Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunks between the Verizon Business IP Contact Center (IPCC) Services Suite and Avaya SIP Telephony Using Domain Based Routing – Issue 1.2

Abstract

These Application Notes describe the steps required for configuring SIP trunk connections between the Verizon Business (VzB) IP Contact Center (IPCC) Services Suite and Avaya's SIP enabled telephony solutions consisting of Avaya Aura™ Communication Manager (version 5.2), Avaya Aura™ SIP Enablement Services (version 5.2), and various Avaya telephones. Domain Based Routing is also discussed.

Verizon Business IPCC Services Suite is a portfolio of interaction services that include IP Toll Free (IP TF) and IP Interactive Voice Response (IP IVR). These feature sets add the capability to support SIP terminations over Internet Dedicated Access (IDA) to the existing Verizon Business inbound services suite.

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

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1. Introduction

These Application Notes describe the steps for configuring Session Initiation Protocol (SIP) trunks between the Verizon Business (VzB) IP Contact Center (IPCC) Service and an Avaya SIP telephony solution consisting of Avaya Aura™ SIP Enablement Services (SES), Avaya Aura™ Communication Manager, and various Avaya telephony endpoints. The endpoints that were tested include IP telephones (using SIP and H.323 protocols) and digital telephones. The Avaya VoIP equipment in the "CustomerA" and "CustomerB" locations are designated as Customer Premises Equipment (CPE) by the Verizon network.

Verizon Business IPCC is a portfolio of interaction services that includes IP Toll Free (IP TF) and IP Interactive Voice Response (IP IVR). These feature sets add the capability to support SIP terminations over Internet Dedicated Access (IDA) or Private IP (PIP) to the existing Verizon Business inbound services suite (see **Section 1.1.1**). IDA was used for the reference configuration described in these Application Notes. IP TF is the base service offering, that offers core call routing and termination features. IP IVR is an enhanced service offering that is built on top of IP TF, and includes features such as menu-routing, custom transfer, and additional media capabilities.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network services 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.

Provisioning SIP trunks to a service provider such as Verizon Business allows any Avaya telephony endpoint to receive calls from the 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 a long distance telephone company.

The reference configuration used *Domain Based Routing* to direct outbound calls to the proper Verizon Business IPCC destination. Domain Based Routing utilizes SES DNS lookups to direct outbound calls to the proper Verizon Business IPCC destination. This provides the most flexible outbound route provisioning for the CPE as all the call route decisions take place in the Verizon Business IPCC network (see **Section 1.7.1**).

Note – Verizon Business also offers an IP Trunking (IPT) Service. That service is not discussed in these Application Notes. For more information on the Verizon Business IP Trunking (IPT) Service visit http://www.verizonbusiness.com/.

1.1. Verizon Business IPCC Service - Overview

Verizon Business IP Contact Center (IPCC) is a portfolio of internetworking services that tightly couples signaling and functionality from the Advanced Toll Free and IP networks to deliver the intelligent routing and call treatment required by today's contact centers. The IPCC services are network-based and include IP Toll Free and IP Interactive Voice Response (IP IVR).

Verizon's IPCC is standards-compliant and provides single-call service that allows PSTN-originated Toll Free calls to seamlessly terminate and transfer to a SIP or TDM endpoints, without call re-originations that tie up CPE port capacity. IPCC service includes advanced toll free features –

including automatic ISDN User Part and SIP Error overflow for reliable termination to SIP or TDM devices anywhere; as well as customer-driven pre/post call routing and/or call treatment and queuing for customers using Cisco ICM and Genesys Intelligent Call Routing.
As an extension of the Advanced Toll Free network, IPCC provides seamless service for hybrid (TDM and IP) terminations and transfers which makes it possible for customers to migrate their contact centers to IP at their own pace, without disrupting contact center operations or service levels. The customer continues to dial an 8XX number (TDM) and IPCC converts the Toll Free call into Voice over Internet Protocol, or back to TDM if required, and delivers it to the contact center(s) over standard access methods such as Internet Dedicated Access (IDA), Private IP, or ISDN.
These IP Contact Center products are part of an evolving portfolio of specialized functionality to help Contact Center Managers improve customer service, productivity and business strategy, as well as control costs

- Cost control can be achieved from efficiencies in IP Networking, Application deployment and administration and from Customer Premise Equipment (CPE).
- Productivity enhancements can result from the integration of voice and data, with multi-site, multimedia routing; the centralized management of distributed operations with virtual agents and high levels of redundancy for business continuity and disaster recovery planning.
- Efficiency improvements can be derived from the continued utilization of existing contact center systems, which leverage IPCC hybrid termination, transfer and overflow features, to integrate with new Contact Center installations to support multi-site operations and enable smoother technology upgrades.
- Customer Service improvements can come from integration of multiple customer touch points, the use of User-to-User data transfers and collaborative applications such as advanced conferencing, presence and interactive video, as well as from more consistent contact handling and prioritization across all media channels.

1.1.1 Verizon Business IP Toll Free

IP Toll Free (IP TF) provides Customers carrier-grade internetworking service between the Verizon SS7 (PSTN) and IP Networks for their Contact Centers. Customers who are in the process of migrating their Contact Centers from TDM to IP infrastructure, require seamless internetworking with the Public Switched Telephone Network to support their business, i.e.: from the same type of SS7 features (Caller-ID, Toll Free Service, and others), which they use today, and they need them to work the same way for a seamless migration.

IP Toll Free extends the capabilities of Verizon Toll Free service by converting traditional PSTN-originated Toll Free calls to RTP streams (Real-Time Transport Protocol, the standard for transmitting real-time data such as audio and video) and using SIP (Session Initiation Protocol) signaling to terminate calls to SIP enabled devices connected IDA and PIP Terminations. Thus, IP TF can now provide single-call service for Toll Free calls and deliver comprehensive QOS (Quality of Service) from Public Switched Telephone Network (PSTN) originations to IP Network terminations.
IP TF includes a full set of Advanced Toll Free features such as percentage allocation, time-of-day, and geographic or point of origin routing (listed below) which is compatible with traditional Toll Free service and make it possible for customers to migrate to IP without having to recreate their call routing plans. IP TF advanced features support includes Intelligent Contact Routing - Gateway to

provide near real-time and call-by-call routing control capability based on customer-provided routing parameters.

Moreover, in its continuing effort to further the internetworking of Advanced Toll Free and IP for Contact Centers, Verizon has now added Network Call Redirect to IP Toll Free. Network Call Redirect is an Advanced Toll Free feature is used to redirect or overflow calls in real-time according to outage, busy, or other customer-specific conditions, and it can now be used with IP Toll Free (IPTF) and IP Interaction Voice Response (IPIVR) services to provide seamless routing and overflow for both TDM and SIP end-points. By adding IP Toll Free and IPIVR terminations to their NCR tables, customers can redirect traffic using both SIP error and ISUP (ISDN User Part) failure conditions. NCR's call-by-call overflow functionality is programmed via Toll Free Network Manager (TFNM), a web-based application.

Terminations provisioned with NCR can support up to 5 overflow attempts (aka hops); for each hop, customers can define a different routing choice for each individual TDM cause value or SIP error code. In addition, when customers provision their alternate routing choices, they also have the option of using any of the standard Verizon Toll Free routing decision nodes available (listed below), such as percent allocation, day of week routing, time of day routing, etc. This means that customers can use Percentage Allocation routing to distribute traffic evenly across 3 sites and, when a site is down or experiencing a time out, they can use an NCR hop with a percent allocation decision node to automatically load-balance between their two remaining sites.

The Verizon Business IP TF solution delivers a superior single-call and intelligent routing carrier-grade alternative to Customer-owned VoIP Gateway implementations used today to support IP Contact Centers. IP TF provides the necessary Packet-Voice Gateway functionality at the Network level and provides a high level of flexibility and scalability, as well as usage-based costs.

- **IP TF Service Features**
  - North American Numbering Plan 8XX Originations.
  - International Toll Free Service Originations (ITFS / UFIN).
  - 8xx Gateway DAP translations supported.
  - International IP terminations - where permitted by international and country-specific regulatory/ correspondent agreements.
  - IP and TDM Termination Route Plans.
  - SIP Refer Call transfers (RFC 3515) (Single Channel Transfers).
  - Small Office / Home Office (SOHO) SIP Device Registration.
  - NGSN IP (SIP) Terms (IP Terminations for existing ECR applications).
  - Intelligent Call Routing Gateway (pre/post call routing) for Cisco and Genesys Intelligent Call Routers.
  - User to User Interface support for Avaya's standards (RFC) recommendation.
  - Advanced Toll Free Features:
    - Network Call Redirect (ISUP and SIP Error overflow plans)
    - Extended Call Coverage
    - Tailored Call Coverage/Call Area Selection
    - Geographic/Point of Call Routing
    - Cross Corp ID Routing (out)
    - Payphone Blocking
    - Time of Day/Time Interval Routing

- Day of Week Routing
- Day of Year/Holiday Routing
- Alternate Routing
- Exchange Routing
- Percentage Allocation
- Dialed Number Service (DNIS may be delivered via SIP URI and/or From header -not SIP standard)
- Calling Party Number (Treated as ANI and translated to URI)
- Disconnect Message Referral
- IP Interactive Voice Response (IP IVR)
- Verizon Enterprise Center
- National Toll-Free Listing

1.1.2 Verizon Business IP Interactive Voice Response

IP IVR provides customizable treatment, routing and transfers for IP Toll Free (IP TF) calls that allow seamless internetworking between IP or TDM Customer's Contact Centers and the Verizon SS7 (PSTN) and IP Networks. The key benefits are:

- Network-level IP IVR functionality, native-IP call processing in the network.
- Hybrid Call Processing, seamless call treatment, routing and transfers for TDM and IP.

Like PSTN network-based IVR applications such as ECR, IP IVR provides significant call processing cost control by pre-processing calls in the network before they tie up CPE and agent resources. Indeed, IP allows the customer greater levels of virtual call processing, due to the inherent separation of the signaling and media (speech, video etc.); however, the native IVR functionality of IP IVR continues to provide cost control and value because it extends the application capabilities of a customer's IP Call Center without protocol translations particularly for multiple contact centers.

Notice that:

- It is now possible for existing ECR Customers to support IP Enabled Contact Centers – without having to use IPIVR, by adding Toll Free terminations to their existing applications via NGSN IP Terms. ECR applications with TDM and NGSN IP Terms support existing ECR features with TDM and IP functionality –depending of the termination type used.
- When NGSN IP Terminations are ordered for the call flow of a new or existing ECR application, NGSN retains TDM call control and runs the application using existing ECR feature functionality. In addition, the IP functionality of calls that are extended to IP Terminations is supported by IPTF –under the control of the Verizon Business IP Enabled Contact Center network architecture. This means that once extended to the IP network, a call can be transferred by NGSN via TNT using DTMF or it can be transferred using SIP Refer IP via IPTF.

1.1.2.1 IP IVR Hybrid Call Processing

In the same way that IP TF provides seamless internetworking between IP and PSTN, IPIVR provides seamless call treatment, routing and agent transfers; across hybrid, TDM and IP enabled, contact center locations. IP IVR provides unifying access to all agent and information resources and makes it possible for customers to migrate to IP at their own pace while continuing to deliver the highest service levels. IP IVR is tightly integrated with PSTN and IP TF and extends the capabilities of Verizon Toll Free service – including Network Call Redirect to overflow traffic using both SIP error and ISUP (ISDN User Part) failure conditions. It provides single-call treatment and comprehensive QOS (Quality of Service) whether calls terminate to TDM or SIP enabled locations.

In addition, IPECC customers can now use IP-ICRI to control of network based IPIVR resources to manage the treatment, routing and queuing of their calls –before, during or after they are handled by their Contact Center Agents. When using IP-ICRI, Customers can control the way in which Verizon's IPIVR answers each of their calls, allow calls to be parked within the IP environment until an operator is available, route a given call to a specified operator, or transfer a given call to another operator or specified destination. ICR-I is also know as Third Party Call Control (3PCC) because it allows the Verizon Business IPIVR Application Server (AS) platforms to connect and communicate with in-network and/or customers premise router equipment to provide call parking and out of band call control features. ICR-I currently supports the use of Intelligent Routing CPE.

The Verizon Business IP IVR solution delivers carrier-grade IVR functionality to complement or replace Customer-owned implementations. IP IVR provides robust call treatment, routing and transferring functionality at the Network level and provides a high level of flexibility and scalability, as well as usage-based costs.

- **IP IVR Service Features**
  - North American Numbering Plan 8XX Originations.
  - International Toll Free Service Originations (ITFS / UFIN).
  - 8xx Gateway DAP translations supported.
  - International IP terminations - where permitted by international and country-specific regulatory/ correspondent agreements.
  - IP and TDM Termination Route Plans.
  - Advanced Toll Free Features, including Network Call Redirect.
  - SIP Refer Call transfers (RFC 3515) (Single Channel Transfers).
  - Small Office / Home Office (SOHO) SIP Device Registration.
  - Intelligent Call Routing Gateway (pre/post call routing) for Cisco and Genesys. Intelligent Call Routers.
  - IP Intelligent Call Routing Integration (pre/post call routing, treatment and queuing) for Cisco and Genesys Intelligent Call Routers.
  - User to User Interface support for Avaya's standards (RFC) recommendation.
  - Full compatibility with Enhanced Call Routing Features:
    - Menu Routing
    - Message Announcement
    - Busy/Ring-No-Answer (B/RNA) w/Custom-Treatment
    - Standard Database Routing
    - Network Database Routing
    - Announced Connect
    - Caller TakeBack & GiveBack
    - TakeBack and Transfer (TnT)
    - Standard Reports
    - Customized Call Records (CCRs)
    - Survey (CCRs only)
    - Dealer Connect Dealer Connect (Network Database

1.2. Avaya/Verizon Business IPCC SIP Trunk Reference Configuration

A simulated enterprise site consisting of an Avaya SIP telephony solution supporting SIP trunks was connected to a laboratory version of the Verizon Business IPCC 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 trunk interoperability compliance test. All calls involved various Avaya telephones, the Verizon Business IPCC Service and various PSTN telephones.

- Incoming calls to Verizon Business provided DID numbers from PSTN telephones.
- Outgoing calls (Transfers and REFERs) to PSTN and between Avaya Aura™ Communication Manager/SES environments.
- Calls using Avaya 9600 Series IP Telephones with the H.323 firmware configurations.
- Calls using Avaya 9600 Series IP Telephones with the SIP firmware configurations.
- G.729A and G.711MU codecs.
- DTMF tone transmission.
- Direct IP-to-IP media (also known as "shuffling") with IP telephones.
- Avaya Aura™ Communication Manager Network Call Redirection (NCR)
- Avaya Aura™ Communication Manager User to User Information (UUI) based on IETF Internet Draft (draft-johnston-sipping-cc-uui)
- Verizon Network Call Redirection.
- Verizon Business IP TF and IP IVR services.
- Single site Avaya Aura™ Communication Manager/SES environment (firewall with NAT).
- Dual site Avaya Aura™ Communication Manager/SES environment (no firewall)

Figure 1 illustrates the reference configuration with the Avaya SIP telephony solution connected to Verizon Business IPCC Service using SIP trunks. This reference configuration was used during the DevConnect compliance testing and for these Application Notes.

Two customer locations were simulated; CustomerA and CustomerB. Both of these simulated customer locations accessed the Verizon Network via the Internet through a single edge router.

In the reference configuration, the Verizon Business IPCC Service typically includes managed IP access service. This managed service may include Verizon Business provided and administered routers and firewalls. The edge router was configured, per Verizon Business specifications, to provide IPSEC tunnels over the Internet between the CPE router and the Verizon Business IPCC network. Separate access lists were defined for each simulated customer.

The CustomerA site included a Cisco PIX firewall providing Network Address Translation (NAT) and SIP Application Layer Gateway (ALG) functionality. No firewall was used at the CustomerB location.

Verizon Business provided a Fully Qualified Domain Name (FQDN) for each simulated customer. These FQDNs were used to define the SIP and DNS domains in the SES (**Section 4.1.7**) as well as the Authoritative Domain in the Avaya Aura™ Communication Manager local SIP Trunk Signaling Group (**Section 3.1.7**).

Note - For security reasons the actual FQDNs are not specified in these Application Notes. Sample domain names are used instead.

- **CustomerA – customer.ay.com**

CustomerA represents a single site location that incorporates a firewall. The firewall provides static and port based Network Address Translation (NAT).

  - An Avaya S8300 Server running co-resident Avaya Aura™ Communication Manager/Avaya Aura™ SIP Enablement Services (SES) 5.0
  - Avaya G700 Media Gateway controlled by Avaya Aura™ Communication Manager
  - Digital line card
  - Avaya 4600 and 9600 Series IP telephones using SIP software.
  - Avaya 4600 and 9600 Series IP telephones using H.323 software.
  - Avaya 6400 Series digital phones.

- **CustomerB – customer.be.com**

CustomerB represents a customer with two locations. Both locations are controlled by a single Avaya Aura™ Communication Manager.

  - Site 1
    - Avaya S8720 Servers running Avaya Aura™ Communication Manager 5.1.1
    - Avaya S8720 Server running Avaya Aura™ SIP Enablement Services (SES) 5.1.1
    - Avaya G650 Media Gateway controlled by Avaya Aura™ Communication Manager
    - Control LAN (C-LAN - signaling) and Media Processor (MedPro – media) cards.
    - Digital line card.
    - Avaya 9600 Series IP telephones using SIP software.
    - Avaya 6400 Series digital phones.
  - Site 2
    - Avaya G650 Media Gateway controlled by Avaya Aura™ Communication Manager
    - Control LAN (C-LAN - signaling) and Media Processor (MedPro – media) cards.
    - Avaya 4600 Series IP telephones using H.323 software.
    - Avaya 6400 Series digital phones.

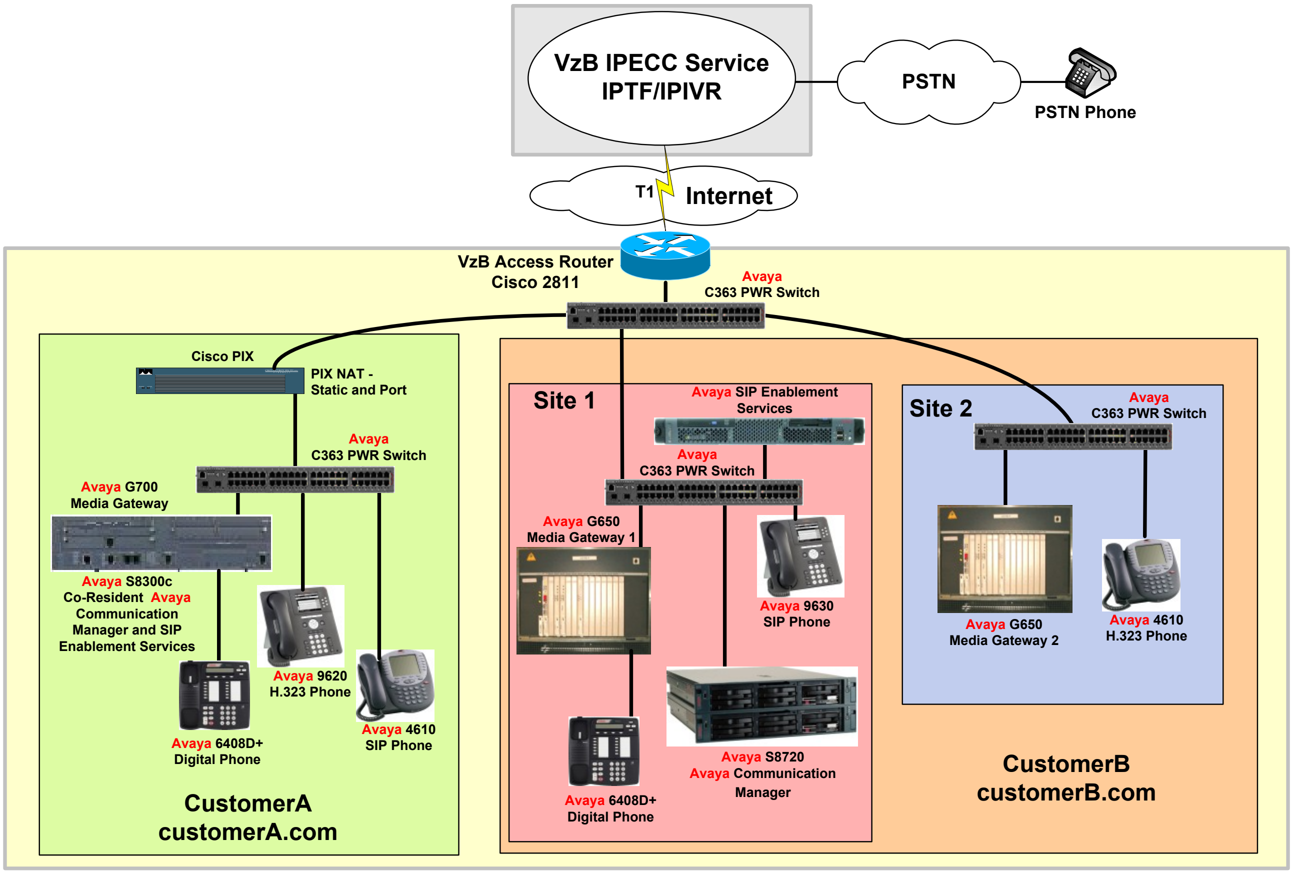

Figure 1: Avaya/Verizon Business IPCC Service Reference Configuration.

1.2.1 FQDNs and Domains

The terms FQDN and Domain are sometimes used interchangeably even though they have different meanings. Typically *domain* means just that – a domain name such as *company.com*. An FQDN typically includes a *host name* with the domain – *big.company.com*. In the reference configuration, Verizon supplied FQDNs to be used by both CustomerA and CustomerB as well as supplying an FQDN for the Verizon service node for use in DNS queries. For security reasons these FQDNs are not specified in these application notes and examples are used instead.

1.3. Verizon Business IP TF and IP IVR Call Flows

The Verizon Business IPCC products (IP TF and IP IVR) work by extending the capabilities of the Toll Free network. The customer continues to dial a Toll Free Number (8XX, ITFS or UFIN) into the Public Switched Telephone Network (PSTN); however, instead of routing the call to a TDM Logical Termination (LogTerm), the call is directed to a new termination type called a Universal Resource Indicator (URI).

When the Verizon Network receives the Toll Free call, the processing Switch sends a routing request (including call information such as Dialed Number, ANI, etc.) to the network Service Control Point (DAP). The DAP collects the switch information and the route plan for the dialed number before returning a routing decision. If no URI is involved the call proceeds as a normal Toll Free call. If the route plan includes a URI, the DAP stores the call information in a transient database (Transient Data Store) assigns it token value and routes the call to a Packet Voice Network Gateway.

When the Network Gateway receives the call request and the assigned token value, it sends a SIP INVITE message to the Service Controller (SC), which sends an acknowledgement to the Gateway, retrieves the call information from the transient database (via the assigned token value), and uses the information to make a routing decision, i.e.: send call to a customer's SIP end point or the IP IVR for treatment before termination to a customer location.

Upon receiving the SC routing decision, the Network Gateway, establishes a two way media channel with the terminating SIP end point, after the appropriate coding is negotiated between the SIP end points (G.711 or G.729a) the processing of media (speech) packets begins.

Calls are physically internetworked (SS7 to IP and IP to SS7) by the IP CC Network Gateways proxied by the Service Controller. The Network Gateways convert calls to SIP (RTP packets) and activate media connections to IP IVR or customer SIP terminations (IDA, PIP) that are compliant with IETF RFC 3261, and which may be located in many locations around the world (subject to regulatory restrictions)

The Verizon Business IP TF service only supports inbound dialing (PSTN to CPE). However IP TF does support outbound calls from the CPE if they are part of either an attended transfer or an unattended (blind) transfer.

Attended transfer is when the transferring party (CPE) establishes a dialog to the third party, then completes the transfer and drops off. In this case two voice paths (PSTN to CPE and CPE to third party) have been established prior to the completed transfer. Attended transfers are typically made from a CPE attendant's phone.

Unattended transfers occur when the transferring party (CPE) completes the transfer and drops off prior to the third party answering the call. In this case only one voice path (PSTN to CPE) has been established prior to the completed transfer. Unattended transfers may also be initiated via the Avaya Network Call Redirection (NCR) feature or via a CPE phone.

Attended and unattended transfers are not permitted out to PSTN.

1.3.1 Verizon Business IP Toll Free (IP TF) Call Flow

The following details a PSTN Toll Free Call to a SIP Termination, including the set up of an IP TF route plan with an IP termination in INCP setup via NetCAP:

1. Customer defines route plan with an IP termination using Toll Free Network Manager.
2. User places a PSTN Toll Free (8xx) call.
3. 8xx call is sent from the Class-5 Switch with calling party number.
4. Class-3 Switch queries the Verizon SCP (DAP) for routing instructions on the Verizon-owned 8xx number. Customer has Route Plan with an assigned IP termination. Info about customer and SS7 signaling (token) is stored in temporary database (TDS)
5. DAP returns a termination switch/trunk along with the assigned token value to the Class-3 Switch.

6. Class-3 Switch routes the call with the information returned from the DAP to an IP CC Network Gateway.
7. The Network Gateway formulates a SIP message with the calling ANI and Called (assigned token) party numbers
8. The Verizon Business IP network routes the SIP message to Verizon Service Controller
9. The Service Controller uses the assigned token back to get the TDS call info.
10. Information (token) temporarily stored in the TDS is returned to the SIP Service Controller. Info includes Originating Switch/Trunk, Intended URI for termination and Originating Line Information.
11. The service controller formulates a new SIP message with the intended customer URI and sends it back to the IP network.
12. The Verizon Business IP network queries a DNS server to obtain the intended customer URI termination.
13. The Verizon Business IP network will route the call to the intended customer termination.

1.3.2 Verizon Business IP Integrated Voice Response (IP IVR) Call Flow

The following details a PSTN inbound (IP TF) Call to IP IVR. This is a leg-1 Example. Leg-three examples would include: IP IVR Outdials to IP terminations (same as IP TF call extension) and IP IVR Outdials to DAL and DDD terminations which require IP to TDM protocol conversion.

1. The Initial 8xx call terminates on Class-3 (C-3) Switch which does a DAP Lookup. The DAP resolves the 8xx number to an EVS LT that terminates at an IPCC site, and proceeds with token processing
2. The DAP saves token information to TDS
3. The call is routed to a Network Gateway (GW) with the assigned token value
4. The GW proxies the call to a Service Controller (SC)
5. The SC resolves token data and determines the call is destined for IP IVR. The SC proxies the call through CMS (Common Media Services) routing app
6. CMS sends the call to the IPIVR Application Server. IPIVR accepts the Call. CMS acquires resources. CMS sends an Invite to a MS to secure resources.
7. The MS responds with OK which CMS forwards through to the Service controller completing set up of the inbound leg.
8. The application now can control media interaction on the inbound leg in order to prompt the user and interact to collect information.
9. When the application determines an outbound (agent) leg is required, it makes a request through CMS for an outbound call.
10. CMS will interact with the media server to obtain the necessary resources.
11. Once resources are acquired, CMS then sends a SIP invite to the terminating party via the SC
12. The SC proxies the request/response to GW and back to C-3. Once call setup is competed, CMS sends an indication to IPIVR and IVR with the agent leg can begin.
13. Typically, IPIVR will conference the calling party and agent together while monitoring for transfer sequences (either DTMF or SIP refer initiated)

1.4. Verizon Network Call Redirection (NCR)

Verizon's advanced Toll Free feature used to redirect or overflow calls in real-time according to outage, busy, or other customer-specific conditions, can now be used with IP Toll Free (IPTF) and IP Interaction Voice Response (IPIVR) services to provide seamless routing and overflow for both

TDM and SIP end-points. By adding IP Toll Free and IPIVR terminations to their NCR tables, customers can redirect traffic using both SIP error and ISUP (ISDN User Part) failure conditions. NCR's call-by-call overflow functionality is programmed via Toll Free Network Manager (TFNM), a web-based application.

Terminations provisioned with NCR can support up to 5 overflow attempts (aka hops); for each hop, customers can define a different routing choice for each individual TDM cause value or SIP error code. In addition, when customers provision their alternate routing choices, they also have the option of using any of the standard Verizon Toll Free routing decision nodes available (listed below), such as percent allocation, day of week routing, time of day routing, etc.

This means that customers can use Percentage Allocation routing to distribute traffic evenly across 3 sites and, when a site is down or experiencing a time out, they can use an NCR hop with a percent allocation decision node to automatically load-balance between their two remaining sites.

The example above is simple but representative of the customer value that can be provided by the new Verizon NCR capabilities. Previously, NCR did not support IP end points and could only be provisioned to allow a single alternate route per error code so that customers could not support IP end points seamlessly and, when an endpoint failed, only one site could be used as an alternate route. Release 7.2 of NCR provides IP support, multiple routing alternatives for each call hop and thus allows customers to optimize their contact center resources, whether for domestic or global "follow the sun" applications.

In the reference configuration, IP IVR was provisioned so that the PSTN dials an IP IVR toll free access numbers to reach the CPE. Once a conversation is established, the CPE may transfer the call by dialing an access code. This access code is intercepted by the Verizon Business IP IVR service application server, the PSTN user is placed on hold, and the CPE may dial a new destination (attended) or Avaya NCR may dial a new destination (unattended). Manual attended and unattended transfer calls are also possible, however IP IVR must be provisioned to permit them.

1.5. Avaya Network Call Redirection (NCR)

Avaya NCR is a feature enabled on Avaya Aura™ Communication Manager via licensing and system provisioning. Avaya NCR utilizes vectoring to generate an announcement followed by the call redirection. The call redirection is accomplished with a SIP REFER.

1.6. Dialed Digit Manipulation

Customers may provision various private dial plans. To simulate this, CustomerA was provisioned with a 4 digit private dial plan and CustomerB was provisioned with a 5 digit private dial plan in the reference configuration. Avaya Aura™ Communication Manager provides all digit manipulation between the Verizon Business DID numbers and the local CustomerA and CustomerB dial plans (see **Sections 3.1.2, 3.1.9, 3.2.4, and 3.2.7**). It should be noted that although the PSTN phone uses 11 digit dialing (1+10 digit DID number) to access both Verizon Business IP TF/IP IVR destinations, Verizon Business IP TF service delivers 7 digits to the CPE while the Verizon Business IP IVR service delivers 10 digits to the CPE.

The dial plans used in the reference configuration are as follows.

- PSTN inbound dialing Verizon Business IP TF/IP IVR DID numbers→ 11 digit dialing – 1xxx-xxx-xxxx
- Verizon Business IP TF inbound INVITE to CPE → 7 digits
- CPE Avaya NCR outbound REFER to IP TF → 7 digits
- CPE Avaya outbound Attended Transfer INVITE to IP TF→ 7 digits
- Verizon Business IP IVR inbound INVITE to CPE → 10 digits
- CPE Avaya NCR outbound REFER to IP IVR → 10 digits
- CPE outbound Attended Transfer INVITE to IP IVR→ 10 digits
- CompanyA local extensions – 4 digits
- CompanyB local extensions – 5 digits

1.7. Avaya SIP Trunks

Avaya SIP trunks provisioned to the Verizon Business IPCC Service works with any combination of Avaya analog, digital, H.323 and SIP telephones. In the Avaya SIP trunk architecture, the Avaya Aura™ SIP Enablement Services acts as a SIP proxy through which all incoming and outgoing SIP messages flow to the Verizon Business IPCC Service. There is no direct SIP signaling path between the Verizon Business IPCC Service and Avaya Aura™ Communication Manager or Avaya SIP endpoints.

For incoming calls, the Avaya Aura™ SIP Enablement Services uses Media Server Address Maps to direct the incoming SIP messages to the appropriate Avaya Aura™ Communication Manager. Once the message arrives at Avaya Aura™ 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 Aura™ Communication Manager and are subject to outbound features such as Automatic Route Selection, digit manipulation and class of service restrictions. Once Avaya Aura™ Communication Manager selects a SIP trunk for the outbound call, the SIP signaling is routed to the Avaya Aura™ SIP Enablement Services. Based on how the Avaya Aura™ Communication Manager SIP trunk is defined, the SES may use Host Address Maps, Outbound Proxy, or DNS lookups (independently or in conjunction with each other) to direct the outbound SIP messages.

1.7.1 Domain Based Routing for Outbound calls

In the reference configuration, the SES uses DNS lookups to direct outbound calls to proper the Verizon Business IPCC destination. This method is referred to as *Domain Based Routing*. When Domain Based Routing is used, Avaya Aura™ Communication Manager specifies a foreign SIP Fully Qualified Domain Name (FQDN) in the outbound SIP trunk provisioning. The FQDN used for the outbound SIP trunk was the domain of the Verizon Outbound Proxy server. This foreign domain is specified as the destination for the INVITE sent by Avaya Aura™ Communications Manager to the SES. The SES then does a DNS query for this foreign domain and forwards the INVITE based on the DNS response. When Domain Based Routing is used, no other outbound call route provisioning is required for the SES other than provisioning a DNS IP address(s). This provides the most flexible outbound route provisioning for the CPE as all the call route decisions take place in the Verizon Business IPCC network.

Note – There is an alternative method for configuring Domain Base Routing where Avaya Aura™ Communication Manager that *does* use Avaya Aura™ SIP Enablement Services Host Maps. This alternative method for configuring Domain Base Routing is described in reference [9].

a. Provisioning Domain Based Routing

i. Avaya Aura™ Communication Manager

1. Avaya Aura™ Communication Manager has the *Authoritative Domain* field of the *ip-network-region* form provisioned with the FQDN of the SES *SIP Domain* (see **Section 4.1.2**). This is the region associated with IP telephones or the region associated with the CLAN/Procr for legacy phones (see **Section 3.1.5**).
    a. Avaya Aura™ Communication Manager will use this value in the INVITE *From* header.
2. The Avaya Aura™ Communication Manager outbound trunk has a *foreign FQDN* specified in the Far-End Domain field of the associated Signaling Group form (see **Section 3.1.7**). *Foreign* means that the FQDN *does not* match the FQDN specified in the *SIP Domain* field of the SES. Avaya Aura™ Communication Manager will use this foreign FQDN in the *Request URI* of the INVITE.
3. Avaya Aura™ Communication Manager sends the INVITE to the SES.

ii. SES

1. The SES compares the *Request URI* of the INVITE received from Avaya Aura™ Communications Manager to its *SIP Domain* and sees that it is *not* authoritative for the foreign FQDN. Therefore, the SES will *not* look at any provisioned *Host Maps*, making them unnecessary.
2. The SES will then do a *DNS SRV lookup* based on the *DNS server(s)* that has been provisioned in the SES (see **Section 4.1.7**. The SES will send the call to the destination specified in the SRV response.
    - **Note** – Alternatively the SES can send the call to an *Outbound Proxy* if one has been provisioned on the SES. If no DNS server or Outbound has been provisioned, the SES will deny the call.

Note – By default the SES will use port 5060 for outbound calls. If a different destination port is required, then Domain Based Routing *should not* be used and SES Host Maps *must be used* to specify this non-standard port. See [4] and [8] for more information on provisioning Avaya Aura™ Communication Manager and the SES to utilize Host Maps for outbound calls.

1.8. Reference Configuration Sample Call Flows

To better understand how calls are routed between the PSTN and the CPE sites used in the reference configuration (see **Figure 1)**, several call flows are described in this section.

Note – The following sections are examples. Many other combinations of these call flow scenarios are possible. Not all protocol exchanges are described.

1.8.1 Verizon Business IP TF Inbound Call

Given:

- Verizon Business IP TF toll free numbers are defined and the Verizon Business IP TF network is provisioned to deliver calls to the appropriate Avaya CPE environments.

Then:

1. PSTN calls a CustomerA IP TF toll free number.
2. Verizon delivers an INVITE to CustomerA Avaya Aura™ SIP Enablement Services which forwards the call to CustomerA Avaya Aura™ Communication Manager.
3. Avaya Aura™ Communication Manager rings appropriate station.
4. PSTN connected to CustomerA station.

1.8.2 Verizon Business IP TF Unattended Transfer (Inbound and Outbound call) with Avaya Network Call Redirection (NCR)

Given:

- Avaya Aura™ Communication Manager at the CustomerA location is provisioned with a Vector Directory Number (VDN) and a vector that first generates an announcement saying "Welcome to CustomerA" and then generates a REFER specifying the DID number of a CustomerB station in the *Refer-To* header.
- CustomerA Avaya Aura™ Communication Manager *inbound* SIP trunk form is provisioned with Avaya Network Call Redirection enabled.
- Verizon Business IP TF toll free numbers are defined and the Verizon Business IP TF network is provisioned to deliver calls to the appropriate Avaya CPE environments.

Then:

1. PSTN calls a CustomerA IP TF toll free number.
2. Verizon delivers an INVITE to Customer A Avaya Aura™ SIP Enablement Services which forwards the call to CustomerA Avaya Aura™ Communication Manager.
3. Avaya Aura™ Communication Manager maps the incoming call to the VDN/vector.
4. The vector generates the announcement to PSTN.
5. The vector generates a REFER specifying a CustomerB station and Avaya Aura™ Communication Manager sends the REFER to the Avaya Aura™ SIP Enablement Services.
6. The Avaya Aura™ SIP Enablement Services forwards the REFER to the Verizon network.
7. The Verizon network accepts the REFER and generates an INVITE to CustomerB for the station specified in the REFER.
8. CustomerB Avaya Aura™ SIP Enablement Services forwards the call to CustomerB Avaya Aura™ Communication Manager.
9. CustomerB Avaya Aura™ Communication Manager rings the appropriate station.
10. PSTN connects with CustomerB station.

1.8.3 Verizon Business IP TF Attended Transfer (Inbound and Outbound call) – CPE Station

Two Attended Transfer methods are possible. One is based on a new INVITEs being sent to Verizon. The other is based on a REFER (with replaces) being sent to Verizon.

1.8.3.1 Attended Transfer INVITE method

Given:

- Verizon Business IP TF toll free numbers are defined and the Verizon Business IP TF network is provisioned to deliver calls to the appropriate Avaya CPE environments.

Then:

1. PSTN calls a CustomerA IP TF toll free number.
2. Verizon delivers an INVITE to Customer A Avaya Aura™ SIP Enablement Services which forwards the call to CustomerA Avaya Aura™ Communication Manager.
3. Avaya Aura™ Communication Manager rings station 1.
4. PSTN is connected to the CustomerA station 1.
5. The station 1 presses the transfer button, the PSTN phone is placed on hold, and the station 1 receives dial-tone.
6. The station 1 dials another IP TF number associated with CustomerB station 2.
7. Avaya Aura™ Communication Manager sends an INVITE to Avaya Aura™ SIP Enablement Services.
8. The Avaya Aura™ SIP Enablement Services forwards the INVITE to the Verizon network.
9. The Verizon network sends the INVITE to CustomerB.
10. CustomerB Avaya Aura™ SIP Enablement Services forwards the call to CustomerB Avaya Aura™ Communication Manager.
11. CustomerB Avaya Aura™ Communication Manager rings station 2.
12. Stations 1 and 2 are connected.
13. Station 1 presses the transfer button again and hangs up.
14. PSTN and station 2 are connected.

1.8.3.2 Attended Transfer REFER with Replaces method

Note - This method is currently in development and will be available in the next release of Avaya Aura™ Communication Manager. It is described here for illustrative purposes.

Given:

- CustomerA Avaya Aura™ Communication Manager *outbound* SIP trunk form is provisioned with Network Call Redirection enabled.
- Verizon Business IP TF toll free numbers are defined and the Verizon Business IP TF network is provisioned to deliver calls to the appropriate Avaya CPE environments.

Then:

1. PSTN calls a CustomerA IP TF toll free number.
2. Verizon delivers an INVITE to CustomerA SES which forwards the call to CustomerA Avaya Aura™ Communication Manager.
3. CustomerA Avaya Aura™ Communication Manager rings station 1.
4. PSTN is connected to the CustomerA station 1.
5. The station 1 presses the transfer button, the PSTN phone is placed on hold, and the station 1 receives dial-tone.
6. The station 1 dials another IP TF number associated with CustomerB station 2.
7. Avaya Aura™ Communication Manager sends an INVITE to the SES.
8. The SES forwards the INVITE to the Verizon network.
9. The Verizon network sends the INVITE to CustomerB.
10. CustomerB SES forwards the call to CustomerB Avaya Aura™ Communication Manager.

11. CustomerB Avaya Aura™ Communication Manager rings station 2.
12. Stations 1 and 2 are connected.
13. Station 1 presses the transfer button again.
14. CustomerA Avaya Aura™ Communication Manager sends a REFER with Replaces to the Verizon Network.
15. Verizon sends an INVITE with Replaces to CustomerB station 2
16. Verizon sends a BYE to CustomerA station 1
17. PSTN and CustomerB station 2 are connected.

1.8.4 Verizon Business IP IVR - Inbound call

Given:

- Verizon Business IP IVR toll free numbers are defined and the Verizon Business IP TF network is provisioned to deliver calls to the appropriate Avaya CPE environments.
- Verizon Business IP IVR service is provisioned to "intercept" a call when a *8 DTMF dial code is detected.

Then:

1. PSTN calls a CustomerA IP IVR toll free number.
2. Verizon delivers an INVITE to Customer A Avaya Aura™ SIP Enablement Services which forwards the call to CustomerA Avaya Aura™ Communication Manager.
3. Avaya Aura™ Communication Manager rings station 1.
4. PSTN is connected to the CustomerA station 1.
5. The station 1 presses *8 plus a CustomerB IP TF number associated with CustomerB station 2. The PSTN phone is placed on hold.
6. The Verizon network sends the INVITE to CustomerB.
7. CustomerB Avaya Aura™ SIP Enablement Services forwards the call to CustomerB Avaya Aura™ Communication Manager.
8. CustomerB Avaya Aura™ Communication Manager rings station 2.
9. Stations 1 and 2 are connected.
10. Station 1 hangs up.
11. PSTN is taken off hold and is connected to station 2.

1.8.5 Verizon Network Call Redirection (NCR) – Inbound dialing

Verizon NCR is an application that allows customers to customize Verizon network responses to various network call messaging (e.g, busy signal, SIP call clearing codes, etc). Since all the possible interactions are beyond the scope of this document, the following basic Verizon NCR scenario is described.

Given:

- Verizon NCR provisioned to call the IP TF DID number for a CustomerB station when CustomerA returns a SIP ***403 forbidden*** message.

Then:

1. PSTN calls an IP TF number that is not assigned on CustomerA Avaya Aura™ Communication manager.
2. CustomerA Avaya Aura™ Communication manager returns a SIP ***403 forbidden*** message back to the Verizon network.
3. The Verizon NCR application detects the 403 and redirects the call by sending an INVITE to the CustomerB station IP TF DID number.

1.8.6 History Info and Diversion Headers

The Verizon Business IPCC Services suite does not support SIP History Info Headers or Diversion Headers. Therefore, in the reference configuration Avaya Aura™ Communication Manager was provisioned ***not*** to send History Info Headers or Diversion Headers (see **Section 3.1.7**).

1.9. Known Limitations

The following limitations are noted for the reference configuration described in these Application Notes:

- These Application Notes describe the Domain Based Routing method for directing outbound calls from Avaya Aura™ SIP Enablement Services. In certain configurations this method will result in the absence of a PAI header for INVITEs sent by Avaya Aura™ SIP Enablement Services. If the PAI header is required for network admission control, then the Avaya Aura™ SIP Enablement Services Host Map method of Domain Based Routing described in [**11**] should be used.
- Avaya Aura™ Communication Manager sends SIP 180 RINGING messages with SDP. Although this does not meet the Verizon Business Product Integration Requirements [**8**], no impact to call processing was observed.
- Verizon Business IPCC Services suite does not support fax.
- Verizon Business IPCC Services suite does not support History Info or Diversion Headers.
- Verizon Business IP Trunking service does not support G.729B codec.

2. Equipment and Software Validated

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

Equipment	Firmware	Software
Avaya S8720 Server	-	-
Avaya S8300c Server	-	-
Avaya G650 Media Gateway IPSI – TN2312BP CLAN – TN799DP MedPro – TN2302AP	 HW12 FW46 HW1 FW32 HW20 FW120	 - - -
Avaya Aura™ Communication Manager		R015x.02.0.947.3 SP1 - 02.0.947.3-17294
Avaya Aura™ SIP Enablement Services (SES)	-	ses-5.2.0.0-947.3b
Avaya 4610 SW IP Telephones	-	2.9.1 (H.323)
Avaya 9620 and 9630 IP Telephones	-	2..4.1.0 (SIP)
Avaya 6408D+ Digital Phones	-	-

Table 1: Equipment and Software Used in the Tested Configuration

Note - The solution integration validated in these Application Notes should be considered valid for deployment with Avaya Aura™ Communication Manager release 5.2.1 and Avaya Aura™ SIP Enablement Services release 5.2.1. Avaya agrees to provide service and support for the integration of Avaya Aura™ Communication Manager release 5.2.1 and Avaya Aura™ SIP Enablement Services release 5.2.1 with Verizon Business IPCC Services Suite, in compliance with existing support agreements for Avaya Communication Manager release 5.2 and Avaya SIP Enablement

Services release 5.2, and in conformance with the integration guidelines as specified in the body of this document.

3. Configure Avaya Aura™ Communication Manager SIP Trunks

This section describes the steps for configuring Avaya Aura™ Communication Manager with the necessary signaling and media characteristics for the SIP trunk connection with the Verizon Business IPCC Service.

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

As described in **Section 1**, two customer locations were created for the reference architecture, CustomerA and CustomerB.

CustomerA used an Avaya Aura™ Communication Manager Co-Resident (CoRes) configuration where the Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services applications both run on an S8300 server located in an Avaya G700 Media Gateway. This location contained Avaya SIP, H.323, and digital endpoints.

CustomerB used Avaya Aura™ Communication Manager running on an Avaya S8720 dual server platform. Collocated with these servers is an Avaya G650 Media Gateway containing a C-LAN signaling processor card, a MedPro media processor card, and an IPSI controller card for communicating to the Avaya S8720 servers. CustomerB also used a second site that contained a second Avaya G650 Media Gateway containing a C-LAN signaling processor card, a MedPro media processor card, and an IPSI controller card for communicating to the S8720 servers in the primary site. Both of these sites contained Avaya SIP, H.323, and digital endpoints.

Note - As the CustomerB configuration is more complex, these Application Notes will focus on the CustomerB provisioning. Unless otherwise noted, the configuration of the CustomerA Avaya Aura™ Communication Manager is similar. See **Section 3.2** for CustomerA specific provisioning of Avaya Aura™ Communication Manager.

Note – The Avaya Aura™ Communication Manager commands described in these Application Notes were administered using the System Access Terminal (SAT). SAT commands are shown in ***bold italics***. SAT form names and options are shown in **bold**. SSH was used connect to SAT via the appropriate IP address, login and password.

3.1. System Capacity and Features

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

3.1.1 Verify System Capacity and Features

On **Page 2** of the ***display system-parameters customer-options*** form, verify that the **Maximum Administered SIP Trunks** is sufficient for the combination of trunks to the Verizon Business IPCC Service, trunks for SIP endpoints and any other SIP trunk based applications.

Be aware that for each call from a non-SIP endpoint to the Verizon Business IPCC Service one SIP trunk is used for the duration of the call. However, due to Avaya SIP endpoint configuration, each Avaya SIP endpoint on a call with the Verizon Business IPCC Service requires two SIP trunks for the duration of the call.

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

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 100     0
           Maximum Concurrently Registered IP Stations: 12000 1
             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     40
  Maximum Administered Ad-hoc Video Conferencing Ports: 0       0
   Maximum Number of DS1 Boards with Echo Cancellation: 0       0
                             Maximum TN2501 VAL Boards: 10      0
                     Maximum Media Gateway VAL Sources: 10      1
           Maximum TN2602 Boards with 80 VoIP Channels: 128     0
          Maximum TN2602 Boards with 320 VoIP Channels: 128     2
   Maximum Number of Expanded Meet-me Conference Ports: 0       0

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

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

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

```
display system-parameters customer-options                     Page   3 of  10
                                OPTIONAL FEATURES

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

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

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

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

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

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

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

On Page 6 of the **System-Parameters Customer-Options** form, verify that the **ACD** and the indicated **Vectoring** features highlighted in **Figure 5** are enabled.

```
display system-parameters customer-options                    Page   6 of  10
                          CALL CENTER OPTIONAL FEATURES

                           Call Center Release: 5.0

                                ACD? y                        Reason Codes? n
                        BAvaya Aura™ Communication ManagerS (Basic)? n
Service Level Maximizer? n
         BAvaya Aura™ Communication ManagerS/VuStats Service Level? n
Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? n     Service Observing (Remote/By FAC)? y
                  Business Advocate? n              Service Observing (VDNs)? y
                    Call Work Codes? n                             Timed ACW? n
      DTMF Feedback Signals For VRU? n                    Vectoring (Basic)? y
                  Dynamic Advocate? n                 Vectoring (Prompting)? y
       Expert Agent Selection (EAS)? y            Vectoring (G3V4 Enhanced)? y
                           EAS-PHD? y             Vectoring (3.0 Enhanced)? y
                  Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? y
              Least Occupied Agent? n     Vectoring (G3V4 Advanced Routing)? y
          Lookahead Interflow (LAI)? n                     Vectoring (CINFO)? y
Multiple Call Handling (On Request)? n      Vectoring (Best Service Routing)? y
    Multiple Call Handling (Forced)? n                  Vectoring (Holidays)? y
  PASTE (Display PBX Data on Phone)? n                 Vectoring (Variables)? y

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

Figure 5: System-Parameters Customer-Options Form – Page 6

3.1.2 CustomerB Dial Plan

In the reference configuration CustomerB uses 5 digit extensions, 3 digit and 1 digit access code formats.

The dial plan is modified with the ***change dialplan analysis*** command.
On Page 1 of the form:

- For extensions:
  - In the **Dialed String** field enter **3**
  - In the **Total Length** field enter **5**
  - In the **Call Type** field enter **ext**
- For Trunk Access Codes (**TAC**):
  - In the **Dialed String** field enter **1**
  - In the **Total Length** field enter **3**
  - In the **Call Type** field enter **dac**
- For Feature Access Codes (**FAC**) – Feature access from stations:
  - In the **Dialed String** field enter **6**
  - In the **Total Length** field enter **3**
  - In the **Call Type** field enter **fac**.
- For Feature Access Codes (**FAC**) – ARS access:
  - In the **Dialed String** field enter **9**
  - In the **Total Length** field enter **1**
  - In the **Call Type** field enter **fac**

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

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     1            3    dac
     3            5    ext
     6            3    fac
     9            1    fac
```

Figure 6: Change Dialplan Analysis Form – Page 1

3.1.3 Determine Node Names

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

3.1.3.1 CustomerB – Avaya Aura™ Communication Manager and SIP Enablement Services

As described in **Section 1**, CustomerB uses a pair of Avaya S8720 Servers as the Avaya Aura™ Communication Manager Platform. Only one of the servers is active at any given time. Each of these servers has its own IP address for its NIC interface, but there is also a logical "active server" IP address that can be accessed regardless of which server is active. When the ***display node-names IP*** command is issued, the NIC IP address of the active server is displayed as **procr**.

Server 1 NIC IP Address	Server 2 NIC IP Address	Active Server Logical IP Address
10.10.10.11	10.10.10.12	10.10.10.10

Table 2 – CustomerB Avaya Aura™ Communication Manger Server IP Addresses

- **Procr** and **10.10.10.11** are the **Name** and **IP Address** of the active Avaya Aura™ Communication Manager server. Note - If the other server was active, 10.10.10.12 would be displayed.

- **GW1-CLAN1** and **10.10.10.13** are used below as the **Name** and **IP Address** of the C-LAN signaling processor in the G650 Media Gateway at CustomerB Site 1.

- **GW1-MEDPRO1** and **10.10.10.14** are used below as the **Name** and **IP Address** of the Media Processor in the G650 Media Gateway at CustomerB Site 1.

- **GW2-CLAN1** and **11.11.11.11** are used below as the **Name** and **IP Address** of the C-LAN signaling processor in the G650 Media Gateway at CustomerB Site 2.

- **GW2-MEDPRO1** and **11.11.11.12** are used below as the **Name** and **IP Address** of the Media Processor in the G650 Media Gateway at CustomerB Site 2.

- **SES** and **10.10.10.9** are used below as the **Name** and **IP Address** of the Avaya SES located in CustomerB Site 1.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
GW1-CLAN1           10.10.10.13
GW1-MEDPRO1         10.10.10.14
GW2-CLAN1           11.11.11.11
GW2-MEDPRO1         11.11.11.12
SES                 10.10.10.9
default             0.0.0.0
procr               10.10.10.11
```

Figure 7: CustomerB IP Node Names Form

3.1.4 IP-Network-Region Assignments

CompanyB has two locations. Company B Site 1 used ip-network-region 1 and Site 2 used ip-network-region 2. Ip-network-region 3 was used in the provisioning of the inbound and outbound SIP trunks for CustomerA and CustomerB (see **Section 3.1** and **Section 3.2**).

3.1.4.1 CustomerB – IP-Network-Regions

Avaya equipment located in CustomerB Site 1 is assigned to ip-network-region 1 and Avaya equipment in Site 2 is assigned to ip-network-region 2. See **Section 3.1.5** and **Section 3.2.3** for ip-network-region provisioning.

Avaya Component	IP_Network-Region
Site 1 CLAN	1
Site 1 MedPro	1
Site 2 CLAN	2
Site 2 MedPro	2

Table 3 – CustomerB IP Network Regions

In addition ip-network-regions are defined in the SIP Signaling Group Far-end regions (see **Section 3.1.7** and **Section 3.2.5**).

Avaya SIP Trunk Signaling Group	Far-End Network-Region
Trunk 1 – Local SES (SIP Phones)	1
Trunk 2 – Outbound Calls to Verizon (call transfers)	3
Trunk 4 – Inbound Calls from Verizon	3

Table 4 – CustomerB Signaling Group Far-end Regions

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

```
list ip-interface all
                                IP INTERFACES
                                                                        Net
ON Type   Slot  Code Sfx Node Name/      Subnet Mask     Gateway Address Rgn VLAN
                         IP-Address
-- ----   ----  ---- --- --------------- --------------- --------------- --- ----
y C-LAN   01A02 TN799  D GW1-CLAN1       255.255.255.0    10.10.10.1     1   n
                         10.10.10.13
y MEDPRO 01A03 TN2602    GW1-MEDPRO1     255.255.255.0    10.10.10.1     1   n
                         10.10.10.14
y C-LAN   02A02 TN799  D GW2-CLAN1       255.255.255.0    11.11.11.1     2   n
                         11.11.11.11
y MEDPRO 02A03 TN2602    GW2-MEDPRO1     255.255.255.0    11.11.11.1     2   n
                         11.11.11.12
```

Figure 8: IP-Interface IP-Network-Region Assignments – CustomerB

The network-region for an ip-interface may be modified with the ***change ip-interface x*** command where **x** is the board location (e.g., **01a02**).

Note – Board locations can be found by issuing the ***list configuration all*** command.

```
change ip-interface 01a02                                         Page   1 of   3
                                 IP INTERFACES
                 Type: C-LAN
                 Slot: 01A02       Target socket load and Warning level: 400
          Code/Suffix: TN799  D          Receive Buffer TCP Window Size: 8320
     Enable Interface? y                          Allow H.323 Endpoints? y
                 VLAN: n                           Allow H.248 Gateways? y
       Network Region: 1                            Gatekeeper Priority: 5

                                IPV4 PARAMETERS
            Node Name: GW1-CLAN1
          Subnet Mask: /24
    Gateway Node Name: Gateway001
        Ethernet Link: 1
        Network uses 1's for Broadcast Addresses? Y
```

Figure 9: IP-Interface IP-Network-Region Assignment.

3.1.5 Define IP Network Region Parameters

The **IP Network Region** form specifies the parameters used by the Avaya Aura™ Communication Manager components (as described in **Section 3.1.4** and **Section 3.2.3**) and how components defined to different regions interact with each other. The following ip-network-region assignments were used in the reference configuration. Other combinations are possible. Note that the local SIP trunk used region 1 and the inbound and outbound SIP trunks to Verizon used region 3.

Avaya Component	IP-Network-Region
Avaya Aura™ Communication Manager Site 1	Region 1
Avaya Aura™ Communication Manager Site 2	Region 2
Local SIP trunk to SES	Region 1
Outbound SIP Trunk to Verizon Network	Region 3
Inbound SIP Trunk from Verizon Network	Region 3

Table 5: IP-Network-Region Assignments

In addition, specific codecs are used to communicate between these regions. See **Section 3.1.6** for the Codec form configurations.

Inter Region Pair	IP-Codec form used
Region 1 to Region 1	Codec form 1
Region 1 to Region 2	Codec form 1
Region 1 to Region 3	Codec form 2
Region 2 to Region 3	Codec form 2

Table 6: Inter Region Codec Form Assignments

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

3.1.5.1 IP-Network-Region 1

The network regions are modified with the ***change ip-network-region x*** command, where x is the network region number (**Figure 10**).
On Page 1 of the **IP Network Region** form:

- Configure the **Authoritative Domain** field to match the **SIP Domain** name configured on the Avaya Aura™ SIP Enablement Services **System Properties** field (see **Section 4.1.2**). In this configuration, the CustomerB domain name is **customer.be.com**.

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

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

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

Figure 10: IP Network Region 1 – Page 1

On Page 3 of the **IP Network Region** form:

- Define the **Codec Set** used for inter-region communications. **Codec Set *1*** is entered for communications with region **2**. **Codec Set *2*** is used for inter-region communication with region **3**.
- Set the **direct WAN** field to **y**, indicating that devices in each region can directly communicate with each other.
- Set the **WAN-BW-Limits** fields to **NoLimit** indicating that the Inter Network Region Connections are not constrained by bandwidth limits.
- Set the **IGAR** (Inter-Gateway-Alternate-Routing) field to **n** because this field is not used in these Application Notes.

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

                   Inter Network Region Connection Management

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

Figure 11: IP Network Region 1 – Page 3

3.1.5.2 IP-Network-Region 2

Provisioning is the same as **Section 3.1.5.1** except:

On Page 3 of the **IP Network Region** form:

- Define the **Codec Set** used for inter-region communications. **Codec Set 1** is entered for communications with region **1**. **Codec Set 2** is used for inter-region communication with region **3**.

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

                     Inter Network Region Connection Management

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

Figure 12: IP Network Region 2 – Page 3

3.1.5.3 IP-Network-Region 3

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

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

```
change ip-network-region 3                                  Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location:         Authoritative Domain:
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                     RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 13: IP Network Region 3 – Page 1

3.1.5.4 IP-Network-Region 3

Provisioning is the same as **Section 3.1.5.1** except:
On Page 3 of the **IP Network Region** form:

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

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

                    Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video      Intervening      Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC IGAR AGL
 3   1    2     y    NoLimit                                             n
 3   2    2     y    NoLimit                                             n
 3   3    2                                                                  all
 3   4
 3   5
```

Figure 14: IP Network Region 3 – Page 3

3.1.6 Define IP Codec Sets

Two codec sets are defined in the reference configuration. One for local intra customer location calls (CustomerA and CustomerB Site 1), inter customer location calls (CustomerB Site 1 and Site 2), and calls between the Customer locations and the Verizon network.

IP-Codec Form	Codecs Defined (primary/secondary)
Codec Form 1	G.711MU/G.729A
Codec Form 2	G.729A/G.711MU

Table 7: Codec Form Codec Assignments

Note – In this configuration a call to a 46xx SIP telephone will use G.711MU since the 46xx SIP telephones do not support G.729A.

3.1.6.1 Intra Customer Location –IP-Codec-Set 1

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

```
display ip-codec-set 1                                          Page   1 of   2
                          IP Codec Set
    Codec Set: 1

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

Figure 15: Intra Customer location IP Codec Set Form

3.1.6.2 Inter Customer Locations – IP-Codec-Set 2

The choice of codecs here is dependent on the connectivity and available bandwidth between the customer sites. G.729a was picked as the first option with G.711U as the second choice. This order could be reversed and additional codecs can be specified if required. This codec set is associated with ip-network-region 2.

The **IP-Codec-Set** form is modified with the ***change ip-codec x*** command, where ***x*** is the codec form number.
On Page 1 of the form:

- Configure the **Audio Codec** field **1** to **G.729A.**
- Configure the **Audio Codec** field **2** to **G.711MU.**
- Let all other fields default.

```
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
```

Figure 16: Inter Customer Location IP Codec Set Form

3.1.7 Define SIP Trunk Groups

Three SIP trunk groups are necessary to support local, inbound, and outbound call. For each trunk group it is necessary to define a separate SIP signaling group as well a SIP trunk group.

NOTE: For Verizon Business customers utilizing either Verizon's **IP Contact Center** or **IP-IVR** service offers, at least one **Elite Agent license is required** to support the ability to utilize the Network Call Redirection capabilities of those services with Avaya Aura(TM) Communication Manager. This license is required to enable the **System-Parameters Customer-Options** form which contains the **"ISDN/SIP Network Call Redirection"** feature that must be turned **ON** to support Network Call Redirection. Additional details on how to configure Network Call Redirection in Avaya Aura(TM) Communication Manager can be found within the supporting text and figures contained within this section.

3.1.7.1 Define the Local SIP Trunk Group

Create a signaling group and a trunk group (Signaling Group 1 and Trunk Group 1 below) to support the local SIP phone registration and for voice calls between the local Avaya SIP stations and phones (H.323/DCP/analog phones) defined on Avaya Aura™ Communication Manager.

Using the ***add signaling-group x*** 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 Aura™ Communication Manager and

Avaya Aura™ SIP Enablement Services, not the transport method used to the Verizon Business IPCC Service.

- Specify the C-LAN used for SIP signaling (node name **GW1-CLAN1**) and the Avaya Aura™ SIP Enablement Services (node name **SES**) as the two ends of the signaling group in the **Near-end Node Name** and **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in **Figure 7.**
- Specify **5061** in the **Far-end Listen Port** fields.
- Enter the value **1** into the **Far-end Network Region** field. This value is for the **IP Network Region** defined in <u>**Section 3.1.5**</u> **Table 5**.
- Enter the domain name of Avaya Aura™ SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is **customer.be.com**.
- The **Direct IP-IP Audio Connections** field should be set to **y** to allow RTP voice paths to be established directly between IP telephones and the Verizon Business IPCC Service. When compatible conditions exist, this allows the voice packets to follow a direct path between the telephones and the network edge, potentially reducing media processor resources and network usage.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Avaya Aura™ Communication Manager to send DTMF tones using RFC 2833.
- The default values for the other fields may be used.

```
add signaling-group 1                                        Page   1 of  1
                                SIGNALING GROUP
 Group Number: 1              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n
   Near-end Node Name: GW1-CLAN1           Far-end Node Name: SES
 Near-end Listen Port: 5061              Far-end Listen Port: 5061
                                       Far-end Network Region: 1
Far-end Domain: customer.be.com
                                             Bypass If IP Threshold Exceeded? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                      Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Figure 17: Local Signaling Group 1 for SIP Telephones

Configure the **Trunk-Group** form as shown in **Figure 18** and **Figure 19** using the ***add trunk-group x*** command. Trunk Group number 1 is defined for local SIP phone registration and for voice calls between the local Avaya SIP stations and phones (H.323/DCP/analog phones) defined on Avaya Aura™ Communication Manager.

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

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name** such as **Local to SES.**
- Specify an available trunk access code (**TAC**) such as **101**.
- Set the **Service Type** field to **tie.**
- Enter **1** as the **Signaling Group** number. This value was previously determined during the **Signaling Group** configuration specified in **Figure 17.**
- Specify the **Number of Members** supported by this SIP trunk group (e.g., **10**).

```
add trunk-group 1                                          Page   1 of  21
                                TRUNK GROUP
Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: Local_To_SES               COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie          Auth Code? n
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

Figure 18: Local Trunk Group 1 for SIP Telephones – Page 1

On Page 3 of the **Trunk Group** form:

- Set the **Numbering Format** field to **public.** This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 1                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y
                     Numbering Format: public
                                               UUI Treatment: service-provider
                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
```

Figure 19: Local Trunk Group 1 for SIP Telephones – Page 3

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

- Set the **Telephone Event Payload Type** to **101** to match the configuration on the Verizon Business IPCC Services suite offer.
- Set **Network Call Redirection** to **Y**. While this parameter is usually set to support Avaya Network Call redirection features such as REFER, it also enables SIP signaling to be sent when an Avaya station presses Hold (Media Attribute=SendOnly).
- Set **Support Request History?** to **N**.
- Let all other values default (Note - Send Diversion Header defaults to No).

```
add trunk-group 1                                             Page   4 of  21
                              PROTOCOL VARIATIONS
                        Mark Users as Phone? n
               Prepend '+' to Calling Number? n
         Send Transferring Party Information? n
                    Network Call Redirection? y
                       Send Diversion Header? n
                     Support Request History? n
                Telephone Event Payload Type: 101
```

Figure 20: Inbound Trunk Group 1 – Page 4

3.1.7.2 Configure the Public Outbound Trunk Group

Create a signaling group and a trunk group (Signaling Group 2 and Trunk Group 2 below) to support outbound voice calls with the Verizon Business IPCC Service. As mentioned in **Section 1.3**, outbound calls will be the second leg of an Attended or Unattended station transfer.

Using the ***add signaling-group x*** 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 Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services, not the transport method used to the Verizon Business IPCC Service.
- Specify the CLAN used for SIP signaling (node name **GW1-CLAN1**) and the Avaya Aura™ SIP Enablement Services (node name **SES**) as the two ends of the signaling group in the **Near-end Node Name** and **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in **Figure 7**.
- Specify **5061** in the **Far-end Listen Port** fields.
- Enter the value **3** into the Far-end Network Region field. This value is for the **IP Network Region** defined in **Section 3.1.5** **Table 5**.
- Enter the domain name in the **Far-end Domain** field with the domain name provided by Verizon Business Service. In this configuration, the domain name is **vziptf.dns.com.** This FQDN is the domain for the Verizon Outbound Proxy server (supplied by Verizon).
- The **Direct IP-IP Audio Connections** field should be set to ***y*** to allow RTP voice paths to be established directly between IP telephones and the Verizon Business IPCC Service. When compatible conditions exist, this allows the voice packets to follow a direct path between the telephones and the network edge, potentially reducing media processor card resources and network usage.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Avaya Aura™ Communication Manager to send DTMF tones using RFC 2833.
- 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
  IMS Enabled? n
   Near-end Node Name: GW1-CLAN1           Far-end Node Name: ASM
 Near-end Listen Port: 5061              Far-end Listen Port: 5061
                                      Far-end Network Region: 3
Far-end Domain: vziptf.dns.com
                                           Bypass If IP Threshold Exceeded? n
         DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                   IP Audio Hairpinning? n
         Enable Layer 3 Test? y                   Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```

Figure 21: Signaling Group 2 for Public Outbound Calls

Configure the **Trunk-Group** form as shown in **Figure 22** and **Figure 23** using the ***add trunk-group x*** command. Trunk Group number 2 is defined for outbound calls originating from station transfers.

On Page 1 of the Trunk Group form:

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name** such as **Outbound**.
- Specify an available trunk access code (**TAC**) such as **102**.
- Set the **Service Type** field to **public-ntwrk**.
- Enter **2** as the **Signaling Group** number. This value was previously determined during the Signaling Group configuration specified in **Figure 21.**

- Specify the **Number of Members** supported by this SIP trunk group (e.g., **10**).

```
add trunk-group 2                                            Page   1 of  21
                                TRUNK GROUP
Group Number: 2                    Group Type: sip           CDR Reports: y
  Group Name: Outbound               COR: 1        TN: 1        TAC: 102
   Direction: two-way       Outgoing Display? n
 Dial Access? n                              Night Service:
Queue Length: 0
Service Type: public-ntwrk                Auth Code? n
                                                   Signaling Group: 2
                                                 Number of Members: 10
```

Figure 22: Trunk Group 2 for Public Outbound Calls – Page 1

On Page 3 of the **Trunk Group** form:

- Set the **Numbering Format** field to **public.** This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 2                                            Page   3 of  21

TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                     Maintenance Tests? y
                     Numbering Format: public
                                               UUI Treatment: service-provider
                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
 Show ANSWERED BY on Display? y
```

Figure 23: Trunk Group 2 for Public Outbound Calls – Page 3

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

- Set the **Telephone Event Payload Type** to **101** to match the configuration on the Verizon Business IPCC Services suite offer.
- Set **Network Call Redirection** to **Y**. While this parameter is usually set to support Avaya Network Call redirection features such as REFER, it also enables SIP signaling to be sent when an Avaya station presses Hold (Media Attribute=SendOnly).
- Set **Support Request History?** to **N**.
- Let all other values default (Note - Send Diversion Header defaults to No).

```
add trunk-group 1                                    Page   4 of  21
                              PROTOCOL VARIATIONS
                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? y
                    Send Diversion Header? n
                  Support Request History? n
             Telephone Event Payload Type: 101
```

Figure 24: Trunk Group 2 for Public Outbound Calls – Page 4

3.1.7.3 Configure the Public Inbound Trunk Group

Create a signaling group and trunk group (Signaling Group 4 and Trunk Group 4 below) to support inbound calls from the Verizon Business IPCC Service.

The configuration information is similar to that used for the outbound trunk group above. Only the differences are outlined below.

Using the ***add signaling-group x*** command, configure the inbound signaling group form similar to the outbound signaling group described above. The notable differences are:

- Specify **3** as the value for **Far-end Network Region** field**.** This value is for the **IP Network Region** defined in **Section 3.1.5** **Table 5**.
- Leave the **Far-end Domain** field blank. Avaya Aura™ Communications Manager examines the domain specified in the SIP *From* header in the inbound INVITE. This value must match the value specified in the **Far-end Domain** field or the call is denied. Since Verizon may originate the INVITE from many service nodes, the field is left blank which allows any SIP From header domain value.

```
add signaling-group 4
                               SIGNALING GROUP

Group Number: 4              Group Type: sip
                       Transport Method: tls

  Near-end Node Name: GW1-CLAN1             Far-end Node Name: SES
Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                       Far-end Network Region: 3
      Far-end Domain:
                                        Bypass If IP Threshold Exceeded? n
        DTMF over IP: rtp-payload        Direct IP-IP Audio Connections? y
                                                   IP Audio Hairpinning? n
        Enable Layer 3 Test? n
Session Establishment Timer(min): 3
```

Figure 25: Signaling Group 4 for Public Inbound Calls

Configure the **Trunk-Group** form as shown in **Figure 26** and **Figure 27** using the ***add trunk-group x*** command. Trunk Group number 4 is defined for inbound calls originating from the Verizon network.

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

- Choose a descriptive **Group Name** such as **Blank**.
- Specify an available trunk access code (**TAC**) such as **104**.
- Enter **4** as the **Signaling Group** number. This value was previously determined during the Signaling Group configuration specified in **Figure 25.**

```
add trunk-group 4                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 4                    Group Type: sip          CDR Reports: y
  Group Name: BLANK                       COR: 1        TN: 1        TAC: 104
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk                  Auth Code? n

                                                      Signaling Group: 4
                                                    Number of Members: 10
```

Figure 26: Trunk Group 4 for Public Inbound Calls – Page 1

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

- Set the **Telephone Event Payload Type** to **101** to match the configuration on the Verizon Business IPCC Services suite offer.
- Set **Network Call Redirection** to **Y**. While this parameter is usually set to support Avaya Network Call redirection features such as REFER, it also enables SIP signaling to be sent when an Avaya station presses Hold (Media Attribute=SendOnly).
- Set **Support Request History?** to **N**.
- Let all other values default (Note - Send Diversion Header defaults to No).

```
add trunk-group 1                                             Page   4 of  21
                              PROTOCOL VARIATIONS
                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? y
                    Send Diversion Header? n
                  Support Request History? n
             Telephone Event Payload Type: 101
```

Figure 27: Trunk Group 4 for Public Inbound Calls – Page 4

3.1.8 Configure Public Unknown Numbering

In these Application Notes, the extensions on CustomerB Avaya Aura™ Communication Manager use a 5 digit dialing plan with extensions between 300xx. The **Public-Unknown-Numbering** form allows Avaya Aura™ Communication Manager to use these extensions as the calling party number for outbound calls. Otherwise *Anonymous* is displayed as the calling number.

Figure 28 shows the use of the ***change public-unknown-numbering x*** command to send the correct calling party number. The entries below indicate that 5-digit extension numbers beginning with 300 will send the corresponding digits via trunk groups 1 through 4.
Configure the **Public-Unknown-Numbering** form shown below as follows:

- Set the **Ext Len** field to **5**.
- Set the **Ext Code** field to **300**.
- Set the **Trk Grp(s)** field to **1-4**.
- Set the **Total CPN Len** field to **5**.

```
change public-unknown-numbering 3                              Page   1 of   2

                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk        CPN          CPN
Len Code            Grp(s)     Prefix       Len
                                                      Total Administered: 1
 5  300             1-4                     5            Maximum Entries: 9999
```

Figure 28: Public/Unknown Numbering Form

3.1.9 Configure Call Routing

3.1.9.1 Outbound Calls

As described previously, the only outbound calls permitted are for the second leg of a station transfer. In these Application Notes, the Automatic Route Selection feature is used to route calls via the SIP trunk to the Verizon Business IPCC Service, which in turn completes the calls to the destination.

1. Verify that the appropriate extensions are defined in the **Public-Unknown-Numbering** form (see **Section 3.1.8**).
2. Use the ***change dialplan analysis*** command to add **9** as a feature access code (**fac**).
   - Set **Dialed String** to **9**.
   - Set **Total Length** to **1**.
   - Set **Call Type** to **fac**.

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

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
          9        1    fac
```

Figure 29: Dialplan Analysis Form

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

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

Figure 30: Feature-Access-Codes Form – Page 1

4. Use the ***change ars analysis*** command to configure the route pattern selection rule based upon the number dialed following the ARS access digit "9". In this sample configuration, outbound calls are either placed to an IP TF number (1555888xxxx, 1555777xxxx, or 1555852xxxx) or an IP IVR number (1555666xxxx).

- For **IP TF** calls the matching number 1555 is entered on the form, the length of the called number is set to 11 digits, and the call is to be routed to route pattern 2.
    - Set the **Dialed String** field to **1555**.
    - Set the **Total Min** field to **11**.
    - Set the **Total Max** field to **11**.
    - Set the **Route Pattern** field to **2**.
    - Set the **Type** field to **hnpa**.

- For **IP IVR** calls the matching number 1555666 is entered on the form, the length of the called number is set to 11 digits, and the call is to be routed to route pattern 6.
    - Set the **Dialed String** field to **1555666**.
    - Set the **Total Min** field to **11**.
    - Set the **Total Max** field to **11**.
    - Set the **Route Pattern** field to **6**.
    - Set the **Type** field to **hnpa**.

```
change ars analysis 18                                      Page   1 of   2
                           ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    1
         Dialed           Total     Route    Call   Node  ANI
         String          Min  Max  Pattern   Type   Num   Reqd

    1555                 11   11     2       hnpa         n
    1555666              11   11     6       hnpa         n
```

Figure 31: ARS Analysis Form

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

   - For IP TF calls the dialed number 1555xxxxxxx must be reduced to a 7 digit number (IP TF network routing uses a 7 digit format). For example if 15558883221 is dialed, Avaya Aura™ Communication Manager must send 8883221 to the Verizon Business IP TF network. Calls are then sent to trunk 2 which is defined as the outbound SIP trunk (see **Section 3.1.7**).
     - Set the **Grp No** field to **2**.
     - Set the **FRL** field to **0**.
     - Set the **No. Del Dgts** field to **4**.

```
change route-pattern 2                                          Page   1 of   3
                    Pattern Number: 2   Pattern Name: IP TF_7_Digits
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 2    0                    4                                          n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user
     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                          none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
```

Figure 32: Route Pattern 2 – IP TF 7 Digit Dialing

- For IP IVR calls the dialed number 1555666xxxx must be reduced to a 10 digit number (IP TF network routing uses a 10 digit format). For example if 15556663221 is dialed, Avaya Aura™ Communication Manager must send 5556663221 to the Verizon Business IP TF network. Calls are then sent to trunk 2 which is defined as the outbound SIP trunk (see **Section 3.1.7**).
  - Set the **Grp No** field to **2**.
  - Set the **FRL** field to **0**.
  - Set the **No. Del Dgts** field to **1**.

```
change route-pattern 6                                          Page   1 of   3

                    Pattern Number: 6   Pattern Name: IP IVR_10_digits
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 2    0                    1                                        n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

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

Figure 33: Route Pattern 2 – IP TF 10 Digit Dialing

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

```
change locations
                                   LOCATIONS

              ARS Prefix 1 Required For 10-Digit NANP Calls? y

Loc  Name           Timezone Rule  NPA                               Proxy Sel
No                   Offset                                            Rte Pat
1:  Main            + 00:00   0                                         1
```

Figure 34: Locations Form Administration

3.1.9.2 Incoming Calls

This step configures the mapping of incoming numbers to the proper extensions. All incoming calls are processed by Trunk 4 (see **Section 3.1.7**).

As mentioned above, Verizon Business IP TF network will deliver 7 digit numbering and IP IVR will deliver 10 digit numbering in the incoming INVITE messages.

In the reference configuration, the incoming digits received do not match the intended Avaya Aura™ Communication Manager station extension. Therefore, incoming call handling treatment is used to modify the incoming number to the desired extension.

The following mappings were defined in the reference configuration.

	Incoming Digits	Mapped to Extension	Extension Use
IP TF	9995245	30001	Digital station
	9995250	30002	H.323 station
	9995259	30003	SIP station
	9995264	30005	VDN
IP IVR	5556664249	30001	Digital station
	5556664251	30002	H.323 station
	5556664253	30003	SIP station
	5556664256	30005	VDN

Table 8: Incoming Digit Mapping

1. The example below illustrates the technique to assign the incoming IP TF number 9995245 to
   extension 30001 (digital station). All other IP TF numbers are provisioned in the same manner.
   Use the ***change inc-call-handling-trmt trunk-group x*** command to administer this assignment.
   - Enter **7** into the **Called Len** field to match the length of the incoming digits.
   - Enter **9995245** into the **Called Number** field as the digit pattern to be matched.
   - Enter **7** into the **Del** field as the number of digits that should be deleted from the end of the incoming digits.
   - Enter **30001** into the **Insert** field.

2. The example below illustrates the technique to assign the incoming IP IVR number 5556664249 to extension 30001 (digital station). All other IP IVR numbers are provisioned in the same manner.

   Use the ***change inc-call-handling-trmt trunk-group x*** command to administer this assignment.
   - Enter **10** into the **Called Len** field to match the length of the incoming digits.
   - Enter **5556664249** into the **Called Number** field as the digit pattern to be matched.
   - Enter **10** into the **Del** field as the number of digits that should be deleted from the end of the incoming digits.
   - Enter **30001** into the **Insert** field.

```
change inc-call-handling-trmt trunk-group 4                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
 public-ntwrk    10 5556664249       10  30001
 public-ntwrk    10 5556664251       10  30002
 public-ntwrk    10 5556664253       10  30003
 public-ntwrk    10 5556664256       10  30005
 public-ntwrk    7  9995245          7   30001
 public-ntwrk    7  9995250          7   30002
 public-ntwrk    7  9995259          7   30003
 public-ntwrk    7  9995264          7   30005
```

Figure 35: Incoming Call Handling Treatment

3.1.10 Avaya Network Call Redirection (NCR)

Avaya Aura™ Communication Manager supports NCR by using a SIP REFER frame. This REFER is generated by a Vector that is associated with a Vector Directory Number (VDN). An inbound call to the VDN number triggers the vector.

3.1.10.1 Avaya NCR to Verizon Business IP TF

In the following Avaya NCR provisioning example, an incoming call is received by Avaya Aura™ Communication Manager, converted to a VDN number, triggers a Vector, and a REFER is issued back to Verizon containing the new destination IP TF number of 8883221.

1. Configure the **VDN** form as shown in **Figure 36** using the ***add vdn x*** command where **x** is an available extension.

   On Page 1 of the **VDN** form:
   - Choose a descriptive **Name** such as **REFER vector**.
   - Specify an available **Vector Number** such as **101**.
   - All other values are default.

```
   add vdn 30005                                                  Page   1 of   3
                               VECTOR DIRECTORY NUMBER
                                Extension: 30005
                                    Name*: REFER vector
                            Vector Number: 101
                    Meet-me Conferencing? n
                      Allow VDN Override? n
                                      COR: 1
                                      TN*: 1
                                 Measured: none
          VDN of Origin Annc. Extension*:
                               1st Skill*:
* Follows VDN Override Rules
```

Figure 36: Add VDN

2. List available announcements by entering the ***list announcement*** command. An announcement from this list will be used in **Step 4**.

Note – If no announcements are present or you wish to add a new announcement, follow the procedures described in **[7]**

```
   list announcement
                              ANNOUNCEMENTS/AUDIO SOURCES
   Announcement                                                Source      Num
   of
   Extension              Type       Name                     Pt/Bd/Grp
   Files
   21001                  integrated REFER                     001V9        1
   21002                  integrated REFER_fail                001V9        1
```

Figure 37: List Announcements

3. In **Section 3.1.7**, SIP trunk 4 is provisioned as the trunk for inbound calls. Verify that the incoming call handling treatment for SIP trunk 4 maps the called IP TF number 9996264 to the VDN extension 30005.

```
display inc-call-handling-trmt trunk-group 4                    Page   1 of  30
                      INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
 public-ntwrk    10 5556664249       10  30001
 public-ntwrk    10 5556664251       10  30002
 public-ntwrk    10 5556664253       10  30003
 public-ntwrk    10 5556664256       10  30005
 public-ntwrk    7  9995245          7   30001
 public-ntwrk    7  9995250          7   30002
 public-ntwrk    7  9995259          7   30003
 public-ntwrk    7  9995264          7   30005
```

Figure 38: Incoming Call handling Treatment Trunk-Group 4

4. Configure the **Vector** form as shown in **Figure 39** using the ***change vector x*** command, where **x** is the **Vector Number** specified in the **VDN** form shown in **Figure 36**.

   On Page 1 of the **Vector** form:
   - Choose a descriptive **Name** such as **REFER to IP TF**.
   - OPTIONAL - On line **01** enter a description using the format → # **<text>**
     The # character indicates this line is descriptive and not an executable function. In the example, # ***announcement answers before REFER sent*** is entered.
   - On line **02** enter an announcement selected in **Step 2** (e.g, **21001**). This announcement will be heard by the original caller.

Note – Playing an announcement at this point is required. A REFER cannot be generated if an RTP stream is not first established to the caller.

   - OPTIONAL - On line **04** enter a description using the format → # **<text>**
     In the example, # ***REFER occurs*** is entered.
   - On line 05 enter **route-to** and then press the *Tab* key.
   - The word **number** will appear.
   - Press the *Tab* key again and enter **~r<number>**, where <number> is the new destination number you want the call redirected to (e.g, **9995245**)

```
change vector 101                                                 Page   1 of   6
                                  CALL VECTOR

    Number: 101              Name: REFER_to_IP TF
                                            Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 #    annoucement answers bfore Refer sent
02 announcement 21001
03
04 #    Refer occurs
05 route-to     number ~r8883221   with cov n if unconditionally
06
```

Figure 39: Change Vector – Verizon Business IP TF

The previous provisioning results in the following scenario.

a. PSTN dials an IP TF number 15559996264 and the Verizon network sends an INVITE to Avaya Aura™ Communication Manager containing 9996264.
b. Avaya Aura™ Communication Manager receives the call on SIP trunk 4 (inbound trunk) and converts the IP TF called number 9996264 to the VDN extension 30005 via the SIP trunk 4 incoming call handling treatment.
c. VDN 30005 triggers Vector 101.
d. Vector 101 plays announcement 21001 back to PSTN, establishing the required RTP path between Avaya Aura™ Communication Manager and PSTN.
e. Vector 101 generates a REFER containing the IP TF number 8883221, based on the Route-To statement in the Vector.

Note – As described in **Section 3.1.9**, the Verizon Business IP TF network uses a 7 digit dial plan. That is why 8883221 is specified instead of a 11 digit number (15558883221) or a 10 digit number (5558883221).

f. The REFER is issued to Verizon out the same SIP trunk that the INVITE was received in **Step a** (inbound trunk 4). This is why Avaya NCR must be enabled on the inbound SIP Trunk (see **Figure 27**). The REFER contains the IP TF destination number specified in the Vector Route-To statement (e.g, **8883221**).
g. Verizon redirects the call to the IP TF destination 8883221.

3.1.10.2 Avaya NCR to Verizon Business IP IVR

Configuring Avaya NCR for a Verizon Business IP IVR number is the same procedure as described in **Section 3.1.10.1**. However since IP IVR uses a 10 digit dial plan, a 10 digit IP IVR number must be specified in the Vector.

1. Configure a **VDN** using the same procedure shown in **Section 3.1.10.1** specifying the new vector number (e.g **102**).

2. Configure the **Vector** form as shown in **Figure 40** using the ***change vector x*** command, where **x** is the **Vector Number** specified in **Step 1**.

On Page 1 of the **Vector** form:

- On line 05 enter **route-to** and then press the *Tab* key.
- The word **number** will appear.
- Press the *Tab* key again and enter **~r<number>**, where <number> is the new IP IVR destination number you want the call redirected to (e.g, **5556665245**)

```
change vector 102                                                   Page   1 of   6
                                  CALL VECTOR
    Number: 102             Name: REFER_to_IP IVR
                                             Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 #    annoucement answers bfore Refer sent
02 announcement 21001
03
04 #    Refer occurs
05 route-to     number ~r5556665245   with cov n if unconditionally
06
```

Figure 40: Change Vector – Verizon Business IP IVR

3.1.10.3 Avaya NCR to Verizon PSTN Gateway

The Verizon network supports call redirection out to PSTN via Verizon network gateways. Verizon identifies calls to their gateways by a leading "+" character in the dial string.

Configuring Avaya NCR for Verizon PSTN access is the same procedure as described in **Section 3.1.10.1**. However since IP IVR uses a 10 digit dial plan, a 10 digit IP IVR number must be specified in the Vector.

1. Configure a VDN using the same procedure shown in **Section 3.1.10.1** specifying the new vector number (e.g,, **103**).

2. Configure the **Vector** form as shown in **Figure 41** using the ***change vector x*** command, where **x** is the **Vector Number** specified in **Step 1**.

   On Page 1 of the **Vector** form:

   - On line 05 enter **route-to** and then press the *Tab* key.
   - The word **number** will appear.
   - Press the *Tab* key again and enter ***~r<number>***, where **<number>** is the new IP IVR destination number you want the call redirected to (e.g,, +17325551212)

```
display vector 103                                                   Page   1 of   6
                                  CALL VECTOR
    Number: 103             Name: REFER_to_PSTN
                                             Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 #    annoucement answers then NCR Refer
02 announcement 21001
03
04 #    Refer occurs
05 route-to     number ~r+17325551212   with cov n if unconditionally
06
```

Figure 41: Change Vector – Verizon PSTN Gateway

3.1.11 Avaya NCR - User To User Information (UUI)

Avaya Aura™ Communication Manager supports the passing of UUI in SIP signaling. This information may be displayed on an Avaya station using a provisioned button. When a call comes in, and the UUI button is pressed, the UUI information is displayed on the station.

Avaya NCR can be used to generate UUI information and send it out as part of a REFER. This UUI information is defined via a variable and an entry in a vector.

In the following Avaya NCR provisioning example, a string variable is defined and then the variable is used to define the UUI data in the vector. The vector processing is the same as described in **Section 3.1.10.1**.

1. Configure a **VDN** using the same procedure shown in **Section 3.1.10.1** specifying the new vector number (e.g,, **104**).

2. Configure the **Variables** form as shown in **Figure 42** using the ***change variables*** command.

   On Page 1 of the **Variables** form:
   - In the **Description** field enter a name such as **UUIREFER**.
   - In the **Type** field specify **asaiuui**.
   - All other values are default (note that the length field defaults to**16**).

```
change variables                                                Page   1 of  39
                              VARIABLES FOR VECTORS
Var Description                 Type    Scope Length Start Assignment       VAC
A   UUIREFER                    asaiuui L     16     1
B
```

Figure 42: Change Variables

3. Configure the vector form as shown in **Figure 43** using the ***change vector x*** command, where **x** is the vector number specified in **Step 1**. The vector is provisioned in a similar manner to those described in **Section 3.1.10.1** with the following additions. Adding comment lines is optional.

   On Page 1 of the **Vector** form:
   - On line 02 enter **wait-time** and then press the *Tab* key twice.
   - Change the default **silence** setting to **ringback**.
   - Press the *Tab* key again and enter **set**, and then press the *Tab* key.
   - Change the default **digits** setting to **A** and press the *Tab* key twice.
   - Change the default **ADD** setting to **CATR** and press the *Tab* key.
   - Enter a **16** digit UUI information string (e.g, **1234567890123456**)

     Note - 16 is the default length specified in the variable form (**Figure 42**).
   - Enter an **announcemen**t line and a **route-to** line as described in **Section 3.1.10.1**.

```
display vector 104                                              Page   1 of   6
                                  CALL VECTOR
    Number: 104             Name: REFER_UUI
                                             Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 #    NCR REFER with UUI
02 wait-time    2   secs hearing ringback
03
04 #    Define UUI
05 set          A      = none   CATR  1234567890123456
06
07 #    play announcement
08 announcement 21001
09
10 #    REFER
11 route-to     number ~r8883221        with cov n if unconditionally
12
```

Figure 43: Change UUI Vector

Based on the provisioning examples above, when vector 104 is triggered it will issue a REFER to IP TF number **8883221** (IP TF uses 7 digits) and will contain the string **1234567890123456** as UUI data.

3.1.11.1 UUI – Station Button

A UUI display button may be provisioned on a station to display incoming UUI information.

1. Modify the **Class of Restriction** (**COR**) form with the ***change cor x*** command where **x** is the COR assigned to the phone (1 is default).

   On page 2 of the form:
   - Enter **Y** in the Station-Button Display of UUI IE Data field.

```
change cor 1                                                   Page   2 of  23
                              CLASS OF RESTRICTION
                     MF Incoming Call Trace? n
               Brazil Collect Call Blocking? n
                     Block Transfer Display? n
Block Enhanced Conference/Transfer Displays? y
                     Remote Logout of Agent? n
 Station Lock COR: 1       TODSL Release Interval (hours):
  Outgoing Trunk Disconnect Timer (minutes):
      Station-Button Display of UUI IE Data? y
      Service Observing by Recording Device? n
                           ERASE 24XX USER DATA UPON
           Dissociate or unmerge this phone: none
          EMU login or logoff at this phone: none
           Mask CPN/NAME for Internal Calls? n
```

Figure 44: COR - Station Button UUI Data Enabled

2. Add a UUI display button to the station form using the ***change station x*** command.

   On page 4 of the form:
   - Under the **Button Assignments**, enter **uui-info** in an available button field (e.g, **4:**).

```
change station 30003                                          Page   4 of   5
                                     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: cpn-unblk
 2: call-appr                           6: cpn-blk
 3: call-appr                           7: auto-in           Grp:
 4: uui-info                            8: manual-in         Grp:
```

Figure 45: Station Button UUI Info Defined

3.1.12 Call Center Provisioning

Avaya Aura™ Communication Manager supports the provisioning of Call Center agents. Many Call Center functions are supported with Avaya Aura™ Communication Manager; however they are beyond the scope of these Application Notes. The following is a basic example of a Call Center configuration.

1. Add an agent login ID as shown in **Figure 46** with the ***add agent-loginID x*** command, where **x** is an available extension (different than the extension already assigned to the station).
   On page 1 of the form:
   - Enter a name in the **Name field** such as **James Bond**.
   - Enter a password in the **Password** field and again in the **Password (enter again)** field (e.g, **123456**).
   - Let all other fields default.
   - Note – You must log out of SAT and log in again for changes to this form to take effect.

```
add agent-loginID 30007                                    Page   1 of   2
                                AGENT LOGINID
               Login ID: 30007                                  AAS? n
                   Name: James Bond                           AUDIX? n
                     TN: 1                             LWC Reception: spe
                    COR: 1                    LWC Log External Calls? n
          Coverage Path:                    AUDIX Name for Messaging:
          Security Code:
                                         LoginID for ISDN/SIP Display? n
                                                            Password: 123456
                                              Password (enter again): 123456
                                                         Auto Answer: station
                                                   MIA Across Skills: system
                                          ACW Agent Considered Idle: system
                                          Aux Work Reason Code Type: system
                                            Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                            Forced Agent Logout Time:   :
     WARNING:  Agent must log in again before changes take effect
```

Figure 46: Define Agent ID – Page 1

On page 2 of the form:
- On line 1, enter **1** in the **SN** (skill number) and ***1*** in the **SL** (skill level) fields.

- Let all other fields default.

```
add agent-loginID 30007                                         Page   2 of   2
                                 AGENT LOGINID
      Direct Agent Skill:
Call Handling Preference: skill-level              Local Call Preference? n
    SN      SL          SN      SL          SN      SL          SN      SL
 1: 1       1       16:                 31:                 46:
 2:                 17:                 32:                 47:
 3:                 18:                 33:                 48:
 4:                 19:                 34:                 49:
 5:                 20:                 35:                 50:
 6:                 21:                 36:                 51:
 7:                 22:                 37:                 52:
 8:                 23:                 38:                 53:
 9:                 24:                 39:                 54:
10:                 25:                 40:                 55:
11:                 26:                 41:                 56:
12:                 27:                 42:                 57:
13:                 28:                 43:                 58:
14:                 29:                 44:                 59:
15:                 30:                 45:                 60:
```

Figure 47 Define Agent ID – Page 2

2. Configure the **VDN** form as shown in **Figure 48** using the ***add vdn x*** command where **x** is an available extension.

   On Page 1 of the **VDN** form:
   - Choose a descriptive **Name** such as **Avaya Call Center**.
   - Specify an available **Vector Number** such as **110**.
   - All other values are default.

```
add vdn 30008                                                   Page   1 of   3
                             VECTOR DIRECTORY NUMBER
                              Extension: 30008
                                 Name*: Avaya Call Center
                         Vector Number: 110
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: internal
        Acceptable Service Level (sec): 20
        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
* Follows VDN Override Rules
```

Figure 48: Define Call Center VDN

3. Configure the **Vector** form as shown in **Figure 49** using the ***change vector x*** command, where **x** is the **Vector Number** specified in the **VDN** form shown in **Figure 48**.

   On Page 1 of the **Vector** form:

- Choose a descriptive **Name** such as **Call Center**.
- On line **01** enter **wait-time** and press the *Tab* key. The line **2 seconds hearing silence** will appear.
  - Change the value **2** to a **5** and press *Tab* twice.
  - Change the value **silence** to **ringback**.
- On line **02** enter **queue-to** and press *Tab*,
  - Enter **skill** and press *Tab*,
  - Enter **1** and press *Tab*. The value **pri** (priority) will appear. Press *Tab* again.
  - Enter **m** (medium priority)
- On line **03** enter **wait-time** and press the *Tab* key. The line **2 seconds hearing silence** will appear.
  - Change the value **2** to a **10** and press *Tab* twice.
  - Change the value **silence** to **music**.
- On line **04** enter **stop**.

```
change vector 110                                            Page   1 of   6
                                  CALL VECTOR

    Number: 110              Name: Call Center
                                              Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? Y

01 wait-time    5   secs hearing ringback
02 queue-to     skill 1    pri m
03 wait-time    10  secs hearing music
04 stop
05
06
```

Figure 49: Define Call Center Vector

4. Configure the **feature-access-codes** form as shown in **Figure 50**. The access code format is defined in the dial plan described in **Section 3.1.2.**

   On Page 5 of the **feature-access-codes** form:
   - In the **Login Access Code** field enter **650**
   - In the **Logout Access Code** field enter **651**
   - In the **Manual-in Access Code** field enter **652**
   - In the **Auto-in Access Code** field enter **653**
   - In the **Aux Work Access Code** field enter **654**

```
change feature-access-codes                                     Page   5 of   8
                               FEATURE ACCESS CODE (FAC)

                         Automatic Call Distribution Features
                    After Call Work Access Code:
                             Assist Access Code:
                            Auto-In Access Code: 653
                           Aux Work Access Code: 654
                              Login Access Code: 650
                             Logout Access Code: 651
                          Manual-in Access Code: 652
      Service Observing Listen Only Access Code:
      Service Observing Listen/Talk Access Code:
          Service Observing No Talk Access Code:
                    Add Agent Skill Access Code:
                 Remove Agent Skill Access Code:
             Remote Logout of Agent Access Code:
```

Figure 50: Add Feature Access Codes

5. Configure the station to be used as an agent by using the ***change station x*** command, as shown in **Figure 51**, where **x** is the extension of an existing station.

   On Page 4 of the **change station** form:
   - In the button **5:** field enter **aux-work**
   - In the button **6:** field enter **manual-in**
   - In the button **6:** field enter **auto-in**

```
change station 30003                                            Page   4 of   5
                                     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: aux-work
 2: call-appr                             6: manual-in          Grp:
 3: call-appr                             7: auto-in            Grp:
 4: uui-info                              8:
```

Figure 51: Add Station Buttons for Agent Access

3.1.13 Save Avaya Aura™ Communication Manager Provisioning

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

3.2. CustomerA (CoRes environment) Specific Provisioning For Avaya Aura™ Communication Manager

The following Avaya Aura™ Communication Manager provisioning is required for the CustomerA Co-Resident (CoRes) environment. All other provisioning is common with CustomerB.

3.2.1 CustomerA Firewall NAT Addressing

As described in **Section 1**, the CustomerA location uses a Cisco PIX firewall to NAT the private addressing to public addresses. The following address mapping was used by the PIX in the sample configuration.

Device	Private Address	Public NAT Address
Avaya Co-Resident Avaya Aura™ Communication Manager and the Avaya Aura™ SIP Enablement Services	192.168.1.100	65.65.65.100
Avaya G700 VoIP Processor	192.168.1.101	65.65.65.101
All other CustomerA Avaya equipment	192.168.1.xxx	65.65.65.98

Table 9: CustomerA NAT Addressing

3.2.2 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.

Note - The Co-Resident Avaya Aura™ Communication Manager and the Avaya Aura™ SIP Enablement Services share the same IP address **192.168.1.100**.

- **Procr** and **192.168.1.100** are used below as the **Name** and **IP Address** of the Avaya Aura™ Communication Manager at the enterprise site.

- **Co-Res-SES** and **192.168.1.100** are used below as the **Name** and **IP Address** of the Avaya SES at the enterprise site.

```
 change node-names ip                                          Page   1 of   2
                                 IP NODE NAMES
    Name              IP Address
Co-Res-SES          192.168.1.100
default             0.0.0.0
procr               192.168.1.100
```

Figure 52: CustomerA IP Node Names Form

3.2.3 IP-Network-Region Assignments

The reference configuration used ip-network-region 1 for all Avaya network components in CompanyA. Network region assignments for Media Gateways may be verified with the ***list media-gateway*** command.

```
list media-gateway

                             MEDIA-GATEWAY REPORT

Num Name                     Serial No/        IP Address/        Type     NetRgn Reg?
                             FW Ver/HW Vint    Cntrl IP Addr               RecRule

1   G700                     066626011162      192.168.1.102   g700        1       y
                             26 .31 .0  /5     192.168.1.100               none
```

Figure 53: Verify Media Gateway IP-Network-Region Assignment – CustomerA

The network-region for a Media Gateway may be modified with the ***change media-gateway x*** command where x is the Media Gateway ID number.

```
change media-gateway 1                                         Page   1 of   1
                               MEDIA GATEWAY
         Number: 1                            Registered?  y
           Type: g700            FW Version/HW Vintage: 26 .31 .0  /5
           Name: G700                   MGP IP Address: 192.168.1.102
      Serial No: 066626011162    Controller IP Address: 192.168.1.100
   Encrypt Link? y                         MAC Address: 00:04:0d:ef:29:a1
 Network Region: 1
       Location: 1                           Site Data:
  Recovery Rule: none

Slot   Module Type             Name
 V1:
 V2:   MM711                   ANA MM
 V3:
 V4:   MM710                   DS1 MM

 V8:
 V9:   gateway-announcements   ANN VMM
```

Figure 54: Change Media Gateway IP-Network-Region Assignment – CustomerA

3.2.4 Dial Plan

Like CustomerB, CustomerA uses 3 digit and 1 digit access code formats. However CustomerA uses 4 digit extensions beginning with 1 and 3 digit TAC codes beginning with 2.

The dial plan is modified with the ***change dialplan analysis*** command.
On Page 1 of the form:

- For extensions:
    - In the **Dialed String** field enter**1**
    - In the **Total Length** field enter **4**
    - In the **Call Type** field enter **ext**
- For Trunk Access Codes (TAC):
    - In the **Dialed String** field enter **2**

- In the **Total Length** field enter **3**
- In the **Call Type** field enter **fac**

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

      Dialed    Total  Call    Dialed    Total  Call     Dialed    Total  Call
      String    Length Type    String    Length Type     String    Length Type
    1             4    ext
    2             3    fac
    9             1    fac
```

Figure 55: Change Dialplan Analysis Form – Page 1

3.2.5 Configure SIP Signaling Groups

The SIP signaling groups are provisioned differently in a CoRes environment. In a CoRes environment the Avaya Aura™ Communication Manager and SES processes both share the same IP address. To differentiate SIP signaling between the processes, ports 6001 and 5061 are used.

Using the ***add signaling-group x*** 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 Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services, not the transport method used to the Verizon Business IPCC Service.
- Set **Co-Resident SES?** to **y** .
- Specify the procr for the **Near-end Node Name**.
- Specify **Co-Res-SES** for the **Far-end Node Name**.
- With **Co-Resident SES?** parameter set to **y,** the **Near-end Listen Port** value will be automatically set to **6001**.
- Specify **5061** in the **Far-end Listen Port** field.
- Enter the value **1** into the **Far-end Network Region** field.
- Enter the domain name of Avaya Aura™ SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is **customer.ay.com**.
- The **Direct IP-IP Audio Connections** field should be set to ***y*** to allow RTP voice paths to be established directly between IP telephones and the Verizon Business IPCC Service. When compatible conditions exist, this allows the voice packets to follow a direct path between the telephones and the network edge, potentially reducing media processor resources and network usage.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Avaya Aura™ Communication Manager to send DTMF tones using RFC 2833.
- The default values for the other fields may be used.

```
add signaling-group 10                                         Page   1 of   1
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
                        Transport Method: tls
                         Co-Resident SES? y

   Near-end Node Name: procr                 Far-end Node Name: Co-Res-SES
 Near-end Listen Port: 6001                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
       Far-end Domain: customer.ay.com

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3               Alternate Route Timer(sec): 6
```

Figure 56: Signaling Group 10 for SIP Telephones

3.2.6 Configure Public-Unknown-Numbering

In these Application Notes, the extensions on CustomerA Avaya Aura™ Communication Manager use a 4 digit dialing plan with extensions 11xx.

Figure 57 shows the use of the ***change public-unknown-numbering x*** command to send the correct calling party number. The entries below indicate that 4-digit extension numbers beginning with 11 will send the corresponding digits via trunk groups 1 through 4.
Using the **change public-unknown-numbering** command, configure the **Public-Unknown-Numbering** form shown below as follows:

- Set the **Ext Len** field to **4**.
- Set the **Ext Code** field to **11**.
- Set the **Trk Grp(s)** field to **1-4**.
- Set the **Total CPN Len** field to **4**.

```
display public-unknown-numbering 11                            Page   1 of   2

                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                    Total Administered: 1
 4  11           1-4                       4          Maximum Entries: 9999
```

Figure 57: Public/Unknown Numbering Form

3.2.7 Configure Call Routing

CustomerA routes calls in a similar fashion to CustomerB however the dial plans and associated DID numbers are different.

3.2.7.1 Outbound Calls

Outbound calls are provisioned the same as for CustomerB (see **Section 3.1.9**).

3.2.7.2 Incoming Calls

This step configures the mapping of incoming numbers to the proper extensions. Please refer to **Section 3.1.9** for details on Verizon Business IP TF and IP IVR digit strings.

The following mappings were defined in the reference configuration for CustomerA.

	Incoming Digits	Mapped to Extension	Extension Use
IP TF	8883221	1101	Digital station
	8882380	1102	H.323 station
	7770107	1103	SIP station
	7772649	1104	VDN
IP IVR	5556664246	1101	Digital station
	5556664250	1102	H.323 station
	5556664254	1103	SIP station
	5556664284	1104	VDN

Table 10: CustomerA Incoming Digit Mapping

Figure 58 shows the incoming call handling treatment for SIP Trunk 4 (inbound calls) used for CustomerA.

```
display inc-call-handling-trmt trunk-group 4                  Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called       Del Insert
 Feature        Len      Number
 public-ntwrk    10 5556664246       10  1101
 public-ntwrk    10 5556664250       10  1102
 public-ntwrk    10 5556664254       10  1103
 public-ntwrk    10 5556664284       10  1104
 public-ntwrk    7  8523221          7   1101
 public-ntwrk    7  8522380          7   1102
 public-ntwrk    7  7770107          7   1103
 public-ntwrk    7  7772649          7   1104
```

Figure 58: CustomerA Incoming Call Handling Treatment Trunk 4 (Inbound)

3.2.8 Save Avaya Aura™ Communication Manager Provisioning

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

3.3. SIP Endpoint Configuration

This section describes the administration of Avaya SIP telephones and requires that the preceding SIP trunk configuration be completed beforehand. SIP telephones are optional and not required for the Verizon Business IPCC Service. CustomerB is used in the following examples. CustomerA provisioning is similar.

3.3.1 Add a Station for the SIP Endpoint

The first step in adding a station for Avaya SIP telephones registered with Avaya Aura™ SIP Enablement Services is to assign a station as shown in **Figure 59**. In the following example an Avaya 96xx SIP telephone is provisioned.

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

- Set the station **Type** field to **9600SIP**.
- Enter a **Name** for the station (e.g, **9630 SIP**).
- The **Security Code** is left blank for SIP stations. (Note: SIP phone passwords are administered within the SES **Add User** screen. See **Section 4.2**).
- Let all other fields default.

```
add station 30002                                            Page   1 of   6
                                     STATION

Extension: 30002                          Lock Messages? n              BCC: 0
     Type: 9600SIP                          Security Code:                 TN: 1
     Port: S00003                        Coverage Path 1: 1             COR: 1
     Name: 9630 SIP                  Coverage Path 2:               COS: 1
                                         Hunt-to Station:
STATION OPTIONS
                                             Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                   Message Lamp Ext: 30002
            Speakerphone: 2-way                 Mute Button Enabled? y
        Display Language: english                    Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal                Media Complex Ext:
   Survivable Trunk Dest? y                            IP SoftPhone? n

                                              Customizable Labels? y
```

Figure 59: 9630 SIP Station Administration – Page 1

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

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

Note – By default the **Restrict Last Appearance** value is **y** and reserves the last call appearance for outbound calls only (e.g, emergency calls). By limiting inbound calls to two appearances, situations may arise where conference and transfer calls may not function properly if a third appearance is not available to complete these call functions. See **Figure 60**.

```
add station 30002                                              Page   2 of   6
                                     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
                                                               EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
     MWI Served User Type:                        Display Client Redirection? n
             AUDIX Name:                          Select Last Used Appearance? n
                                                    Coverage After Forwarding? s

                                                  Direct IP-IP Audio Connections? y
  Emergency Location Ext: 30002         Always Use? n IP Audio Hairpinning? n
```

Figure 60: 9630 SIP Station Administration – Page 2

By default 3 call appearances (**call-appr**) are provided on page 4 of the **Station** form for the phone type 9640SIP.

```
add station 30002                                              Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                              Headset? n
       Jack:                                              Speaker? n
      Cable:                                             Mounting: d
      Floor:                                          Cord Length: 0
   Building:                                            Set Color:

ABBREVIATED DIALING
     List1:                   List2:                    List3:

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

Figure 61: 9630 SIP Station Administration – Page 4

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

3.3.2 Configure Off-PBX Telephone Station Mapping

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

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 62**, enter the following:

- In the **Station Extension** field, enter the extension number from the station defined in **Section 3.3.1** (e.g, **30002**).

- 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 and matches the station extension (e.g, **30002**).
- Set the **Trunk Selection** field to **1**, which is the number assigned to the SIP trunk group used for the SIP stations (see **Section 3.1.7**).
- Let all other fields default.

```
change off-pbx-telephone station-mapping 30002                   Page   1 of   2

                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station         Application Dial   CC  Phone Number      Trunk       Config
 Extension                   Prefix                       Selection   Set
 30002               OPS       -          30002               1         1
```

Figure 62: Off-PBX Telephone Station Mapping for 9630 SIP Telephone – Page 1

On page 2 of the form:

1. Set the **Call Limit** field to the maximum number of calls that may be active simultaneously at the station. In this example, the **Call Limit** is set to **3**, which corresponds to the number of call appearances configured on the station form (See **Figure 61**).
2. Accept the default values for the other fields.

```
change off-pbx-telephone station-mapping 30002                   Page   2 of   2

                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Call       Mapping      Calls       Bridged         Location
 Extension      Limit      Mode         Allowed     Calls
 30002            3        both           all         none
```

Figure 63: Off-PBX Telephone Station Mapping for 9630 SIP Telephone – Page 2

3.3.3 Save Avaya Aura™ Communication Manager Changes

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

4. Configure Avaya Aura™ SIP Enablement Services (SES)

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

This section is divided into two parts: **Section 4.1** provides the steps necessary to configure SIP trunks to the Verizon Business IPCC Service. **Section 4.2** provides the steps necessary to complete the administration for optional SIP endpoints.

As described in **Section 1**, CustomerB uses a standalone SES provisioned as an Edge/Home. CustomerA uses a Co-Resident (CoRes) Avaya Aura™ Communication Manager/SES platform. The following section describe the CustomerB standalone SES configuration. See **Section 4.3** for CustomerA CoRes specific SES provisioning.

Note – All SES web interface settings are shown in ***bold/italic***.

4.1. Provisioning SIP Trunks to the Verizon Business IPCC Service

4.1.1 Log in to Avaya Aura™ SIP Enablement Services

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

Log in with the appropriate credentials and the screen shown in **Figure 64** is displayed.

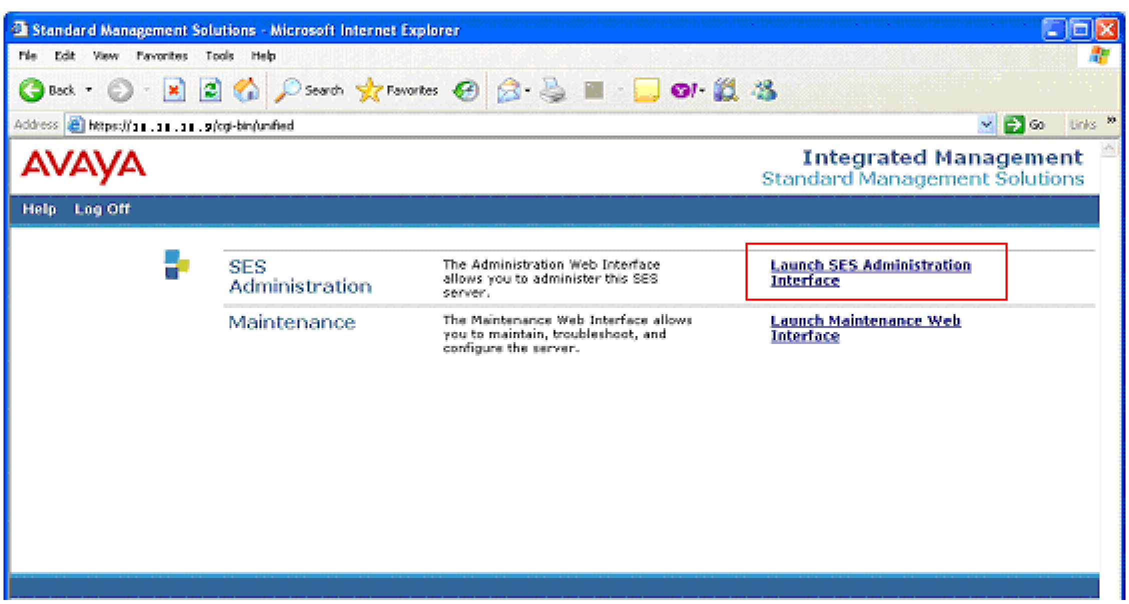

Figure 64: Avaya SES Management Main Page

Click on **Launch SES Administration Interface** for the administration home page and the **Top SIP Server Management** page shown in **Section 4.1.2** will open.

4.1.2 Verify System Properties

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

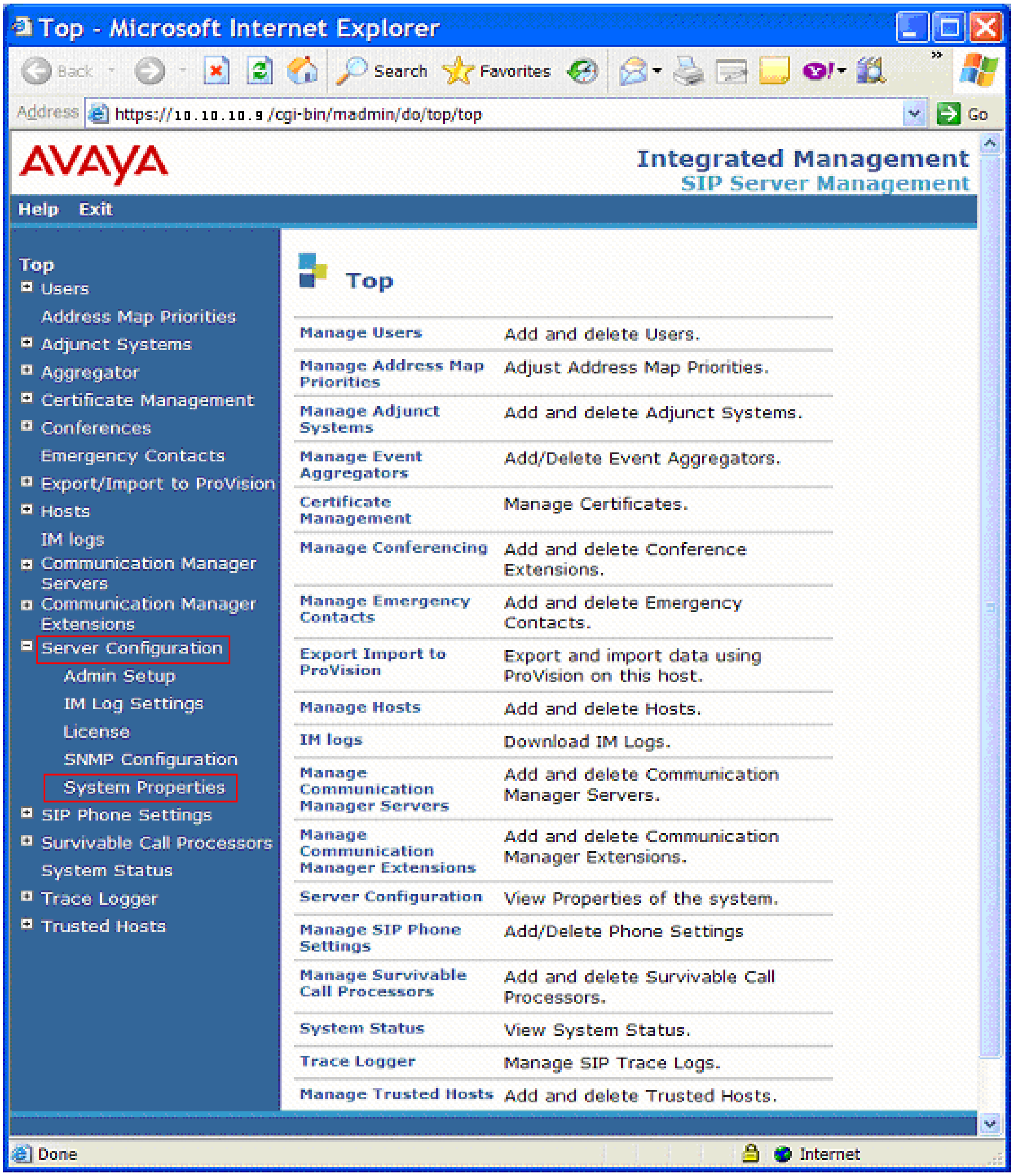

Figure 65: SES Administration Top Page

The **View System Properties** page (**Figure 66**) displays the **SES Version** and the **Network Properties** entered during the installation process.

In the **View System Properties** page:

1. Verify the SES Host information using the **Edit Host** page. In these Application Notes the SES **Host Type** is **SES combined home/edge** (defined during installation)**.**
2. Enter the **SIP Domain**. For CustomerB the SIP Domain ***customer.be.com*** is used in these Application Notes (see **Section 1.2**).
3. Select **Update**.

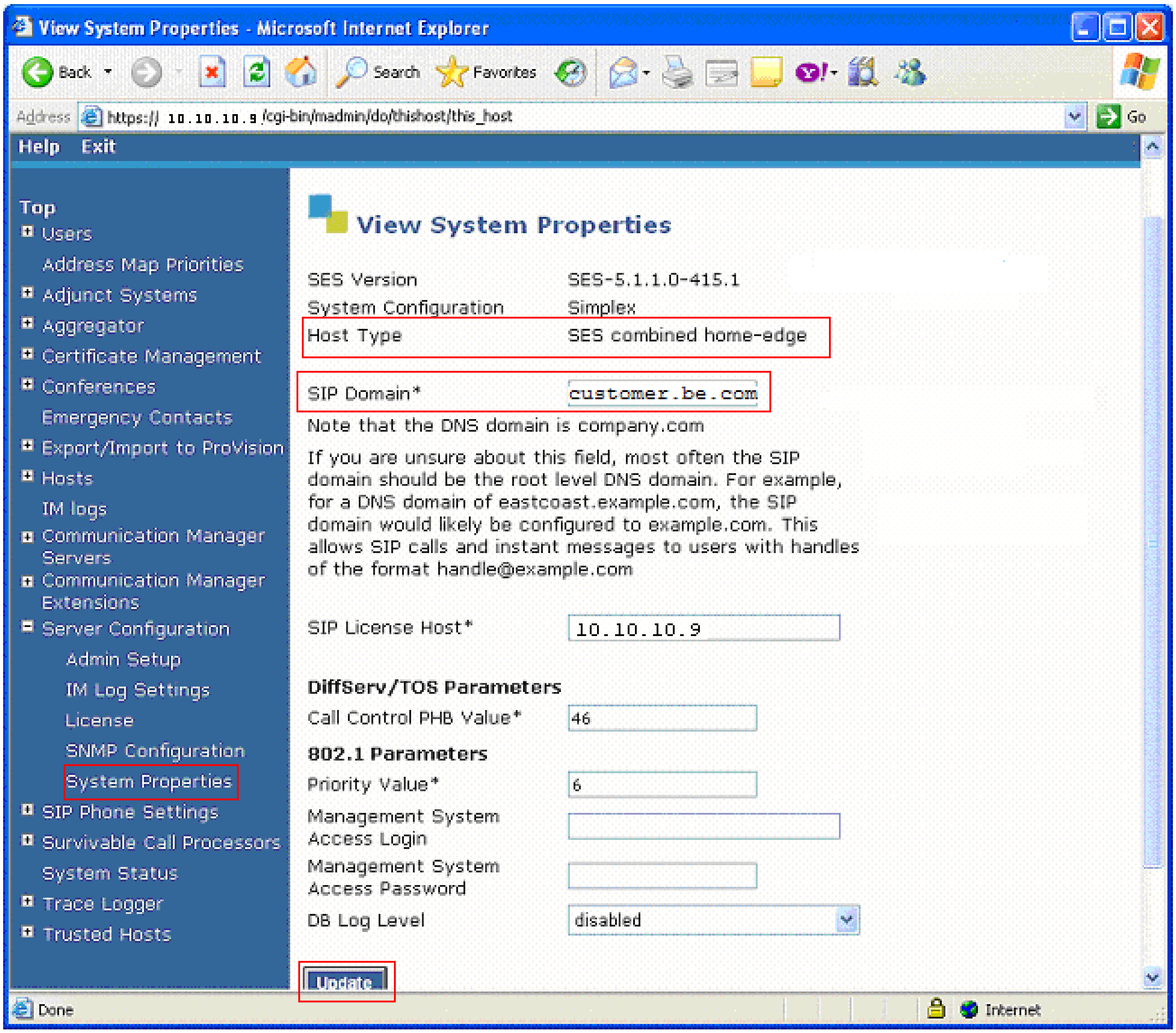

Figure 66: SES System Properties

4.1.3 Verify the Avaya Aura™ SIP Enablement Services Host Information

Display the **Edit Host** page by following the **Hosts** link in the left navigation pane and then clicking on the **Edit** option under the **Commands** section of the **List Hosts** screen (**Figure 67**).

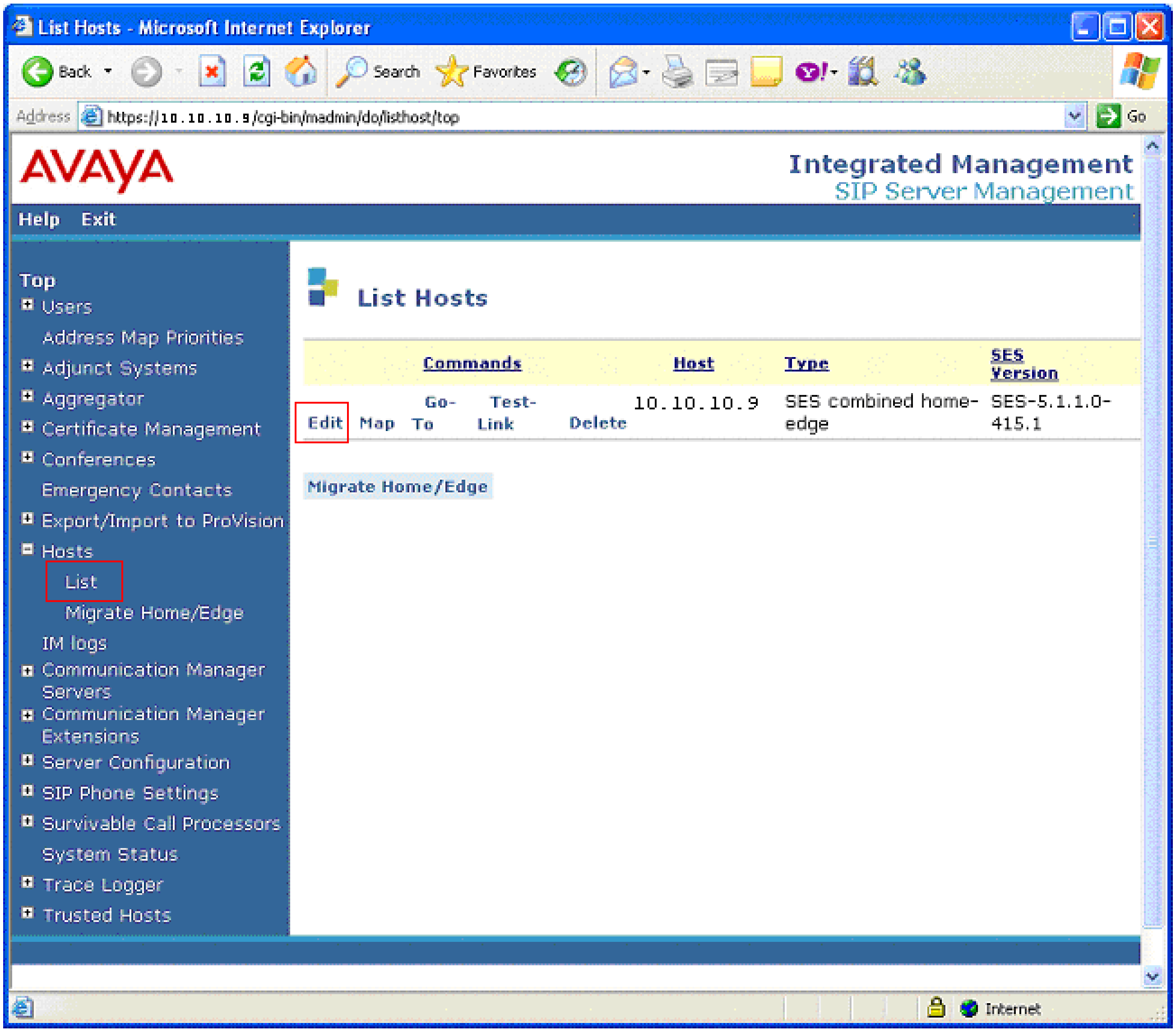

Figure 67: SES List Host

On the **Edit Host** screen (**Figure 68**):

- Verify that the IP address of this combined SES Home/Edge server is in the **Host IP Address** field (e.g, **10.10.10.9**).
- Do not modify the **Profile Service Password** fields.
- Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols** (TCP is enabled by default).
- Verify that **TLS** is selected as the **Link Protocol**.
- Verify that the **Outbound Routing Allowed** options **Internal** and **External** are checked.
- Leave the **Outbound Proxy** and **Outbound Direct Domains** fields blank.
- 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.

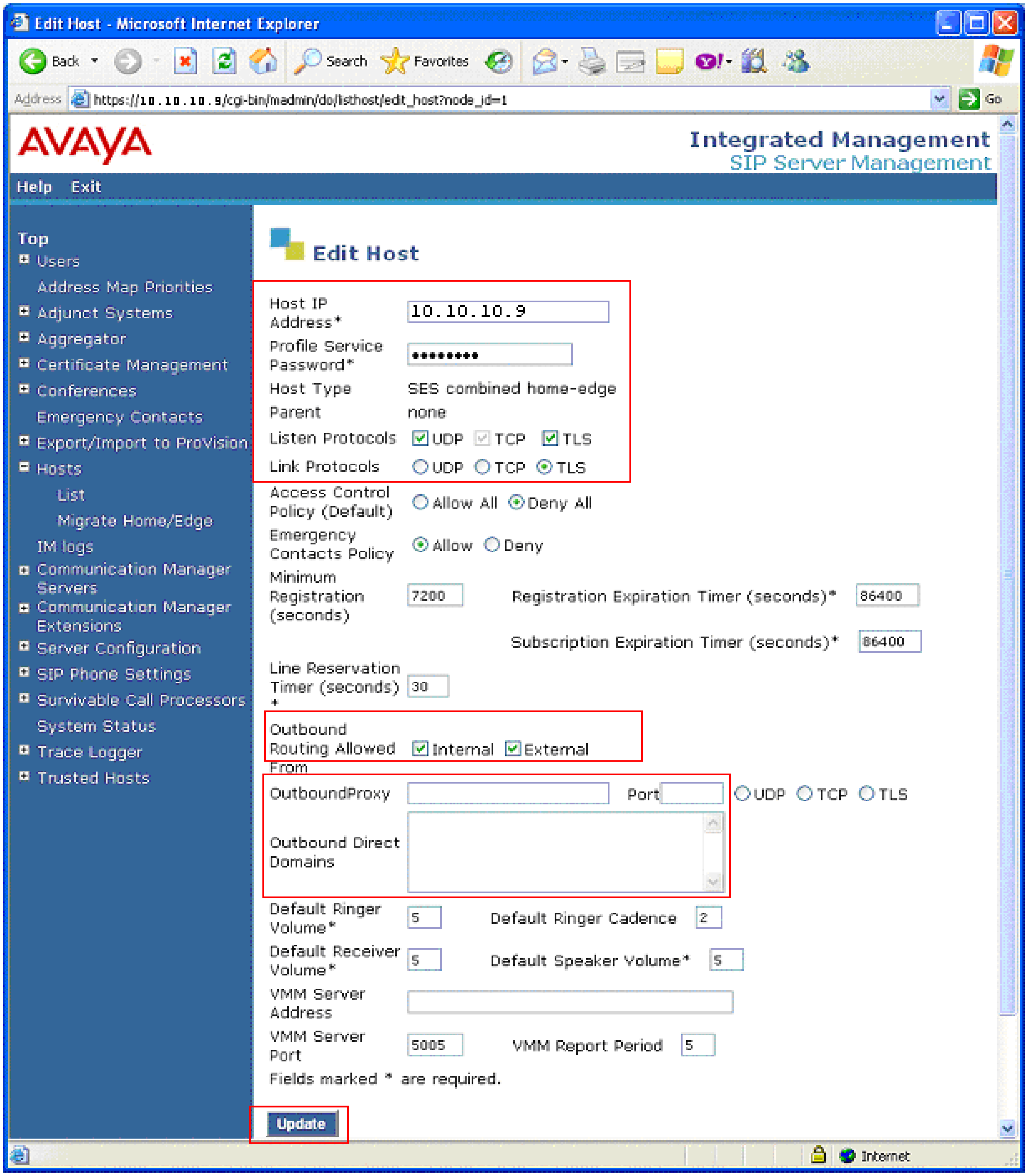

Figure 68: SES Edit Host

4.1.4 Configure Avaya Aura™ Communication Manager Server Interfaces

Expand the **Communication Manager Servers** option within any SES **SIP Server Management** page, and select **Add** to add a new **Communication Manager Server** or **List** to display or edit an existing **Communication Manager Server** configuration (**Figure 69**).

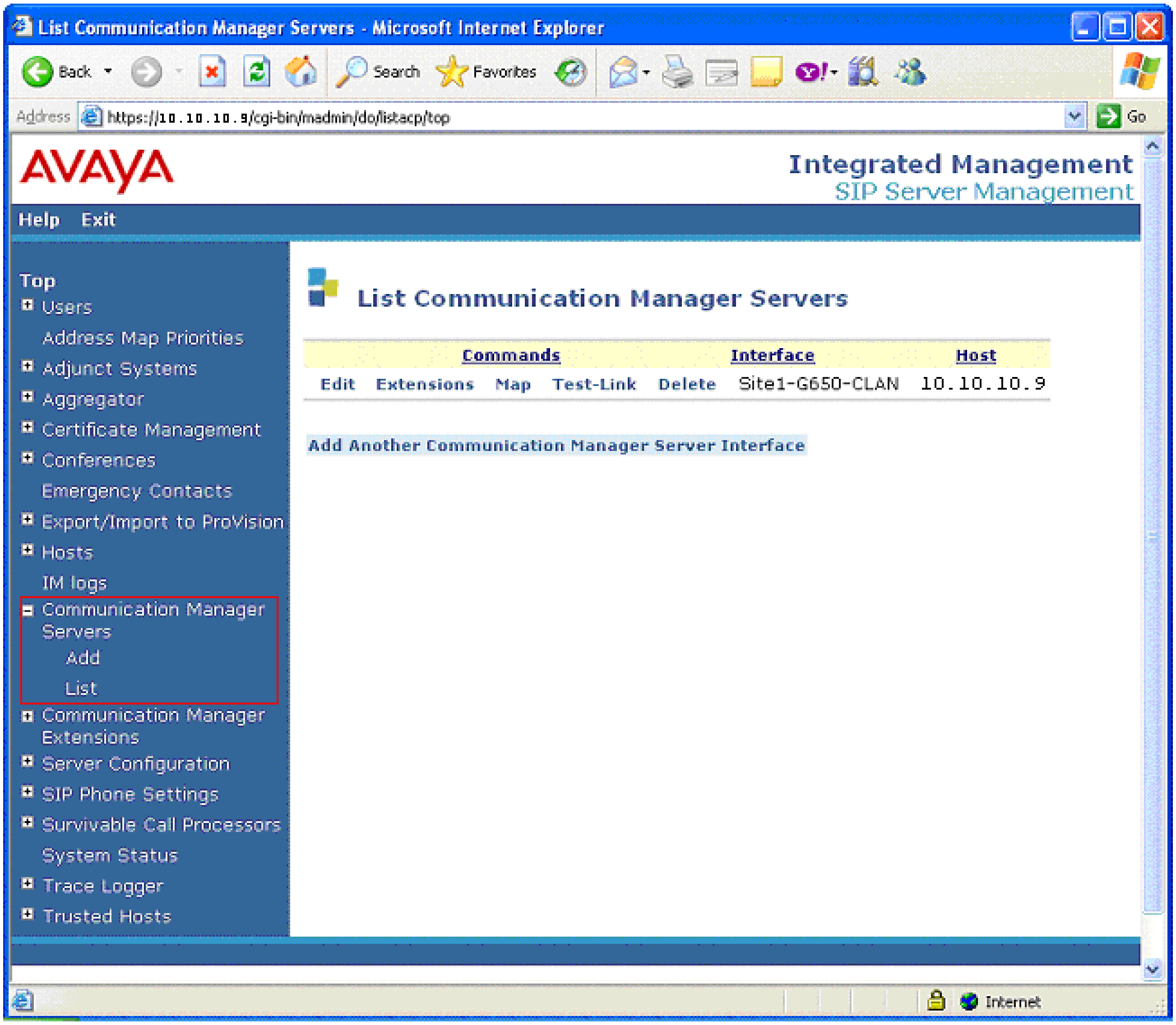

Figure 69: SES List Media Server

Figure 70 shows an existing configuration, which matches Signaling Group 1 configuration (SIP Telephones trunk) on Avaya Aura™ Communication Manager (see **Section 3.1.7**).

On the **Edit Host** screen:

- Specify a **Communication Manager Server Interface Name** (e.g, **Site1-G650-CLAN**).
- Verify that **TLS** is selected as the **Link Protocol**.
- Specify the IP address of the Avaya Aura™ Communication Manager SIP trunk interface. This is the IP address of the **Near-end Node Name** specified in the Avaya Aura™ Communication Manager Signaling Group form for the associated SIP Trunk (e.g, **10.10.10.13**).
- Specify the IP address of Avaya Aura™ Communication Manager in the **Communication Manager Server** Admin Address field (e.g, **10.10.10.10**).
- Specify ***5022*** (SSH) for the **Communication Manager Server** Admin Port field.
- Specify the proper credentials for the **Communication Manager Server** Admin Login and Password fields.

- Verify that **SSH** is selected for **SMS Connection Type**.
- Click the **Update** button only if changes are necessary. Otherwise exit the **Edit Host** page by selecting the **Top** link on the left navigation bar.

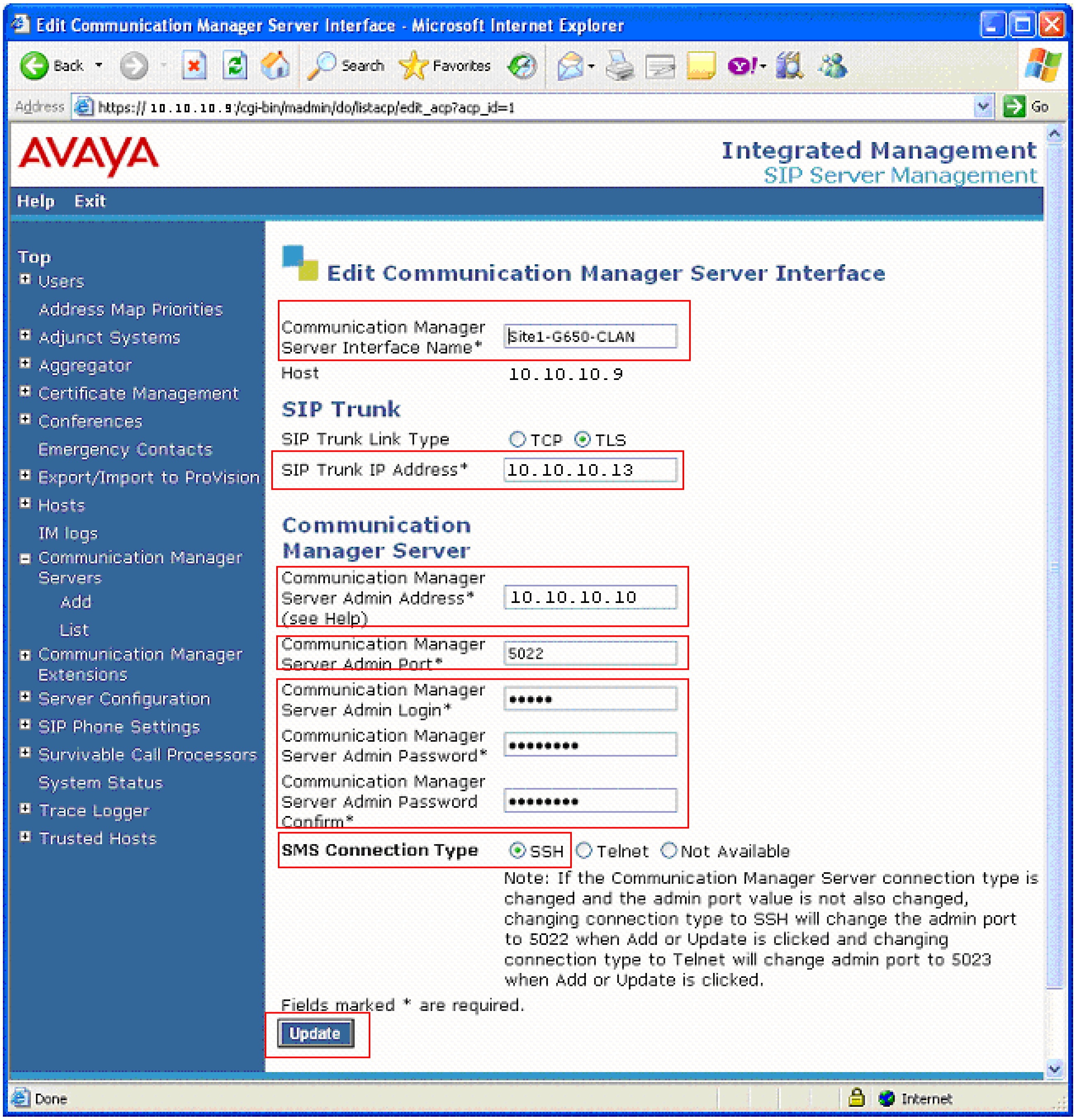

Figure 70: SES Edit Communication Manager Server Interface

4.1.5 Configure Trusted Hosts

Avaya Aura™ SIP Enablement Services will deny inbound calls from unknown foreign nodes. Therefore the Verizon service nodes must be specified as Trusted Hosts in Avaya Aura™ SIP Enablement Services.

Expand the **Trusted Hosts** option and select Add to add a new **Trusted Host** or select **List** to display or edit an existing **Trusted Host** configuration (**Figure 71**).

From the Avaya Aura™ SIP Enablement Services "**Top**" web page, select **Trusted Hosts**.

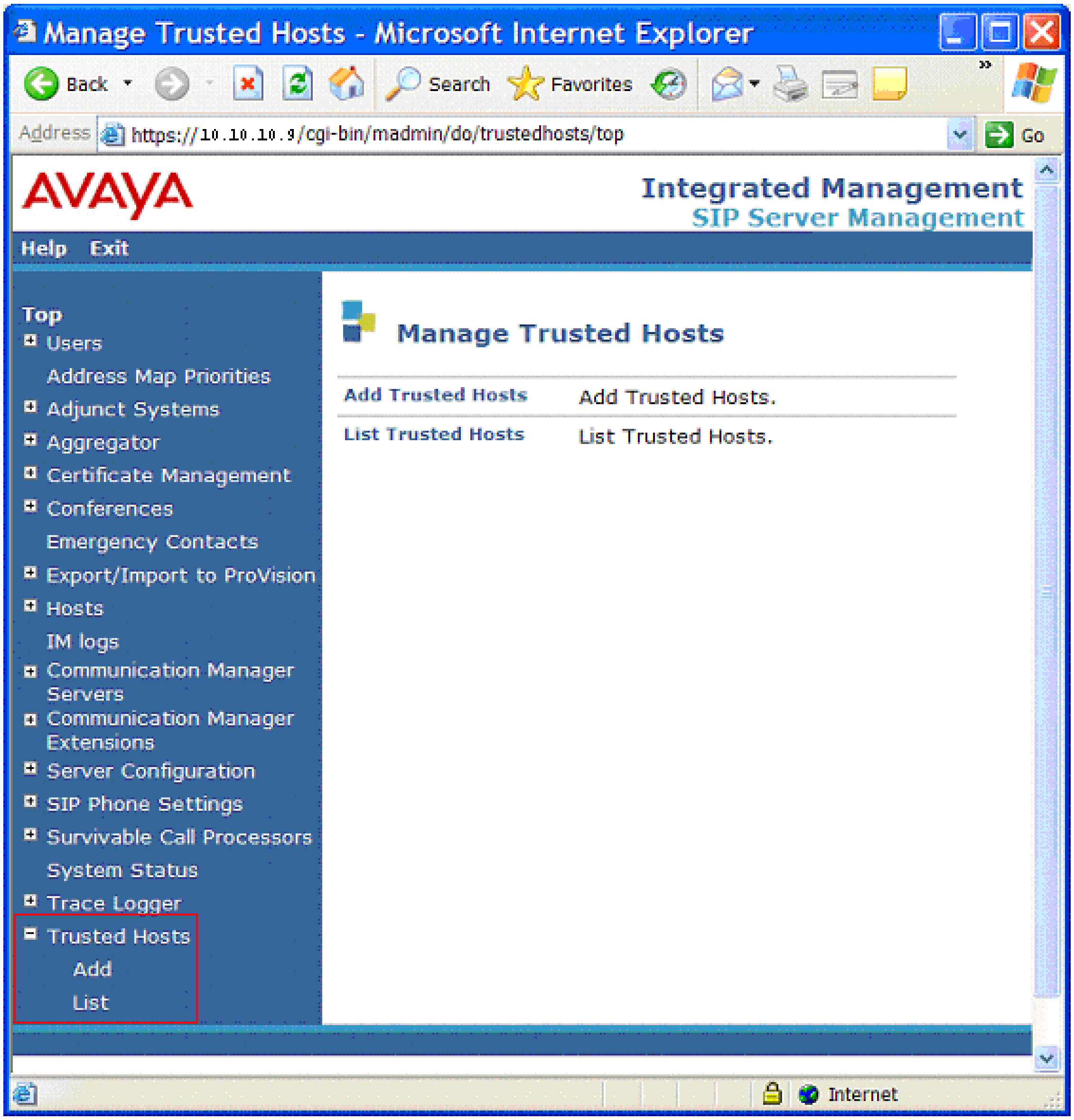

Figure 71: SES Trusted Hosts

On the **Edit Host** screen:

- Specify the IP address of the **Trusted Host** (e.g, **99.88.77.66**).
- Specify a description of the **Trusted Host** entry (e.g, **IP TF**).
- Select **Update** to save changes.

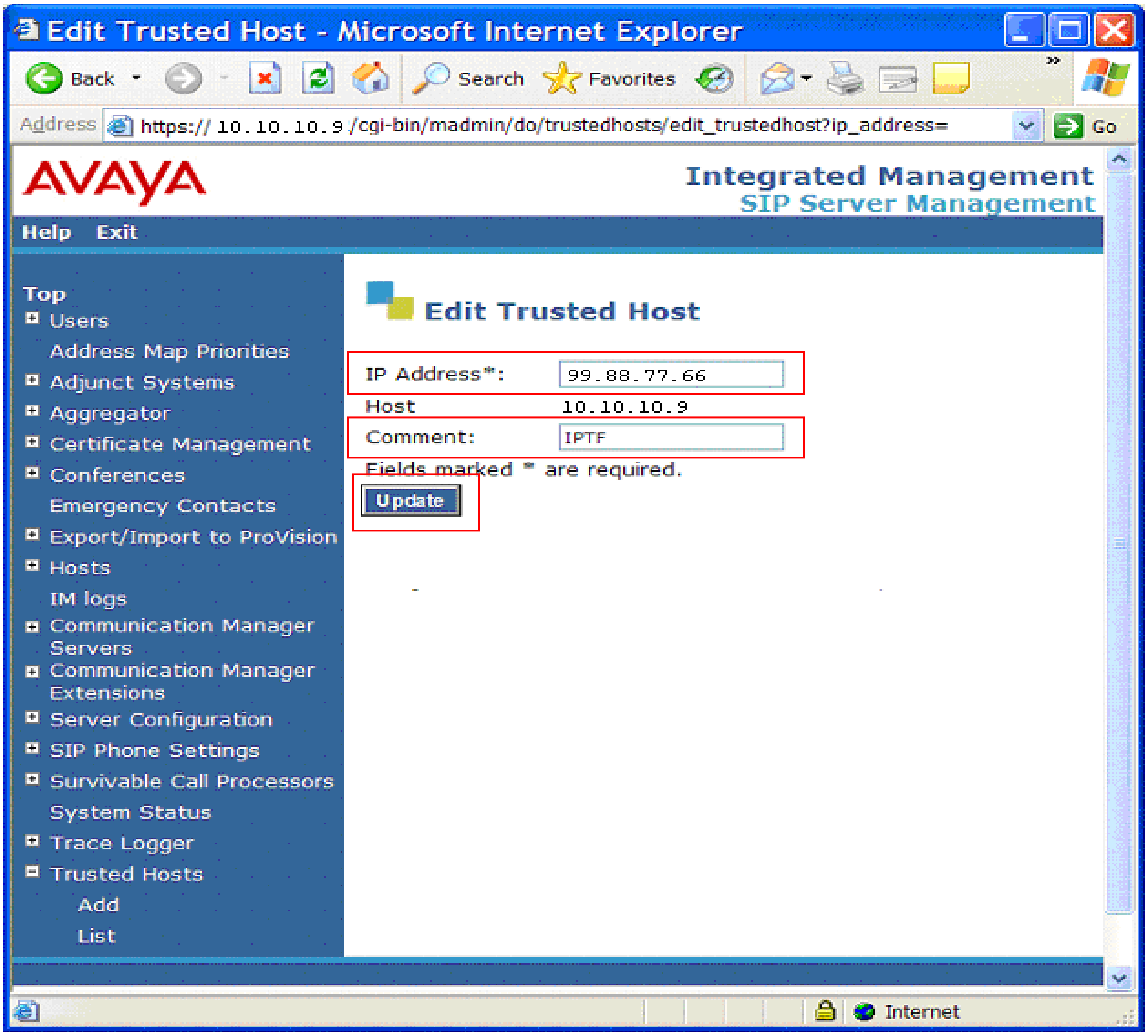

Figure 72: SES Edit Trusted Hosts

Note - Since a Trusted Host must be entered for each service node that may deliver an inbound INVITE, the Trusted Host may be lengthy. It is suggested that a list of possible nodes be provided by the Service Provider so that provisioning may be completed prior to service turn-up.

4.1.6 Configure Call Routing

The SES functions as a SIP proxy server for the SIP trunks connections to the Verizon Business IPCC Service. In this role, for outbound calls the SES must direct SIP messages originating from Avaya Aura™ Communication Manager to the Verizon SIP Service Controllers. In a similar manner for incoming DID calls, the SES must route messages received from the Verizon SIP network to the proper signaling interface on Avaya Aura™ Communication Manager.

4.1.6.1 Domain Based Routing – Outbound Calls

As described in **Section 1.7.1**, Domain Based Routing is used to send outbound calls from Avaya Aura™ Communication Manager to the SES.

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

In the reference configuration Avaya Aura™ Communication Manager specifies the FQDN of the Verizon Outbound Proxy server (***vziptf.dns.com***) in the Far-End Domain field of the outbound SIP Trunk Signaling Group (Trunk 4 in the reference configuration, see **Section 3.1.7**). The SES receives an INVITE from Avaya Aura™ Communication Manager with the destination URL of ***<callednumber>@vziptf.dns.com,*** and with no Outbound Proxy specified in the SES (see **Figure 68**), the SES will issue a DNS SRV record query for the foreign domain. The Verizon network will then determine the proper destination based on the called number and return the appropriate destination IP address. The SES will then send the INVITE to the IP address provided in the SRV response.

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

Note – By default the SES will use port 5060 for outbound calls. If a different destination port is required, then Domain Based Routing *should not* be used and SES Host Maps *must be used* to specify this non-standard port. See **[4]** and **[8]** for more information on provisioning Avaya Aura™ Communication Manager and the SES to utilize Host Maps for outbound calls.

4.1.6.2 Inbound Calls

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

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

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.

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

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

4.1.6.2.1 Inbound Call Routing

Inbound calls may come from Verizon's IP TF or IP IVR services. In the Reference configuration, an Address Map is provisioned for each. Since both IP TF and IP IVR calls go to the same Avaya Aura™ Communication Manager trunk, both Address Maps use the same Contact.

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

- Expand the **Communication Manager Servers** link in the left navigation menu of any **SIP Server Management** page. Select **List** to display the **List Communication Manager Servers** page as shown in **Figure 73.**
- Click on the **Map** link to display the **List Communication Manager Server Address Map**.

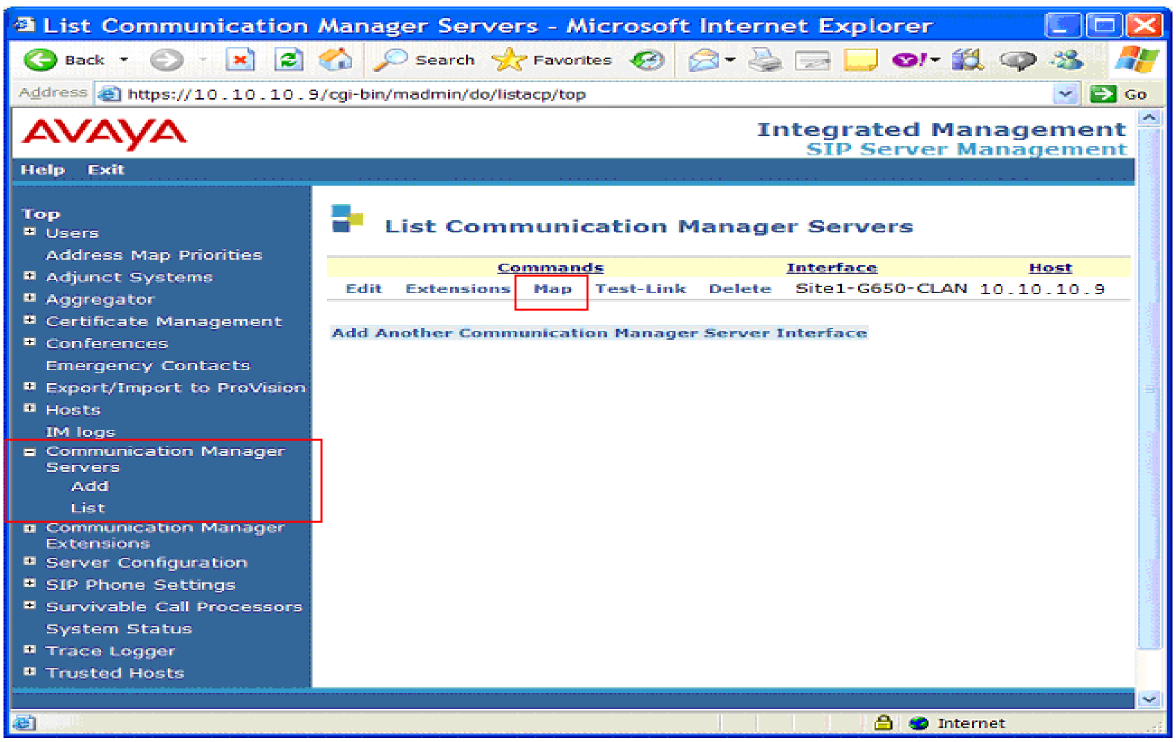

Figure 73: SES List Communication Manager Servers

Figure 74 shows the IP TF and IP IVR maps. The IP IVR map is called ***VZ_IPIVR*** and the IP TF map is called ***VZ_IPTF_999***.

1. **IP IVR Map provisioning.**
    - Click on the ***Edit*** button next to the **VZ_IPIVR** name.

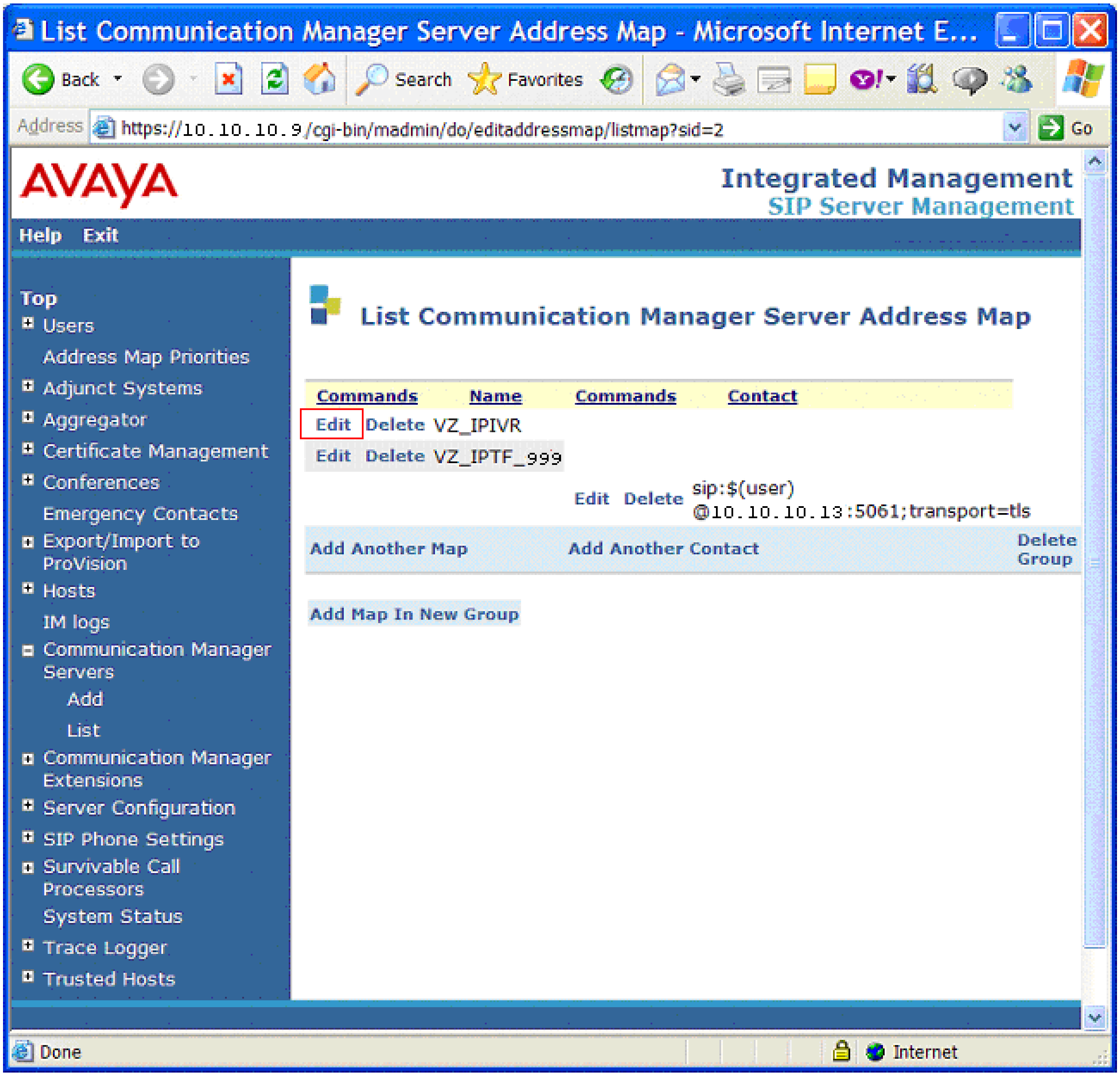

Figure 74: SES Communication Manager Server Address Maps

The **Edit Communication Manager Map Entry** window will open (**Figure 75**).

- Enter a name for the map. (e.g, ***VZ_IPIVR***)
- Enter the **Address Map Pattern** for incoming IP IVR calls into the **Pattern** field.
    - All IP IVR numbers begin with 555666 and are 10 digits long. The entry *^sip:5556661[0-9]{4}* means match on any number beginning with 555666, followed by any digit string, 4 digits in length.
- Verify the *Replace URI* option is checked.
- Click the **Update** button once the form is completed.

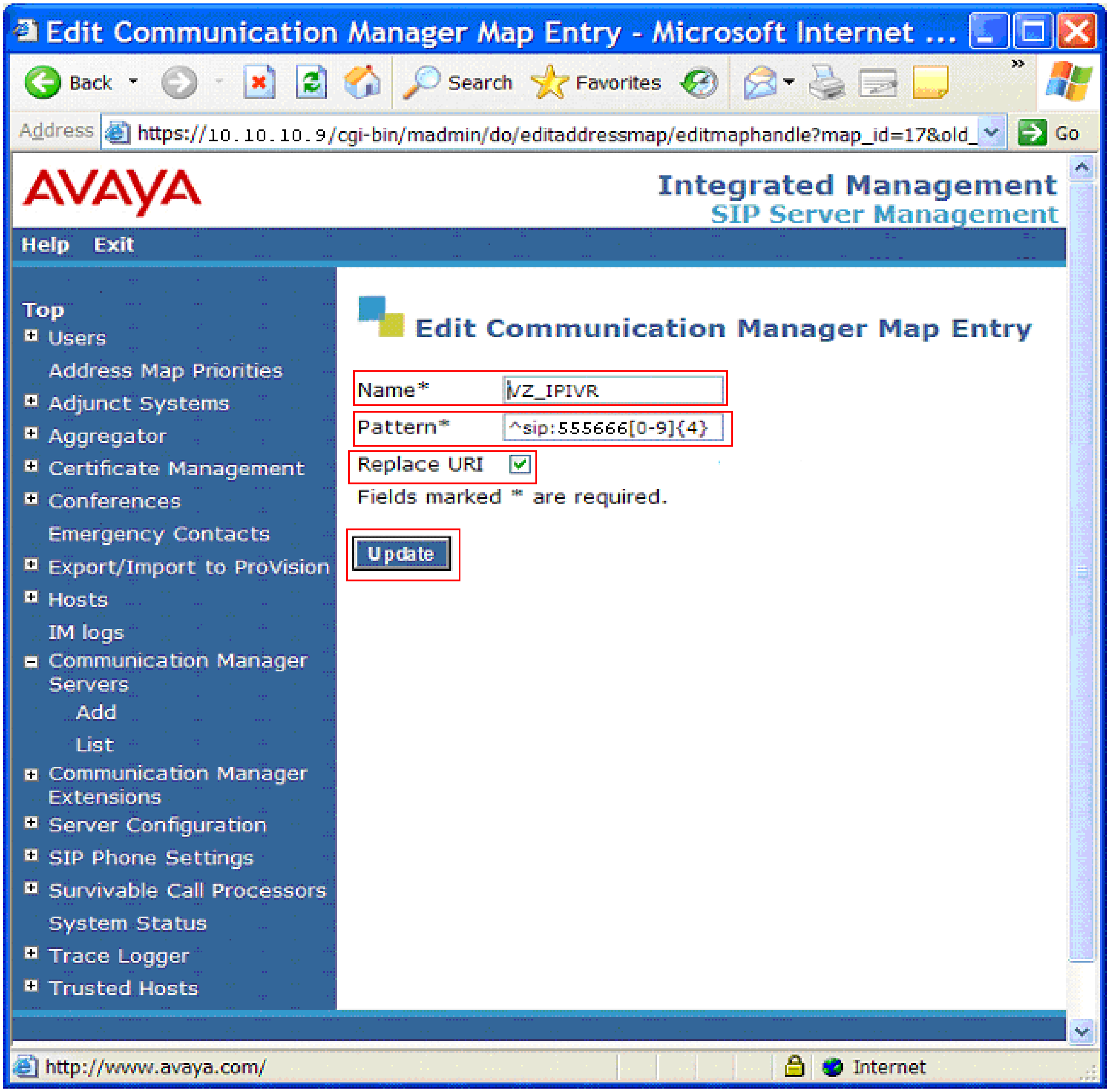

Figure 75: SES IP IVR Address Map

2. **IP TF Map provisioning.**
   - Go back to the List Communication Manager Server Address Map (**Figure 73**) and click on the *Edit* button next to the ***VZ_IPTF_999*** name and the **Edit Communication Manager Map Entry** window will open (**Figure 76)**.

     - Enter a name for the map. (e.g, ***VZ_IPTF_999***)
     - Enter the **Address Map Pattern** for incoming IP IVR calls into the **Pattern** field.
       - IP TF numbers for CustomerB begin with 999 and are 7 digits long. The entry *^sip:999** means match on any number beginning with 999.
     - Verify the *Replace URI* option is checked.

Click the **Update** button once the form is completed.

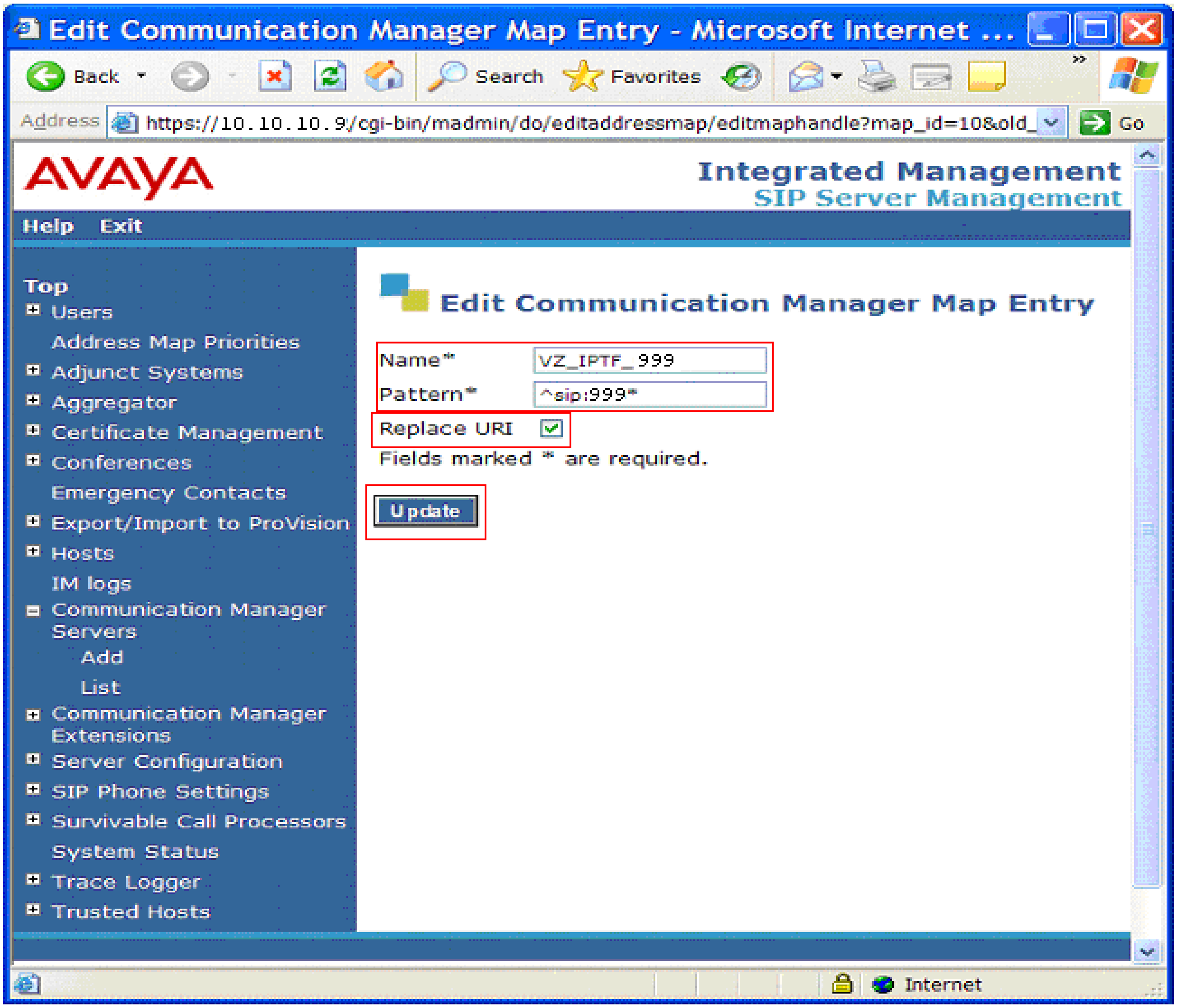

Figure 76: SES IP TF Address Map

As shown in **Figure 76** above, after the first **Communication Manager Address Map** is created, a corresponding media server **Contact** entry is created automatically.

sip:$(user)@10.10.10.13:5061;transport=tls

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

If necessary, the Contact information can be modified by clicking on the ***Edit*** button. The Edit **Communication Manager Contact** window will open (see **Figure 77**). Click the **Submit** button to submit the changes.

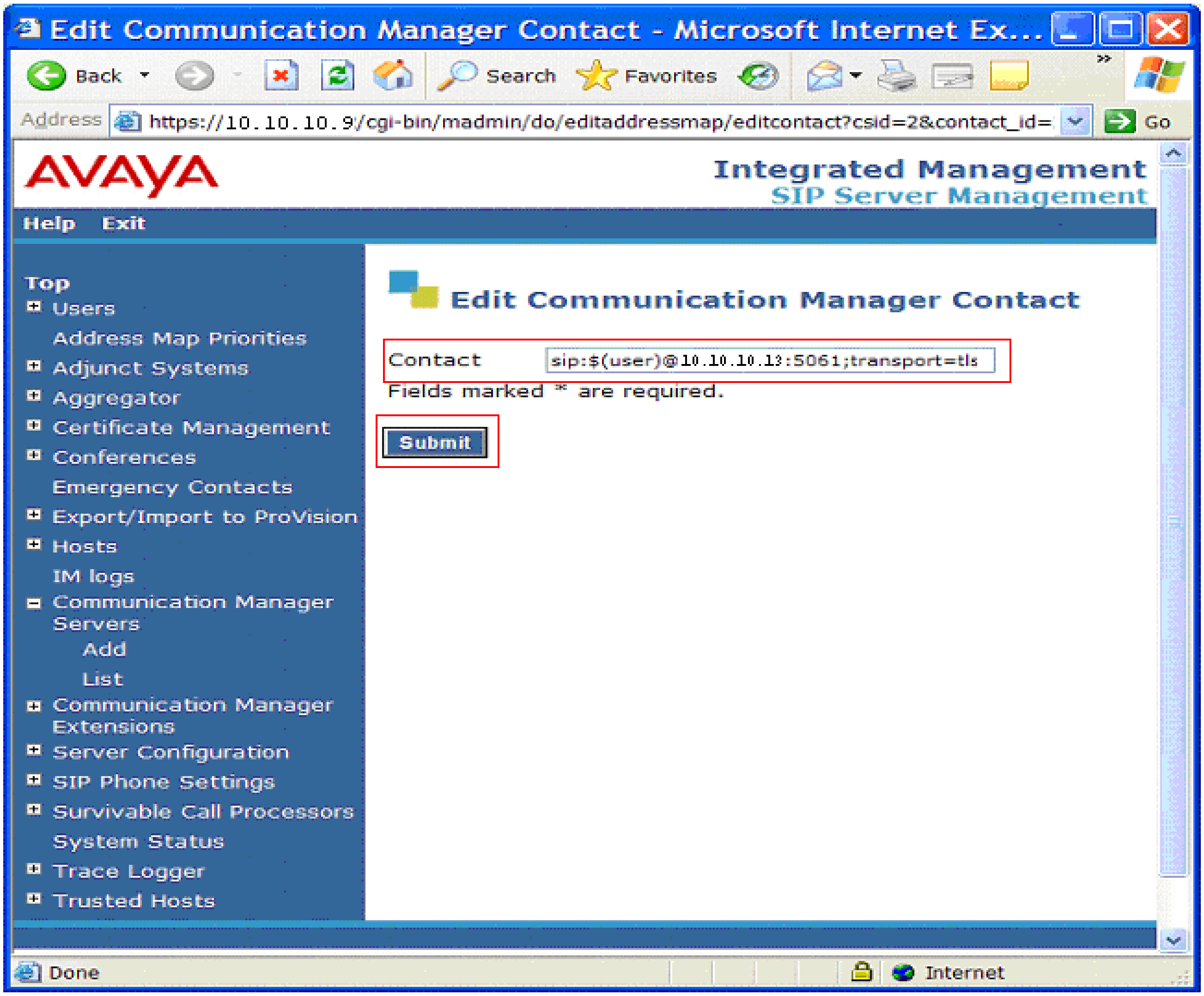

Figure 77: SES Communication Manager Contact

4.1.7 Verifying/Configuring SES DNS Information

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

Note – The following procedures are only applicable for a stand-alone SES platform. See **Section 4.3.1** for configuring DNS on an Avaya Aura™ Communication Manager/SES Co-Resident platform.

The IP address of the DNS server to be used by the SES is typically specified during the SES installation process. The DNS address can be verified by logging on to the Avaya SES Linux shell (using SSH and the appropriate credentials). Go to the **/etc** directory and enter the ***cat resolv.conf*** command. The current DNS entry is displayed (**Figure 78**).

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

Figure 78: SES Display Current DNS Configuration

If the DNS entry needs to be changed, connect to the Avaya SES Linux shell (any directory) and enter the ***initial_setup*** command (the setup process may execute several background functions that may take a few minutes). The screen showed in **Figure 79** will be displayed.

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

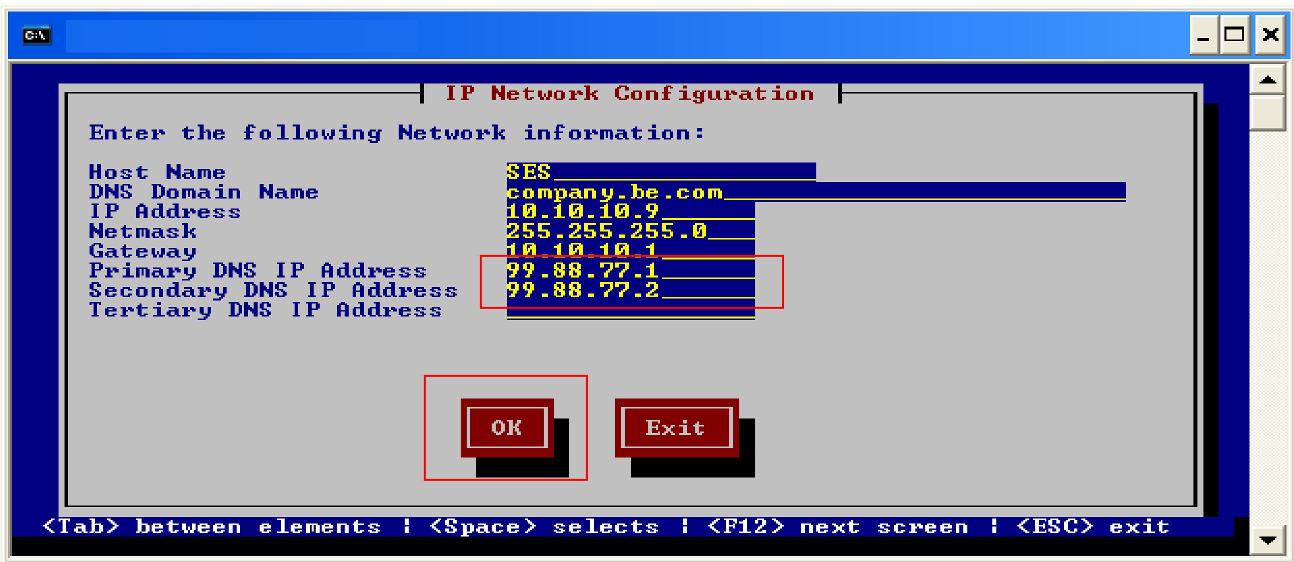

Figure 79: SES Changing the DNS IP Address

After the ***OK*** button is pressed (**Figure 79**), the screen showed in **Figure 80** will be displayed. Leaving the default value selected (e.g, **Simplex**), press the *Tab* key until the ***OK*** button is highlighted. Press the *Enter* key.

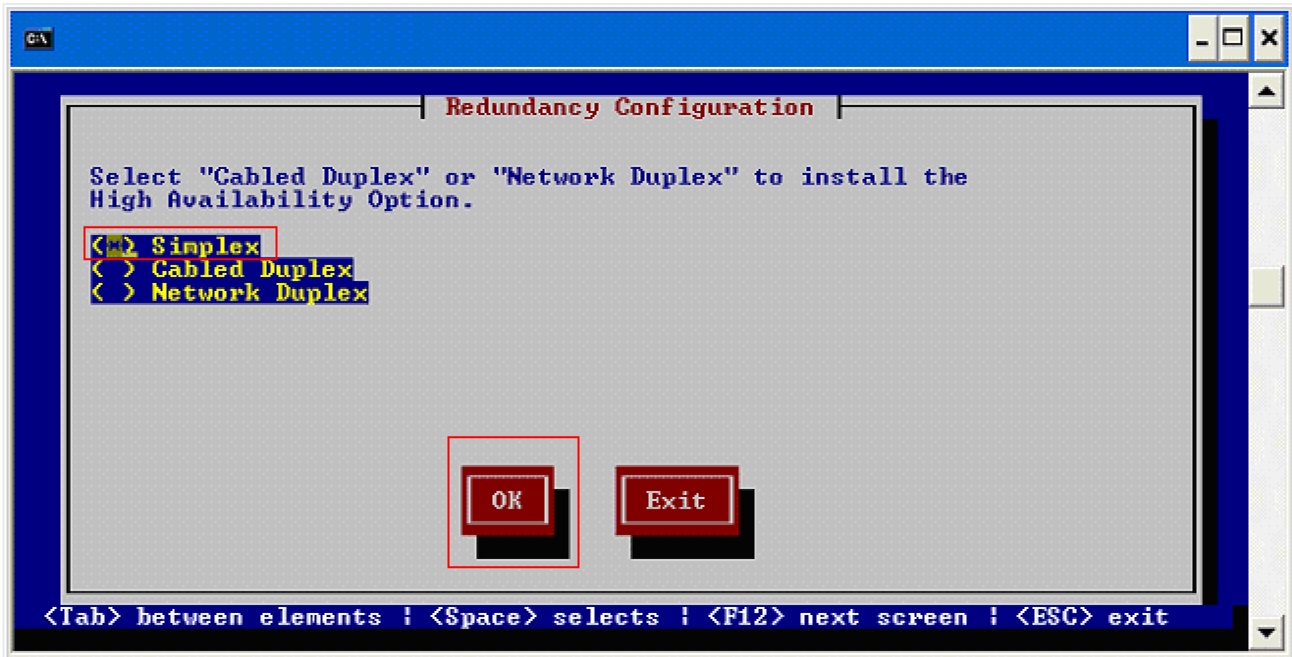

Figure 80: SES Default Redundancy Configuration

After the ***OK*** button is pressed (**Figure 80**), the screen shown in **Figure 81** will open. Verify the ***Finish*** button is highlighted (use the *Tab* key to change the selection) and press the *Enter* key.

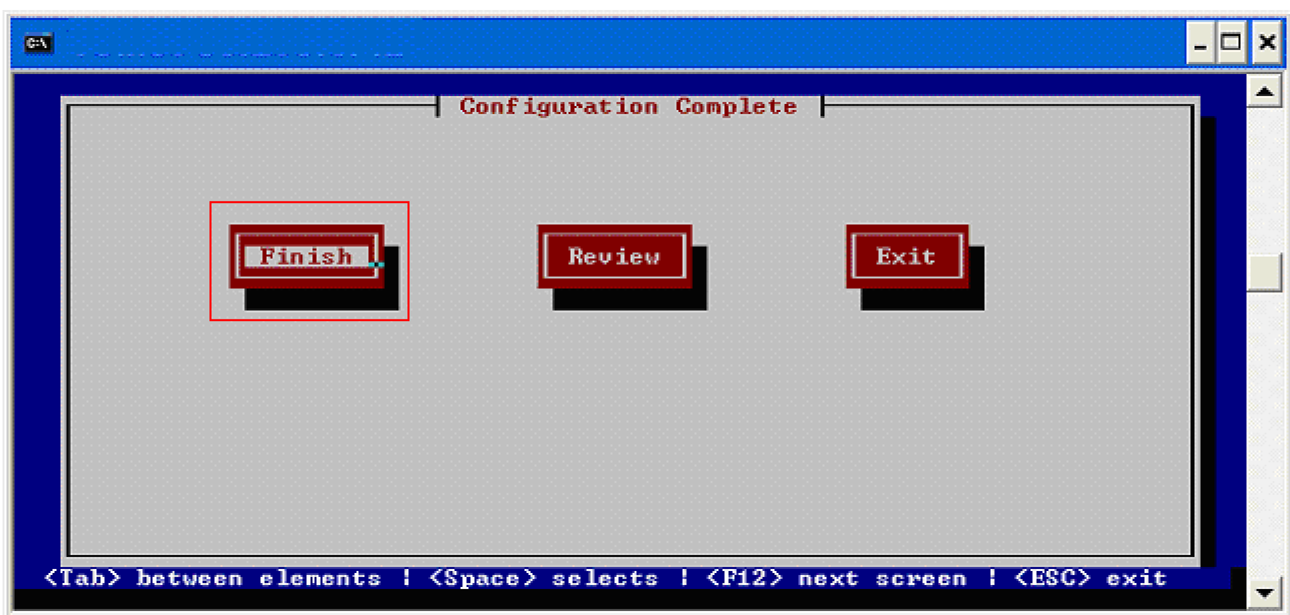

Figure 81: SES Finish Changing the DNS IP Address

Next, the screen shown in **Figure 82** will open asking to stop the SES processes. Verify that the option **[y]** is offered. If not, enter ***y***. Press the *Enter* key to accept the default value **[y]**.

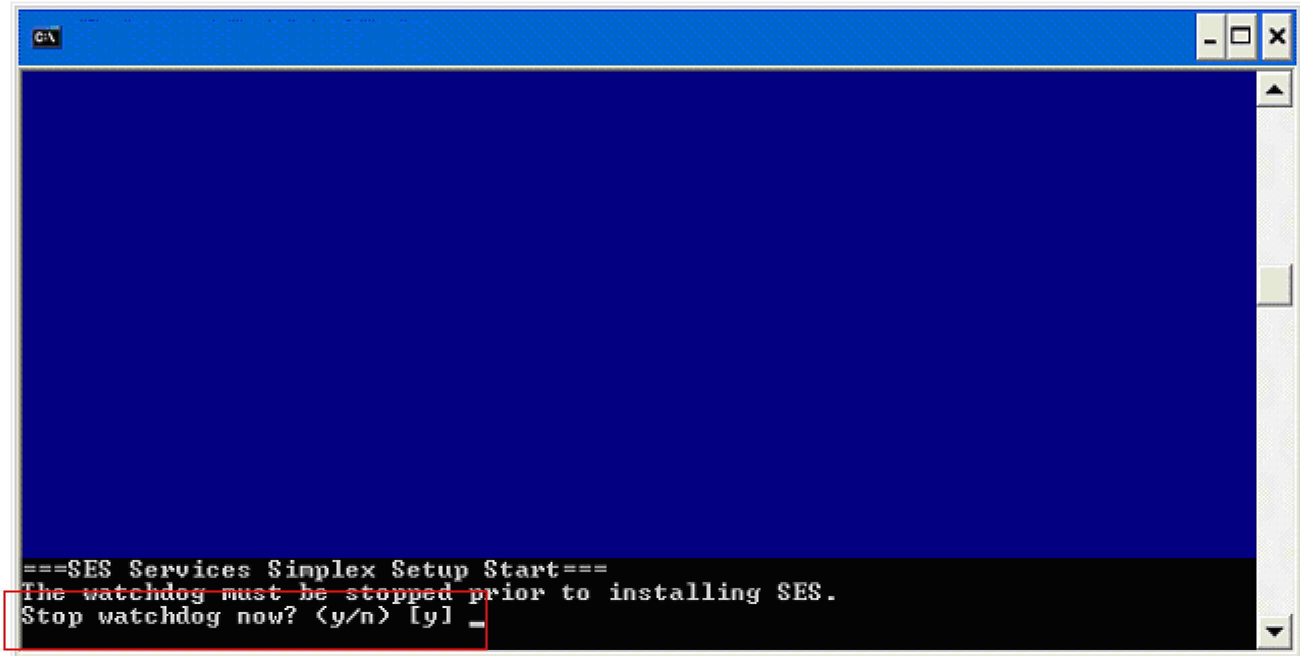

Figure 82: SES Changing the DNS IP Address – Stopping the SES Processes

Next the screen shown in **Figure 83** will open regarding the SES database. Verify that the option **[n]** is offered. If not, enter ***n***. Press the *Enter* key to accept the default value **[n]**.

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

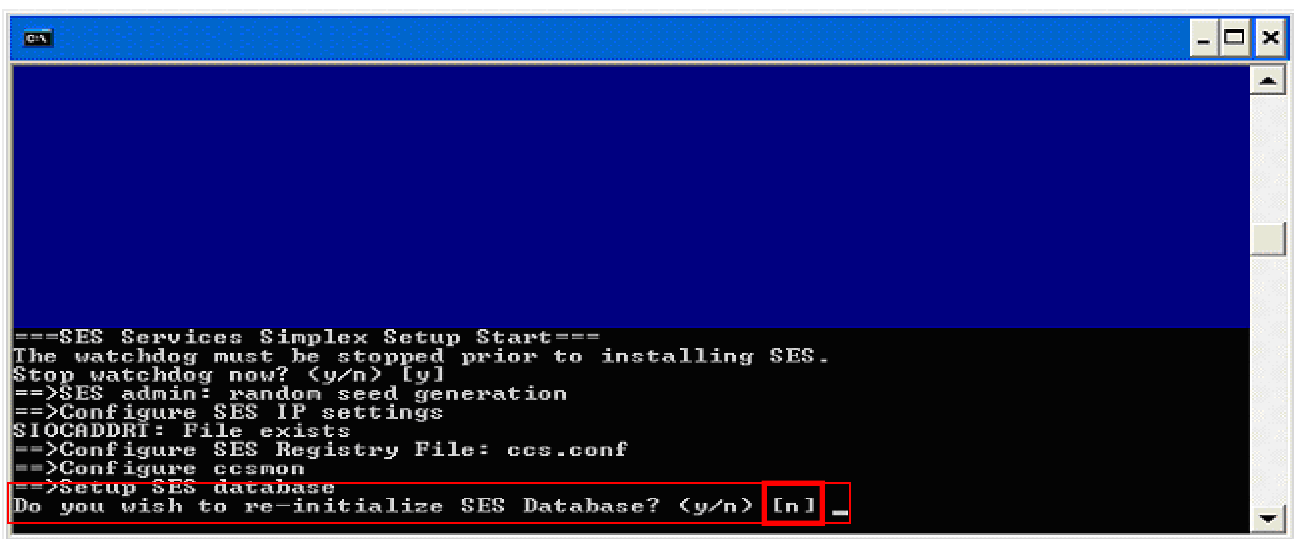

Figure 83: <u>Do Not</u> Initialize the SES Database

Next the screen shown in **Figure 84** will open asking to restart the SES processes. Verify that the option **[y]** is offered. If not, enter ***y***. Press the *Enter* key to accept the default value **[y]**.

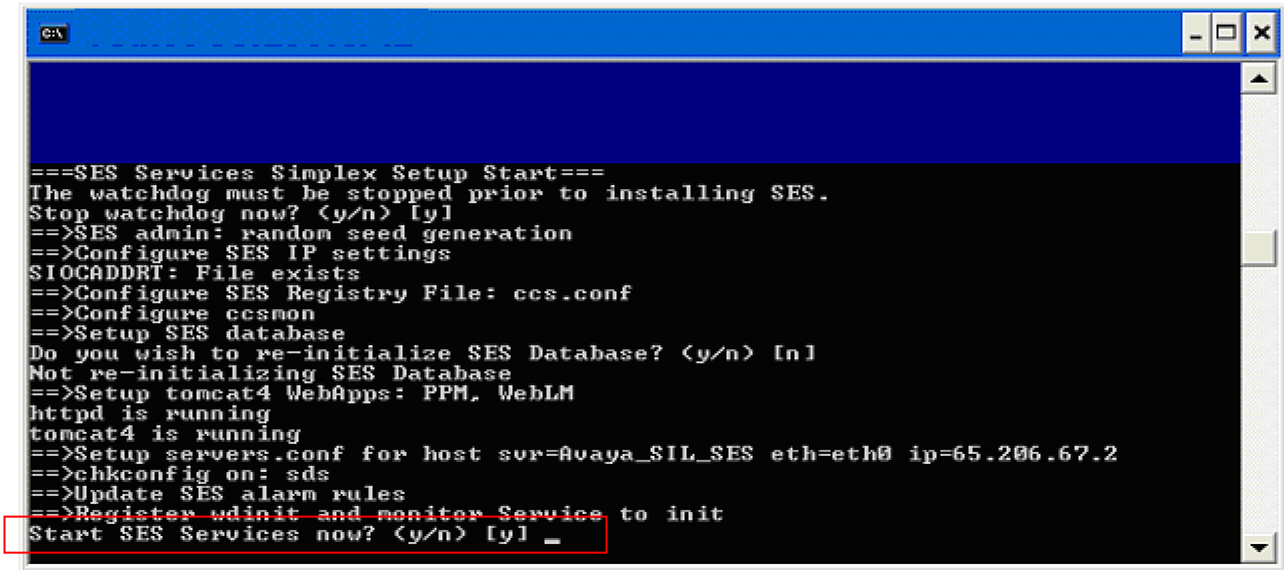

Figure 84: Changing the DNS IP Address – Restarting SES

The SES will then restart with the new DNS parameters.

4.2. SES Configuration for SIP Telephones

This section provides basic instructions for completing the administration to use Avaya 9600 Series IP Telephones (SIP) with the described configuration. Additional features are beyond the scope of these Application Notes.

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:

- From the **Top** page of the SES **SIP Server Management** web interface (**Figure 77**), expand the **Users** link in the left side blue navigation bar and click on the **Add User** link. The Add User window will open (**Figure 85**).

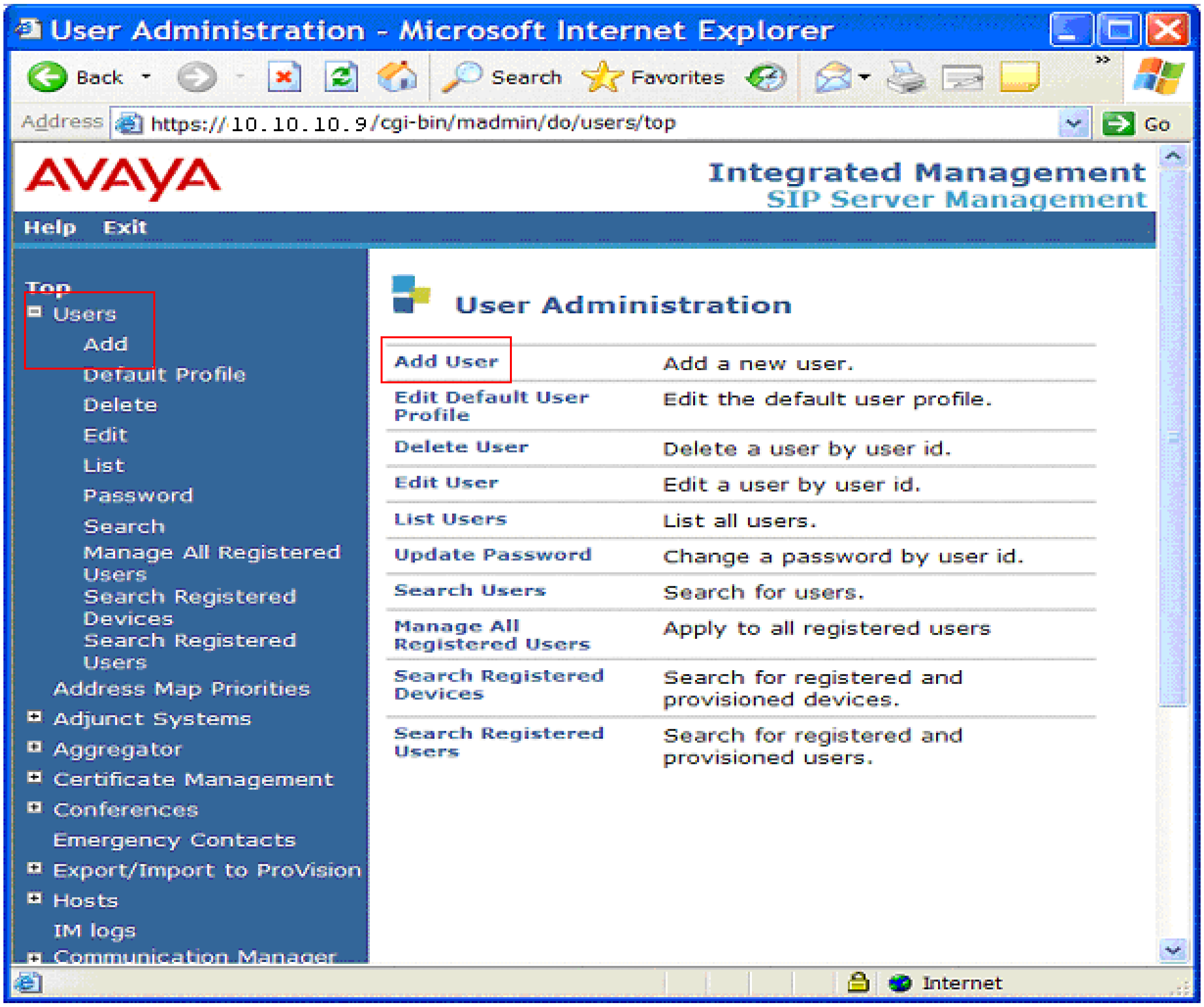

Figure 85: SES User Administration

- In the **Add User** page (**Figure 86**), enter the extension number for the SIP telephone in the **Primary Handle** and the **User ID** fields.
- Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
- Enter the **First Name** and **Last Name** of the user.
- Select the **Add Media Server Extension** checkbox. This associates an Avaya Aura™ Communication Manager extension with this SIP User. Calls from this user will be provided features and routing via Avaya Aura™ Communication Manager.
- In the **Host** field drop down menu, select the Avaya Aura™ SIP Enablement Services server hosting the domain (**10.10.10.9**) for this user.
- Click the **Add** button. This will cause a confirmation screen to appear.
- Click **Continue** on the confirmation screen. The **Add Communication Manager Extension** window will open (**Figure 87**).

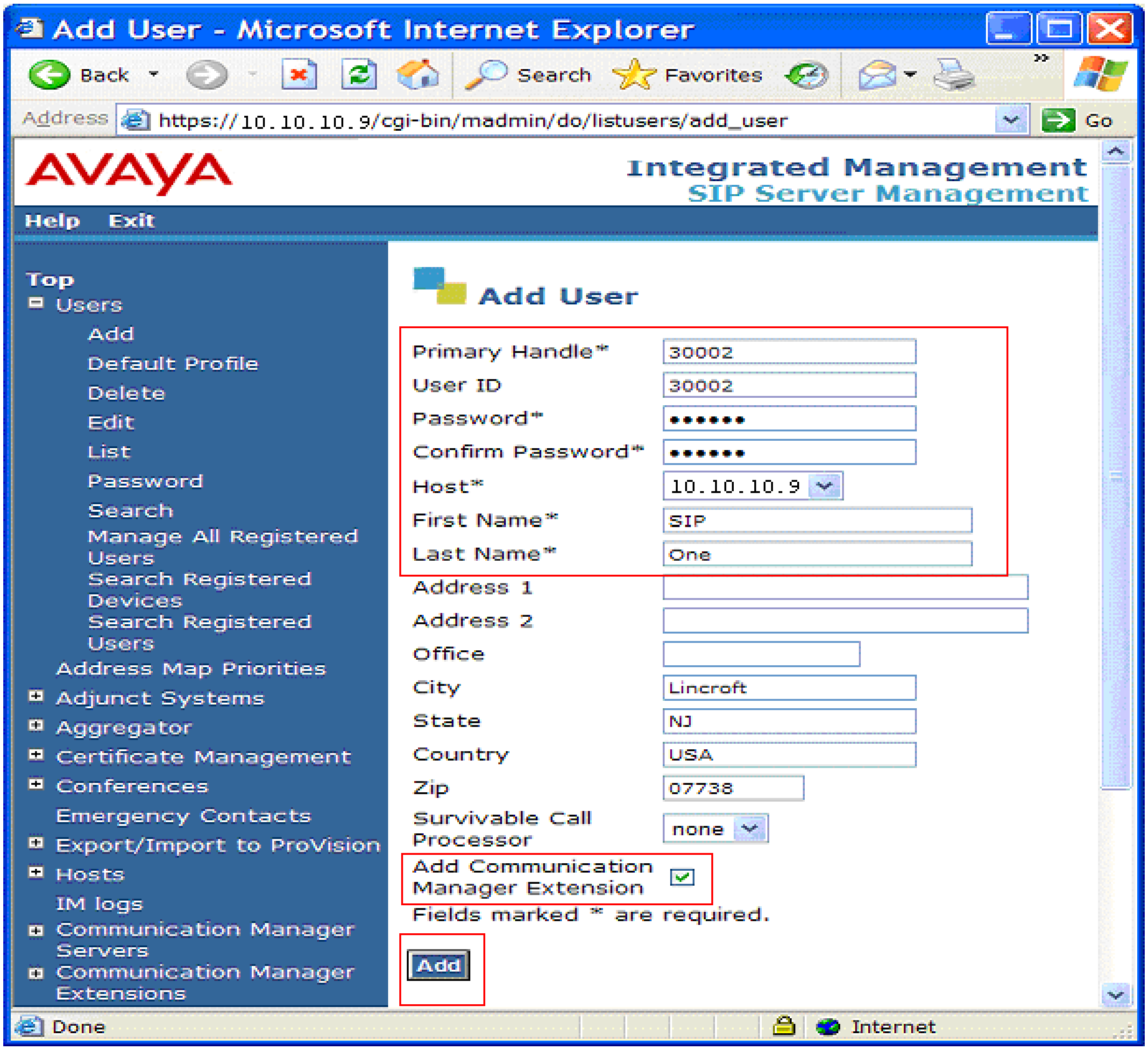

Figure 86: SES Add User

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

- In the **Add Communication Manager Extension** page, enter the extension number (e.g,, ***30002***) corresponding to the extension previously configured on Avaya Aura™ Communication Manager. The **Communication Manager Extension** and the **Primary Handle** are usually the same, but are not required to be.
- From the drop down menu, select the **Communication Manager Server** interface assigned to this extension.
- Click the **Add** button.

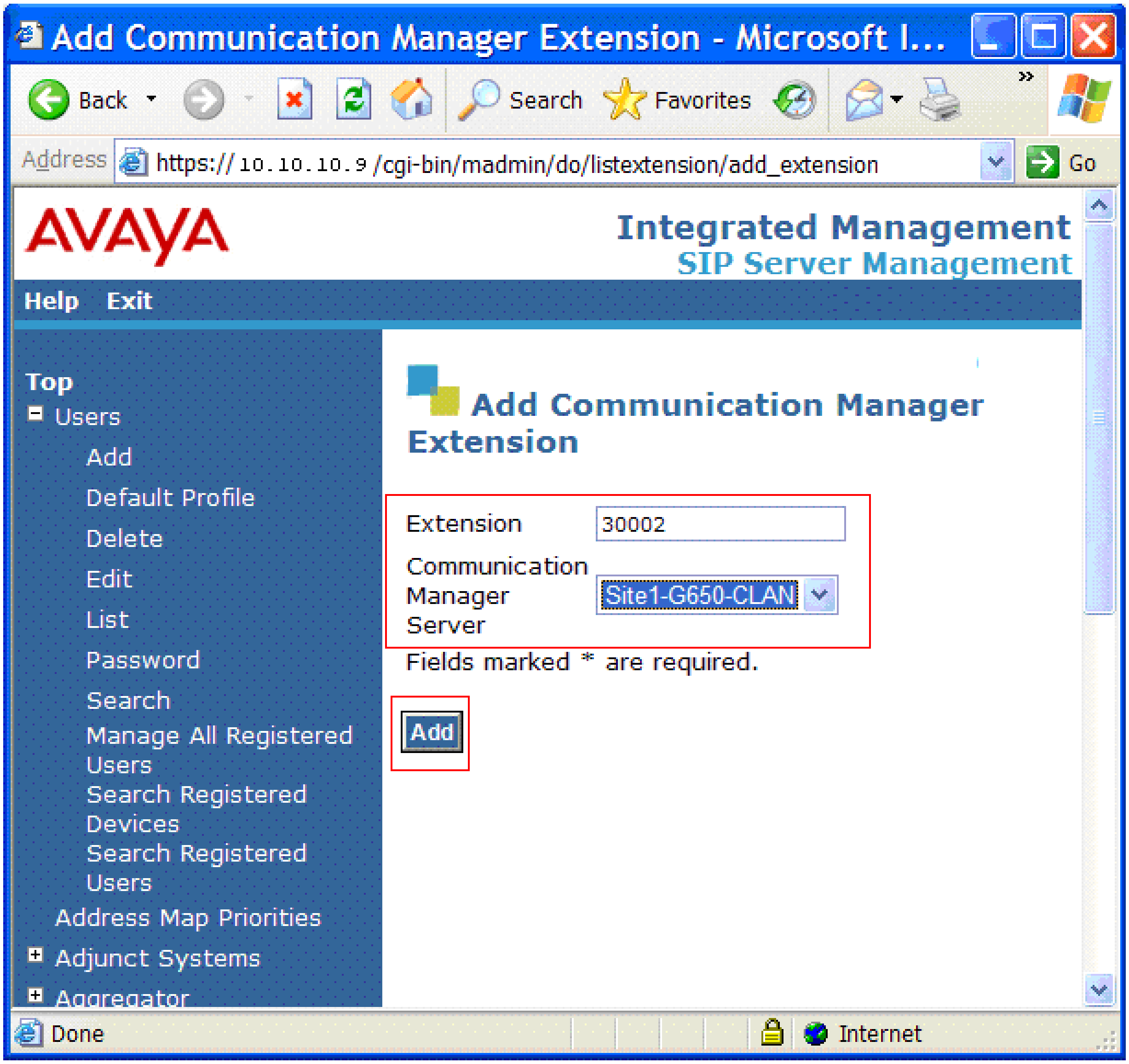

Figure 87: SES Add Communication Manager Extension

4.3. CustomerA CoRes SES Specific Provisioning

The following provisioning is specific to the Avaya Aura™ Communication Manager/SES Co-Resident platform.

The SES provisioning in the CustomerA CoRes platform is similar to the CustomerB SES provisioning with the exceptions described in the following sections.

4.3.1 Verifying/Configuring SES DNS Information

Note – The ***initial_setup*** procedure described in **Section 4.1.7** for provisioning DNS on the stand-alone SES is not available in the Avaya Aura™ Communication Manager Co-Resident platform.

Since the Avaya Aura™ Communication Manager and SES processes share the same platform, the Avaya Aura™ Communication Manager Management web interface is used to modify the DNS entry.

- From a web browser entre the IP address of Avaya Aura™ Communication Manager. After entering the appropriate credentials the window shown in **Figure 88** will open.
- Click on **Launch Maintenance Web Interface**. The Integrated Management window will open (**Figure 89**).

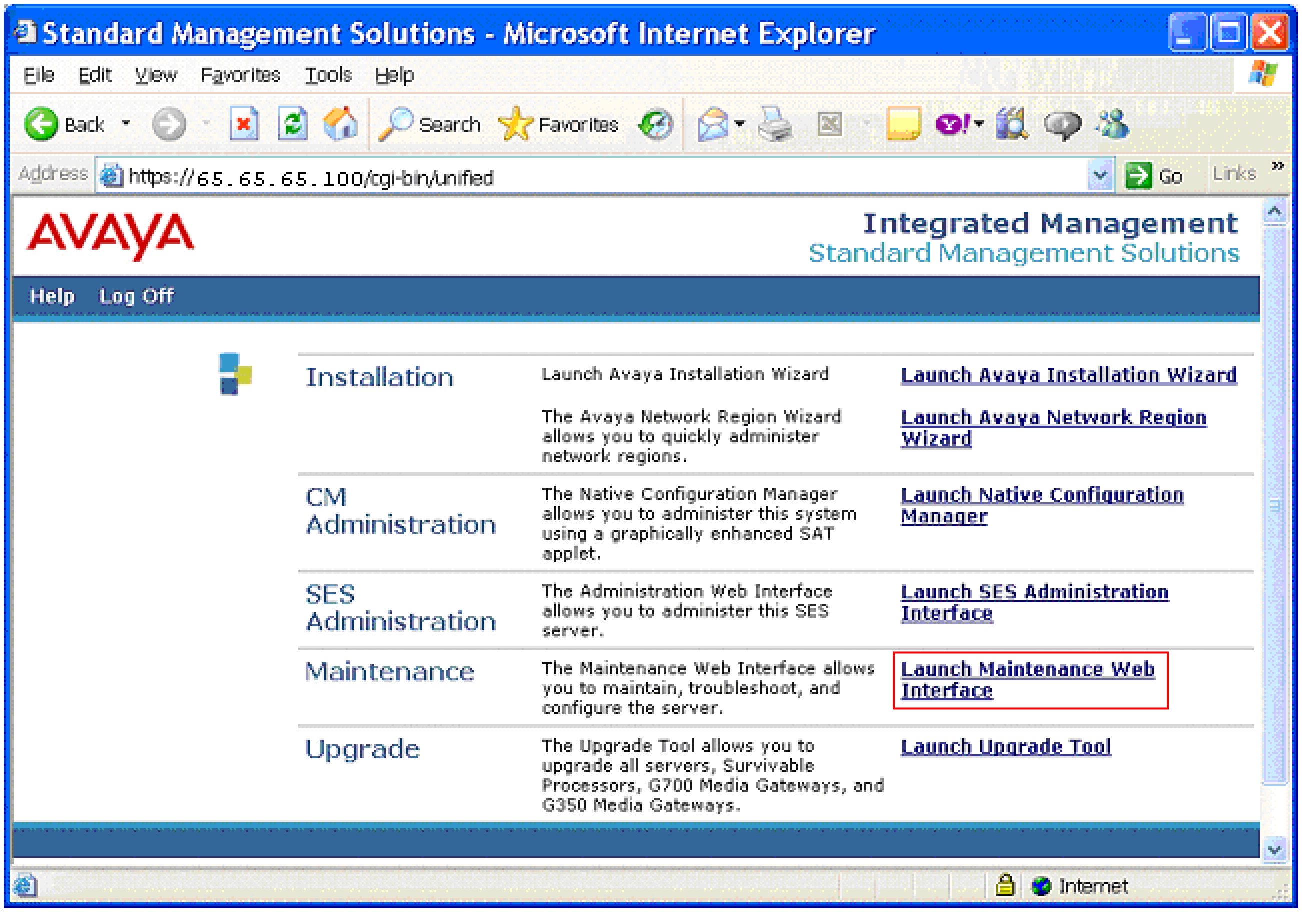

Figure 88: Avaya Aura™ Communication Manager Management Web Interface (CoRes)

- In the left hand column, click on **Configure Server**. (**Figure 89**).

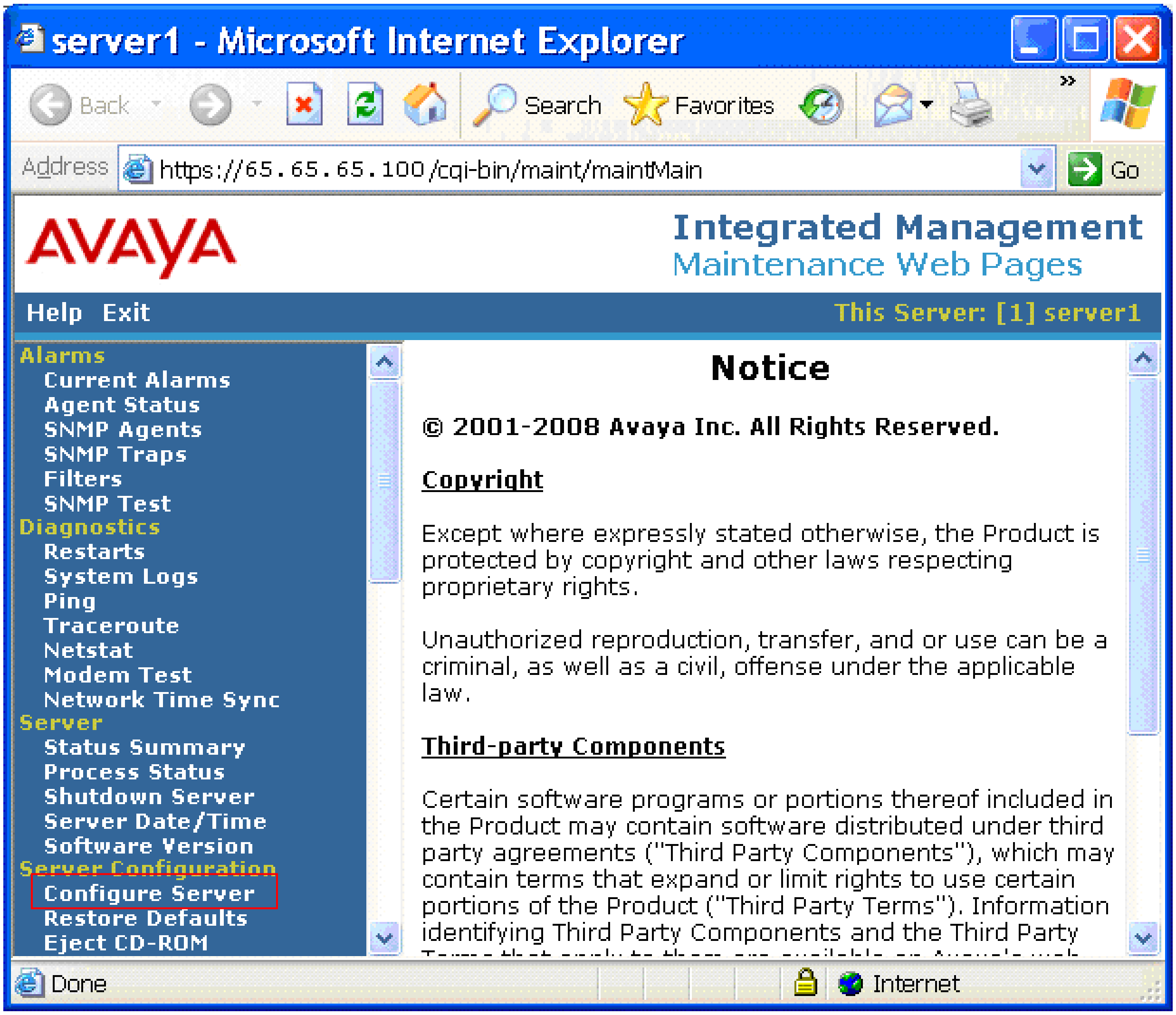

Figure 89: Avaya Aura™ Communication Manager Configure Server (CoRes)

- Two new screens will open. When prompted, enter press the **Continue** button on each.
- Then the **Configure Server** window shown in **Figure 90** will open. Select **Configure individual services** and click on **Continue**.

Figure 90: Avaya Aura™ Communication Manager Configure Individual Services (CoRes)

- Click on **Set DNS/DHCP** and the screen in **Figure 91** is displayed. Configure the **Name Servers** as provided by Verizon Business.

Figure 91: Avaya Aura™ Communication Manager Set DNS/DHCP (CoRes)

- Enter the IP address(s) of the DNS server(s) then click on the **Change** button (**Figure 92**).

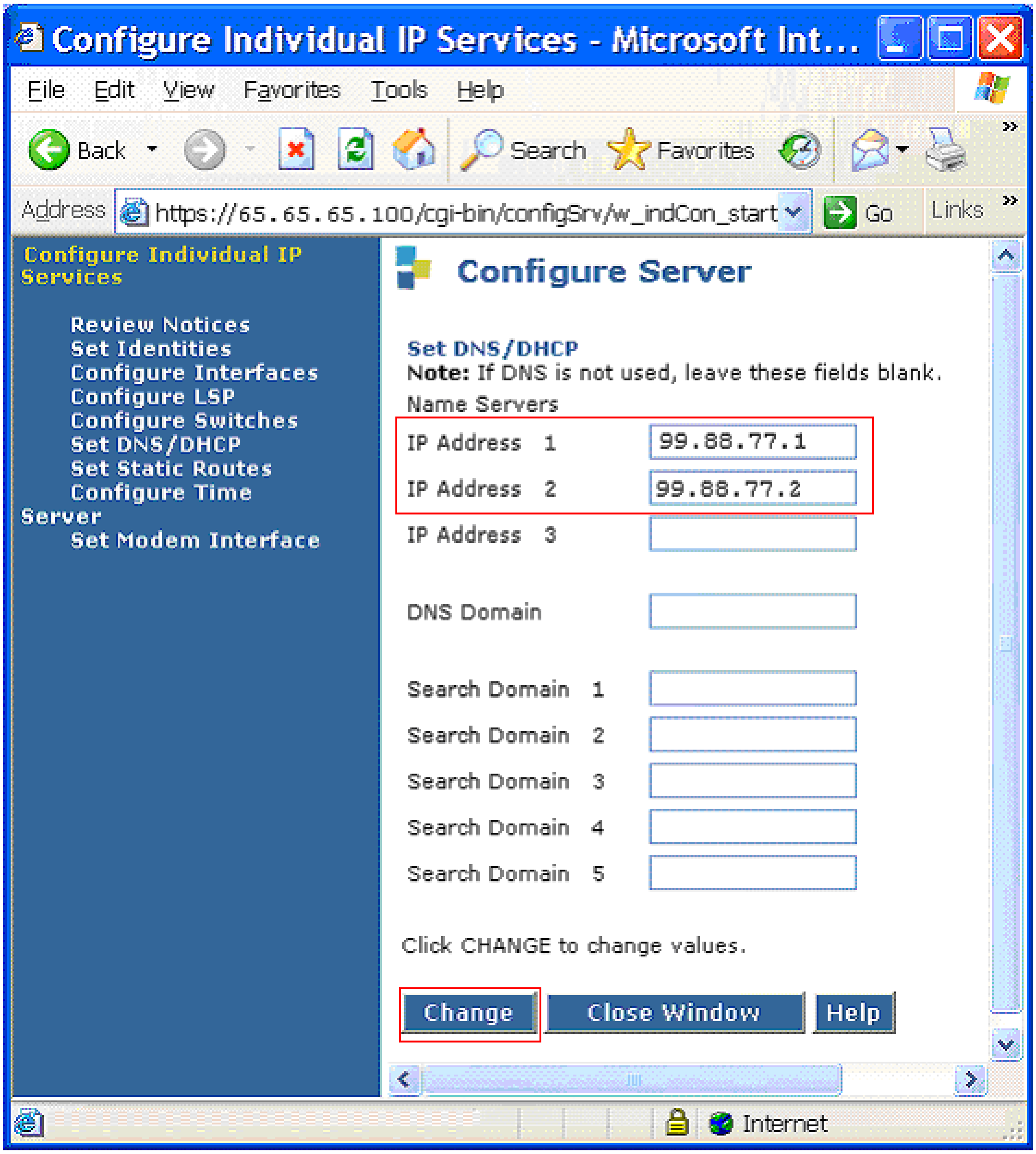

Figure 92: Avaya Aura™ Communication Manager DNS Server Entries (CoRes)

- When the changes are accepted successfully, the window shown in **Figure 93** will open. Select the Close Window Button. Then exit the browser session.

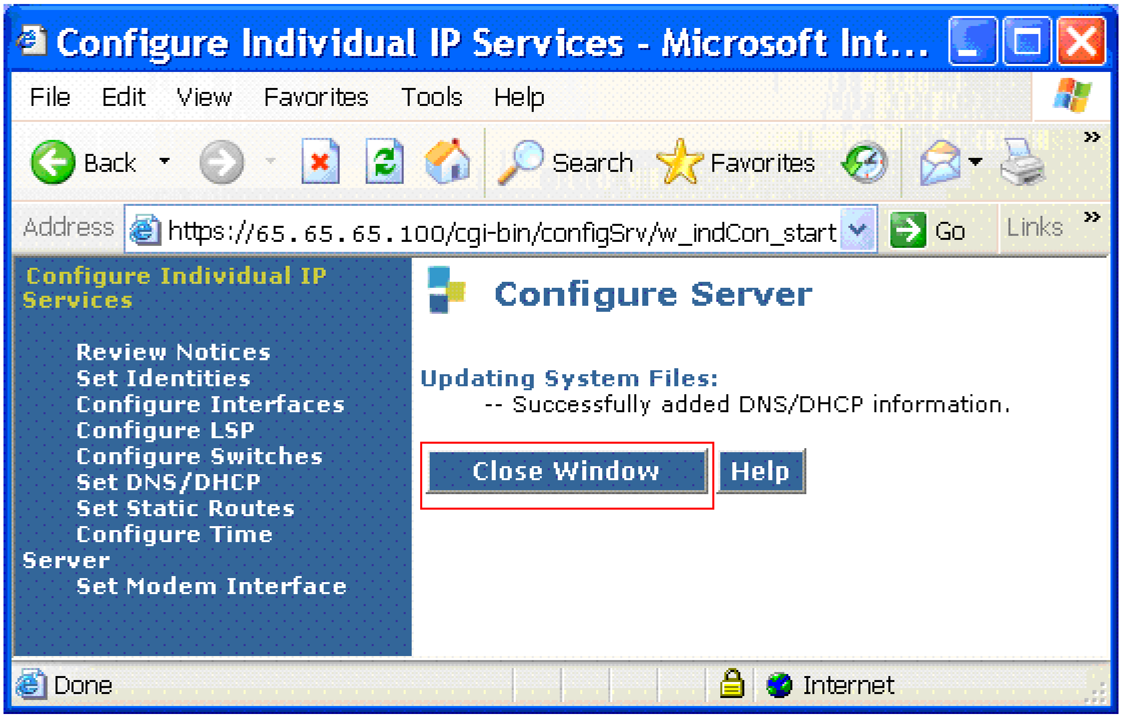

Figure 93: Avaya Aura™ Communication Manager DNS Server Successful Update (CoRes)

4.3.2 CoRes SES Web Interface

Since the Avaya Aura™ Communication Manager and SES processes share the same platform, the SES web provisioning is accessed via the Avaya Aura™ Communication Manager Management web interface.

- From a web browser entre the IP address of Avaya Aura™ Communication Manager. After entering the appropriate credentials the window shown in **Figure 94** will open.
- Click on **Launch SES Administration Interface** to access the CoRes SES web interface.

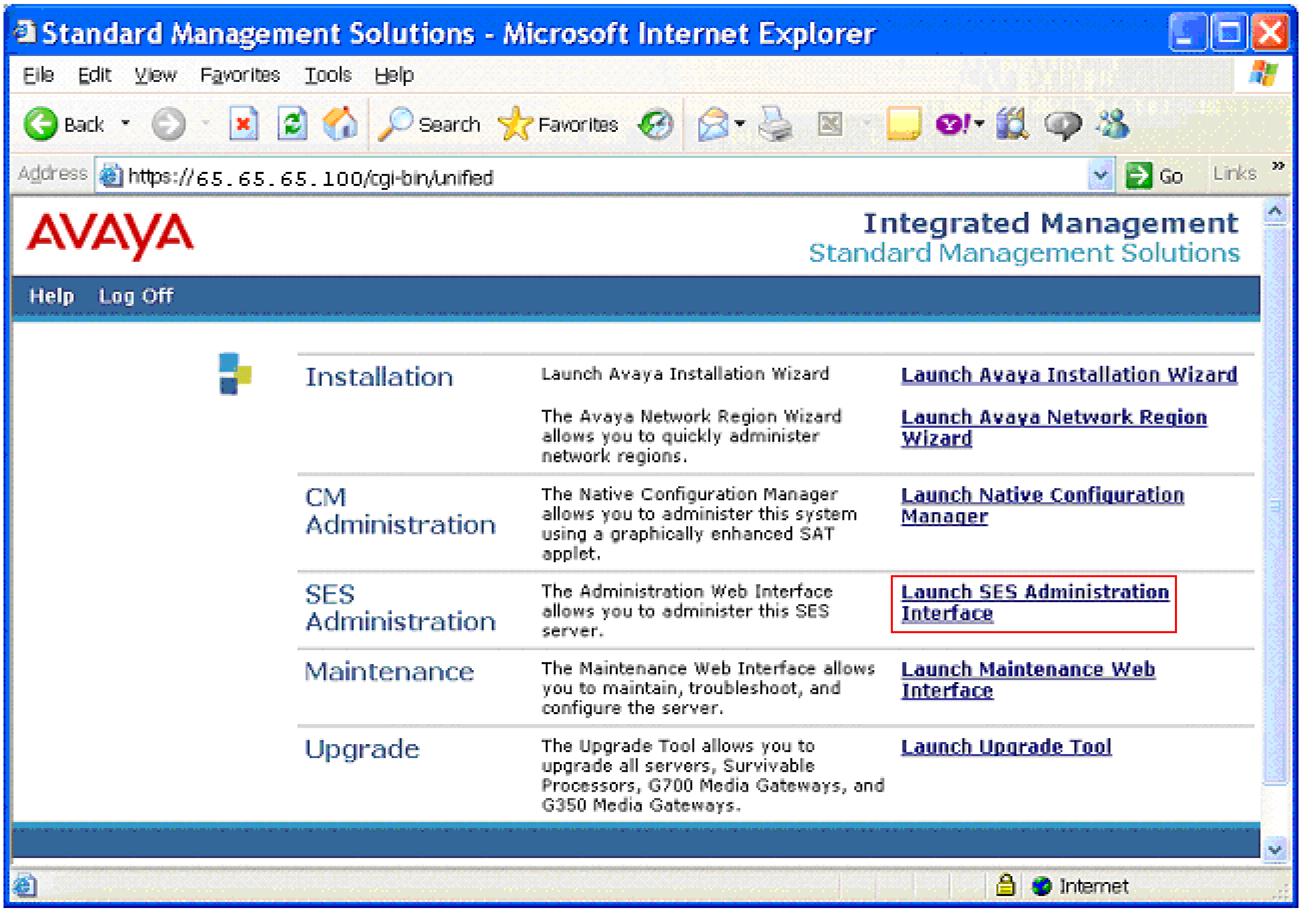

Figure 94: Access SES Administration Web Interface (CoRes)

All SES provisioning is the same as for a stand alone SES platform as shown in **Section 4.1** and **Section 4.2** with the exception of the Map Contact port. This difference is described in the next section (**Section 4.3.3**).

4.3.3 CoRes Communication Manager Map Contact Port

Since the Avaya Aura™ Communication Manager and SES processes share the same platform, the SES and Avaya Aura™ Communication Manager share the same IP address. For SIP transactions between the two processes, port 6001 is used. Therefore when SES Communication Manager Map Contacts are defined (see **Section 4.1.6**) for a CoRes platform, the following format is used.

sip:$(user)@192.168.1.100:6001;transport=tls

5. Verizon Business IPCC Service Configuration

In order to use Verizon Business IPCC Service, a customer must order service from Verizon Business by contacting their corporate web site at http://verizonbusiness.com or by contacting a Verizon Business sales representative.

Verizon Business provided the information contained in the following tables to complete the configuration in these Application Notes. This information was necessary to complete the Avaya

Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services administration discussed in the previous sections.

5.1. IP TF and IP IVR DID numbers

The following sample DID numbers were used for the CustomerA and CustomerB environments in the reference configuration. Actual DID numbers would be provided by Verizon in a customer configuration.

	DID numbers
IP TF	5558883221
	5558882380
	5557770107
	5557772649
IP IVR	5556664246
	5556664250
	5556664254
	5556664284

Table 11: CustomerA DID numbers

	Incoming Digits
IP TF	5559995245
	5559995250
	5559995259
	5559995264
IP IVR	5556664249
	5556664251
	5556664253
	5556664256

Table 12: CustomerB DID Numbers

5.2. Local FQDNs

The following sample FQDNs were used for the CustomerA and CustomerB environments in the reference configuration.

	FQDN
IP TF	customer.ay.com
IP IVR	customer.be.com

Table 13: Local FQDNs

5.3. Verizon DNS Information

The following sample DNS information was used for the CustomerA and CustomerB environments in the reference configuration. Actual DNS information would be provided by Verizon in a customer configuration.

DNS	
IP address	99.88.77.1, 99.88.77.2
FQDN	vziptf.dns.com

Table 14: DNS Information

5.4. Verizon Service Nodes

Verizon provided a list of all of their SIP service nodes that would send SIP call requests (INVITEs) to the CustomerA and CustomerB environments. This information was required for the Trusted Host list provisioned in the SES servers of both environments (see **Section 4.1.5**).

5.5. Verizon Business IP TF and IP IVR Service Router Access

Verizon provided specific network and security access information that was provisioned in the local Internet access edge router of the reference configuration. The edge router was configured, per Verizon Business specifications, to provide IPSEC tunnels over the Internet between the CPE router and the Verizon Business IPCC network. Separate access lists were defined for each simulated customer. For security reasons this information is not described in these Application Notes.

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Verizon Business IP Contact Center (IPCC) 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 the production Verizon Business IP Contact Center (IPCC) Service via Internet access facilities (**Figure 1**). 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 IP Contact Center (IPCC) Service and PSTN telephones.

- Verizon Business IP Toll Free (IPTF) service access.
- Verizon Business Interactive Voice Response (IP IVR) service access.
- Verizon Business Network Call Redirection (NCR) service access.
- Avaya Aura™ Communication Manager Network Call Redirection (NCR) feature.
- Verizon IP Incoming calls to Verizon Business provided DID numbers from PSTN telephones.
- Outgoing calls to PSTN telephones (transfers and redirection of incoming calls only).
- Calls using Avaya 9600 Series IP Telephones with the H.323 firmware configurations.
- Calls using Avaya 9600 Series IP Telephones with the SIP firmware configurations.
- 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 codec for voice calls.
- DTMF tone transmission using RFC 2833.
- Telephone features such as hold and transfer
- Direct IP-to-IP media (also known as “shuffling”) with IP telephones

6.2. Test Results

Interoperability testing of the Verizon Business IP Contact Center (IPCC) Service with the Avaya SIP trunking configuration was completed with successful results.

As illustrated in these Application Notes, Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services can be configured to interoperate successfully with Verizon Business's IP Contact Center services suite inclusive of VoIP Inbound, IP Contact Center, IP-IVR SIP trunk services. This solution provides users of Avaya Aura™ Communication Manager the ability to support inbound toll free calls over a Verizon Business VoIP Inbound SIP trunk service connection. In addition, these application notes further demonstrate that the Avaya Aura™ Communication Manager's implementation of SIP Network Call Redirection (SIP-NCR), can work in compliment with Verizon's Business's IP Contact Center and IP-IVR services implementation of SIP-NCR to support call redirection over SIP trunks. This capability includes support of outbound calls for the specific call redirection scenarios documented in this application note.

Please note that the sample configurations shown in these application notes are representative of a basic enterprise customer configuration and as such are intended to provide configuration guidance to supplement other Avaya product documentation. Finally, the test results indicated in these application notes are based upon formal interoperability compliance testing that was conducted as part of the Avaya DevConnect Service Provider program. As part of this program, compliance testing of this solution was conducted with the full support and collaboration with Verizon's CPE Systems Interoperability Test Lab.

7. Verification Steps

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

7.1. Call scenarios

- Dial IP TF and IP IVR DID numbers from PSTN and verify the incoming voice calls are received successfully on the associated Avaya Aura™ Communication Manager station, there is 2-way audio, and that the calls remain connected for at least 3 minutes.
- After an inbound call is received, perform attended and unattended transfers from the station to another IP TF or IP IVR destination.
- After an inbound IP IVR call is received, enter the appropriate Verizon Business IP IVR access code from the Avaya Aura™ Communication Manager station and redirect the call to a new destination by entering an IP TF DID.
- Verify that calls are properly disconnected when either end disconnects.
- Verify Verizon Business NCR is provisioned to redirect a call based on a SIP call clearing code (e.g, 403). From PSTN call an IP TF or IP IVR DID that is ***not*** associated with a station provisioned on Avaya Aura™ Communication Manager. Verify that Verizon Business NCR redirects the call on receipt of the 403 from Avaya Aura™ Communication Manager.
- From PSTN call an IP TF or IP IVR DID that is associated with an Avaya NCR vector provisioned on Avaya Aura™ Communication Manager. Verify that the call is redirected to the provisioned destination.

7.2. Troubleshooting

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

The following Avaya Aura™ Communication Manager and SES commands may be used to troubleshoot call completion issues.

7.2.1 Avaya Aura™ Communication Manager

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

7.2.2 Avaya SES

- SES Trace Logger – The trace logger is accessed via the SES web GUI.
  1. Connect to the SES web GUI using a web browser and appropriate credentials. The screen shown in **Figure 95** will open.
  2. Select ***Launch SES Administration Interface***

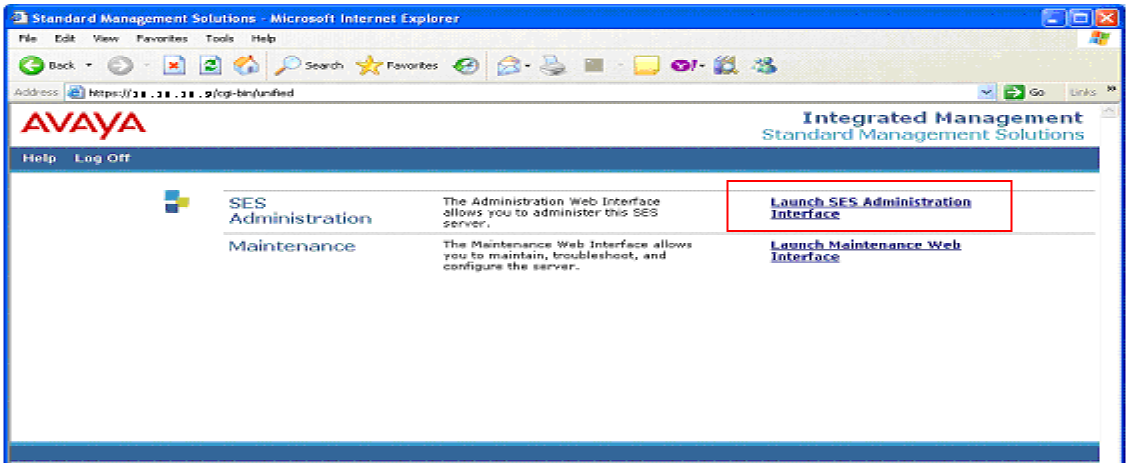

Figure 95: SES Administration/Maintenance Web Interface

3. The "Top" window will open (**Figure 96**). Select ***Trace Logger*** and ***Configure Filters***.

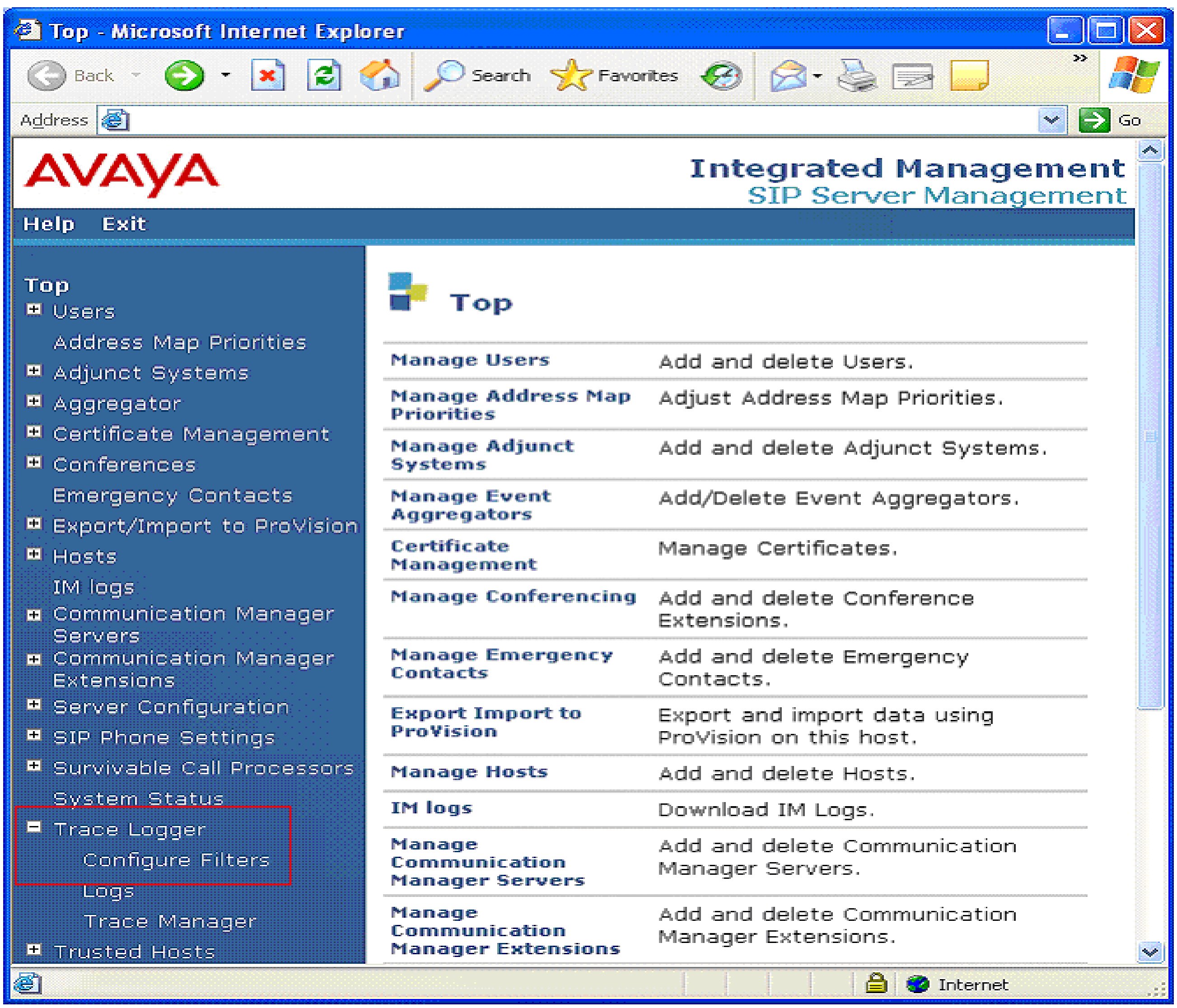

Figure 96: SES "Top" Screen

4. The Filter Configuration window will open (**Figure 97**). Select ***Add New Rule To Filter***.

Figure 97: Create Trace Filter

5. Create a filter to capture all traffic as shown in **Figure 98**.
   - **Filter Label** → Enter a name for the filter.
   - **Message Type** → Enter ***Any***
   - **From** → Enter **.***
   - Let all other fields default.
   - Select **Update** to save the filter. The select **Continue** when prompted.

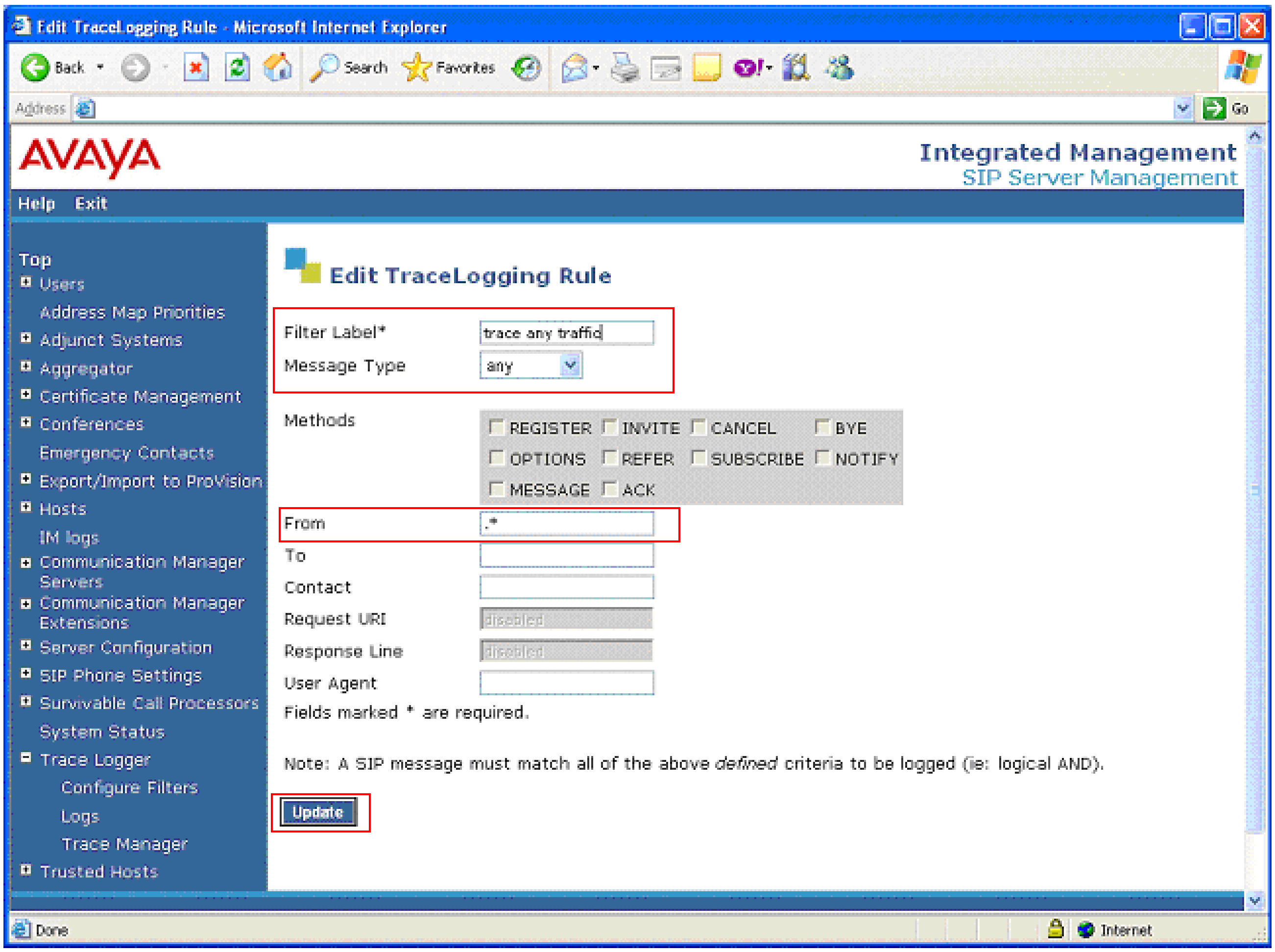

Figure 98: Filter to Capture All Traffic

6. The new filter will be displayed (**Figure 99**).

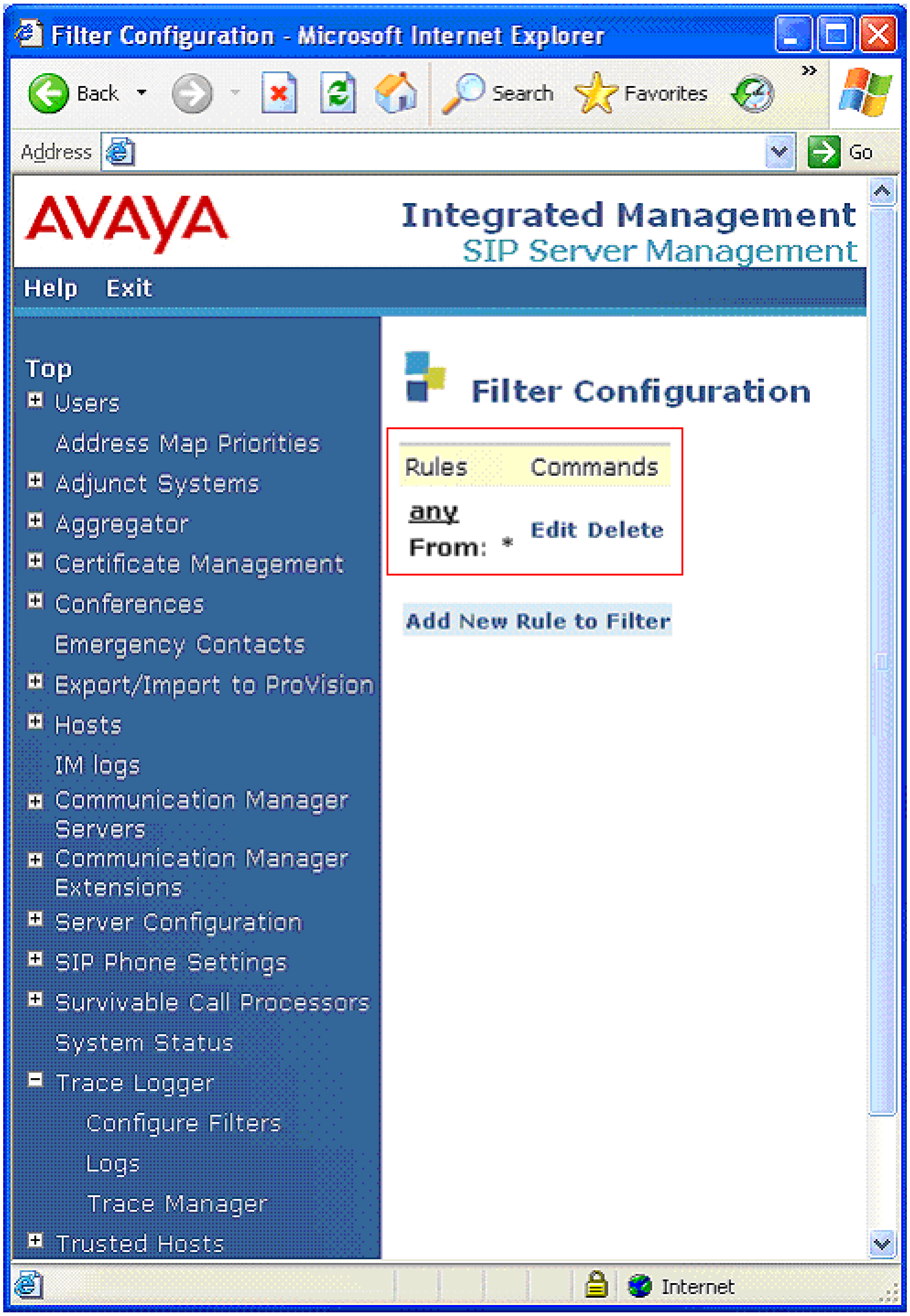

Figure 99: Filter to Capture All Traffic

7. Select ***Trace Manager*** and the **Trace Manager** window is displayed (**Figure 100**). Select ***Start Tracing*** (the screen will update to display **Tracing is on** and **Stop Tracing**). Run the test. When the test is completed select ***Stop Tracing***.

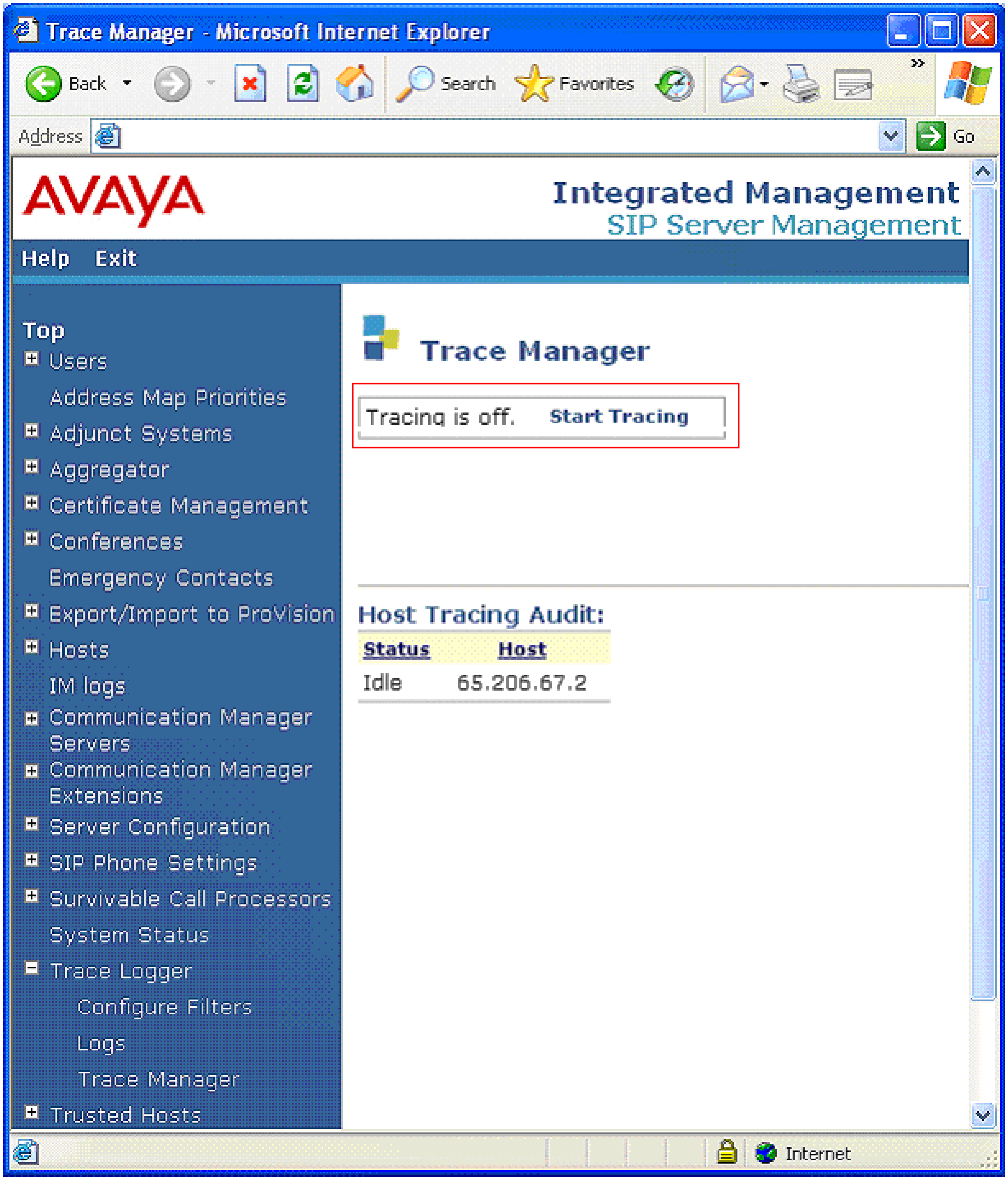

Figure 100: Start Trace

8. When the tracing is complete select Logs and the TraceLogs window will open displaying the captured trace (**Figure 101**). Select ***Down load*** and then ***OK*** when prompted. You will be asked to save the file. Once the file is saved it may be viewed with any test editor.

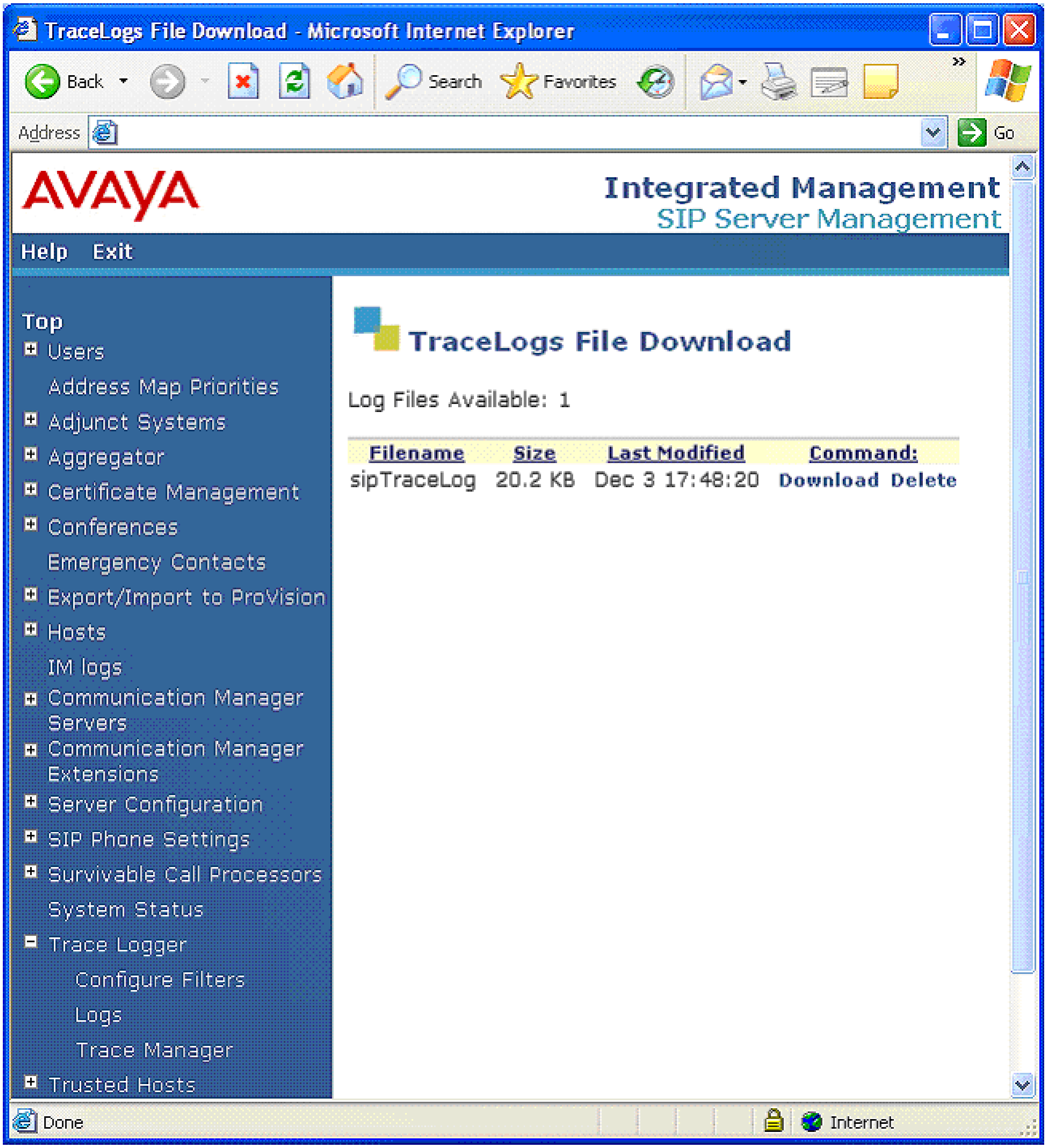

Figure 101: Start Trace

9. You will be asked to save the file. Once the file is saved it may be viewed with any test editor. **Figure 102** shows sample output from an SES Trace Log.

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

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

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

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


-----------------------------------------------------------------
```

Figure 102: Sample SES Trace Log.

8. Support

8.1. Avaya

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

8.2. Verizon

For technical support on Verizon Business IPCC Service, visit their online support at http://www.verizonbusiness.com/us/customer/

9. References

9.1. Avaya

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

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

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

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

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

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

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

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

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

[9] *Application Notes for Configuring Alternate Methods of Domain Based Routing in an Avaya SIP Trunk Architectur*e

9.2. Verizon

Documentation on Verizon Business IPCC services may be obtained by contacting your Verizon Business representative or visit http://www.verizonbusiness.com/products/contactcenter/ip/ for more information.

10. APPENDIX A: Specifying Pattern Strings in Address Maps

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

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

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

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

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

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

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

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

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

10.1. Address Map Caveats

10.1.1 Over-lapping Address Map Strings

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

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

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

And the following dialed string is received:

```
555121211111
```

The Avaya Aura™ SIP Enablement Services may send the call to 10.10.10.10 instead of the intended destination of 20.20.20.20 by matching on 5551212.

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

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

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

And the following dialed string is received:

```
5551212121211111
```

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

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

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

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

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

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

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

11. APPENDIX B: SES TimerB Value and Multiple DNS SRV Response Entries.

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

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

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

Where:

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

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

30 > 4 * 7000

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

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

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

Note – There is another method for configuring Domain Base Routing where Avaya Aura™ Communication Manager uses SES Host Maps. This method for configuring Domain Base Routing may be used as an alternative to modifying TimerB. See reference **[9]**

12. APPENDIX C: Verizon Business IPCC Service Portfolio

12.1. IP Enabled Contact Center Portfolio

Feature	Description	Benefit to the Customer
Domestic Toll Free 8XX (NANP) Origination	Supports Traditional U.S. and Canada Toll Free numbers.	Leverages advanced features, route plans and carrier-grade characteristics of traditional Toll Free network.
International Toll Free (ITFS/UFIN) Origination	Supports Traditional International Toll Free numbers.	Leverages features and carrier-grade infrastructure of International network.
IP (URI) and TDM Terminations	Supports Traditional TDM (PSTN) and new IP terminations.	Benefits of IP and helps improve utilization of existing TDM Contact Center installations.
Network Call Redirect (NCR)	Redirect or overflow calls in real-time according to outage, busy, or other customer-specific conditions using both SIP error and ISUP (ISDN User Part) cause factors.	Provide seamless routing and overflow for both TDM and SIP end-points. NCR's call-by-call overflow functionality is programmed via Toll Free Network Manager (TFNM), a web-based application.
Network-based TDM to IP Conversion	TDM-to-IP protocol conversion that is traditionally performed by a customer premises Gateway.	Can eliminate the capital, scalability and operational concerns associated with customer-owned premises Gateways.
IP TF and IP IVR Service	Continued support for the combined Toll Free and IVR services required by traditional contact centers.	Full integration of Inbound TDM originations, IP conversion, IP call treatment and the capability to terminate calls to IP and TDM call centers - seamlessly and including agent transfers (IP IVR hybrid transfers as described below).
Session Initiation Protocol (SIP) signaling per IETF RFC 3261, etc.	Open standard to set up, modify, and terminate voice calls which allows for interoperability between SIP-enabled devices.	Compatibility and access to multiple types of termination systems (IP end-points) that are SIP compliant (i.e., SIP phones, PBX/ACD, IVR, etc.).
SIP Registration	REGISTER is a type of SIP request that allows a device to identify its particular address with the network and allows IP to support personal mobility.	Ability to support a SIP phone connectivity from many location (i.e., work from home agents).
SIP (REFER) Transfer	A SIP REFER request enables the sender to instruct the receiver to contact a third party - with the contact details of the third party included in the request.	IP CC supports SIP REFER transfers (blind transfers) between SIP devices. Note: SIP REFER Transfers must be initiated through SIP signaling (not DTMF), must terminate to another SIP device and cannot provide progress messages or error recovery.
IP IVR Hybrid Transfer	Capability to transfer calls via DTMF and SIP (REFER) Transfer commands.	Support agent transfers to IP or TDM terminations, where TDM terminations can initiate transfer via DTMF and IP terminations can initiate transfer via SIP (REFER) Transfer or DTMF.
Customer Test Center Support	Verizon Business live network environment for customers and vendors to test and get hands-on experience with proposed network, hardware, and software solutions.	Capability to test and verify critical IP functionality, actual IP TF and IP IVR terminations and transfers with the customer's associated CPE, before deployment in mission critical environments.

Service Controller	Key system responsible for routing of the SIP calls, managing sessions, tracking bandwidth utilization, and registration of SIP phones and trusted entities (enterprise gateways, IP-ACDs, IP-PBXs, etc.).	Provides common infrastructure that simplifies application deployment and will support future IP services such as "presence."
Session Border Controllers	Systems which control real-time session traffic at the signaling, call control, and packet layers across network borders or between network segments.	Enable the use of MPLS (Private IP) for access; will support other protocols (such as H.323) in 2008.
SODEE (Service on Demand Execution Environment) Architecture	Network capabilities that separate the application from the network and can provide more choice for the customer.	Provides future migration path to IP Multimedia Subsystem (IMS) for customer IP investments and new application solutions.
NGSN IP Terms	SIP-compliant connectivity to a Customer's IP Contact Center, using Verizon's IP Toll Free service, for Toll Free calls that are processed by the ECR TDM (NGSN) platform	Offers SIP terminations for an existing network IVR (ECR) application and provides seamless service from both TDM and IP CPE, which allows customers to upgrade CPE at their own pace.
Intelligent Call Routing Gateway	Provides a customer real-time capability to control the routing of each call to their Contact Center Agents.	Allows the customer's Cisco or Genesys Intelligent Routers to provide routing for each call into their IP or TDM contact Centers.
IP Intelligent Call Routing Integration	Provides a customer real-time capability to control network based IPIVR resources to manage the treatment, routing and queuing of their calls, before, during or after they are handled by their Contact Center Agents.	Allows the customer's Cisco or Genesys Intelligent Routers to provide skills based routing and handling for each call in their IP or TDM contact Centers, using the Verizon Business IP network infrastructure.
User to User Interface (UUI)	A method of passing information from one user to another in a SIP telephony environment. This information is transmitted via a single call transfer and performed utilizing either a new or existing SIP Header or a URI (Universal Resource Identifier).	Verizon's UUI implementation is consistent with the Avaya recommendation for the international standard but is not vendor specific and does not require any order entry or provisioning.

12.2. IP Toll Free

IP Toll Free provides TDM to IP conversion and advanced feature functionality for domestic (TF) and international (ITFS and UFIN) Toll Free calls using Session Initiation Protocol (SIP) signaling per IETF RFC 3261 over Internet Dedicated Access (IDA) or Private IP (PIP) terminations.

Feature	Description	Benefit to the Customer
Feature parity with Verizon Toll Free Service	Advanced features allow customers to tailor the origination, routing and termination of their Verizon Business IP Toll Free calls.	Requires little migration redesign and allows the use of existing route plans to route calls via advanced features: TOD, DOW, Geographic, ANI, etc.
Network Call Redirect (NCR)	Redirect or overflow calls in real-time according to outage, busy, or other customer-specific conditions using both SIP error and ISUP (ISDN User Part) cause factors.	Provide seamless routing and overflow for both TDM and SIP end-points. NCR's call-by-call overflow functionality is programmed via Toll Free Network Manager (TFNM), a web-based application.
Toll Free Network Manager (TFNM) compatibility	TFNM allows customers to modify their IP Toll Free routing structures from their desktops.	Provides near real-time customer control over hybrid (TDM and IP) routing scenarios.

Voice over IP conversion in the network	Network-based Gateways that convert TDM to VoIP.	Can eliminate the burden of owning and managing costly gateway equipment.
Call transfers via SIP (REFER) Transfer	A SIP (REFER) Transfer request enables the sender to instruct the receiver to contact a third party - with the contact details of the third party included in the request.	Enables agents to make unattended transfers via IP without the use of DTMF tones and in a more cost effective manner.
Automatic Device Registration	REGISTER is a SIP method used by a user-agent client to log in and identify its address with a SIP server, thereby letting the registrar know the address at which the user can be reached.	Provides automatic resolution of SIP devices such as phones and ACD/PBX's (devices auto configure when connected to the network), allowing for simpler moves/adds/changes/deletes.
Compression G.729 (Note: Music, DTMF or fax tones cannot be transported reliably with this codec)	G.729 is a compression algorithm that compresses voice audio in chunks of 10 milliseconds, operates at 8 kbit/s (37 kbit/s including overhead) and is commonly used in VoIP.	When G.729 is used in IP TF calls it increases data bandwidth efficiency (typically allows 41 concurrent calls vs. 23 utilizing TDM on the same T1).
Bandwidth control	Bandwidth management tools to control the service, throughput and the efficiency of network facilities.	Allows administrators to manage Quality of Service (QoS) and committed access rate (CAR).
NGSN IP Terms	SIP-compliant connectivity to a Customer's IP Contact Center, using Verizon's IP Toll Free service, for Toll Free calls that are processed by the ECR TDM (NGSN) platform	Offers SIP terminations for an existing network IVR (ECR) application and provides seamless service from both TDM and IP CPE, which allows customers to upgrade CPE at their own pace.
Intelligent Call Routing Gateway	Provides a customer real-time capability to control the routing of each call to their Contact Center Agents.	Allows the customer's Cisco or Genesys Intelligent Routers to provide routing for each call into their IP or TDM contact Centers.
User to User Interface (UUI)	A method of passing information from one user to another in a SIP telephony environment. This information is transmitted via a single call transfer and performed utilizing either a new or existing SIP Header or a URI (Universal Resource Identifier).	Verizon's UUI implementation is consistent with the Avaya recommendation for the international standard but is not vendor specific and does not require any order entry or provisioning.

12.3. IP Interactive Voice Response (IP IVR)

IP IVR provides network-level IVR call treatment and routing for IP Contact Centers, using Session Initiation Protocol (SIP) signaling per IETF RFC 3261, as well as seamless call treatment and transfers for "hybrid" (TDM and IP) call center environments.

Feature	Description	Benefit to the Customer
Carrier-grade IP infrastructure for call treatment and routing	Toll-Quality voice, redundant and managed network capacity.	Provides high-capacity, native-IP IVR functionality that extends the benefits of IP CPE deployments without added protocol conversions.
Robust IVR call treatment and routing capabilities	Support currently available network IVR features such as menu routing, transfer, message announcement and others.	Allows the re-use of existing Network IVR (Hosted IVR-ECR) application call flow logic with few migration design requirements.
Hybrid terminations	Terminate to and transfer from both, TDM	Provides seamless integration of TDM and IP

(TDM and IP) and transfers	and IP end-points.	call types and allows customer to migrate to IP at their own pace.
Network Call Redirect (NCR)	Redirect or overflow calls in real-time according to outage, busy, or other customer-specific conditions using both SIP error and ISUP (ISDN User Part) cause factors.	Provide seamless routing and overflow for both TDM and SIP end-points. NCR's call-by-call overflow functionality is programmed via Toll Free Network Manager (TFNM), a web-based application.
Basic SIP REFER Transfer, Full SIP Transfer, and Take Back and Transfer capabilities	Agent-driven selection from available transfer types via SIP or DTMF signaling.	Allows comprehensive management control and cost effective call transfers.
Simple system for migrating Hosted IVR - ECR applications	Dedicated team and proven process to migrate network-IVR (ECR and other carrier) applications.	Allows existing customers to add IP functionality without rewriting complex applications and scripts.
NGSN IP Terms	SIP call terminations connecting Toll Free calls that are processed by the ECR TDM platform (NGSN) to a Customer's IP Contact Center using Verizon's IP Toll Free service.	Offers terminations for an existing network IVR (ECR) application.
G.711 Codec Support (Recommended codec for Speech Recognition, Music, DTMF and fax tones)	G.711 is a standard for audio companding. It represents 8 bit compressed PCM samples for signals of voice frequencies, sampled at the rate of 8000 samples/second and 8 bits per sample.	Provides acceptable Toll-Quality Voice and leverages 64kbps bit rate (approximately 80kbps with overhead) to provide converged access with seamless DTMF and SIP transfers as well with reliable transmission for Speech Recognition and Fax applications.
IP Intelligent Call Routing Integration	Provides a customer real-time capability to control network based IPIVR resources by management of the treatment, routing and queuing of their calls, before, during or after they are handled by their Contact Center Agents.	Offers capability of utilizing a customer's current premise based equipment (Cisco & Genesys Intelligent Routers).
User to User Interface (UUI)	Method of passing information from one user to another in their telephony environment. This information is completed via a single call transfer and performed utilizing either a URI (Universal Resource Identifier) or SIP Header.	Offers the capability of passing information from one user agent to another via SIP or URI and does not require any order entry or provisioning.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect program at devconnect@avaya.com.