Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the Verizon Business VoIP Service and an Avaya SIP Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure SIP trunking between the Verizon Business VoIP Service and an Avaya SIP telephony solution consisting of Avaya Communication Manager, Avaya SIP Enablement Services, and various Avaya telephones.

The Verizon Business VoIP Service with IP trunking used 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.

Verizon Business is a member of the Avaya Developer*Connection* Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions.

1. Introduction

These Application Notes describe the steps for configuring Session Initiation Protocol (SIP) trunking between the Verizon Business VoIP Service and an Avaya SIP telephony solution consisting of Avaya SIP Enablement Services (SES), Avaya Communication Manager and various Avaya telephony endpoints. These endpoints included IP telephones (using SIP and H.323 protocols) and analog and digital phones.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [7] is the primary specification governing this protocol. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network service offered by Verizon Business. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

SIP trunking to a service provider such as Verizon Business allows any Avaya telephony endpoint to place and receive calls with Public Switched Telephone Network (PTSN) endpoints over a Verizon Business provided Internet Protocol (IP) based access facility. This may reduce or eliminate the need for TDM trunks to a local or long distance telephone company.

1.1. Verizon Business VoIP Service - IP Trunking Overview

The Verizon Business VoIP Service with IP trunking used 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.

The service is delivered to the enterprise customer location using either of two Verizon Business access options: Internet Dedicated Access (IDA) or Private IP (PIP). These access options differ with respect to whether public or private IP addressing is used for the Internet access. These access methods may include management of routers and firewalls by Verizon; the managed IDA option was used within these Application Notes.

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 direct-dial calling within the 50 U.S. states. It 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.

Local voice and fax services include unlimited local area calling, plus:

- Directory Assistance
- Enhanced 911 services access
- Primary Directory Listing
- Directory Listing
- Local Number Management
- 900/978 Blocking

A customer's existing local Direct Inward Dialed (DID) numbers can be retained if desired.

Fax and modem transmission are supported over the Verizon Business VoIP Service but are not covered by service level agreements.

1.2. SIP Trunking Configuration

Figure 1 illustrates a sample Avaya SIP telephony solution connected to Verizon Business VoIP Service using SIP trunking. This is the configuration that was used during the Developer*Connection* compliance testing.

The Avaya SIP telephony solution used as a simulated 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 SIP Enablement Services operating on an Avaya S8500B server.
- Avaya 4600 Series IP telephones using the H.323 software bundle.
- Avaya 4600 Series IP telephones using the SIP software bundle.
- Avaya 6400 Series digital and 6200 Series analog telephones.

Although not shown in **Figure 1**, an enterprise customer site will generally also have alternate routes to the PSTN using analog or digital TDM trunks.

In this configuration, the Verizon Business VoIP solution includes managed IP access service. This managed service includes Verizon Business provided and administered Cisco routers and Cisco PIX firewalls. These devices perform Network Address Translation (NAT) and SIP-aware Application Layer Gateway (ALG) functions on the IP packets and SIP messages sent between the Avaya SES and the Verizon Business VoIP gateway. These functions are necessary for SIP trunking to work in this mixed private and public IP address environment. Under this access configuration, Verizon Business is responsible for the configuration of these devices; those details are not included within these Application Notes.

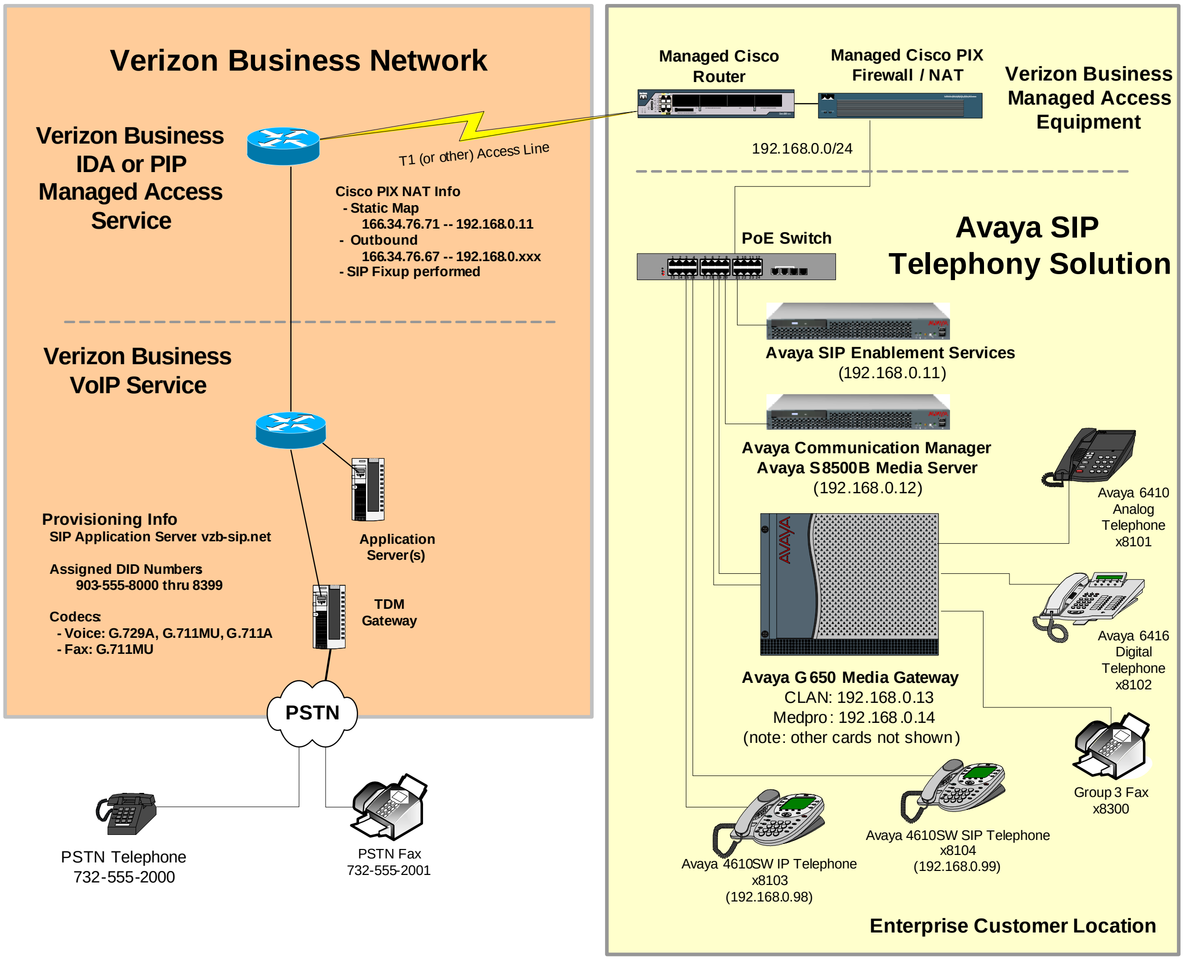

Figure 1: Avaya SIP Telephony Solution using Verizon Business VoIP Service

1.3. Call Flows

To better understand how calls are routed between the PSTN and the enterprise site shown in **Figure 1** using SIP trunks, two call flows are described in this section.

The first call scenario illustrated in **Figure 2** is a PSTN call to the enterprise site terminating on a telephone supported by Avaya Communication Manager.

1. A user on the PSTN dials a Verizon Business VoIP 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 VoIP 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 VoIP Service offers the call to Avaya SES using SIP signaling messages sent over the managed access facility. Note that the assignment of the DID number and the address of the Avaya SES server was previously

established during the ordering and provisioning of the service.

3. Avaya SES routes the call to the Avaya S8500 Media Server running Avaya Communication Manager over a SIP trunk.

4. Avaya Communication Manager terminates the call to the Avaya telephone as shown in **Figure 2**. 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 SES to the SIP telephone (as shown by the 4a arrow.)

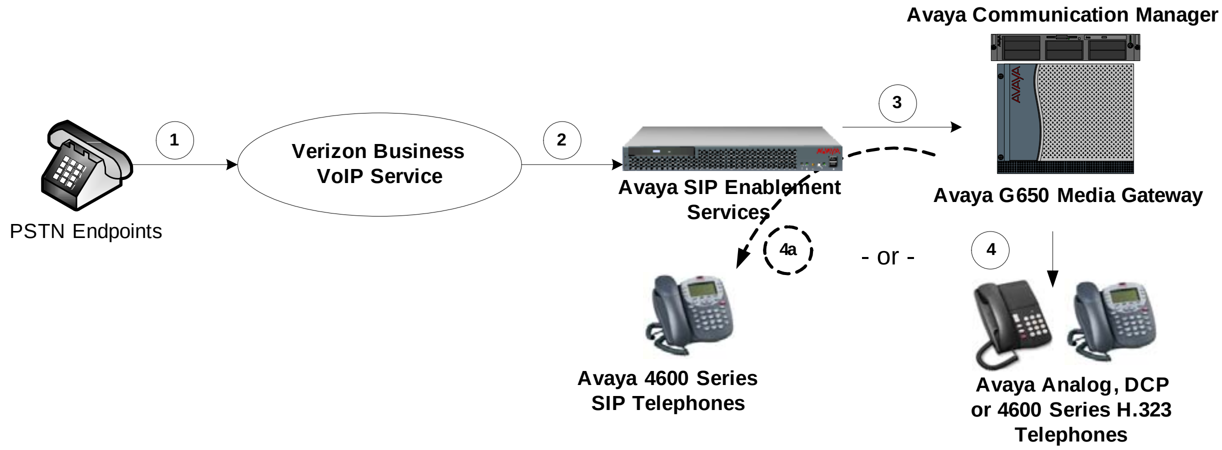

Figure 2: Incoming PSTN Calls to Avaya Communication Manager

The second call scenario illustrated in **Figure 3** is an outgoing call from an Avaya telephone at the enterprise site to the PSTN via the SIP trunk to the Verizon Business VoIP 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 SES (as shown by the 1a arrow) to Avaya Communication Manager.

2. The call request is handled by Avaya Communication Manager 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 SES.

3. Avaya SES routes the call to the Verizon Business VoIP Service.

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

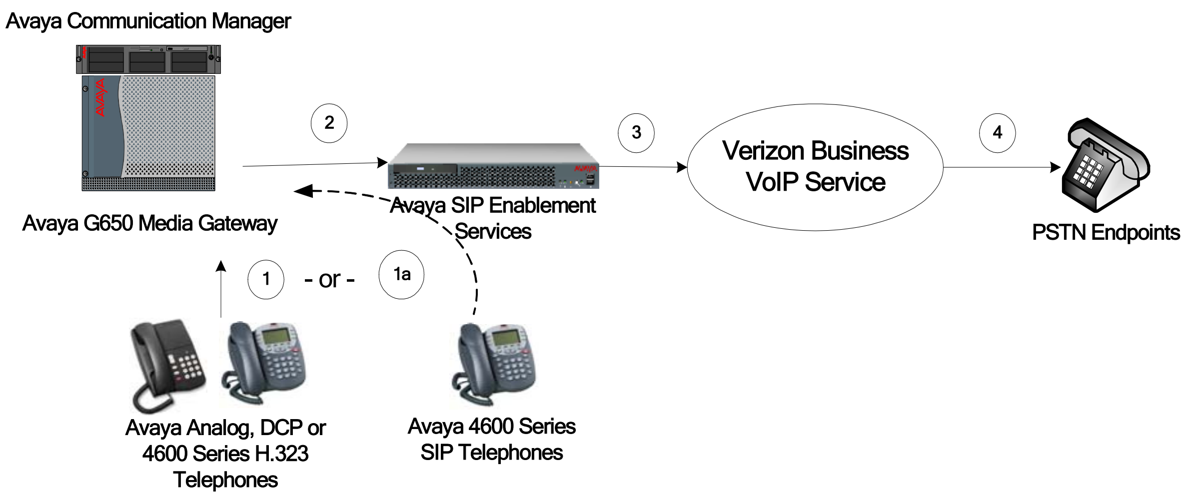

Figure 3: Outgoing Calls from Avaya Communication Manager to the PSTN

Appendix A illustrates examples of the SIP INVITE messages sent on the SIP trunk to begin each call.

1.4. Low Bit Rate Voice and Fax / Modem Calls

The Verizon Business VoIP Service supports both G.729A (low bit rate) codecs in addition to the 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 and modem use; G.711 coding or T.38 fax 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 fax / modem calls may experience degraded performance unless special handing is performed.

Several strategies exist to provide special handling for fax / modem calling:

- Connect fax / modems directly to POTS lines from the Local Exchange Carrier.
- Use DID number assignment and routing techniques to route fax / modems calls via TDM trunks (rather than using VoIP trunks from Verizon Business).
- Assign fax / modem 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 use for fax / modem calls via the Verizon Business VoIP Service.

- Use DID number assignment and outbound routing techniques to force fax / modem lines to use G.711 coding via the Verizon Business VoIP Service.

The use of a dedicated Avaya Media Gateway for fax / modem lines is the most straightforward strategy for fax / modem use via the Verizon Business VoIP 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.

1.5. Dial Plan and Routing Administration for Fax / Modem with the Verizon Business VoIP Service

The current Verizon Business VoIP Service recommends that fax and modem 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 (or modem use).
- Fax / modems 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 SES.
- 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 / modem calls that only use G.711 codecs.
- The fax / modem 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 G.729 codecs.
- 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 SES media server address maps to route fax / modem designated DID numbers to the trunk group supporting G.711 codecs.

2. Equipment and Software Validated

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

Avaya SIP Telephony Solution Components	
Component	**Software Version**
Avaya S8500B Media Server	Avaya Communication Manager 3.1 (R013x.01.0.628.6-11410) (See note below)
Avaya G650 Media Gateway	
TN799DP C-LAN Interface TN2302AP IP Media Processor	HW01 FW017 HW20 FW110
Avaya SIP Enablement Services on S8500B Media Server	SES-3.1.0.0—018.0
Avaya 4620SW SIP Telephones	Release 2.2.2
Avaya 4620SW H.323 IP Telephones	Release 2.3
Avaya 6416 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Verizon Business VoIP Service Components	
Component	**Version**
Verizon VoIP Platform	6.3

Table 1: Equipment and Software Used in the Tested Configuration

The specific configuration above was used for the Verizon Business VoIP Service interoperability testing. Note that this solution is extensible to other Avaya Media Server and Media Gateway platforms running similar versions of Avaya Communication Manager and Avaya SIP Enablement Services.

Note: Use of the G.729A codec with Verizon Business VoIP Service requires Avaya Communication Manager Release 3.1.2 (R013x.01.2.636.1) or later. This release permits the **Telephone Event Payload Type** to be set to ***101***. See section 3.1.6.1 for additional information.

3. Configure Avaya Communication Manager

This section describes the steps for configuring a SIP trunk on Avaya Communication Manager with the necessary signaling and media gateway characteristics for the SIP trunk connection with the Verizon Business VoIP Service.

The SIP trunk configuration also allows a shared trunking solution that supports SIP endpoint devices (such as Avaya 4600 Series SIP telephones) using Avaya Communication Manager in the recommended Avaya Off-PBX Station (OPS) configuration. In this configuration, the SIP endpoints are registered with Avaya SIP Enablement Services but have calling privileges and features managed by Avaya Communication Manager. Avaya Communication Manager acts as a back-to-back SIP user agent for the SIP endpoint when a call involves a SIP trunk to the Verizon Business VoIP Service.

SIP trunking to the Verizon Business VoIP Service works with any combination of Avaya analog, digital, H.323 and SIP telephones. The use of SIP endpoints is optional and the steps discussed in sections 3.2 and 4.2 may be omitted if SIP endpoints are not used.

In the Avaya SIP trunk architecture, the Avaya SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to the Verizon Business VoIP Service. There is no direct SIP signaling path between the Verizon Business VoIP Service and Avaya Communication Manager or Avaya SIP endpoints.

For incoming calls, the Avaya SES uses media server address maps to direct the incoming SIP messages to the appropriate Avaya Communication Manager. Once the message arrives at the 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 and class of service restrictions. Once Avaya Communication Manager selects a SIP trunk, the SIP signaling is routed to the Avaya SES. Within the Avaya SES, host address maps direct the outbound SIP messages to the Verizon Business SIP Application Server.

The dial plan for the configuration described in these Application Notes consists of 1+10-digit dialing for local and long-distance calls over the PSTN. In addition, operator calls (0), directory assistance calls (411) and international calls (011+Country Code) are also supported. Avaya Communication Manager routes all external calls using Automatic Route Selection.

The Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The initial installation and configuration of the S8500B Media Server, G650 Media Gateway and associated circuit packs are presumed to have been previously completed and are not discussed within these Application Notes.

3.1. SIP Trunk Configuration

3.1.1 Verify System Capacity

Using the SAT, verify that sufficient SIP Trunk and Off-PBX Telephone capacities exist by displaying the **System-Parameters Customer-Options** form shown in **Figure 4**. The Avaya Communication Manager license file controls the customer options. Contact an authorized Avaya sales representative for assistance if insufficient capacity exists or a required feature is not enabled.

On Page 1 of the **System-Parameters Customer-Options** form, verify that the number of **OPS** stations available is sufficient for the total number of SIP endpoints to be used. The **Maximum Off-PBX Telephones** field indicates the maximum number available in the system. The **USED** column indicates the number of telephone licenses currently assigned. The difference between the two represents the additional number of SIP endpoints that can be added.

```
display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES
     G3 Version: V13
       Location: 1                            RFA System ID (SID): 1
       Platform: 8                            RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 44000 86
                                      Maximum Stations: 36000 36
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 0     0
                    Maximum Off-PBX Telephones -   OPS: 100   17
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

Figure 4: System-Parameters Customer-Options Form – Page 1

On Page 2, verify that the number of available **SIP Trunks** is sufficient for the combination of trunks to the Verizon Business VoIP Service, trunks for SIP endpoints and any other SIP trunking applications.

Note that each call from a non-SIP endpoint to the Verizon Business VoIP Service uses one SIP trunk for the duration of the call. However, due to the SIP OPS configuration, each SIP endpoint on a call with the Verizon Business VoIP Service requires two SIP trunks for the duration of the call.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 0     0
           Maximum Concurrently Registered IP Stations: 100   2
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     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: 100   50

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 1     1
                    Maximum G250/G350/G700 VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 2     1
          Maximum TN2602 Boards with 320 VoIP Channels: 2     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

Figure 5: System-Parameters Customer-Options Form – Page 2

3.1.2 Determine Node Names

In the **IP Node Names** form, view (or assign) the node names to be used in this configuration using the **`change node-names ip`** command.

`SES-Edge` and ***`192.168.0.11`*** are used below as the **Name** and **IP Address** of the Avaya SIP Enablement Services server at the enterprise site.

`CLAN-1A03` and ***`192.168.0.13`*** below are the **Name** and **IP Address** assigned to the TN799DP C-LAN card used for the SIP signaling connection to the SES. In other media gateways such as the Avaya G350 Media Gateway, the processor address (procr) may be used as the SIP signaling interface instead of the C-LAN.

```
change node-names ip                                               Page   1 of   1

                                 IP NODE NAMES
    Name              IP Address           Name                 IP Address
CLAN-1A03           192.168.0  .13                                .   .   .
default             0   .0   .0   .0                              .   .   .
procr               192.168.0  .12                                .   .   .
medpro-1a04         192.168.0  .14                                .   .   .
SES-Edge            192.168.0  .11                                .   .   .
val-1a05            192.168.0  .15                                .   .   .
                       .   .   .                                  .   .   .
```

Figure 6: IP Nodes Names Form

3.1.3 Define IP Codec Sets

As discussed in sections 1.4 and 1.5, two distinct codec sets are necessary to support differences in voice vs. fax / modem calls when G.729A low bit rate coding is desired as the default for voice calls.

- Define a codec set (**Codec Set 2** below) to support voice calls with the Verizon Business VoIP Service. G.729A, G.711MU and G.711A are the codecs to be used.
- Define a codec set (**Codec Set 3** below) for fax and modem use. G.711MU and G.711A are the codecs to be used.

Use the **`change ip-codec-set`** command (**Figure 7**) to define these two codec sets. On Page 2 of each **ip-codec-set** form, the **Fax**, **Modem** and **TTD/TTY** modes must be ***`off`***.

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

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n


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

```
 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
 3:
 4:
```

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n


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

Figure 7: IP Codec Set Forms for Voice and Fax / Modem Use

3.1.4 Verify Near End IP Network Region

These Application Notes assume that **IP Network Region** 1 (the normal default) is used for the G650 Media Gateway and the C-LAN card chosen for SIP signaling. This is the near-end network region used by IP telephones and interfaces not explicitly mapped to other regions. Use the **`display cabinet`** and **`display ip-interface`** commands to verify the network region assignment of the G650 media gateway and C-LAN card, respectively.

```
display cabinet 1                                                 Page   1 of   1
                                    CABINET
 CABINET DESCRIPTION
                  Cabinet: 1
           Cabinet Layout: G650-rack-mount-stack
             Cabinet Type: expansion-portnetwork


                 Location: 1         IP Network Region: 1

 Rack:               Room:              Floor:             Building:

 CARRIER DESCRIPTION
   Carrier       Carrier Type        Number

      E        not-used             PN  01
      D        not-used             PN  01
      C        not-used             PN  01
      B        not-used             PN  01
      A        G650-port            PN  01
```

Figure 8: Determining IP Network Region for G650 Cabinet

```
display ip-interface 01a03                                        Page   1 of   1
                                  IP INTERFACES


                  Type: C-LAN
                  Slot: 01A03
           Code/Suffix: TN799  D
             Node Name: CLAN-1A03
            IP Address: 192.168.0  .13
           Subnet Mask: 255.255.255.0                             Link: 1
       Gateway Address: 192.168.0  .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? y
```

Figure 9: Determining IP Network Region for C-LAN

3.1.5 Define IP Network Regions

The **IP Network Region** form specifies the parameters used by the SIP trunk groups serving the Avaya SES proxy (used to reach the Verizon Business VoIP Service and any optional SIP endpoints). These parameters also apply to any other elements (such as H.323 telephones, media processor cards, etc.) also assigned to the region.

As discussed in sections 1.4 and 1.5, two additional distinct IP Network Regions are necessary to distinguish the different codec requirements of low bit rate voice vs. fax / modem calls. Below, **IP Network Region 2** is defined for voice calls use, and **IP Network Region 3** is defined for the fax / modem call use.

On Page 1 of each **IP Network Region** form:

- Configure the **Authoritative Domain** field to match the **SIP Domain** name configured on the Avaya SES **System Properties** field shown in **Figure 36**. In this configuration, the domain name is ***customer.com***.
- By default, **IP-IP Direct Audio** (a.k.a., shuffling) is set to ***yes*** to allow audio traffic to be sent directly between SIP endpoints to reduce the use of media resources such as the TN2302AP IP Media Processor (MedPro) card.
- Set the **Codec Set** to the number of the **IP Codec Set** to be used for the corresponding calls within the IP Network Region.
- The **IP Network Region** form is also used to set the packet parameters that provide priority treatment for signaling and audio packets over other data traffic on Verizon Business access facilities. These parameters need to be aligned with the specific values provided by Verizon Business.

In Page 3 of each **IP Network Region** form:

- Define the **Codec Set** used for inter-region communications. **Codec Set *2*** is entered for communications between region ***2*** and ***1***. **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.

The specific **IP Network Region** forms used for voice calls and fax / modem calls within these Application Notes are shown in **Figure 10** and **Figure 11**, respectively.

```
change ip-network-region 2                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location:          Authoritative Domain: customer.com
    Name: VZB Voice
                                Intra-region IP-IP Direct Audio: yes
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: yes
      Codec Set: 2                         IP Audio Hairpinning? y
   UDP Port Min: 2048
   UDP Port Max: 3028                    RTCP Reporting Enabled? y
DIFFSERV/TOS PARAMETERS          RTCP MONITOR SERVER PARAMETERS
 Call Control PHB Value: 46       Use Default Server Parameters? y
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6      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
```

```
change ip-network-region 2                                      Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   Total           Video                         Dyn
 rgn rgn  set   WAN  WAN-BW-limits  WAN-BW-limits  Intervening-regions  CAC IGAR
 2   1    2     y        :NoLimit       :NoLimit                             n
 2   2    2
 2   3    3     y        :NoLimit       :NoLimit                             n
 2   4
 2   5
 2   6
```

Figure 10: IP Network Region Form for Voice Calls

```
change ip-network-region 3                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location:         Authoritative Domain: customer.com
    Name: VZB Fax / modem
                                Intra-region IP-IP Direct Audio: yes
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: yes
      Codec Set: 3                         IP Audio Hairpinning? y
   UDP Port Min: 2048
   UDP Port Max: 3028                    RTCP Reporting Enabled? y
DIFFSERV/TOS PARAMETERS          RTCP MONITOR SERVER PARAMETERS
 Call Control PHB Value: 46       Use Default Server Parameters? y
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6      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
```

```
change ip-network-region 3                                    Page   3 of  19

                  Inter Network Region Connection Management

 src dst codec direct   Total           Video                          Dyn
 rgn rgn  set   WAN  WAN-BW-limits  WAN-BW-limits  Intervening-regions  CAC IGAR
 3   1    3     y        :NoLimit       :NoLimit                              n
 3   2    3     y        :NoLimit       :NoLimit                              n
 3   3    3
 3   4
 3   5
 3   6
```

Figure 11: IP Network Region Form for Fax / Modem Calls

3.1.6 Define SIP Trunk Groups

As discussed in sections 1.4 and 1.5, two distinct SIP trunk groups are necessary to support differences in voice vs. fax / modem calling. For each trunk group it is necessary to define a separate SIP signaling group as well a SIP trunk group.

3.1.6.1 Establish the Voice Trunk Group

Create a signaling group and a trunk group (**Signaling Group 2** and **Trunk Group 2** below**)** to support the voice calling with the Verizon Business VoIP Service.

Using the **`add signaling-group`** command, configure the **Signaling Group** form shown below 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 SES, not the transport method used to the Verizon Business VoIP Service.
- Specify the Avaya C-LAN (node name ***CLAN-1A03***) and the Avaya SES (node name ***SES-Edge***) 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 **Figure 6**. Note for smaller media server platforms, the near end of the SIP signaling group may be the S8300 Media Server processor (procr) rather than the C-LAN.
- Specify ***5061*** in the **Near-end Listen Port** and **Far-end Listen Port** fields.
- Enter the value ***2*** into the **Far-end Network Region** field. This value is for the **IP Network Region** defined in **Figure 10**. Note that since this **Far-end Network Region** value is different from the **Network Region** associated with ***CLAN-1A03***, codec set ***2*** will be selected for the inter-region connectivity.
- Enter the domain name of Avaya SES in the **Far-end Domain** field. In this configuration, the domain name is ***customer.com***. This domain will appear in the Uniform Resource Identifier (URI) of the SIP "To" address in the INVITE message. Mis-configuring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- The **Direct IP-IP Audio Connections** field should be set to ***y*** to allow RTP voice paths to be established directly between IP/SIP telephones and the Verizon Business VoIP Service. When compatible conditions exist, this allows the voice packets to follow a direct path between the telephones and the network edge that may reduce media processor (MedPro) card 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 transmissions using RFC 2833 [8].
- The default values for the other fields may be used.

```
add signaling-group 2                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
                        Transport Method: tls


   Near-end Node Name: CLAN-1A03            Far-end Node Name: SES-Edge
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 2
       Far-end Domain: customer.com

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? y
 Session Establishment Timer(min): 120
```

Figure 12: Signaling Group Form for Voice Use

Configure the **Trunk Group** form as shown in **Figure 13** through **Figure 15** using the **`add trunk-group`** command. **Trunk Group** number 2 is defined for voice use with Verizon Business VoIP Service in these Application Notes.

On Page 1 of the **Trunk Group** form:

- Set the **Group Type** field to ***sip***.
- Choose a descriptive **Group Name** such as ***SIP Voice***.
- Specify an available trunk access code (**TAC**) such as ***102***.
- Set the **Service Type** field to ***tie***.
- Enter ***2*** as the **Signaling Group** number. This value was previously determined during the **Signaling Group** configuration specified in **Figure 12**.
- Specify the **Number of Members** supported by this SIP trunk group.

  Note that one trunk member is required for each call between a non-SIP endpoint and the Verizon Business VoIP Service. Calls involving a SIP endpoint and the Verizon Business VoIP Service will use two trunk members for the duration of the call.

```
add trunk-group 2                                                    Page   1 of  19
                                TRUNK GROUP

Group Number: 2                   Group Type: sip           CDR Reports: y
  Group Name: SIP Voice                  COR: 1        TN: 1        TAC: 102
   Direction: two-way       Outgoing Display? n
 Dial Access? n               Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                                        Signaling Group: 2
                                                      Number of Members: 15
TRUNK PARAMETERS

     Unicode Name? y

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                                   Digital Loss Group: 18
```

Figure 13: Trunk Group Form for Voice Use – Page 1

On Page 2 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 2                                                    Page   2 of  19
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y


                     Numbering Format: public

                                                Replace Unavailable Numbers? n
```

Figure 14: Trunk Group Form for Voice Use – Page 2

On Page 4 of the **Trunk Group** form:

- Set the **Telephone Event Payload Type** to ***101***.

  This sets the RFC 2833 [8] telephone event value sent by the Avaya Communication Manager on outbound calls to 101.

Note: The **Telephone Event Payload Type** administration is required when the G.729A codec is used with the Verizon Business VoIP Service. This is because the Verizon Business VoIP Service does not use RFC 2833 to send DTMF tones when the Avaya default **Telephone Event Payload Type** value (127) is used. DTMF tones may not be successfully received at the far-end when the G.729A codec is used without RFC 2833.

The **Telephone Event Payload Type** administration option is not present in Avaya Communication Manager Release 3.1 (R013x.01.0.628.6-11410). Avaya Communication Manager Release 3.1.2 (R013x.01.2.636.1) is the minimum version that permits the administration of the **Telephone Event Payload Type** value. Contact an authorized Avaya representative if this release is needed.

```
add trunk-group 2                                           Page   4 of  19

                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
           Telephone Event Payload Type: 101
```

Figure 15: Trunk Group Form for Voice Use – Page 4

3.1.6.2 Establish the Fax / Modem (G.711) Trunk Group

Create a signaling group and trunk group (**Signaling Group 3** and **Trunk Group 3** below) to support the fax / modem calling with the Verizon Business VoIP Service.

The configuration information is similar to that used for the voice trunk group above, except for the definition of the signaling group. Only the differences are outlined below.

Using the **add signaling-group** command, configure the fax / modem signaling group form similar to the voice signaling group described above. The notable differences are:

- Specify ***5062*** as the **Near-end Listen Port** and ***5061*** as the **Far-end Listen Port**. This permits the same C-LAN interface to be used for multiple incoming SIP trunk groups having different network region characteristics.
- Specify ***3*** as the value for **Far-end Network Region** field as defined in **Figure 11**.

Using the **`add trunk-group`** command, configure the fax / modem trunk group forms similar to the voice trunk group described above. The notable difference is:

- Specify ***3*** as the value for the **Signaling Group** to use the signaling group defined for fax / modem use in section 3.1.6.2**.**

The completed **Signaling Group** and **Trunk Group** forms for the fax / modem trunks are illustrated in **Figure 16** and **Figure 17** below.

```
add signaling-group 3                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 3              Group Type: sip
                        Transport Method: tls


   Near-end Node Name: CLAN-1A03            Far-end Node Name: SES-Edge
 Near-end Listen Port: 5062               Far-end Listen Port: 5061
                                       Far-end Network Region: 3
       Far-end Domain: customer.com

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? y
 Session Establishment Timer(min): 120
```

Figure 16: Signaling Group Form for Fax / Modem Use

```
add trunk-group 3                                            Page   1 of  19
                                TRUNK GROUP

Group Number: 3                   Group Type: sip          CDR Reports: y
  Group Name: SIP Fax / modem              COR: 1        TN: 1        TAC: 103
   Direction: two-way       Outgoing Display? n
 Dial Access? n               Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                                      Signaling Group: 3
                                                    Number of Members: 10
TRUNK PARAMETERS

     Unicode Name? y

                                             Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
```

```
add trunk-group 3                                            Page   2 of  19
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y


                     Numbering Format: public

                                               Replace Unavailable Numbers? n
```

```
add trunk-group 3                                            Page   4 of  19

                              PROTOCOL VARIATIONS

                           Mark Users as Phone? n
             Prepend '+' to Calling Number? n
       Send Transferring Party Information? n
            Telephone Event Payload Type: 101
```

Figure 17: Trunk Group Form for Fax / Modem Use

3.1.7 Configure Calling Party Number Information

Configure the **public-unknown-numbering** form to send the correct calling party number to the Verizon Business VoIP Service for calls originating from Avaya Communication Manager. The calling party number (sent in the SIP “From” header) must match the value expected by the Verizon Business VoIP Service.

In these Application Notes, the extensions on Avaya Communication Manager use a 4 digit dialing plan with extensions between 8000 and 8399 that match the last four digits of the DID numbers assigned by the Verizon Business VoIP Service. In these Application Notes, the Verizon Business requirement is that the calling party number be 4 digits long between 8000 and 8399. Note this requirement may vary with other numbering plans provided by Verizon Business.

Figure 18 shows the use of the **`change public-unknown-numbering`** command to send the correct calling party number. The entries below indicate that 4-digit extension numbers beginning with 8 will send the corresponding digits via trunk group 2 and 3.

```
change public-unknown-numbering 7                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                   Total                                   Total
Ext Ext    Trk       CPN            CPN Ext Ext    Trk       CPN            CPN
Len Code   Grp(s)    Prefix         Len Len Code   Grp(s)    Prefix         Len

 4   8     2                        4
 4   8     3                        4
```

Figure 18: Public/Unknown Numbering Form

3.1.8 Configure Call Routing

3.1.8.1 Outbound Calls

In these Application Notes, the Automatic Route Selection feature is used to route calls via the SIP trunk to the Verizon Business VoIP Service, which in turn completes the calls to the PTSN destination.

Use the **`change dialplan analysis`** command to add ***9*** as a feature access code (**`fac`**).

```
change dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                                           Percent Full:    3

      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        1       3    dac
        7       4    ext
        8       4    ext
        9       1    fac
        *       3    fac
        #       3    fac
```

Figure 19: Dialplan Analysis Form

Use the **change feature-access-codes** command to specify ***9*** as the access code for external dialing.

```
change feature-access-codes                                Page   1 of   7
                               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: *03
                       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: *10    All: *11    Deactivation: #10
                         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:
                   Contact Closure   Open Code:          Close Code:
                   Contact Closure  Pulse Code:
```

Figure 20: Feature-Access-Codes Form

Now use the **change ars analysis** command to configure the route pattern selection rule based upon the number dialed following the dialed digit “9”. In this sample configuration, the PSTN numbers dialed are all in the form 1AAANXXXXXX. If the area code (AAA) is 732, the call is to be routed to a route pattern containing the SIP trunk groups used for the Verizon Business VoIP Service. Note that further administration of ARS is beyond the scope of these Application Notes but is discussed in References [1] and [2].

```
change ars analysis 173                                        Page   1 of   2
                              ARS DIGIT ANALYSIS TABLE
                                    Location:  all          Percent Full:    3

         Dialed            Total      Route     Call   Node  ANI
         String           Min  Max  Pattern     Type   Num   Reqd
    173                   11   11    1          fnpa         n
```

Figure 21: ARS Analysis Form

Use the **change route-pattern** command to define the SIP trunk groups included in the route pattern that ARS selects.

In these Application Notes, the trunk group order and Facility Restriction Level (**FRL**) is used to force the fax / modem calls to only use trunk group (**Grp No**) ***3***. The **FRL** value assigned to the voice trunk group 2 requires extensions placing voice calls to have a **FRL** value of ***2*** or greater. Fax / modem extensions are assigned an **FRL** of ***1*** to cause the route-pattern to skip over trunk group (**Grp No**) ***2*** when routing a fax call.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: VBZ VoIP
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 2    2       1                                                       n   user
 2: 3    1       1                                                       n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 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                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

Figure 22: Route Pattern Form

Use the **change cor** command to define the rules necessary to prevent the fax / modem calls from using the voice trunk group. The **FRL** value of ***2*** for **COR Number 1** below would allow extensions assigned this COR to use any trunk group (**Grp No**) in **Route Pattern *1***.

```
change cor 1                                                    Page   1 of  22
                              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
 Partitioned Group Number: 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 23: Class of Restriction for Voice Use

The **FRL** value of ***1*** for **COR Number *2*** below would allow extensions assigned with this COR to only use the fax / modem trunk group (**Grp No *3***) in **Route Pattern *1***.

```
change cor 2                                                Page   1 of  22
                              CLASS OF RESTRICTION

               COR Number: 2
          COR Description: Fax / modem Calls

                      FRL: 1                                 APLT? y
  Can Be Service Observed? n           Calling Party Restriction: none
Can Be A Service Observer? n            Called Party Restriction: none
 Partitioned Group Number: 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 24: Class of Restriction for Fax / Modem Use

Using the **`change station`** command, assign each voice extension the **COR** with **FRL** value of ***2*** that was defined above. Note that these extensions must be in the voice extension numbering plan of 8000 – 8299 as defined for these Application Notes.

```
change station 8103                                         Page   1 of   6
                                     STATION

Extension: 8103                           Lock Messages? n          BCC: 0
     Type: 6416D+                         Security Code:             TN: 1
     Port: 001V301                      Coverage Path 1:            COR: 1
     Name: Voice Extension              Coverage Path 2:            COS: 1
                                        Hunt-to Station:

STATION OPTIONS
             Loss Group: 2              Personalized Ringing Pattern: 1
            Data Option: none                       Message Lamp Ext: 8103
            Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: english                    Expansion Module? n
```

Figure 25: Station Administration for Voice Use

In a similar manner, assign each fax / modem extension the **COR** with **FRL** value of ***2*** that was defined above. Note that these extensions must be in the fax / modem extension numbering plan of 8300 – 8399 as defined for these Application Notes.

```
change station 8300                                           Page   1 of   3
                                     STATION

Extension: 8300                            Lock Messages? n          BCC: 0
     Type: 2500                            Security Code:             TN: 1
     Port: 001V401                       Coverage Path 1:            COR: 2
     Name: Fax Machine                   Coverage Path 2:            COS: 1
                                         Hunt-to Station:          Tests? y

STATION OPTIONS
              Loss Group: 1                 Message Waiting Indicator: none
    Off Premises Station? n
```

Figure 26: Station Administration for Fax / Modem Use

Use the **`change locations`** command to designate the SIP trunk route pattern (route pattern ***1*** below) in the **Proxy Sel. Rte. Pat.** field. This provides support for features such as call transfer and URI dialing.

```
change locations                                              Page   1 of   1
                                   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                                   1
```

Figure 27: Location Form Administration

3.1.8.2 Incoming Calls

This step configures the mapping of incoming DID numbers to the proper extensions.

In these Application Notes, the Verizon Business VoIP Service sends exactly four digits in the incoming INVITE message. In general, those digits exactly match the intended extension number and digit manipulation treatment is not necessary.

However, in some cases, the incoming digits received may not match the intended extension. In those cases, incoming call handling treatment is used to modify the number in order to reach the desired extension. The example below illustrates the technique to assign the incoming DID number 8299 to extension 7000.

Use the **`change inc-call-handling-trmt trunk-group`** command to administer this assignment. For this example shown in **Figure 28**:

- Enter ***4*** into the **Called Len** field to match the length of the incoming digits.
- Enter ***8299*** into the **Called Number** field as the digit pattern to be matched.
- Enter ***4*** into the **Del** field as the number of digits that should be deleted from the end of the incoming digits.

- Enter ***7000*** into the **Insert** field as the digits that should be appended to the end of the adjusted incoming digit string (following the deletion step above).

```
change inc-call-handling-trmt trunk-group 2                      Page   1 of   3

                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called        Del  Insert
 Feature        Len      Number
 tie             4       8299           4     7000
```

Figure 28: Incoming Call Handling Treatment – Simple Extension Mapping

3.1.9 Save Avaya Communication Manager Changes

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

3.2. SIP Endpoint Configuration

This section describes the administration of SIP telephones and requires that the preceding SIP trunk configuration be completed beforehand. SIP telephones are optional and not required to use the Verizon Business VoIP Service.

3.2.1 Add a Station for the SIP Endpoint

The first step in adding an Off-PBX station for Avaya SIP telephones registered with Avaya SIP Enablement Services is to assign a station as shown in **Figure 29**.

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

- Set the station **Type** at the default ***6408D+*** value (Note this is the Avaya recommended best practice that will prevent an alarm warning that occurs when 4600 Series IP Telephones are entered).
- Enter ***X*** in the **Port** field to indicate station administration without port hardware.
- Enter a **Name** for the station that will be displayed.
- The **Security Code** is left blank for SIP OPS extensions. (Note: SIP phone passwords are administered within the SES **Add User** screen.)
- Set the **COR** is set to ***1***, the value defined previously for voice extensions.

Configure the remaining fields per normal station administration (beyond the scope of these Application Notes).

```
add station 8104                                            Page   1 of   4
                                     STATION

Extension: 8104                         Lock Messages? n          BCC: 0
     Type: 6408D+                       Security Code:             TN: 1
     Port: X                          Coverage Path 1:            COR: 1
     Name: SIP Phone                  Coverage Path 2:            COS: 1
                                      Hunt-to Station:
STATION OPTIONS
              Loss Group: 2           Personalized Ringing Pattern: 1
             Data Module? n                       Message Lamp Ext: 8104
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: English
                                                 Media Complex Ext:
                                                      IP SoftPhone? n
```

Figure 29: Station Administration – Page 1

On Page 2 of the **Station** form:

- Set the **Restrict Last Appearance** value to ***n***.

  Note: Setting the **Restrict Last Appearance** value to ***y*** reserves the last call appearance for outbound calls. Certain SIP conference and transfer features will not function properly if a third appearance is not available for incoming calls.

```
add station 8104                                            Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe               Auto Select Any Idle Appearance? n
          LWC Activation? y                          Coverage Msg Retrieval? y
  LWC Log External Calls? n                                     Auto Answer: none
             CDR Privacy? n                                Data Restriction? n
   Redirect Notification? y                      Idle Appearance Preference? n
 Per Button Ring Control? n                    Bridged Idle Line Preference? n
   Bridged Call Alerting? n                        Restrict Last Appearance? n
  Active Station Ringing: single              Conf/Trans on Primary Appearance? n

        H.320 Conversion? n          Per Station CPN - Send Calling Number? y
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type: qsig-mwi               Display Client Redirection? n
                                                Select Last Used Appearance? n
                                                  Coverage After Forwarding? s

 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
 Emergency Location Ext: 8104      Always Use? n      IP Audio Hairpinning? y
```

Figure 30: Station Administration – Page 2

On Page 3 of the **Station** form, configure at least 3 call appearances (**call-appr**) for the SIP telephone as shown in **Figure 31**.

```
add station 8104                                              Page   3 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:

BUTTON ASSIGNMENTS
 1: call-appr                           5:
 2: call-appr                           6:
 3: call-appr                           7:
 4:                                     8:
```

Figure 31: Station Administration – Page 3

The same number of call appearances should be configured on the SIP telephone 46xxsettings.txt file (which is beyond the scope of these Application Notes). The parameters to administer call appearances (and many other settings) are described in Reference [1].

3.2.2 Configure Off-PBX Telephone Station Mapping

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

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 32**:

- In the **Station Extension** field, enter the extension number from the station defined in **Figure 29**.
- 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 SES **Add User** form shown in **Figure 59**.
- Set the **Trunk Selection** field to ***2***, which is the number assigned to the SIP trunk group used to route the call to the SIP station. This trunk group number was previous defined in **Figure 13**.
- Set the **Configuration Set** value. In these Application Notes, **Configuration Set *1*** uses the default values of the Off-PBX Telephone Configuration Set form (not shown).

```
change off-pbx-telephone station-mapping 8104                     Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station      Application   Dial   Phone Number      Trunk       Configuration
 Extension                  Prefix                   Selection   Set
 8104           OPS              - 8104               2          1
```

Figure 32: Stations with Off-PBX Telephone Integration – Page 1

On Page 2, 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. Accept the default values for the other fields.

```
change off-pbx-telephone station-mapping 8104                     Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Call       Mapping      Calls       Bridged
 Extension      Limit      Mode         Allowed     Calls
 8104           3          both         all         both
```

Figure 33: Stations with Off-PBX Telephone Integration – Page 2

3.2.3 Save Avaya Communication Manager Changes

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

4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services. Avaya SIP Enablement Services is configured via an Internet browser using **SIP Server Management** screens. It is assumed that Avaya SIP Enablement Services software and the license file have already been installed. During the software installation, the install script is run on the Linux shell of the server to specify the IP network properties of the server including DNS server address(es). For additional information on these installation tasks, refer to Reference [4] .

This section is divided into two parts: section 4.1 provides the steps necessary to configure SIP trunking to the Verizon Business VoIP Service. Section 4.2 provides the steps necessary to complete the administration for optional SIP endpoints in this Avaya SIP telephony solution. SIP endpoints are not necessary to use the Verizon Business VoIP Service.

4.1. SIP Trunking to the Verizon Business VoIP Service

4.1.1 Log in to Avaya SIP Enablement Services

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

Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main page as shown in **Figure 34**.

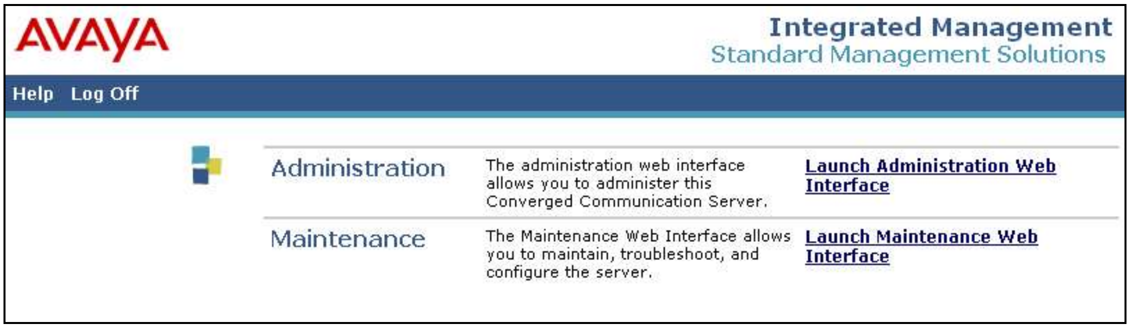

Figure 34: Avaya SES Main Page

The SES administration home page shown in **Figure 35** is displayed.

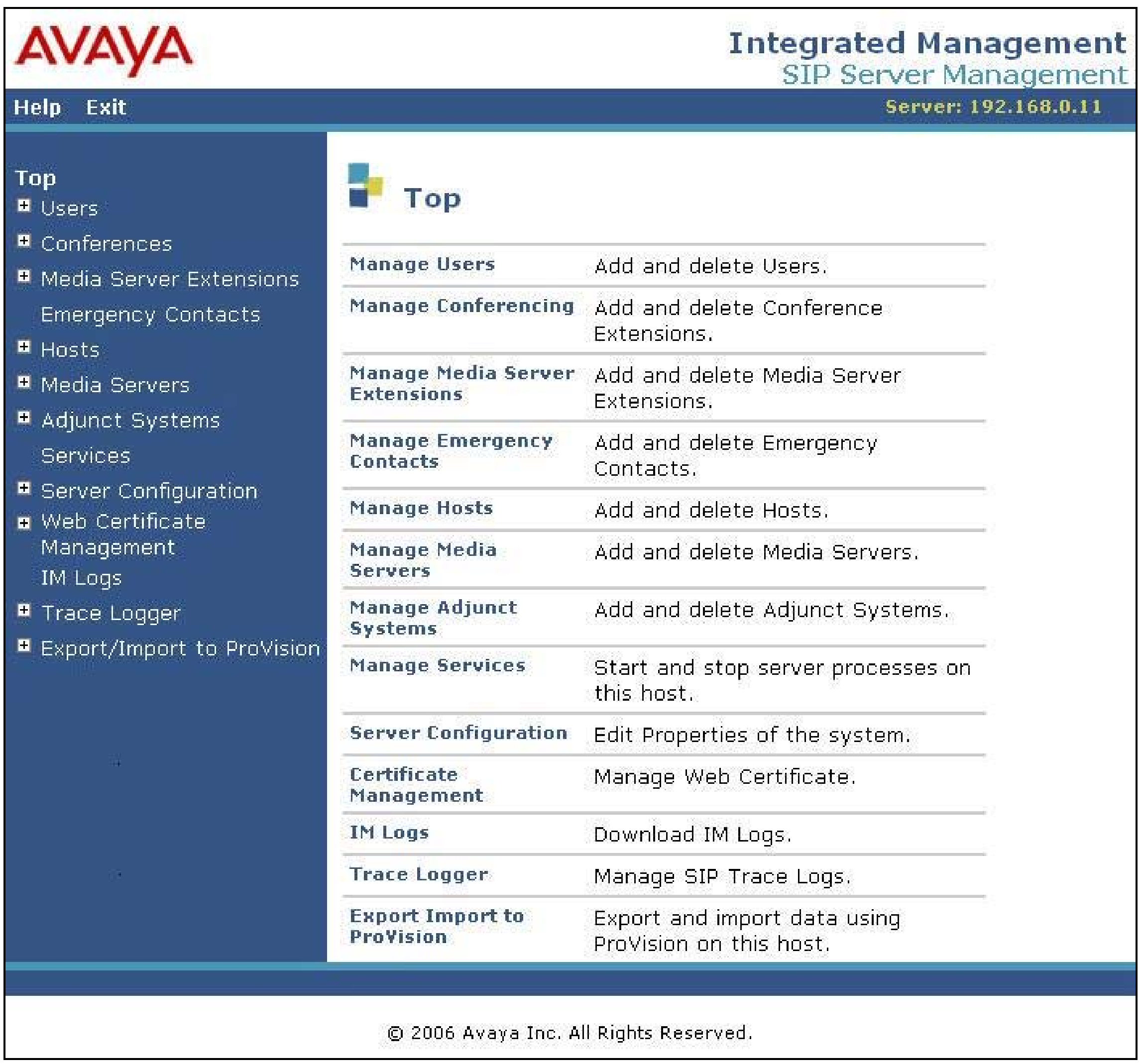

Figure 35: SES Administration Home Page

4.1.2 Verify System Properties

From the left pane of any **SIP Server Management** page, expand the **Server Configuration** option and select **System Properties**. This page displays the **SES Version** and the **Network Properties** entered via the install script during the installation process.

In the **Edit System Properties** page note the **SIP Domain** entered. The **SIP Domain** ***`customer.com`*** is used in these Application Notes in section 3.1.5.

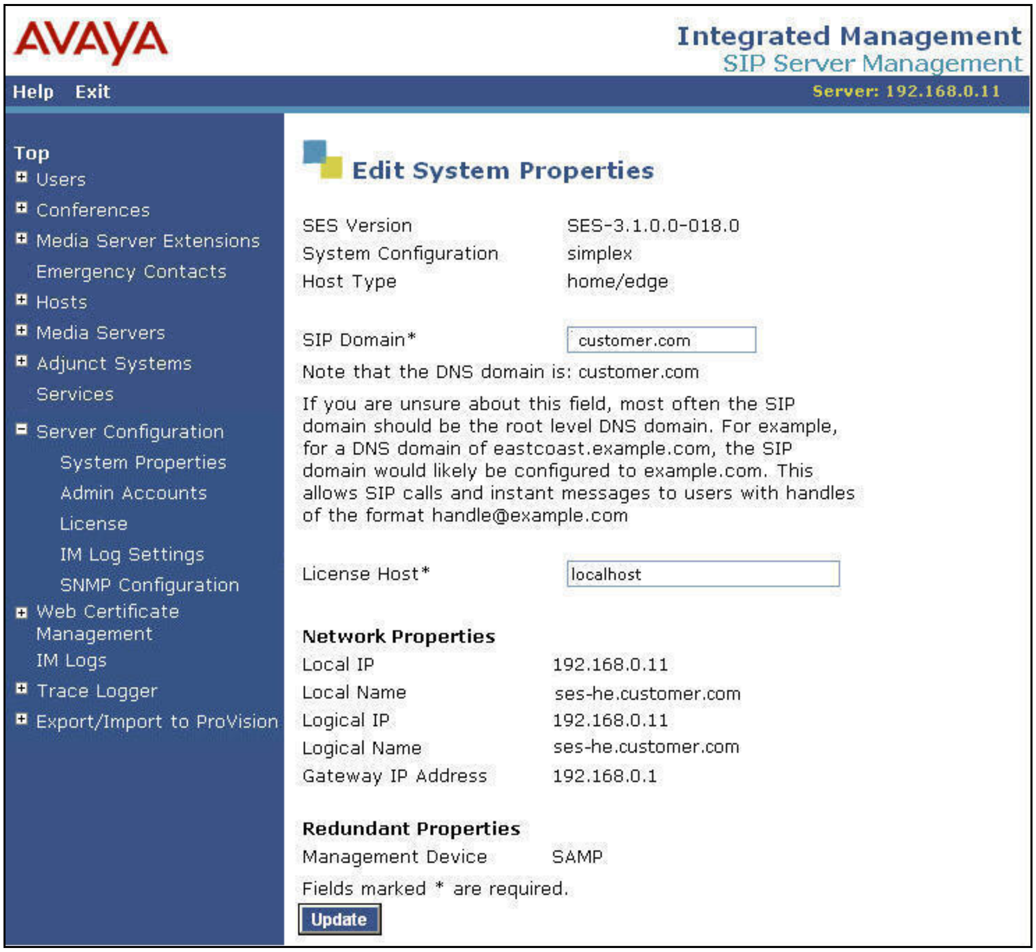

Figure 36: System Properties

4.1.3 Verify the Avaya SES Host Information

Verify the Avaya SES Host information using the **Edit Host** page. In these Application Notes the Avaya SES **Host Type** is a combined ***`home/edge`***. This means that the both the Verizon Business VoIP Service and Avaya Communication Manager are contacting the same SES.

Display the **Edit Host** page (**Figure 37**) 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.

On the **Edit Host** screen:

- Verify that the IP address of this combined SES Home/Edge server is in the **Host IP Address** field.
- Do not modify the **DB Password** or **Profile Service Password** fields. If these fields are changed, exit the form without using the **Update** button. These values must match the values entered during the SES installation; incorrect changes may disable the SES.
- Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols**.
- Verify that **TLS** is selected as the **Link Protocol**.
- Ensure that the Verizon Business VoIP Service SIP Application Server (e.g., ***`vzb-sip.net`***) is not in the **Outbound Proxy** or **Outbound Direct Domains** fields.
- 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.

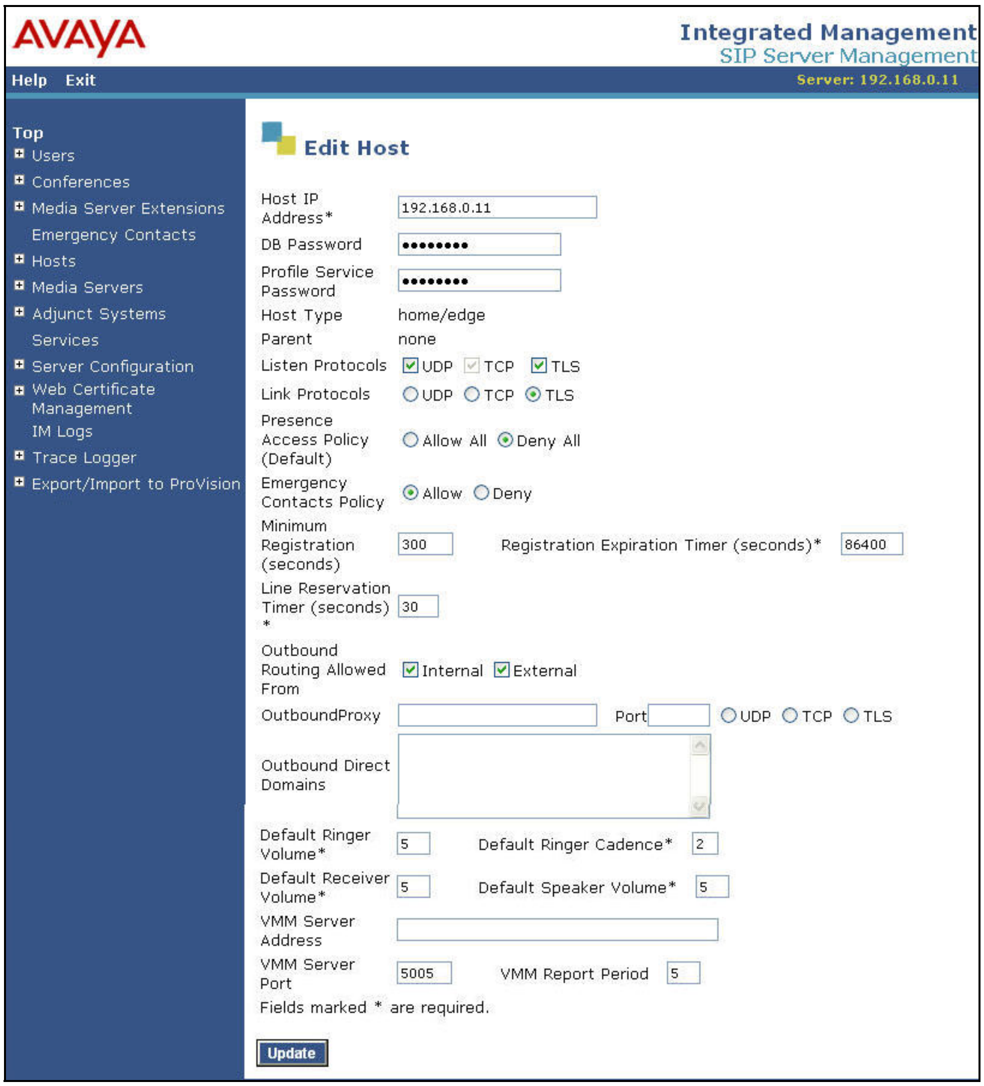

Figure 37: Edit Host

4.1.4 Add Avaya Communication Manager Media Server Interfaces

In these Application Notes, two **Media Server** signaling interfaces will be established with Avaya Communication Manager. The interface named ***CLAN-1A03-5061*** will support voice calls using the G.729A codecs; the interface named ***CLAN-1A03-5062*** will be used for fax / modem calls that require G.711 codecs.

4.1.4.1 Media Server Interface for Voice Calls

Expand the **Media Servers** option within any SES **SIP Server Management** page, and select **Add** to display the **Add Media Server** page (**Figure 38**).

In the **Add Media Server Interface** page, enter information corresponding to the voice **Signaling Group *2*** entry performed in section 3.1.6.1.

- Enter ***CLAN-1A03-5061*** as the descriptive name in the **Media Server Interface Name** field.
- Select the SES home/edge IP address in the **Host** field.
- Select ***TLS*** (Transport Link Security) for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer between Avaya Communication Manager and the SES.
- Enter the IP address of the ***CLAN-1A03*** interface in the **SIP Trunk IP Address** field. Note: This may be the IP address of the media server processor interface in other Avaya Communication Manager configurations.

After completing the **Add Media Server Interface** page, click on the **Add** button.

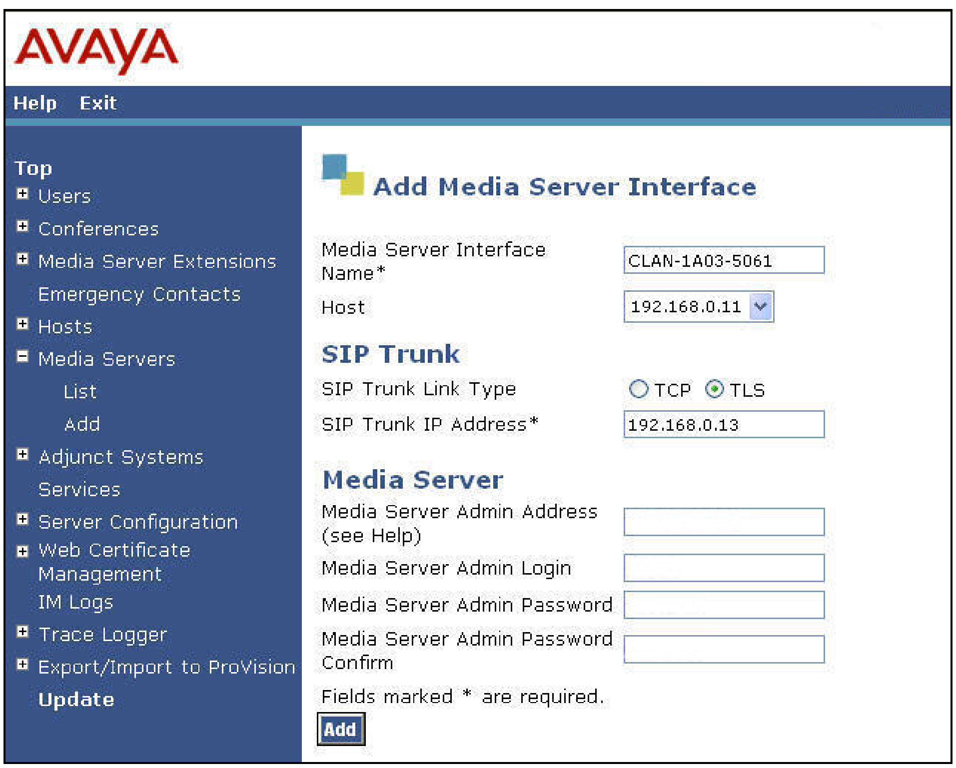

Figure 38: Add Media Server Interface for Voice Calls

4.1.4.2 Media Server Interface for Fax / Modem Calls

Expand the **Media Servers** option in any SES **SIP Server Management** page, and select **Add** to display the **Add Media Server Interface** page (**Figure 39**).

In the **Add Media Server Interface** screen, enter information corresponding to the fax / modem **Signaling Group *3*** entry performed in Section 3.1.6.2.

- Enter ***`CLAN-1A03-5062`*** as the descriptive name in the **Media Server Interface Name** field.
- Configure the remainder of the fields in a similar manner (as was done for the voice media server interface in Section 4.1.4.1).

  Note the **SIP Trunk IP Address** is the same as the voice interface because the signaling groups share the same C-LAN, but are differentiated by the listen port used within Avaya Communication Manager.

After completing the **Add Media Server Interface** page, click on the **Add** button.

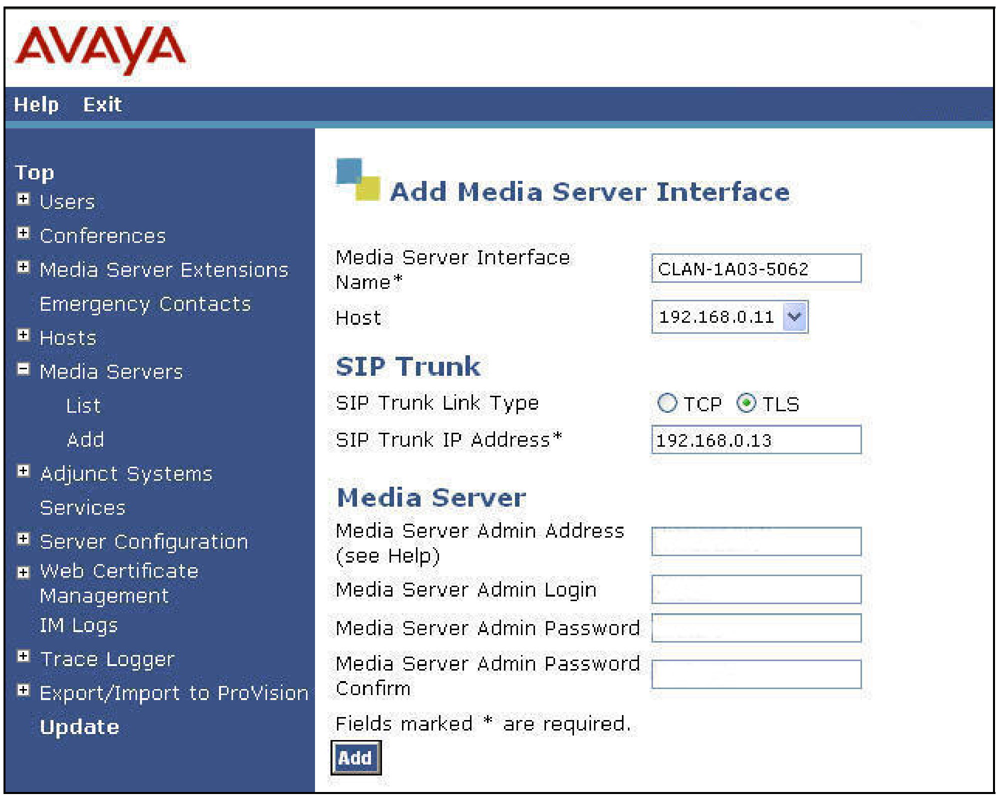

Figure 39: Add Media Server Interface for Fax / Modem Calling

When these operations are completed, the **List Media Servers** page will appear as shown in **Figure 40**.

Figure 40: Completed List Media Servers

4.1.5 Configure Call Routing

Avaya SIP Enablement Services functions as a SIP proxy server for the SIP trunking with the Verizon Business VoIP Service. In this role, for outbound calls the SES must direct SIP messages originating from Avaya Communication Manager to the Verizon SIP Application Server. In a similar manner for incoming DID calls, the SES must route messages received from the Verizon SIP Application Server to the proper signaling interface on Avaya Communication Manager.

In these Application Notes, the SIP message routing will be done for both outbound and inbound calls using Address Maps that examine some or all of the *called number* (using a Pattern) and route to a specific predetermined destination (called a Contact). The outbound proxy and direct domain routing feature is not used due to interactions with the trusted host capabilities.

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@domain*, where *domain* can be a fully qualified domain name or an IP address. The *user* part for SIP trunking in these Application Notes will only contain digits[1].

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

The 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 Address Map Patterns must also be mutually exclusive (non-overlapping) from all other Address Map Patterns used in the SES to ensure proper operation.

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

4.1.5.1 Outbound PSTN Calls

SIP signaling for outbound calls is directed to Avaya SIP Enablement Services based upon rules configured for Automatic Route Selection in Avaya Communication Manager. The choice of trunk group (made within ARS) determines the codec that will be used for voice vs. fax / modem calls. For outbound calls, no further differentiation of voice vs. fax / modem is required within the SES address maps.

Outbound calls require SIP signaling messages to be routed to the Verizon Business SIP Application Server using Host Address Maps within SES. Calls matching Host Address Map Patterns will be directed to the Verizon Business SIP Application Server (e.g., `vzb-sip.net`) contained within the corresponding Contact information.

In these Application Notes, the Avaya SES routing rules will be to send all outbound PSTN traffic to the Verizon Business VoIP Service. **Table 2** contains the rules used to configure the SES Host Address Maps in the following sections.

Type of Call	**Digits Sent**	**Host Address Map Name**	**Host Address Map Pattern**
Local and Long Distance in North American Numbering Plan	Digit 1 plus any 10 digits	***VZB_1plus10***	***^sip:1[0-9]{10}***
Operator and International Calling	Digit 0 with or without additional digits; overall length indeterminate	***VZB_0any***	***^sip:0***
N11 Service Calls	Digits 2 thru 9 followed only by the digits 11	***VZB_N11***	***^sip:[2-9]11***

Table 2: Outbound Host Address Map Rules

Note that additional or more specific pattern matches would be used if necessary to selectively route SIP traffic to different destinations (such as multiple service providers serving different geographic regions). Also note that a user dialed access code (such as 9 to place a PSTN call) has been previously deleted (by ARS) prior to seizing the outbound SIP trunk.

4.1.5.1.1 Outbound Routing - Host Maps

Begin the outbound routing configuration by navigating to the **Add Host Address Map** pages.

- From any **SIP Server Management** page, expand the **Hosts** link and choose the **List** link causing the **List Host** page to appear.
- Select the **Map** link on the **List Host** page causing the **List Host Address Map** page to appear (**Figure 41**).

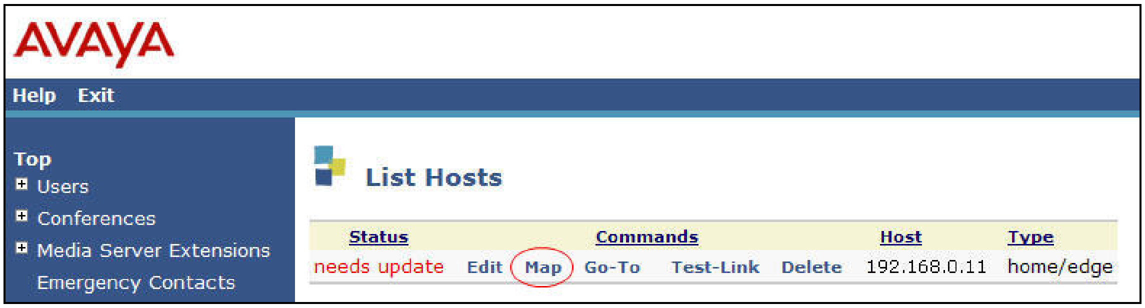

Figure 41: Accessing Host Address Maps from List Hosts Screen

- Select the **Add Map In New Group** link on the **List Host Address Map** page (**Figure 42**). This will display the **Add Host Address Map** screen (**Figure 43**).

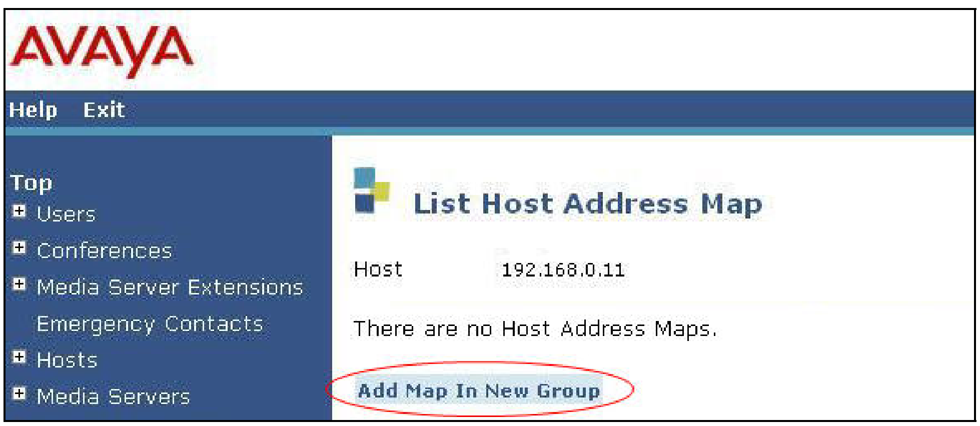

Figure 42: Adding a Host Address Map Group

The configuration of the **Host Address Map** for all 1 plus 10 digit North American calls is shown in **Figure 43**.

- Enter a descriptive **Name** for the map, such as ***`VZB_1Plus10`***.
- Enter the appropriate pattern for the call type. In this example, the pattern used for North American calls is ***`^sip:1[0-9]{10}`*** as noted in **Table 2**.
- Leave the **Replace URI** checkbox selected.
- Click the **Add** button.

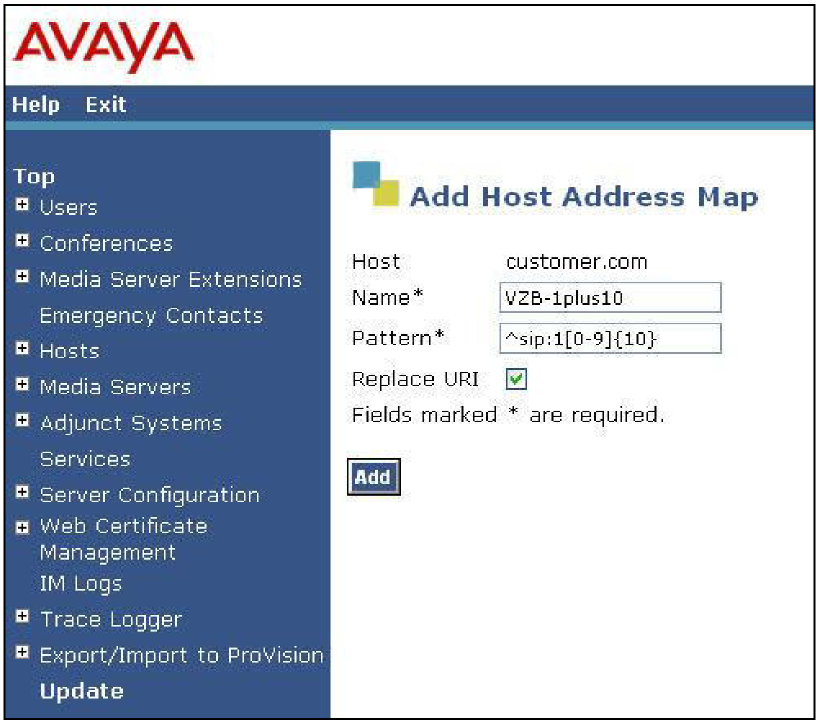

Figure 43: Address Map for 1+10 Digit Dialing

The remaining **Host Address Map** patterns for outbound calling are added in a similar manner. **Figure 44** illustrates the entry for Operator "zero" and "zero-plus" dialing.

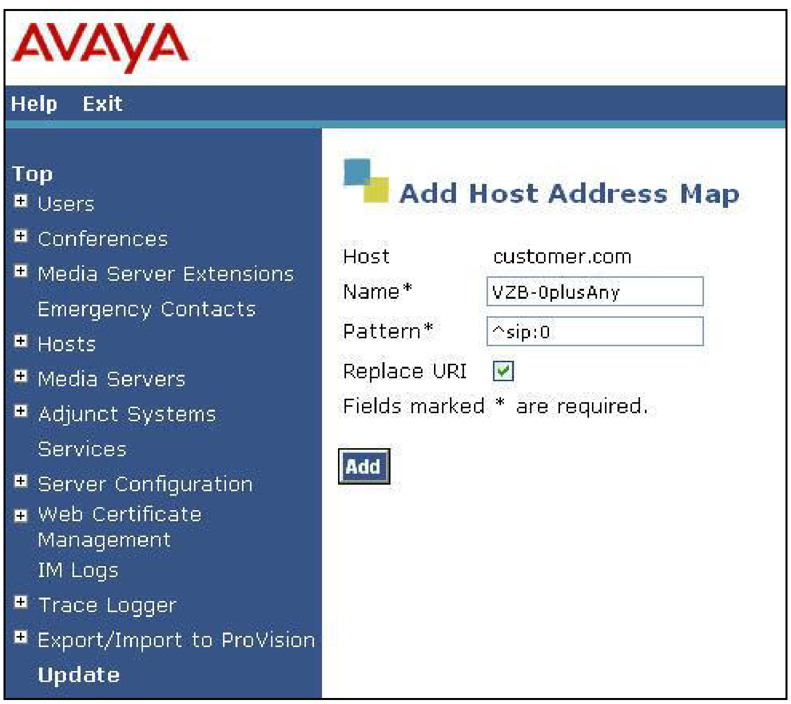

Figure 44: Address Map for 0 and 0+ Dialing

Figure 45 illustrates the host address map for the N11 service codes.

Figure 45: Address Map for N11 Dialing

4.1.5.1.2 Outbound Routing – Verizon Business VoIP Service Gateway

The next step is to enter the **Contact** information for the Verizon Business SIP Application Server. In these Application Notes, the domain name ***vzb-sip.net*** is used and the SES was configured during installation to resolve fully qualified domain names using DNS. This domain name will resolve to a DNS SRV [9] record and may return multiple IP addresses. The customer must obtain the actual domain address and the corresponding IP addresses from Verizon Business.

To enter the Verizon Business SIP Application Server information:

- Access the **Host Address Map** page by expanding the **Hosts** link in the left pane of any **SIP Server Management** page, selecting **List** and then clicking on the **Map** link associated with the appropriate **Host** (e.g., ***192.168.0.11***).

  The **List Host Address Map** page is displayed as shown in **Figure 46**. Notice that the **Contact** information is blank when first creating this entry.

  Note: Should an entry already exist due to prior administration, the entry should be edited or deleted instead of using **Add Another Contact**.

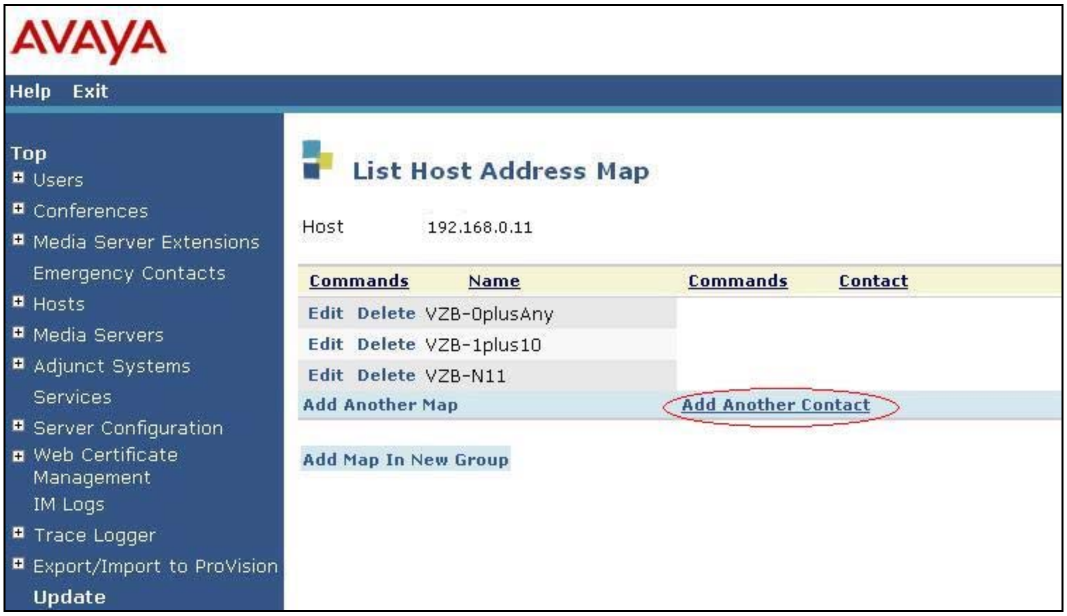

Figure 46: List Host Address Map – Prior to Contact Entry

- Click on the **Add Another Contact** link to open the **Add Host Contact** page.
- In the **Add Host Contact** page, the **Contact** field specifies the destination for the call. In these Application Notes the Contact field is entered as:

  sip:$(user)@vzb-sip.net

  Note that port or transport notation must not be included in this string; otherwise, the proper DNS SVR query will not be performed.

- Click the **Add** button when completed.

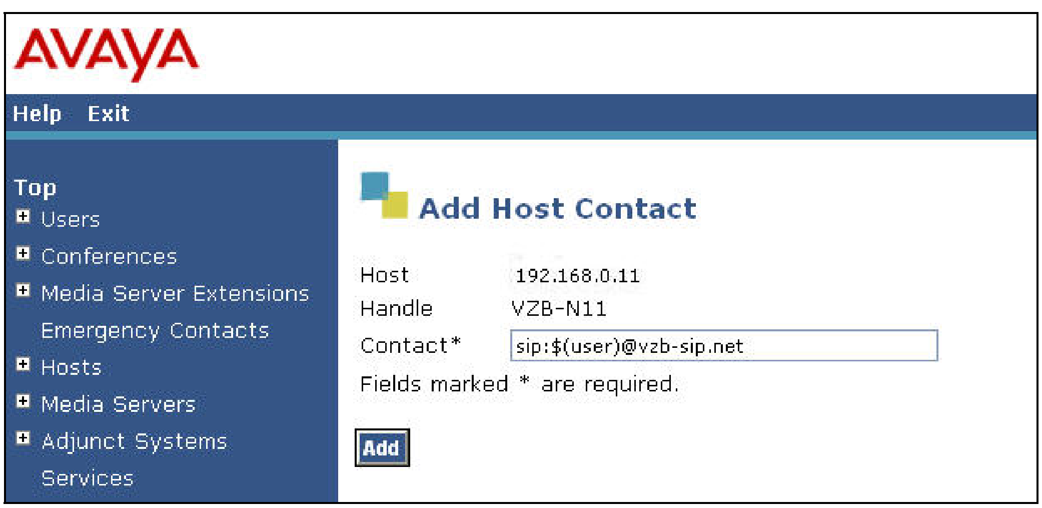

Figure 47: Add Host Contact Entry

After configuring the **Maps** and **Contact** information, the **List Host Address Map** page will appear as shown in **Figure 48**.

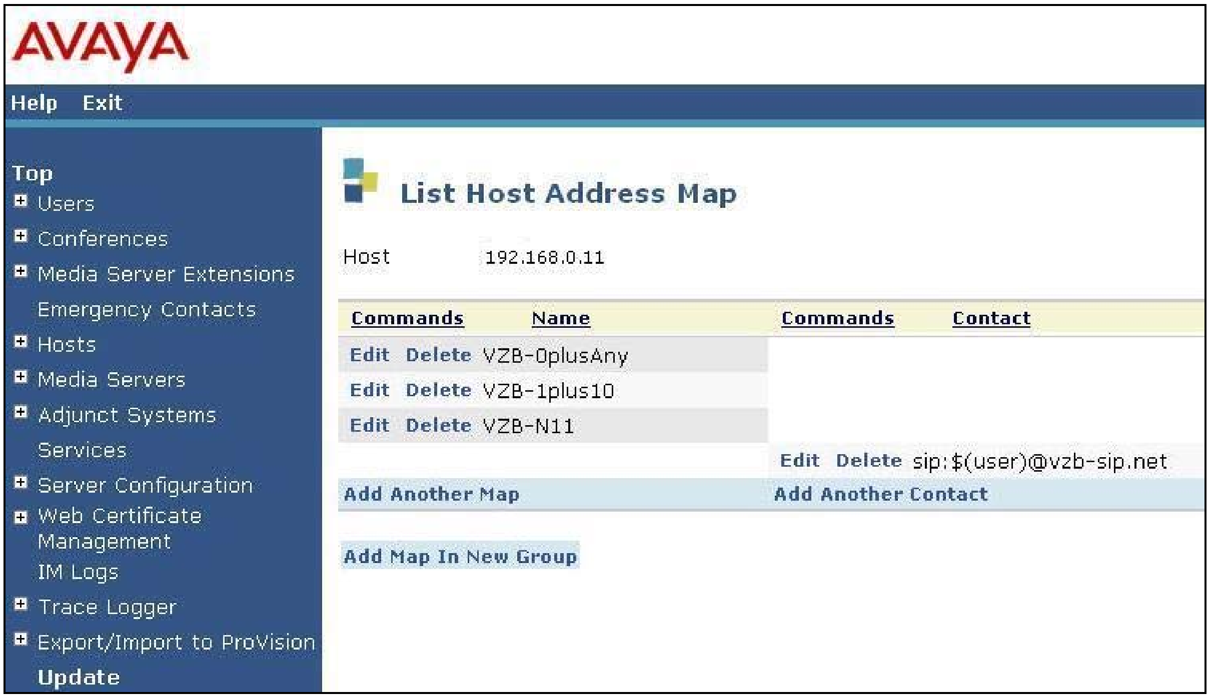

Figure 48: Completed List Host Address Map

4.1.5.2 Inbound Direct Inward Dialed Calls

SIP messages for incoming calls from the Verizon Business VoIP Service are sent to the Avaya SIP Enablement Services server. The SES then routes these messages to the appropriate Avaya Communication Manager **Signaling Group** using an SES **Media Server Address Map**.

In these Application Notes, the incoming PSTN calls from the Verizon Business VoIP Service use **Media Server Address Map Patterns** matching the 4-digit ***called number*** in the ***user*** part of the SIP URI.

An example of a SIP URI in an INVITE message received from the Verizon Business VoIP Service for the DID number 903-555-8009 is:

sip:8009@192.168.0.11

The ***user*** part in this case is the 4-digit number "8009".

In these Application Notes, the DID numbering plan was segmented into two distinct categories of DID numbers to distinguish between voice extensions and fax / modem extensions. Each category of DID number requires a distinct **Address Map Pattern** and a corresponding **Media Server Interface.**

Table 3 below summaries the media server address map strategy used in these Application Notes for incoming calls.

DID Number Range	SIP URI User Portion	Type of Call / Expected Codec	Address Map Pattern	Media Server Interface
903-555-8000 through 8299	8000 through 8299	Voice / G.729A	***^sip:8[0-2][0-9]{2}***	***CLAN-1A03-5061***
903-555-8300 through 8399	8300 through 8399	Fax / G.711MU	***^sip:83[0-9]{2}***	***CLAN-1A03-5062***

Table 3: Incoming DID Address Map Rules

4.1.5.2.1 Voice DID Routing

To configure the **Media Server Address Map** for voice calls:

- Expand the **Media Servers** link in the left navigation menu of any **SIP Server Management** page. Select **List** to display the **List Media Servers** page as shown in **Figure 49**.
- Click on the **Map** link of the ***CLAN-1A03-5061*** **Interface** to display the **List Media Server Address Map** screen associated with the voice signaling interface.

Figure 49: List Media Servers

- Click on the **Add Map In New Group** link. The page shown in **Figure 50** is displayed. The **Host** field displays the name of the media server to which this map applies.
- Enter a descriptive name in the **Name** field, such as ***Voice-8000-8299***.
- Enter the **Address Map Pattern** for incoming voice calls (from **Table 3**) into the **Pattern** field.

In this case, the DID numbers provided by Verizon Business for voice extensions are 903-555-8000 thru 8299. The pattern specification for voice DID numbers is:

^sip:8[0-2][0-9]{2}

This means that URIs beginning with "sip:8" followed the single digit 0 through 2 and then any other 2 digits will match the pattern and be routed to the interface defined as ***CLAN-1A03-5061.***

- Click the **Add** button once the form is completed.

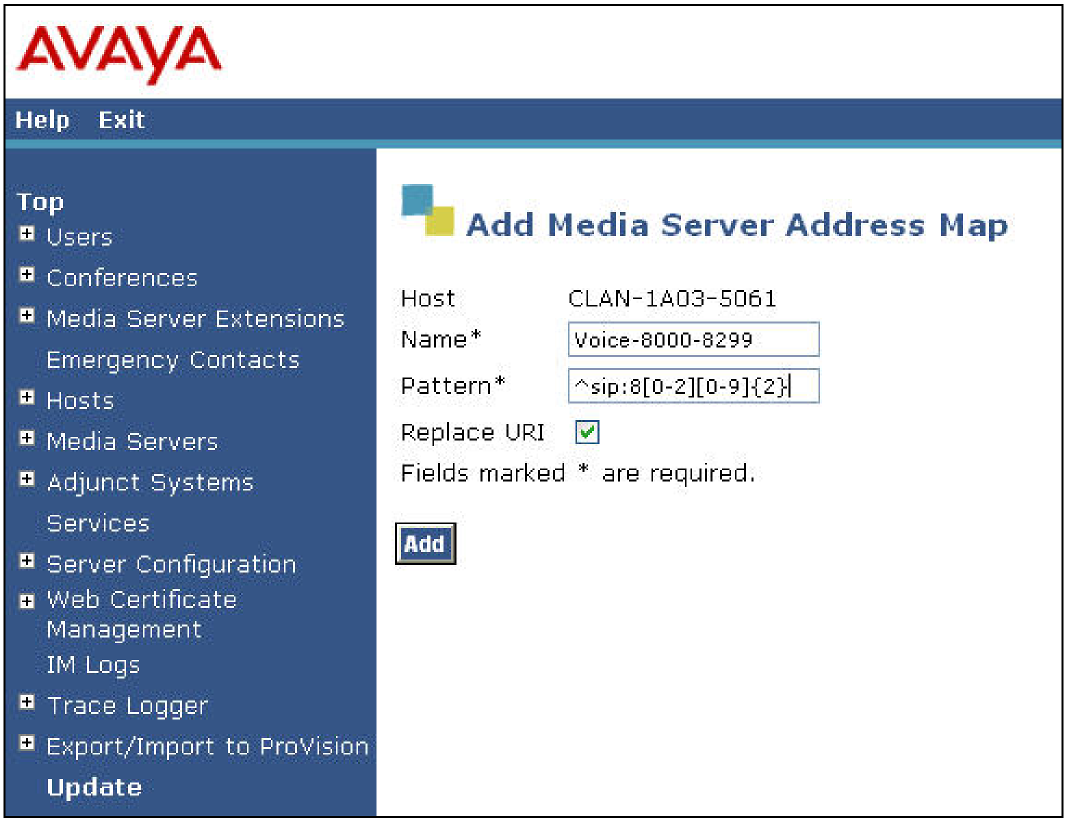

Figure 50: Voice Calls - Media Server Address Map

After configuring the media server address map, the **List Media Server Address Map** page appears as shown in **Figure 51**.

Figure 51: List Media Server Address Map

Note that after the first **Media Server Address Map** is created, a corresponding media server **Contact** entry is created automatically.

`sip:$(user)@192.168.0.13:5061;transport=tls`

This **Contact** entry contains the IP address of the ***`CLAN-1A03`*** interface on Avaya Communication Manager, the port (5061) 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**.

4.1.5.2.2 Fax / Modem DID Routing

The fax / modem **Media Server Address Map** is configured using the information for the fax / modem cases in a similar manner to the one specified in section 4.1.5.2.1.

Note that an additional step is required to update the port number of the **Contact** field in order to match the port value used for the **Signaling Group 3** interface (supporting fax / modem calls) that was previously defined in section 3.1.6.2

- From the **List Media Servers** page, select the **Map** link corresponding to the **Interface** (CLAN-1A03-5062) used for fax / modem calls (**Figure 52)**.

Figure 52: Fax / Modem Calls – List Media Servers

- Enter the **Name** and **Pattern** values from **Table 3** corresponding to the fax / modem interface into the **Add Media Server Address Map** form as shown in **Figure 53**.
- Click the **Add** button.

Figure 53: Fax / Modem Calls – Add Media Server Address Map

- Following the **Add** operation, press **Continue** to display the **List Media Server Address Map** page shown in **Figure 54**.

  Note it now contains an automatically created **Contact** entry. However, the ***5061*** port value does not match port ***5062*** value defined for the fax / modem **Signaling Group 3** interface in section 3.1.6.2.

- Select the **Edit** link (**Figure 54**) associated with the ***CLAN-1A03-5062*** **Contact** information. This will display the **Edit Media Server Contact** screen (**Figure 55**).

Figure 54: Fax / Modem Calls – List Media Server Address Map

- Modify the **Contact** field to use 5062 as the port value, instead of the default 5061 as shown in **Figure 55**. Be careful to not alter the remainder of the string.

Figure 55: Fax / Modem Calls – Edit Media Server Contact Port Value

- **Update** the form and press **Continue** on the following page.
- Verify that the **Contact** entry was correctly modified as shown in **Figure 56**. This completes inbound fax / modem routing configuration.

Figure 56: Fax / Modem Calls – Edit Media Server Contact Port Value

4.1.6 Specify the Verizon Business SIP Application Server as a Trusted Host

The domain name used by the Verizon Business VoIP Service SIP Application Server (e.g., ***vzb-sip.net***) must be added as a trusted host entry in the Avaya SES. As a trusted host, the SES will not issue SIP authentication challenges for incoming requests from the designated IP address.[2]

To configure a trusted host:

- Log in to the Avaya SES Linux shell using the administrative login and password.

- Enter the following **trustedhost** command at the Linux shell prompt.

  trustedhost –a *vzb-sip.net* –n *192.168.0.11* –c *vzb-sipProxy*

  The –a argument specifies the fully qualified domain name or IP address to be trusted; –n specifies the SES host name; –c adds a comment.

- Use the following **trustedhost** command to verify the entry is correct.

  trustedhost –L

[2] If the trusted host step is not done, authentication challenges to incoming SIP messages (such as INVITEs and BYEs) will be issued by the SES. This may cause call setup to fail, active calls to be disconnected after timeout periods, and/or SIP protocol errors.

Figure 57 illustrates the results of the above trustedhost commands.[3]

```
admin@ses-he> trustedhost –a vzb-sip.net –n 192.168.0.11 –c vzb-sipProxy
vzb-sip.net is added to trusted host list.


admin@ses-he> trustedhost -L
Third party trusted hosts.
       Trusted Host       |        CCS Host Name       |         Comment
--------------------------+----------------------------+-------------------------
vzb-sip.net               | 192.168.0.11               |    vzb-sipProxy
```

Figure 57: Results of Trusted Host Commands

- The trusted host configuration will not take effect until the **Update** action done in section 4.1.7 is performed.

  Note: In some cases the **Update** link may not be visible immediately following a trusted host command. Refreshing a **SIP Server Management** page by selecting **Top** from the left navigation menu will cause the **Update** link to appear.

4.1.7 Save the SES Changes

After making any changes within Avaya SES, commit the database changes by using the **Update** link on a **SIP Server Management** page that appears when changes are pending.

Perform this step by clicking on the **Update** link found in the bottom of the blue navigation bar on the left side of any **SIP Server Management** page as shown in **Figure 58**.

[3] For completeness, the –d argument allows the trust relationship to be deleted. For example,

trustedhost –d vzb-sip.net –n 192.168.0.11

removes a previously entered trust relationship.

Figure 58: Update Displayed After an SES Administrative Change

4.2. Configuration for SIP Telephones

This section provides basic instructions for completing the administration to use Avaya 4600 Series SIP telephones with the described configuration. Additional features, such as the use of mnemonic addressing, are beyond the scope of these Application Notes.

As noted previously, SIP telephones are optional; it is not necessary to have SIP telephones to use SIP trunking to the Verizon Business VoIP Service.

The steps below are repeated for each SIP telephone provisioned.

4.2.1 Add a SIP User

Add a SIP User to the SES as follows:

- In Avaya SES **SIP Server Management**, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- In the **Add User** page (**Figure 59**), enter the extension number for the SIP telephone in the **Primary Handle** field.
- Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
- In the **Host** field, select the Avaya SIP Enablement Services server hosting the domain (***`192.168.0.11`***) for this user.

- Enter the **First Name** and **Last Name** of the user.
- 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.
- Click the **Add** button. This will cause a confirmation screen to appear.
- Click **Continue** on the confirmation screen.

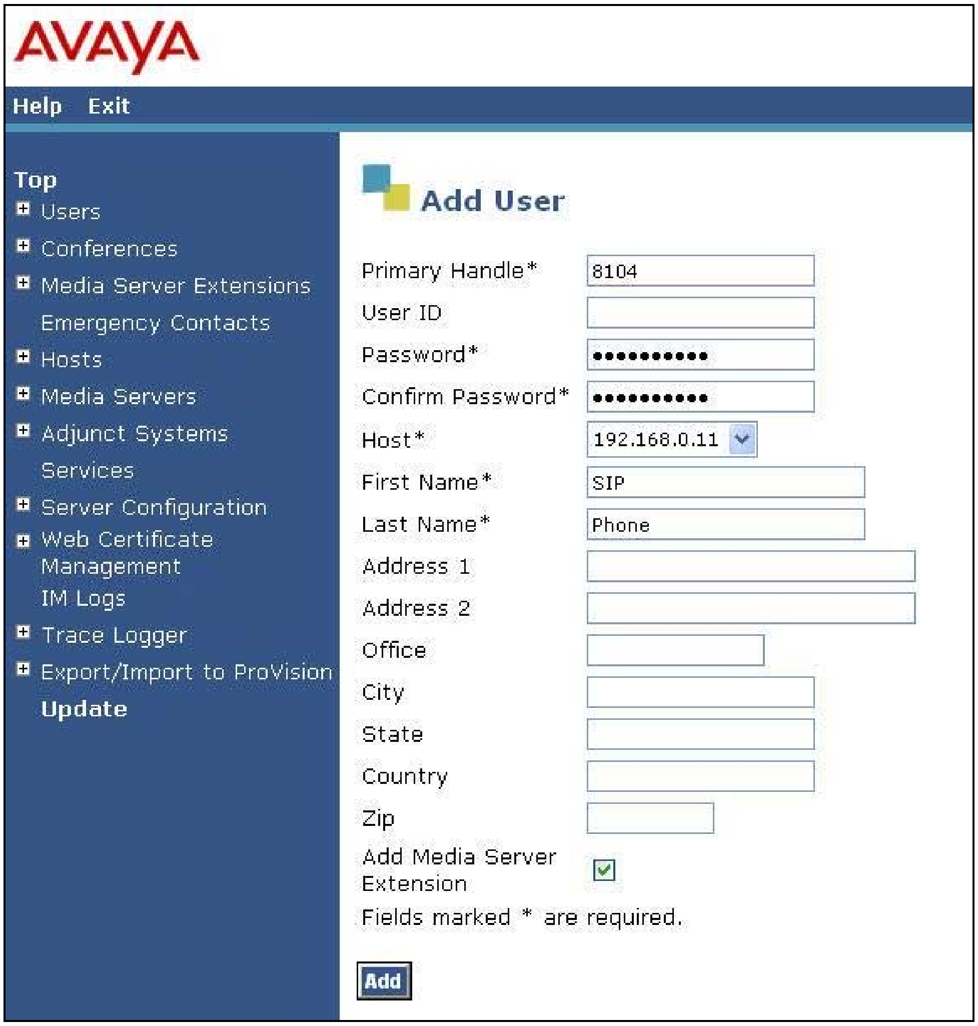

Figure 59: Add User

4.2.2 Specify Corresponding Avaya Communication Manager Extension

The SIP phone handle must now be associated with the corresponding extension on Avaya Communication Manager.

To complete this step:

- In the **Add Media Server Extension** page, enter the extension number (e.g., ***8104***) corresponding to the extension previously configured on Avaya Communication Manager

(**Figure 29**) in section 3.2. The Media Server **Extension** and the SIP Primary Handle are usually the same, but are not required to be.

- Select the **Media Server** interface assigned to this extension.
- Click the **Add** button.

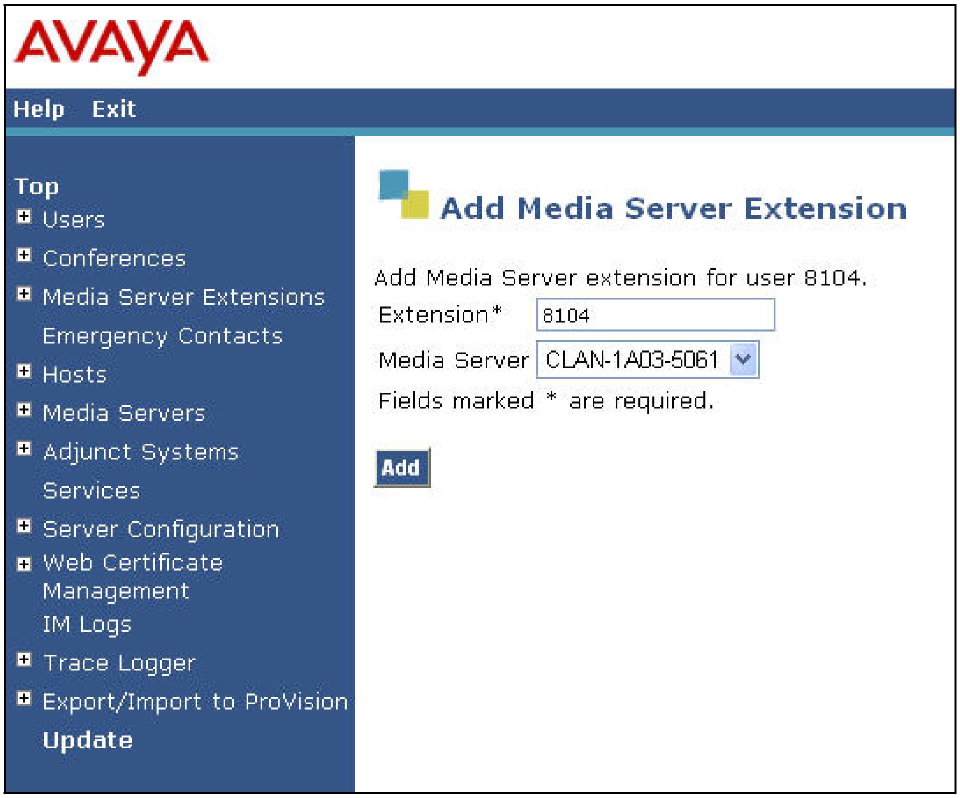

Figure 60: Add Media Server Extension

4.2.3 Save the SES Changes

Perform this step by clicking on the **Update** link that will appear at the bottom of the left side navigation bar of any SES Administration page as previously shown in **Figure 58**.

5. Verizon Business VoIP Service Configuration

In order to use Verizon Business VoIP Service, a customer must order service from Verizon Business using their sales processes. The process can be started by contacting Verizon Business via their corporate web site at http://verizonbusiness.com or by contacting a Verizon Business sales representative.

Verizon Business provided the information contained in **Table 4** to complete the configuration in these Application Notes. This information was necessary to complete the Avaya Communication Manager and Avaya SIP Enablement Services administration discussed in the previous sections.

Required Information	Values Used In These Application Notes
Verizon Business VoIP Service SIP Application Server	***vzb-sip.net***
Assigned DID Numbers	***903-555-8000*** to ***8399***
Incoming Digits Sent in RequestURL	***8000 to 8399***
Codecs Supported	***G.729A, G.711MU, G.711A***

Table 4: Verizon Business VoIP Service Configuration Information

As noted in Section 1.1, Verizon Business provided the managed Internet access service used to reach their VoIP service including the configuration of the Cisco router and firewalls used in the Application Notes.

Table 5 indicates the information that must be known about the Avaya SIP telephony solution used within these Application Notes for Verizon Business to properly configure the Network Address Translation (NAT) for use with SIP trunking. This NAT function must perform SIP-aware Application Layer Gateway functions.

Required Information	Values Used In These Application Notes	Comments
SIP Domain Name	customer.com	DNS SRV Record
SES Edge Private IP Address	192.168.0.11	Requires static NAT mapping to an external IP address
Avaya Communication Manager Media Processor Private IP Address(es) / subnet	192.168.0.14	Requires NAT May be dynamic.
Avaya 4600 Series Telephone Private IP Address(es) / subnet	192.168.0.98 192.168.0.99	Requires NAT if IP-IP Direct Audio (aka “shuffling”) is enabled. May be dynamic.

Table 5: Avaya SIP Solution Information Necessary for NAT/ALG Configuration

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Verizon Business VoIP Service and an Avaya SIP telephony solution using 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 SIP telephony solution supporting SIP trunking was connected to a laboratory version of the Verizon Business VoIP Service via simulated managed access facilities. The enterprise site was configured as if using the generally available service provided by Verizon Business.

The following features and functionality were covered during the SIP trunking interoperability compliance test. All calls involved various Avaya telephones, the Verizon Business VoIP Service and various PSTN telephones.

- Outgoing calls to PSTN telephones
- Incoming calls to Verizon Business provided DID numbers from PSTN telephones
- Calls using Avaya 4600 Series IP Telephones with the H.323 firmware configurations
- Calls using Avaya 4600 Series IP Telephones with the SIP firmware configurations
- G.729A and G.711MU codecs for voice calls
- G.711MU codecs for fax calling
- Distinct fax routing to ensure G.711 use for fax calls
- DTMF tone transmission using RFC 2833 with successful Voice Mail / IVR navigation
- Telephone features such as hold, transfer, conference, and voice mail
- Direct IP-to-IP media (also known as "shuffling") with SIP telephones

6.2. Test Results

Interoperability testing of the sample configuration was completed with successful results.

The following minor issues described in **Table 6** were observed.

Item	Issue Observed	Discussion / Workaround
RFC 2833 Telephone Event Value 127	Outbound calls do not use RFC 2833 for DTMF transmission when the default Avaya telephone event value of 127 is used. Verizon will prohibit the use of G.729 codecs unless RFC 2833 is used.	The value 127 is valid according to RFC 2833. However the use of 127 interacts with other equipment in the Verizon network preventing the use of RFC 2833 for DTMF transmission. This is not a problem if G.711 codecs are used. To support G.729 codecs, upgrading to Avaya Communication Manager to Release 3.1.2 Load 636 or later permits the telephone event value to be administered to the value 101 Verizon expects.
Send Calling Number restrictions	Outbound calls blocked if Send Calling Number is restricted in Avaya Communication Manager. Verizon returns a "604 – Does not exist anywhere" response.	The Verizon service expects to receive digits in the user part of the "SIP From" header. Send Calling Number restriction removes these digits. A Verizon outbound call restriction (controlled per DID number) can be used as an alternative means of avoiding sending caller ID.
SIP endpoint support	SIP endpoints require the SIP Domain (instead of the SES IP address) be sent in the "SIP From" header.	Resolved by ensuring the customer's SIP domain be resolvable within the Verizon Business Service.

Item	Issue Observed	Discussion / Workaround
SIP endpoint calls drop when G.729A codec used	Calls involving Avaya 4600 series SIP telephones connect and then drop when G.729A codec is used and Direct IP-to-IP (a.k.a shuffling) is enabled.	Avaya 46xx SIP phones support G.729B, not G.729A codec. Codec should be restricted to G.711, or Direct IP-to-IP for SIP telephones should be disabled if G.729A used.
Backslash characters at end of station name	Adding a backslash characters "\" at the end of the Station Name field caused the Verizon service to fail to process calls for that extension. Problem appears as no response to INVITE messages.	Don't use backslash characters within the Name field.

Table 6: Summary of Minor Issues Identified During Interoperability Testing

7. Verification Steps

This section provides the verification steps that may be performed to verify operation of the Avaya SIP telephony solution with the Verizon Business VoIP Service.

- Verify outbound voice calls can be placed to PSTN phones and that the calls remain connected for 3 minutes.
- Verify incoming voice calls can be received from PSTN phones and that the calls remain connected for 3 minutes.
- Verify that calls are properly disconnected when either end disconnects first.
- Using the **`list trace station`** command, verify that voice calls are using the expected codec and SIP trunk group (for both outbound and incoming calls).
- Verify outbound and inbound fax calls can be placed to and from PSTN fax machines.
- Using the **`list trace station`** command verify that fax calls are using the expected codec and trunk group (for inbound and outbound calls).
- If Avaya 4600 Series SIP telephones are used with G.711 codecs and Direct IP-to-IP is enabled, verify that the RTP media path is reconfigured once a stable call is established with a PSTN telephone. Verify by using the **`list trace station`** command to observe that the IP address of the SIP phone and Verizon SIP Application Server are the media endpoints after the call is established.

8. Support

For technical support on Verizon Business VoIP Service, contact Verizon Business Customer Service at 1-800-265-2316 or via their online support at http://www.verizonbusiness.com/us/customer/.

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya SIP telephony solution to the Verizon Business VoIP Service using SIP trunking. The Verizon Business VoIP Service is a Voice over IP solution for customers ranging from small businesses to large enterprises. SIP trunking uses the Session Initiation Protocol to connect private company networks to the public telephone network via converged IP access. It provides businesses a flexible, cost-saving alternative to current TDM-based telephony trunk lines.

10. References

This section references documentation relevant to these Application Notes.

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

[1] *Administrator Guide for Avaya Communication Manager*, May 2006, Issue 2.1, Document Number 03-300509.

[2] *Feature Description and Implementation for Avaya Communication Manager*, Issue 4, Document Number 555-245-205

[3] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, June 2005, Issue 9, Document Number 210-100-500.

[4] *Installing and Administering SIP Enablement Services R3.1*, February 2006, Issue 1.5, Document Number 03-600768

[5] *SIP Support in Release 3.1 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700, and S8710 Media Server*, February 2006, Issue 6, Document Number 555-245-206.

[6] *4600 Series IP Telephone R2.4 LAN Administrator Guide*, April 2006, Document Number 555-233-507

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.

[7] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard

[8] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

[9] RFC 2782 - *A DNS RR for Specifying the Location of Services (DNS SRV)*, February 2000, Proposed Standard

Additional information about Verizon Business VoIP Service is available at http://www.verizonbusiness.com/us/voip/trunking/.

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by Verizon Business and the Avaya SIP network at the enterprise site. These INVITE messages may be used for comparison and troubleshooting purposes. Differences in these messages may indicate that different configuration options were selected.

Sample SIP INVITE Message from Verizon Business to Avaya SIP Enablement Services:

```
INVITE sip:8100@customer.com SIP/2.0
Via:SIP/2.0/UDP 166.34.87.74;branch=z9hG4bK-BroadWorks.166.34.87.74-166.34.76.71V5060-0-118288461-2000689704-1157730263193
From:<sip:3363320011@166.34.87.74;user=phone>;tag=2000689704-1157730263193
To:"NA PBX-933-5558100"<sip:8100@customer.com>
Call-ID:BW104423193080906194675896@166.34.87.74
CSeq:118288461 INVITE
Contact:<sip:166.34.87.74:5060>
Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,UPDATE,NOTIFY
Supported:
Accept:application/sdp,application/dtmf
Max-Forwards:10
Content-Type:application/sdp
Content-Length:339

v=0
o=BroadWorks 4754749 1 IN IP4 166.34.87.111
s=-
c=IN IP4 166.34.87.111
t=0 0
m=audio 17866 RTP/AVP 18 0 8 2 101
c=IN IP4 166.34.87.111
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=bsoft: 1 image udptl t38
```

Sample SIP INVITE Message from Avaya SIP Enablement Services to Verizon Business:

```
INVITE sip:17325552000@vzb-sip.net SIP/2.0
Call-ID: 064fe37e52db1e31d44d0b0d300
CSeq: 1 INVITE
From: "Sip Station8100" <sip:8100@customer.com:5061>;tag=064fe37e52db1e21d44d0b0d300
Record-Route: <sip:192.168.0.11:5060;lr>,<sip:192.168.0.13:5061;lr;transport=tls>
To: "17325552000" <sip:17325552000@customer.com>
Via: SIP/2.0/UDP 192.168.0.11:5060;branch=z9hG4bK03030363636343434b02.0,SIP/2.0/TLS
192.168.0.13;psrrposn=2;branch=z9hG4bK064fe37e52db1e41d44d0b0d300,SIP/2.0/UDP
192.168.0.99:5060;psrrposn=1;branch=z9hG4bKe9870d430
Content-Length: 227
Content-Type: application/sdp
Contact: "Sip Station8100" <sip:8100@192.168.0.13:5061;transport=tls>
Max-Forwards: 67
User-Agent: Avaya CM/R013x.01.1.628.7
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
Session-Expires: 240;refresher=uac
Min-SE: 240
History-Info: <sip:3363320011@customer.com>;index=1
History-Info: "3363320011" <sip:3363320011@customer.com>;index=1.1
Supported: 100rel,timer,replaces,join,histinfo
P-Asserted-Identity:"Sip Station8100"<sip:8100@customer.com:5061>

v=0
o=- 1 1 IN IP4 192.168.0.13
s=-
c=IN IP4 192.168.0.14
t=0 0
m=audio 2572 RTP/AVP 18 0 8 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
```

APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the Avaya SES 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 SES:

- 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:17325551638@20.1.1.54:5060;transport=udp SIP/2.0
```