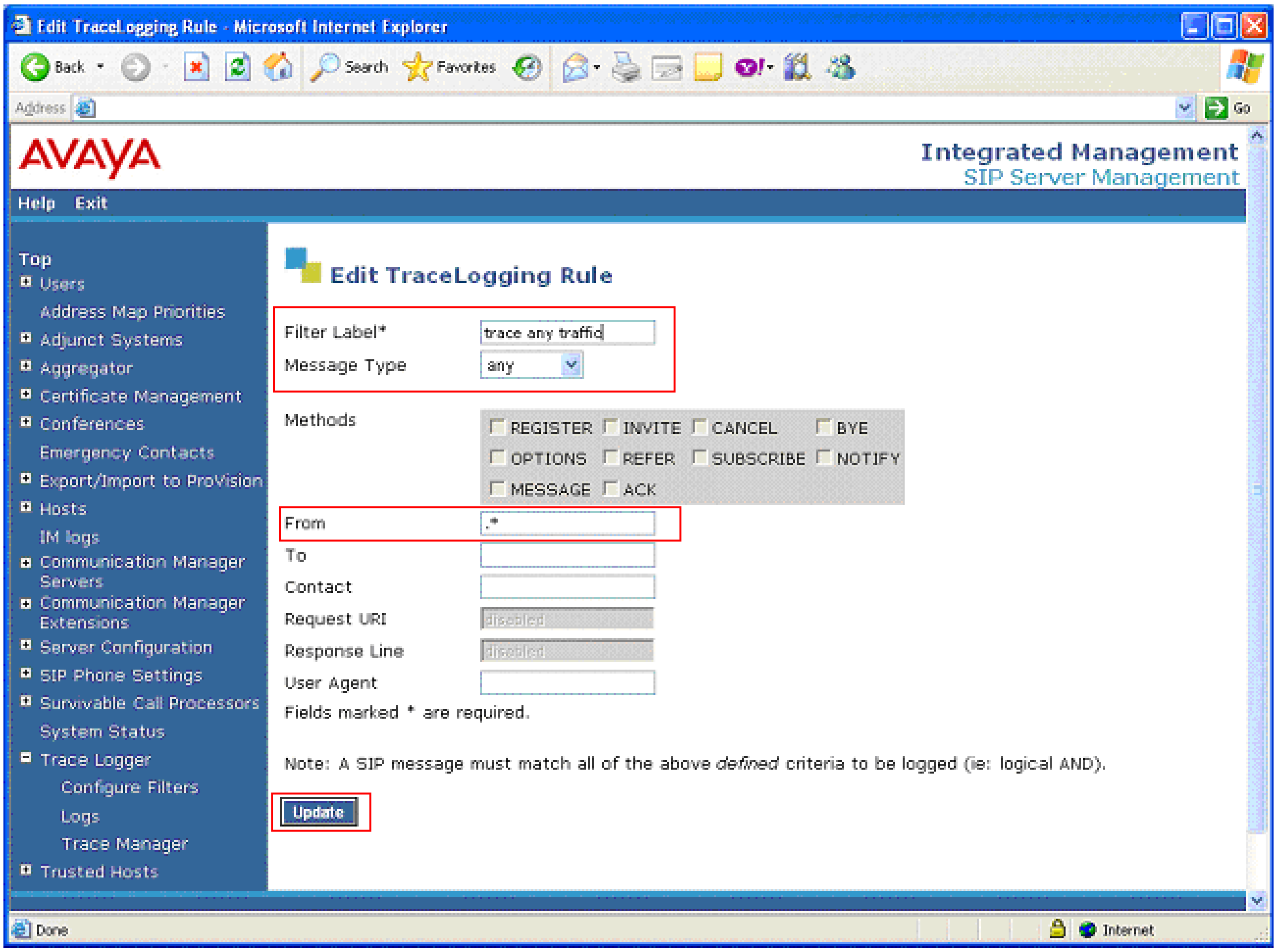

Figure 77: Filter to Capture All Traffic

6. The new filter will be displayed (**Figure 78**).

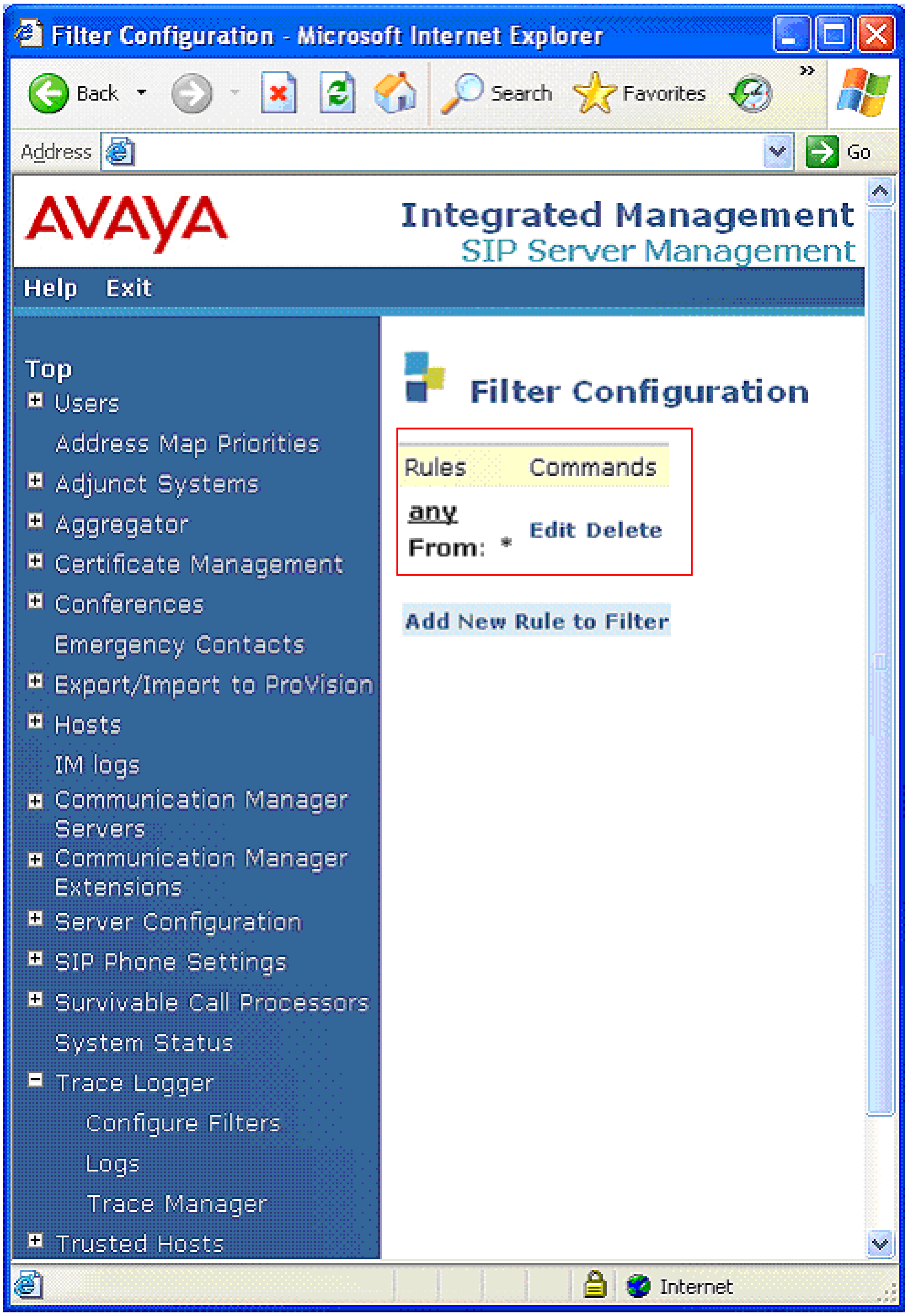

Figure 78: Filter to Capture All Traffic

7. Select ***Trace Manager*** and the **Trace Manager** window is displayed (**Figure 79**). Select ***Start Tracing*** (the screen will update to display **Tracing is on** and **Stop Tracing**). Run the test. When the test is completed select ***Stop Tracing***.

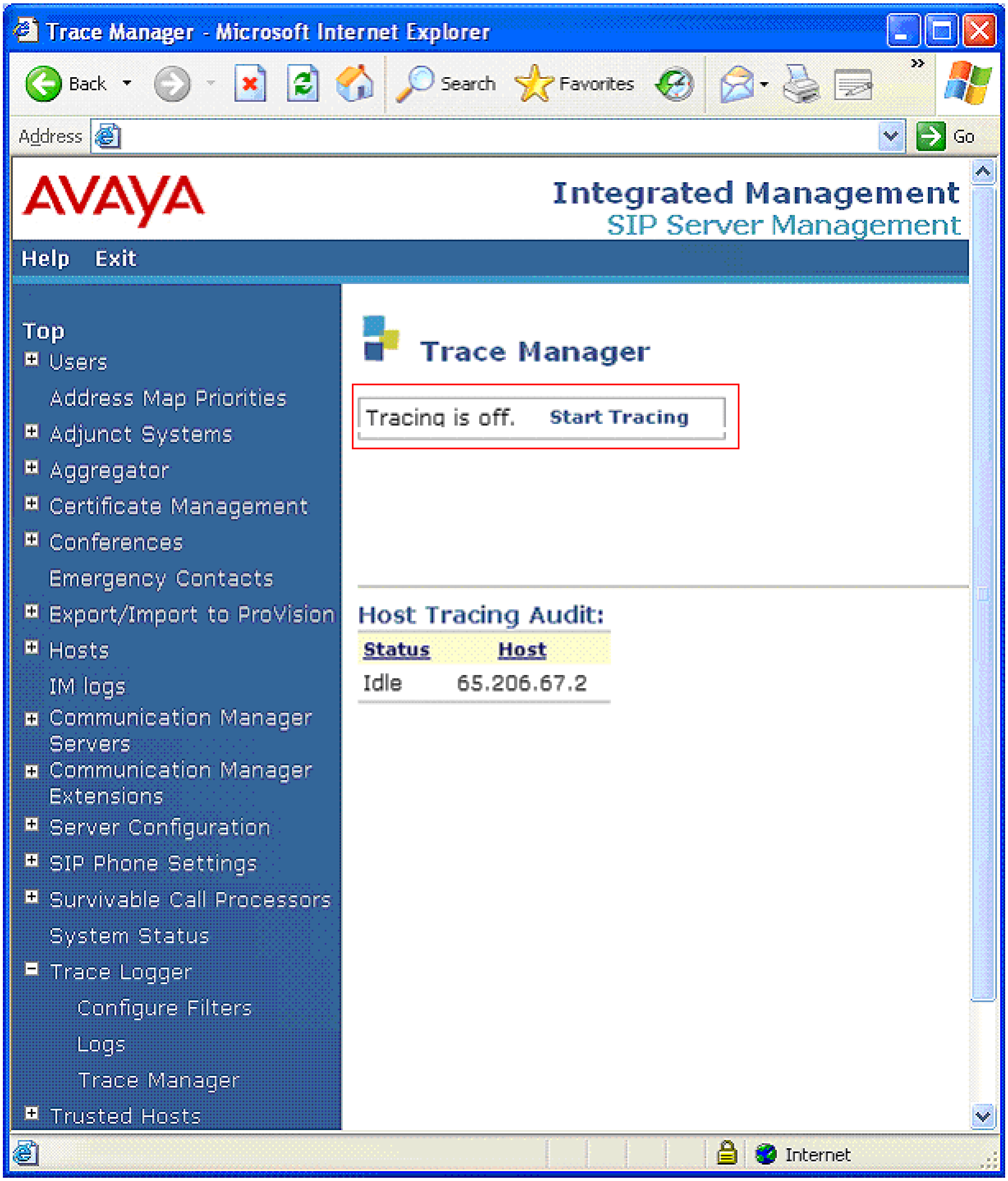

Figure 79: Start Trace

8. When the tracing is complete select Logs and the TraceLogs window will open displaying the captured trace (**Figure 80**). Select ***Down load*** and then ***OK*** when prompted. You will be asked to save the file. Once the file is saved it may be viewed with any test editor.

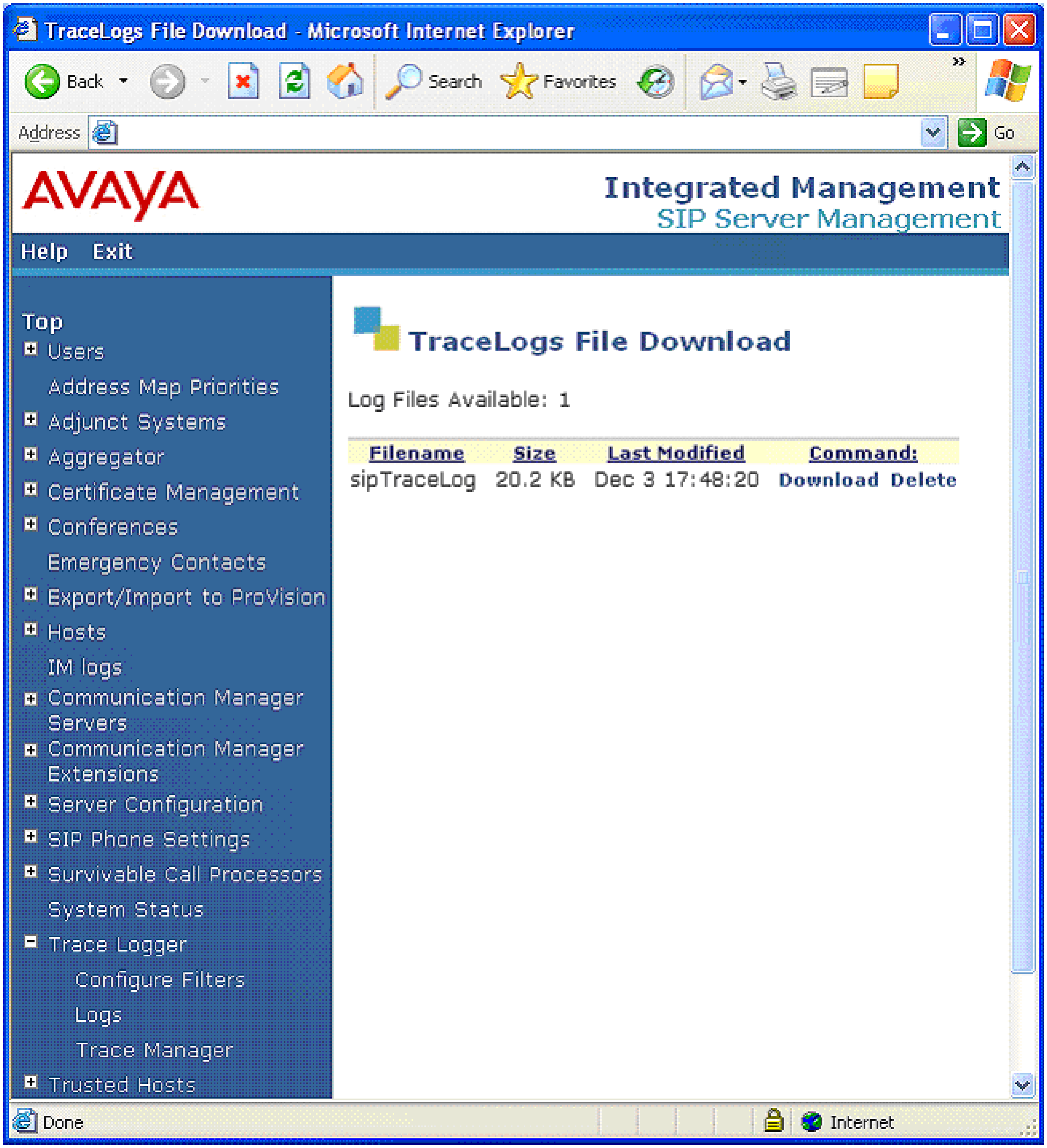

Figure 80: Start Trace

9. You will be asked to save the file. Once the file is saved it may be viewed with any test editor. **Figure 81** shows sample output from an Avaya SIP Enablement Services Trace Log.

```
Dec  3 17:47:57 2008 start remote trace session on Avaya_SIL_SES.bdevc.avaya.21sip.com:
       Using the following filters:
       Field<from> Value<.*>

-----------------------------------------------------------------
Dec  3 17:48:03 2008 matching filter label <any>: Avaya_SIL_SES.bdevc.avaya.21sip.com: [Recv
Request  ]
{connection: host=65.213.126.100 port=5060 protocol=UDP}
INVITE sip:6985245@bdevc.avaya.21sip.com SIP/2.0
Accept-Language: en
Call-ID: 8048282a63d7dd1ec549354c8d00
CSeq: 1 INVITE
From: "Adev_Digital" <sip:1101@adevc.avaya.21sip.com:6001>;tag=8048282a63d7dd1eb549354c8d00
Record-Route: <sip:65.213.126.100:5060;lr>,<sip:65.213.126.100:6001;lr;transport=tls>
To: "6985245" <sip:6985245@bdevc.avaya.21sip.com>
Via: SIP/2.0/UDP 65.213.126.100:5060;branch=z9hG4bK83838303030343433deb.0,SIP/2.0/TLS
65.213.126.100:6001;psrrposn=2;received=65.213.126.100;branch=z9hG4bK8048282a63d7dd1ed549354c8d00
Content-Length: 214
Content-Type: application/sdp
Contact: "Adev_Digital" <sip:1101@65.213.126.100:6001;transport=tls>
Max-Forwards: 68
User-Agent: Avaya CM/R015x.01.1.415.1
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLISH
Supported: 100rel,timer,replaces,join,histinfo
Alert-Info: <cid:internal@bdevc.avaya.21sip.com>;avaya-cm-alert-type=internal
Min-SE: 1200
Session-Expires: 1200;refresher=uac
P-Asserted-Identity: "Adev_Digital" <sip:1101@adevc.avaya.21sip.com:6001>
History-Info: <sip:6985245@bdevc.avaya.21sip.com>;index=1,"6985245"
<sip:6985245@bdevc.avaya.21sip.com>;index=1.1

v=0
o=- 1 1 IN IP4 65.213.126.100
s=-
c=IN IP4 65.213.126.101
b=AS:64
t=0 0
m=audio 2050 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000

-----------------------------------------------------------------
Dec  3 17:48:03 2008 matching filter label <any>: Avaya_SIL_SES.bdevc.avaya.21sip.com: [Send
Response ]
{connection: host=65.213.126.100 port=5060 protocol=UDP}
SIP/2.0 100 Trying
From: "Adev_Digital" <sip:1101@adevc.avaya.21sip.com:6001>;tag=8048282a63d7dd1eb549354c8d00
To: "6985245" <sip:6985245@bdevc.avaya.21sip.com>
Call-ID: 8048282a63d7dd1ec549354c8d00
CSeq: 1 INVITE
Via: SIP/2.0/UDP
65.213.126.100:5060;received=65.213.126.100;branch=z9hG4bK83838303030343433deb.0,SIP/2.0/TLS
65.213.126.100:6001;psrrposn=2;received=65.213.126.100;branch=z9hG4bK8048282a63d7dd1ed549354c8d00
Content-Length: 0
Organization: bdevc.avaya.21sip.com
Server: Avaya SIP Enablement Services
-----------------------------------------------------------------
```

Figure 81: Sample Avaya SIP Enablement Services Trace Log.

9. Support

9.1. Avaya

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

9.2. Verizon

For technical support on Verizon Business IP Trunk service offer, visit their online support at http://www.verizonbusiness.com/us/customer/

10. References

10.1. Avaya

The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administrator Guide for Avaya Communication Manager*, January 2008, Issue 4.0, Document Number 03-300509.

[2] *Feature Description and Implementation for Avaya Communication Manager,* January 2008, Issue 6, Document Number 555-245-205

[3] *Administering SIP Enablement Services on the Avaya S8300 Server*, Document number 03-602508, Issue 1.0, January 2008

[4] *Installing and Administering SIP Enablement Services,* January 2008, Issue 5.0, Document Number 03-600768

[5] *SIP Support in Avaya Communication Manager Running on Avaya S8xxx Servers,* Document Number 555-245-206, Issue 8, January 2008.

[6] *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones,* December 2007, Issue 2, Document Number 16-601944

[7] *Sample Avaya Shared Blade Server 3000 Locally Sourced Announcement and Music-on-Hold Configuration* - Issue 1.0, August 2005

[8] *Application Notes for Configuring SIP Trunking between the Verizon Business Service and an Avaya SIP Telephony Solution – Issue 1.0*

[9] *Avaya one-X™ Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 1.2, 16-300698, Issue 3, January 2007*

[10] *Application Notes for AudioCodes MP-202 Telephone Adaptor with Avaya SIP Enablement Services and Avaya Communication Manager – Issue 1.0*

[11] *Application Notes for Configuring Alternate Methods of Domain Based Routing in an Avaya SIP Trunk Architecture*.

10.2. Verizon

The following documents may be obtained by contacting your Verizon Business Account Representative.

[12] *Verizon Business Product Integration requirement Avaya IP-PBX 5.1 SIP TRUNK Interoperability Testing, Date:10/10/08, Rev 1.1*
[13] *Retail VoIP Interoperability Test Plan version 1.8*
[14] *Network Interface Specification Retail VoIP Trunk Interface (for non-registering devices) Document Version: 3, 2008-08-28*
[15] *Additional information regarding Verizon Business IP Trunk service offer can be found at* http://www.verizonbusiness.com/us/products/voip/trunking/

10.3. RFCs

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: http://www.rfc-editor.org/rfcsearch.html.

[16] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
[17] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

11. APPENDIX A: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the Avaya SIP Enablement Services is a Linux regular expression used to match against the URI string found in the SIP INVITE message.

Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s).

In the pattern matching string used in the Avaya SIP Enablement Services:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
  - A period **.** matches any character once (and only once).
  - An asterisk ***** matches zero or more of the preceding characters.
  - Square brackets enclose a list of any character to be matched. Ranges are designated by using a hyphen. Thus the expression **[12345]** or **[1-5]** both describe a pattern that will match any single digit between 1 and 5.
  - Curly brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' times. Thus **5{3}** matches '555' and **[0-9]{10}** indicates any 10 digit number.
  - The circumflex character **^** as the first character in the pattern indicates that the string must begin with the character following the circumflex.

Putting these constructs together as used in this document, the pattern to match the SIP INVITE string for any valid 1+ 10 digit number in the North American dial plan would be:

^sip:1[0-9]{10}

This reads as: "Strings that begin with exactly **sip:1** and having any 10 digits following will match.

A typical INVITE request below uses the shaded portion to illustrate the matching pattern.

INVITE sip:17325551212@20.1.1.54:5060;transport=udp SIP/2.0

11.1. Address Map Caveats

11.1.1 Over-lapping Address Map Strings

If over-lapping address maps are provisioned, the Avaya SIP Enablement Services may select the first matching address map string in the list, even if there is a more definitive string further down the list. For example:

If the following address map strings are listed in this order:

```
^sip:555[0-9]{4} with a contact of 10.10.10.10
^sip:555[0-9]{4}[0-2]{5} with a contact of 20.20.20.20
```

And the following dialed string is received:

```
55512
```

The Avaya SIP Enablement Services may send the call to 10.10.10.10 instead of the intended destination of 20.20.20.20.

A way to alleviate this situation is to indicate the end of the intended pattern by specifying an **@** character. Using the example above:

If the following address map strings are listed in this order:

```
^sip:555[0-9]{4}@ with a contact of 10.10.10.10
^sip:555[0-9]{4}[0-2]{5} with a contact of 20.20.20.20
```

And the following dialed string is received:

```
55512
```

The Avaya SIP Enablement Services will send the call to the intended destination of 20.20.20.20.

11.1.2 Using a Plus Sign (+) in a Host Map.

Some service providers require that outbound call dial strings begin with a **+** character (e.g. A call to 123-555-1212 would be sent as **+**123-555-1212). When the **+** character is then defined as part of an outbound Host Map, the Avaya SIP Enablement Services will interpret the **+** character as a control character, not as part of a dial string, causing the route to fail. Normally this situation can be fixed by specifying a backslash character **** before the **+** character to indicate that the **+** is not a control character. For example:

```
^sip:\+123555[0-9]
```

However in the 5.1.1 loads of the Avaya SIP Enablement Services, two backslash characters are required to perform this function.

```
^sip:\\+123555[0-9]
```

Note - If you go back to edit the map in the SIP Enablement Services GUI (even if you don't change anything), the SIP Enablement Services may remove a **** character each time. As a result, each time you edit or create a map, ensure that you have the proper number of **** characters before submitting it.

12. APPENDIX B: Configuring Avaya SIP Enablement Services Host Maps

As described in **Section 1.5**, Domain Based Routing was used in the reference configuration for outbound calls. When Domain Based Routing is used, the Avaya SIP Enablement Services uses DNS queries to determine the IP destination for outbound calls. The default port 5060 is used for these calls. The basis of Domain Based Routing is to have Avaya Communication Manager send an outbound INVITE to the Avaya SIP Enablement Services that specifies a foreign domain for which the Avaya SIP Enablement Services is not authoritative (provisioned in the Far-End Domain field of the outbound Signaling Group form, **Section 3.1.6.1**). The Avaya SIP Enablement Services is not authoritative if the domain in the INVITE *does not match* the ***SIP Domain*** of the Avaya SIP Enablement Services (see **Figure 52**).

There may be situations where a Service Provider requires a customer to send outbound calls using a non-standard port (e.g. some Session Border Controller environments). In these cases an alternative outbound method must be used: Avaya SIP Enablement Services Host Maps. Avaya SIP Enablement Services Host Maps are comprised of a Map(s) specifying the matching called number and a Contact that specifies the destination IP address, port, and transport protocol (if no port or protocol is specified, port 5060 and UDP protocol are used by default).

In contrast to Domain Based Routing, the Avaya SIP Enablement Services *must be* authoritative for the domain specified in the INVITEs sent by Avaya Communication Manager in order for the Avaya SIP Enablement Services to use Host Maps (the Avaya Communication Manager Signaling Group Far-End Domain field *does match* the SIP Domain of the Avaya SIP Enablement Services).

Host Maps are provisioned using the Avaya SIP Enablement Services web interface.

Note - In the reference configuration separate SIP Trunks were defined for local calls and outbound calls (**Section 3.1.6**). This is because the local SIP trunk Signaling Groups Far-End Domain fields specified the Avaya SIP Enablement Services SIP Domain while the outbound Signaling Groups specified a foreign domain. As described above, a Host Map configuration also requires outbound Signaling Groups to specify the Avaya SIP Enablement Services SIP Domain in their Signaling Group Far-End Domain fields. Therefore in a Host Map configuration provisioning separate local and outbound SIP trunks is not required.

12.1. Avaya Communication Manager Provisioning for Host Map Routing

1. Using the ***add*** (or *change*) ***signaling-group x*** command, configure the Signaling Group as follows:

   - Enter the domestic Avaya SIP Enablement Services **SIP Domain** of the Primary Avaya SIP Enablement Services server in the **Far-end Domain** field. In the reference configuration, the Avaya SIP Enablement Services server domain name is: **avaya8500.icpiptrunksit2.gsiv.com**.

```
add signaling-group 20
                                SIGNALING GROUP
 Group Number: 20              Group Type: sip
                        Transport Method: tls
   Near-end Node Name: clan-1a03            Far-end Node Name: Primary-SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
       Far-end Domain: avaya8500.icpiptrunksit2.gsiv.com
                                            Bypass If IP Threshold Exceeded? n
         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
         Enable Layer 3 Test? n                        IP Audio Hairpinning? n
 Session Establishment Timer(min): 3             Alternate Route Timer(sec): 6
```

Figure 82: Signaling Group Far-End Domain for Host Map Routing

2. Repeat **Step 1** for all outbound SIP trunk Signaling Groups.

12.2. Avaya SIP Enablement Services Provisioning for Host Map Routing

1. Connect to the Avaya SIP Enablement Services web interface as shown in **Section 4.1.1.1**.
2. Display the **Edit Host** page by following the **Hosts** link in the left navigation pane and then clicking on the **Map** option under the **Commands** section of the **List Hosts** screen (**Figure 83**).

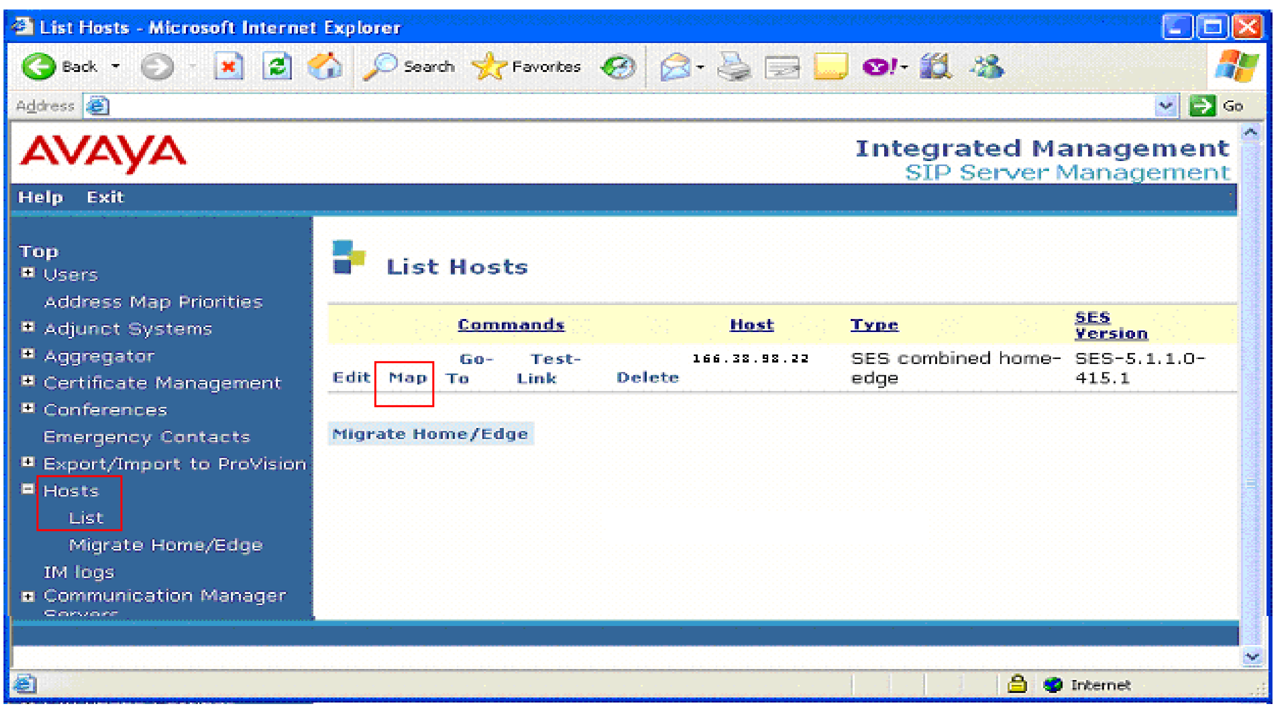

Figure 83: Avaya SIP Enablement Services List Hosts

3. The List Host Address Map page will open (**Figure 84**).
4. Select ***Add Map In New Group***

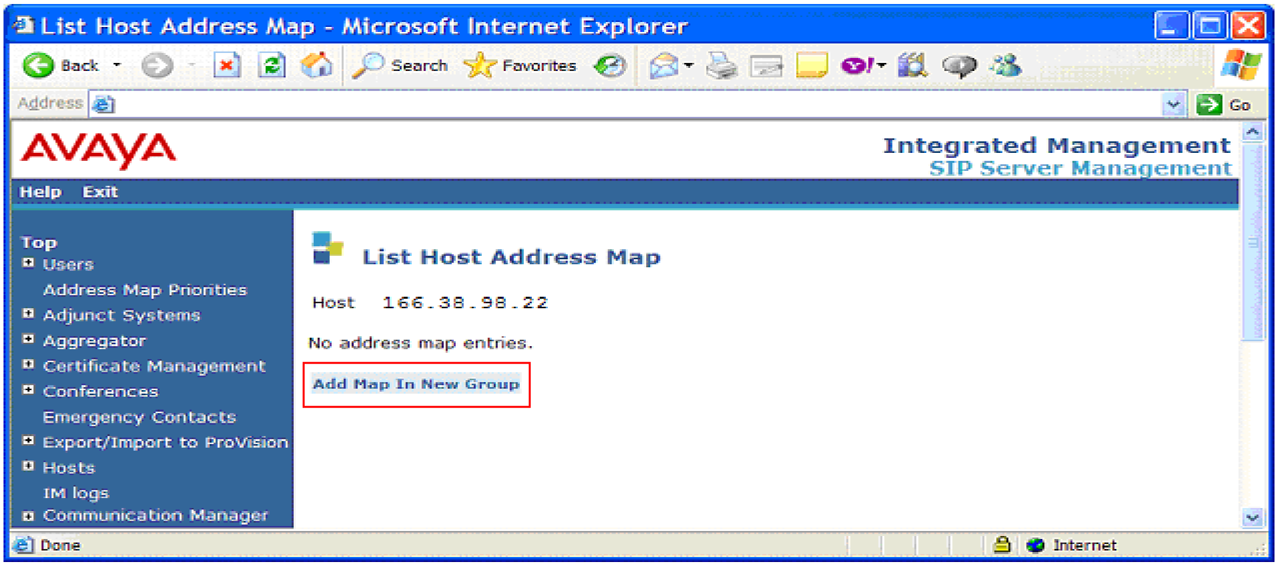

Figure 84: Avaya SIP Enablement Services List Host Address Map

5. The Add Host Address Map window will open (**Figure 85**).
6. In the **Name** field enter a name for the Map (e.g. ***Outbound***).
7. In the **Pattern** field enter a called number string to match on. In this example a pattern to match on all 1800 numbers is entered → ***^sip:1800[0-9]{7}***
8. Verify that the **Replace URI** option is checked.
9. Click on the **Add** button. Then click on **Continue** when prompted.

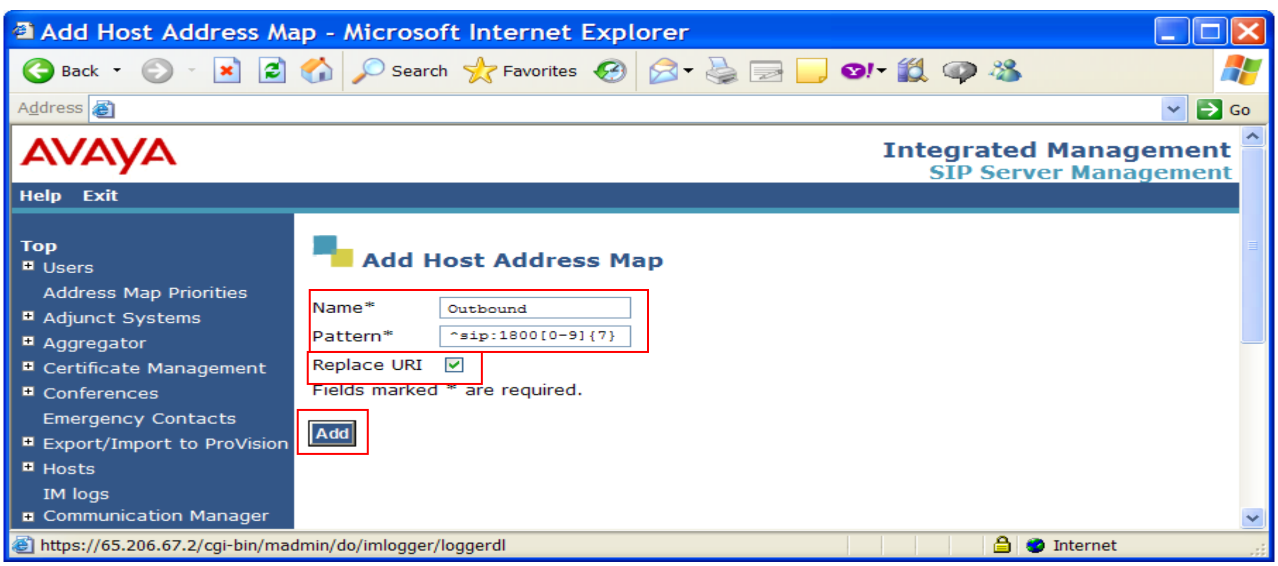

Figure 85: Avaya SIP Enablement Services Add Host Address Map

10. Click on the **Continue** button when prompted. The updated List Host Address Map window will open (**Figure 87**).
11. Select **Add Another Contact** and the **Add Host Contact** window will open (**Figure 86**).

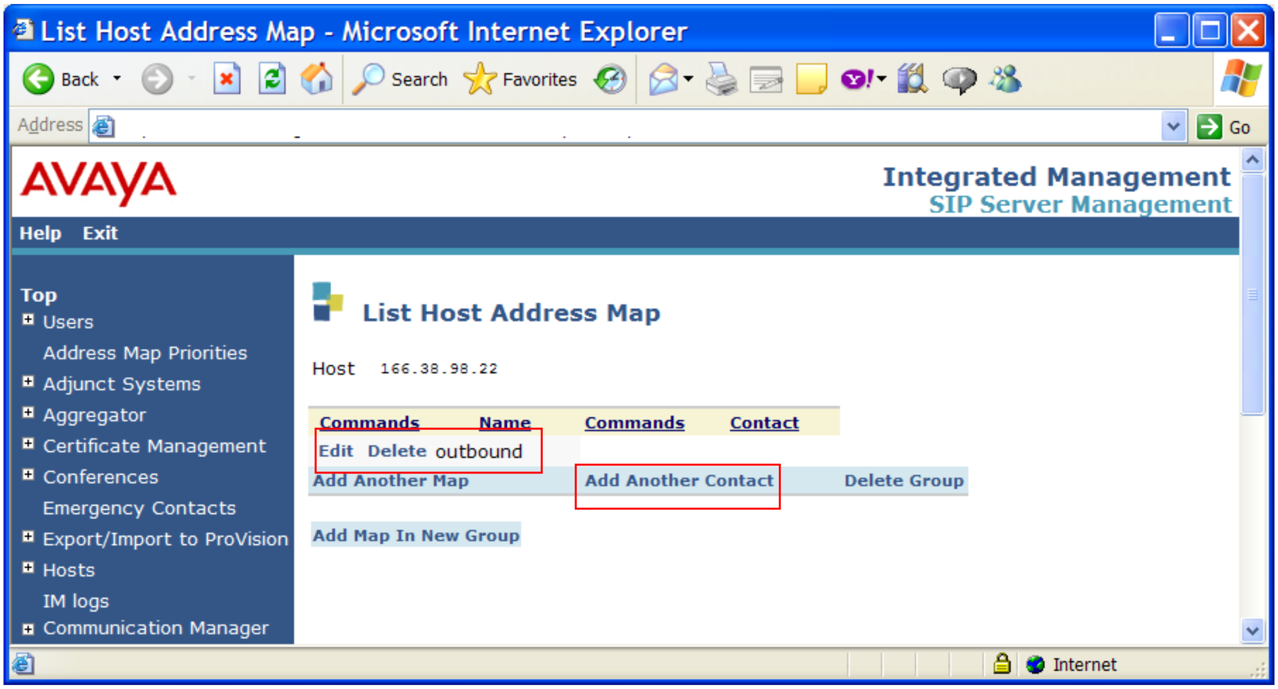

Figure 86: Avaya SIP Enablement Services Updated List Host Address Map

12. In the Contact field (**Figure 87**) enter the destination IP address (e.g. 1566.34.93.140) for this outbound call using the following format: **sip:$(user)@166.34.93.140:5060**

Note – For clarity the default port 5060 was specified as was default transport protocol UDP. A different port and/or transport protocol may be specified if required.

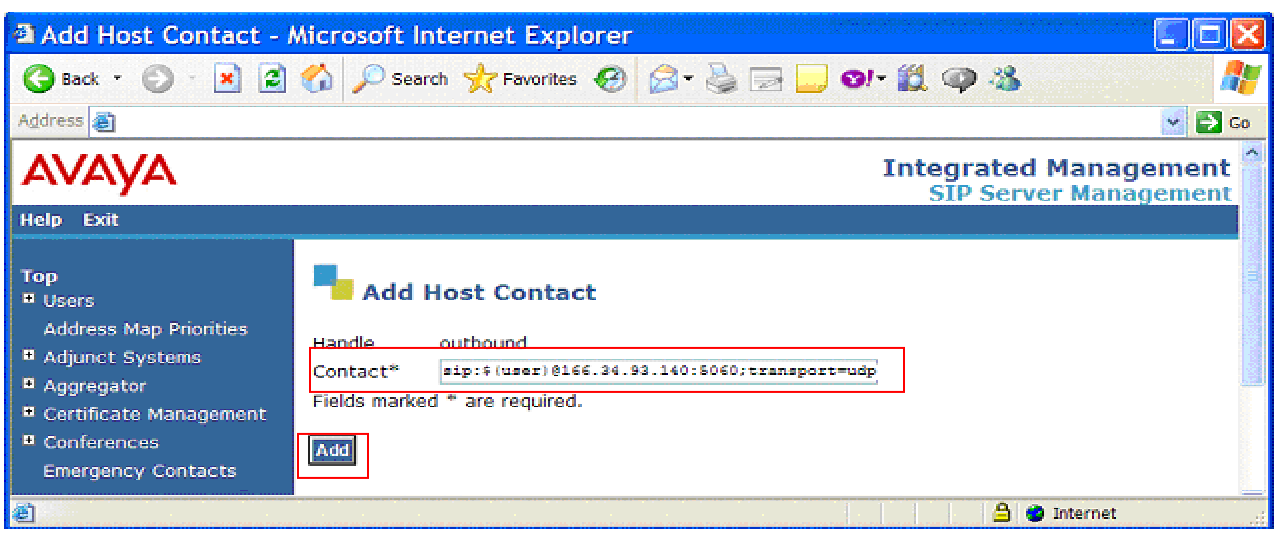

Figure 87: Avaya SIP Enablement Services Add Host Contact

13. Click on **Add** and then **Continue** when prompted. The updated **List Host Address Map** window will open (**Figure 88**). The outbound Host Map is now ready to use.

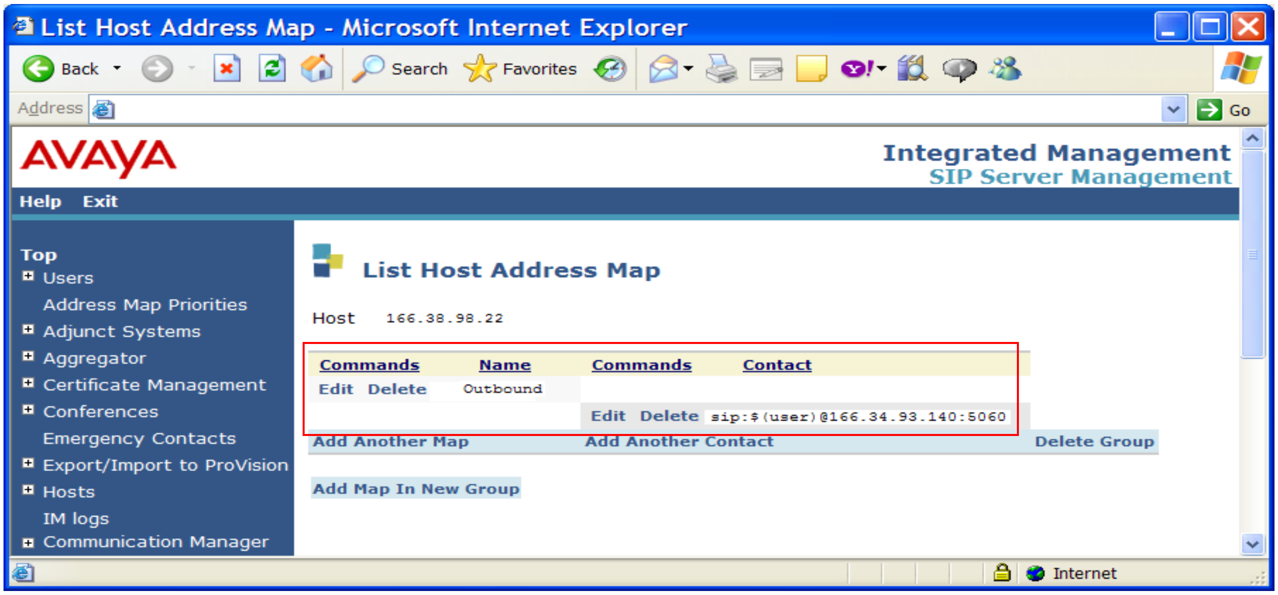

Figure 88: Avaya SIP Enablement Services Updated List Host Address Map

13. APPENDIX C: Avaya SIP Enablement Services TimerB Value and Multiple DNS SRV Response Entries.

As described in **Section 1.5**, the Avaya SIP Enablement Services uses DNS SRV queries to determine where to send the outbound INVITEs during Domain Based Routing. The DNS SRV query response may contain multiple entries in the *Answers* and *Additional Records* fields. When the Avaya SIP Enablement Services receives such a response it will respond by sending an INVITE to the first address listed in these fields. If the first attempt fails, the Avaya SIP Enablement Services tries the next destination in the list until the list is exhausted.

However the Avaya SIP Enablement Services must balance between how long to wait per address attempt versus how long it would take to get to get through all the addresses in the list and thereby delay completion of the call (either successful or unsuccessful). The Avaya SIP Enablement Services uses a calculation of:

Total Wait Time > (**Number of DNS Records** * **TimerB**)

Where:

- **Total Wait Time** - How long the Avaya SIP Enablement Services waits to decide whether to send the INVITE or fail the call. Default value is **30 seconds**.
- **Number of DNS Records** - How many DNS *Answers*/*Additional Records* fields are returned in the DNS SRV query response.
- **TimerB** – The default timer value is **32000 milliseconds** (32 seconds)
- The value of **TimerB** must be modified so that **Number of DNS Records** value times the value of **TimerB** is <u>*less*</u> that the value of **Total Wait Time.**

So if a DNS SRV response contains **4** entries, the maximum value of TimerB must be **7000** (7 seconds).

30 > 4 * 7000

The TimerB value is modified by going to **/usr/impress/sip-server/etc/** in the **ccs.conf** file. Root privileges are required to modify and save the ccs.conf file.

Note – Modification of the TimerB value may cause unexpected results in other call processing. Monitor call processing after changing the timerB value.

Note – **Changing the ccs.conf file must be done with caution as corruption of this file will cause the Avaya SIP Enablement Services to fail.** It is strongly suggested that changes to the ccs.conf file be performed by Avaya support.

Note – There is another method for configuring Domain Base Routing where Avaya Communication Manager uses Avaya SIP Enablement Services Host Maps. This method for configuring Domain Base Routing may be used as an alternative to modifying TimerB. See reference **[11]**

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.