Avaya Solution & Interoperability Test Lab

Application Notes for DiVitas Mobile Unified Communications with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the DiVitas Mobile Unified Communications solution with Avaya Communication Manager and Avaya SIP Enablement Services. DiVitas consists of two key components: the DiVitas Client and the DiVitas Server. The DiVitas Client is installed on a mobile handset, such as the Nokia E- and N-Series, and provides access to DiVitas mobile communications features. The DiVitas Server monitors connections with DiVitas Clients and proactively identifies the optimal network connection for each call. The DiVitas solution provides the seamless convergence of WiFi and cellular networks enabling roaming (back and forth) between the two networks. The DiVitas Client increases user's accessibility by extending their office phone to the WiFi and cellular networks. This is accomplished by mapping the DiVitas Client to a desktop phone on Avaya Communication Manager. Incoming calls can then ring at both phones simultaneously and the call can be answered at either phone. In addition, Avaya Communication Manager and Avaya SIP Enablement Services provides DiVitas Clients with access to the PSTN and to other local stations in the enterprise through a SIP trunk integration.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to integrate the DiVitas Mobile Unified Communications solution with Avaya Communication Manager and Avaya SIP Enablement Services. DiVitas consists of two key components: the DiVitas Client and the DiVitas Server. The DiVitas Client is installed on a mobile handset, such as the Nokia E- and N-Series, and provides access to DiVitas mobile communications features. The DiVitas Server monitors connections with DiVitas Clients and proactively identifies the optimal network connection for each call. The DiVitas solution provides the seamless convergence of WiFi and cellular networks enabling roaming (back and forth) between the two networks. The DiVitas Client increases user's accessibility by extending their office phone to the WiFi and cellular networks. This is accomplished by mapping the DiVitas Client to a desktop phone on Avaya Communication Manager. Incoming calls can then ring at both phones simultaneously and the call can be answered at either phone. In addition, Avaya Communication Manager and Avaya SIP Enablement Services provides DiVitas Clients with access to the PSTN and to other local stations in the enterprise through a SIP trunk integration.

To provide voicemail coverage for the DiVitas Clients using Avaya Modular Messaging, see [4], *Application Notes for DiVitas Mobile Unified Communications with Avaya Modular Messaging*.

1.1. Interoperability Compliance Testing

The focus of the interoperability compliance test was to verify call establishment and basic telephony features between DiVitas Clients registered to the DiVitas Mobile UC Server, telephones on Avaya Communication Manager, and the PSTN. The DiVitas Clients were installed on Nokia E71 mobile handsets and mapped to an H.323 IP station on Avaya Communication Manager. The general test approach was to verify the following functionality:

- Establishing calls between DiVitas Clients and SIP, H.323, digital, and analog stations on Avaya Communication Manager and Avaya SES.
- Establishing calls between the DiVitas Clients and the PSTN.
- Establishing calls with the DiVitas Clients while they were in WiFi and Cellular modes.
- Ability to hold a call, transfer a call, and establish a conference.
- Conferencing using the DiVitas Bridge.
- Conferencing initiated by a station on Avaya Communication Manager.
- Displaying the calling party number on the DiVitas Clients.
- Simultaneous ringing on a desktop IP phone and DiVitas Client when an incoming call is received.
- Ability of DiVitas Clients to roam between the WiFi and Cellular networks.
- Call establishment using G.711mu-law codec.

The serviceability testing focused on verifying the ability of the DiVitas Mobile UC Server to recover from adverse conditions, such as power failures and disconnecting cables to the IP network. In addition, the ability of the solution to recover from Avaya S8730 Server interchange and from cycling power on Avaya SES was also verified.

1.2. Support

For technical support on the DiVitas Mobile Unified Communications Solution and how to configure dual mode handsets connected to it, consult the support pages at http://www.divitas.com/support.html or contact technical support at:

- Telephone: (866) 857-6087
- E-Mail: support@divitas.com

2. Reference Configuration

Figure 1 illustrates a sample configuration consisting of a pair of Avaya S8730 Servers running Avaya Communication Manager, an Avaya G650 Media Gateway, Avaya SIP Enablement Services (SES), and dual-mode wireless telephones registered with DiVitas Mobile Unified Communications. Each DiVitas Client was paired with an H.323 IP telephone on Avaya Communication Manager. The solution described herein is also applicable to other Avaya Servers and Media Gateways. Avaya 4600 Series H.323 IP Telephones, Avaya 9600 Series SIP Telephones, and Avaya analog and digital telephones were included in the configuration to verify calls with the SIP-based DiVitas Mobile UC Server and DiVitas Clients. Calls were also routed from the DiVitas Clients to the PSTN through Avaya Communication Manager and Avaya SES. A SIP trunk was established between the DiVitas Mobile UC Server and Avaya SES and the DiVitas Server was configured as a trusted host in Avaya SES. The Avaya G650 Media Gateway connected to the PSTN via an ISDN-PRI trunk.

Note: While a DiVitas Client is in Cellular mode, it communicates with the DiVitas Mobile UC Server through a Cellular Voice Channel (CVC). When in Cellular mode, the DiVitas Client places a call using a PSTN number assigned to the DiVitas Mobile UC Server. CVC enables the client to make and receive voice calls and use voice features such as hold and resume. CVC supports multiple simultaneous calls and is used when the Cellular Data Channel (CDC) is not available, which requires a public IP address assigned to the DiVitas Mobile UC Server. In this configuration, CVC was used.

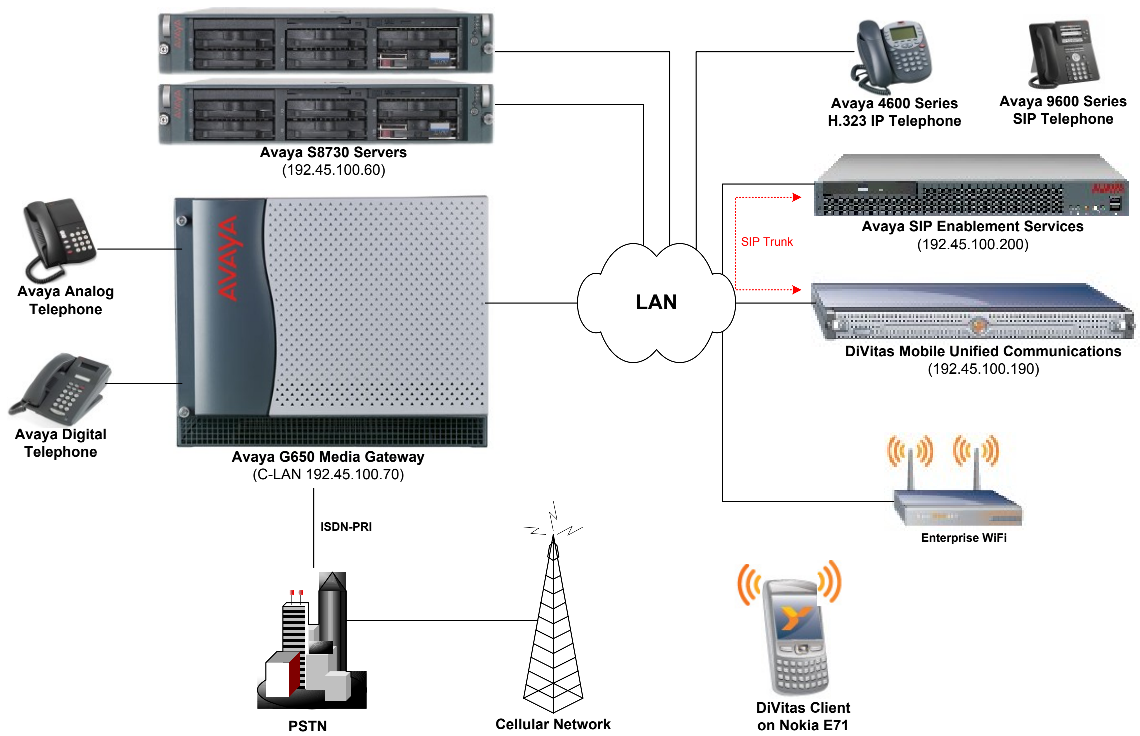


Figure 1: DiVitas Mobile Unified Communications with Avaya SIP-based Network

3. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8730 Server with G650 Media Gateway	Avaya Communication Manager 5.1.1 (R015x.01.1.415.1) with Service Pack 2.01 (Patch 19688)
Avaya SIP Enablement Services	5.1.1. (SES-5.1.1.1-415.1)
Avaya 4600 Series IP Telephones	2.8 (H.323)
Avaya 9600 Series IP Telephones	2.0.4 (SIP)
Avaya 6400 Series Digital Telephones	--
Avaya Analog Telephones	--
DiVitas Mobile Unified Communications	2.7.0.0.18
DiVitas Client on Nokia E71	2.7.0.0.18

Table 1: Equipment and Software Validated

4. Configure Avaya Communication Manager

This section describes the procedure for configuring a SIP trunk to Avaya SES, a local station mapped to a DiVitas Client, and call routing. In addition, the required customer software options are checked and the dial plan is configured. Avaya Communication Manager was configured using the System Access Terminal (SAT). Refer to [1] and [3] for additional details.

4.1. Check Customer Options

Prior to configuring the SIP trunk and stations, verify that the required customer software options are available. Enter the **display system-parameters customer-options** command to verify that the number of EC500 telephones and SIP trunks supported by the system are sufficient. If not, contact an authorized Avaya account representative to obtain additional licenses. The EC500 license also enables CSP (Cellular Service Provider) which is used in this configuration.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES
     G3 Version: V15                             Software Package: Standard
       Location: 1                            RFA System ID (SID): 1
       Platform: 6                            RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 48000 779
                                      Maximum Stations: 36000 261
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 50    2
                    Maximum Off-PBX Telephones -   OPS: 50    33
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0

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

Figure 2: System-Parameters Customer-Options Form – EC500

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES
IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 5000  258
           Maximum Concurrently Registered IP Stations: 12000 6
             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: 10    0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 555   80
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

Figure 3: System-Parameters Customer-Options Form – SIP Trunks

4.2. Configure Dial Plan

In the **Dial Plan Analysis Table**, a dialed string beginning with '2' was configured as extensions for the local stations mapped to DiVitas Clients and for other SIP, H.323, digital, and analog telephones in this configuration. The other entries in bold are used for trunk access codes and the AAR/ARS features access codes. In this configuration, the AAR access code is '8' and the ARS access code is '9'.

```
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
     2            5    ext
     3            5    ext
     4            5    ext
     5            5    ext
     8            1    fac
     9            1    fac
     *            3    fac
     #            3    fac
```

Figure 4: Dial Plan Analysis

4.3. Configure SIP Trunk

This section covers the configuration of the SIP trunk between Avaya Communication Manager and Avaya SES, including the IP node names, IP network region, and IP codec set.

In the **IP Node Names** form, associate a name with the IP addresses of Avaya SES and the C-LAN board in the Avaya G650 Media Gateway.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
clan2               192.45.100.70
default             0.0.0.0
medpro2             192.45.100.71
SIPserver1          192.45.100.200
```

Figure 5: IP Nodes Names

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Avaya SES. In this configuration, the domain name is *example.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between SIP endpoints without using media resources in the Avaya G650 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for local calls and calls routed over the SIP trunk to Avaya SES. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group as shown in **Figure 8**. The IP network region for local and outgoing trunk calls may be different.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: example.com
    Name: Avaya region
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 65531
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? n
 Call Control PHB Value: 34
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        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 6: IP Network Region

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the DiVitas Clients. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown in **Figure 6**. The default settings of the **ip-codec-set** form are shown below. Currently, the DiVitas Mobile UC solution supports the G.711 codec.

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

                          IP Codec Set

    Codec Set: 1

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

Figure 7: IP Codec Set

Prior to configuring a SIP trunk group for communication with Avaya SES, a SIP signaling group must be configured. Configure the **Signaling Group** form as shown in **Figure 8** with the following parameters:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security).
- Specify the C-LAN board in the G650 Media Gateway and Avaya SES as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form shown in **Figure 5**.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- Specify the IP network region to be used for outgoing calls that use this signaling group in **Far-end Network Region** field. The codec type for the outgoing call is derived from the IP codec set specified in the IP network region. In this configuration, IP network region '1' and IP codec set '1' is used which allows the G.711mu-law codec for the call.
- Enter the domain name of Avaya SES in the **Far-end Domain** field. In this configuration, the domain name is *example.com*. This domain is specified in the Uniform Resource Identifier (URI) of the "SIP To Address" in the INVITE message. Mis-configuring this field may prevent calls from being successfully established over the SIP trunk.
- If calls to/from SIP endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to 'y'.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Avaya Communication Manager supports DTMF transmission using RFC 2833. Setting the field to *rtp-payload* implies RFC 2833. The default values for the other fields may be used.

```
change signaling-group 702                                      Page   1 of   1
                                SIGNALING GROUP

 Group Number: 702             Group Type: sip
                        Transport Method: tls

   Near-end Node Name: clan2                Far-end Node Name: SIPserver1
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
      Far-end Domain: example.com
                                           Bypass If IP Threshold Exceeded? n

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

Figure 8: Signaling Group

Configure the **Trunk Group** form as shown in **Figure 9**. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
change trunk-group 702                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 702                  Group Type: sip          CDR Reports: y
  Group Name: To SIPserver1               COR: 1        TN: 1        TAC: 176
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 702
                                                    Number of Members: 20
```

Figure 9: Trunk Group – Page 1

On Page 3 of the trunk group form, set the **Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

```
change trunk-group 702                                          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 10: Trunk Group – Page 3

Configure the **Public/Unknown Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '2' and whose calls are routed over PSTN trunk group '6' or SIP trunk group '702' have their number sent to the far-end for display purposes. The PSTN trunk group is discussed in Section 4.4. In the example shown in **Figure 11**, the **CPN Prefix** field is left blank and the **CPN Len** field is set to '5' indicating that the 5-digit extension corresponding to the calling party will be sent to the far-end. Additional entries may be included to cover other extensions.

```
change public-unknown-numbering 2                               Page   1 of   2
                        NUMBERING - PUBLIC/UNKNOWN FORMAT
                                             Total
Ext Ext            Trk       CPN              CPN
Len Code           Grp(s)    Prefix           Len
                                                      Total Administered: 22
 5  2              6                          5          Maximum Entries: 9999
 5  2              702                        5
```

Figure 11: Public Unknown Numbering Format

4.4. Configure PSTN Trunk

An ISDN-PRI trunk was required to establish calls between Avaya Communication Manager and the PSTN. Follow the standard procedure for configuring an ISDN-PRI signaling group and trunk group. In this configuration, ISDN-PRI trunk group '6' was used. Call routing from the DiVitas Clients to the PSTN over the ISDN-PRI trunk group is described in Section 4.6.1.

4.5. Configure Stations

This section describes how to map a desktop IP phone on Avaya Communication Manager to a DiVitas Client running on a Nokia E71 handset. This would allow a desktop phone and the DiVitas Client to ring simultaneously when a call is received. The call can then be answered by either the desktop phone or the DiVitas Client.

Configure a station as shown in **Figure 12**. In this example, the station maps to an H.323 IP phone with an extension of 24511. The DiVitas Client has an extension of 8524511 and is configured in **Figure 36**. The station and the DiVitas Client *must* have different extensions.

Note: To call a DiVitas Client, the 5-digit extension of the desktop phone mapped to it is dialed, not the 7-digit extension assigned to it on the DiVitas Server. This also applies when a DiVitas Client calls another DiVitas Client.

```
add station 24511                                           Page   1 of   5
                                     STATION

Extension: 24511                         Lock Messages? n               BCC: 0
     Type: 4610                          Security Code: XXXXX            TN: 1
     Port: IP                          Coverage Path 1:                 COR: 1
     Name: DiVitas24511                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: 24511
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                              Customizable Labels? y
```

Figure 12: Station

To allow a call to the H.323 IP phone to be delivered to the DiVitas Client at the same time, the **Stations with Off-PBX Telephone Integration** form must be configured. On this form, specify the extension of the H.323 IP phone in the **Station Extension** field and set the **Application** field to *CSP*, which stands for Cellular Service Provider. The **Phone Number** field is set to the digits to be sent over the SIP trunk. In this case, the 7-digit extension of the DiVitas Client is specified and delivered to the DiVitas Mobile UC server. As previously mentioned, the **Station Extension** and **Phone Number** fields must be different indicating that different extensions for the desk phone and the DiVitas Client are used. Finally, the **Trunk Selection** field is set to '702', the SIP trunk group number. This field specifies the trunk group used to route the call. Another option for routing a call over a SIP trunk group is to use Auto Alternate Routing (AAR) or Auto Route Selection (ARS) routing instead. In this case, the **Trunk Selection** field would be set to *aar* or *ars*. Configuration of other AAR or ARS forms would also be required. Refer to [1] for information on routing calls using AAR or ARS. Repeat this step for each DiVitas Client associated with a desk phone.

```
change off-pbx-telephone station-mapping 24511                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 24511             CSP           -     8524511            702        1
```

Figure 13: Stations with Off-PBX Telephone Integration

4.6. Call Routing

This section describes how to configure call routing on Avaya Communication Manager for calls to the PSTN and for establishing a CVC from the DiVitas Clients in Cellular mode to the DiVitas Server.

4.6.1. Call Routing from DiVitas Clients to the PSTN

To call the PSTN, DiVitas Clients dial the ARS feature access code '9' followed by the 11-digit number. On Avaya Communication Manager, the call is steered to the **ARS Digit Analysis Table** where the call is routed based on the dialed 11-digit number. In this configuration, routing for calls to the 408, 650, and 732 area codes was configured. Configure the **ARS Digit Analysis Table** as shown in **Figure 14**. In this example, calls are routed over route pattern '732'.

```
change ars analysis 14                                          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
    1408                  11   11    732       fnpa         n
    1650                  11   11    732       fnpa         n
    1732                  11   11    732       fnpa         n
```

Figure 14: ARS Digit Analysis Table

Figure 15 shows the route pattern used for PSTN calls. PSTN calls are routed over ISDN-PRI trunk group '6' (not shown). The **Pfx Mrk** field is set to '1' so that the prefix mark is included in the called party number. No other digit manipulation is performed.

```
change route-pattern 732                                        Page   1 of   3
                    Pattern Number: 732 Pattern Name: To PSTN
                             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: 6    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 15: Route Pattern for PSTN Calls

4.6.2. Call Routing from the PSTN to the DiVitas Clients or CVC

An incoming PSTN call to a DiVitas Client arrives on trunk group '6'. In this configuration, the 10-digit PSTN number assigned to a DiVitas Client was already converted to a 5-digit extension. Avaya Communication Manager then terminates the call to the corresponding desktop phone and also routes the call to the DiVitas Client over a SIP trunk through the configuration in **Figure 13**. The PSTN call to the CVC external number also arrives on trunk group '6' and is converted to the 5-digit extension corresponding to the CVC internal number. The CVC call is routed over the SIP trunk as described in Section 4.6.3.

4.6.3. Call Routing for CVC on DiVitas Mobile UC Server

When a DiVitas Client is in Cellular mode, it communicates with the DiVitas Server over the CVC described in Section 2. The CVC is established by dialing an 11-digit PSTN number (external number) assigned to the DiVitas Server. The PSTN number is converted to a 5-digit (internal) number by the time it reaches Avaya Communication Manager. In this example, the internal number is 24555. To route the CVC call to the DiVitas Server, an entry is added in the **Uniform Dial Plan Table** in order to route the call using AAR.

```
change uniform-dialplan 2                                       Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0
  Matching                   Insert              Node
  Pattern       Len Del      Digits     Net Conv Num
24555           5   0                  aar  n
```

Figure 16: Uniform Dial Plan

In the **AAR Digit Analysis Table**, an entry is added for dialed string “24555”, which routes that call over route pattern ’702’.

```
change aar analysis 24                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    1

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    24555                 5    5     702       aar          n
```

Figure 17: AAR Analysis

In **Route Pattern** ‘702’, calls are routed over SIP trunk group ‘702’ with no digit manipulation being performed. The exact dial string is sent to Avaya SES, which route the call based on the 5-digit extension.

```
change route-pattern 702                                        Page   1 of   3
                    Pattern Number: 702 Pattern Name: To SIPserver1
                             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: 702  0                                                             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                                         next
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

Figure 18: Route Pattern

5. Configure Avaya SIP Enablement Services

This section describes the steps for configuring Avaya SES to communicate with Avaya Communication Manager and the DiVitas Mobile Unified Communications Server. This includes configuring:

- SIP trunks to Avaya Communication Manager and the DiVitas Mobile UC Server,
- A Trusted Host for the DiVitas UC Server,
- Host Maps for routing calls to the DiVitas UC Server, and
- Communication Manager Maps for routing calls to the PSTN and other stations on Avaya Communication Manager.

Refer to [3] for additional information on configuring Avaya SES.

Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter *http://<ip-addr>/admin* as the URL in the Internet browser, where <*ip-addr*> is the IP address of Avaya SES. Log in with the appropriate credentials and select the **Launch SES Administration Interface** link.

To verify the system properties, including the SIP domain and the IP address of Avaya SES, select **Server Configuration→System Properties** link on the left pane of the Administration web interface. The **System Properties** are shown in **Figure 19**.

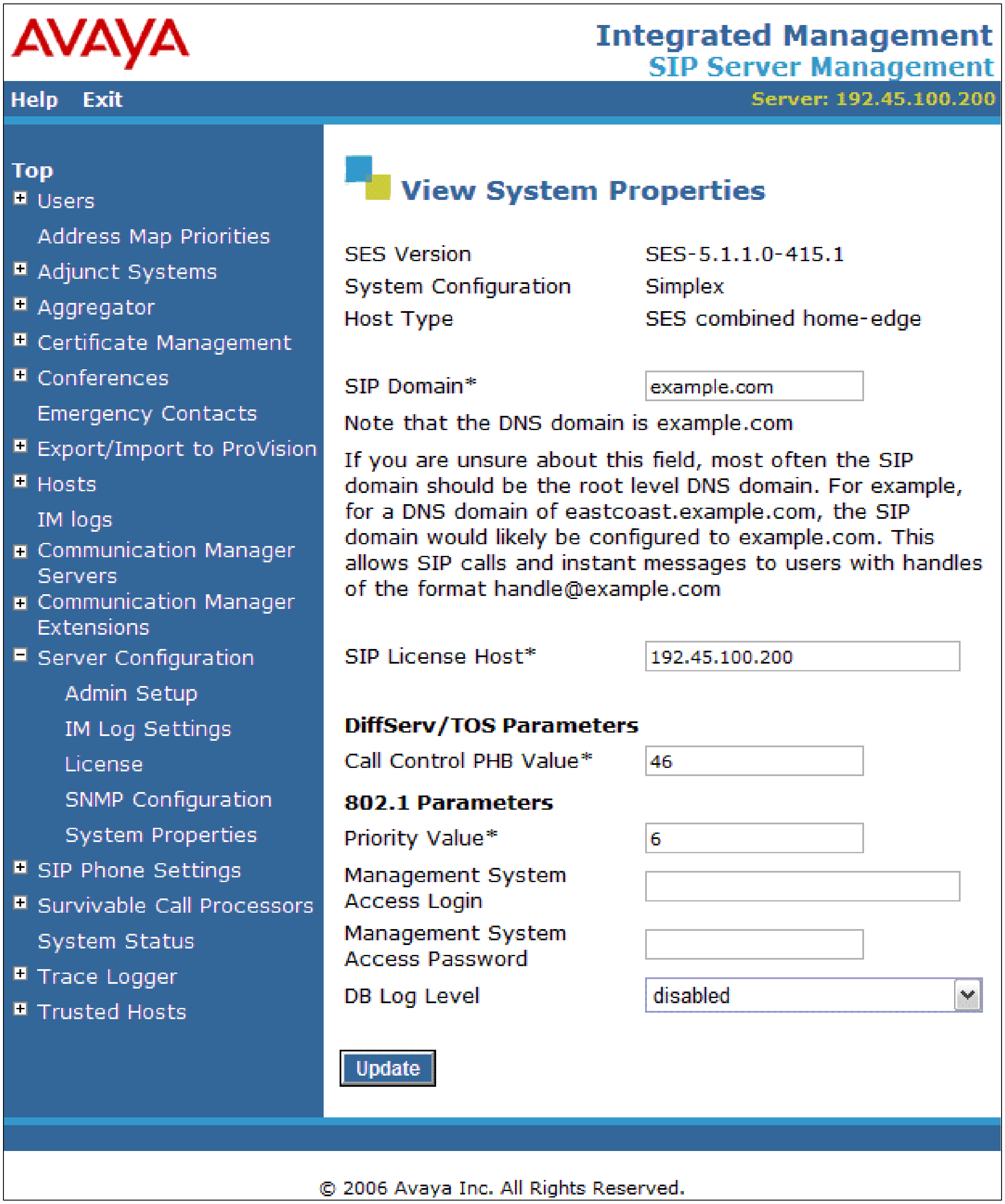


Figure 19: System Properties

To enable secure SIP trunking between Avaya SES and Avaya Communication Manager, add a Communication Manager Server interface. The **Edit Communication Manager Server Interface** screen shown in **Figure 20** displays the following information:

- A descriptive name in the **Communication Manager Server Interface Name** field (e.g., pbx31-clan).
- The IP address of Avaya SES in the **Host** field.
- *TLS* (Transport Link Security) for the **Link Type**. TLS provides encryption at the transport layer.
- The IP address of the C-LAN board in the Avaya G650 Media Gateway in the **SIP Trunk IP Address** field.

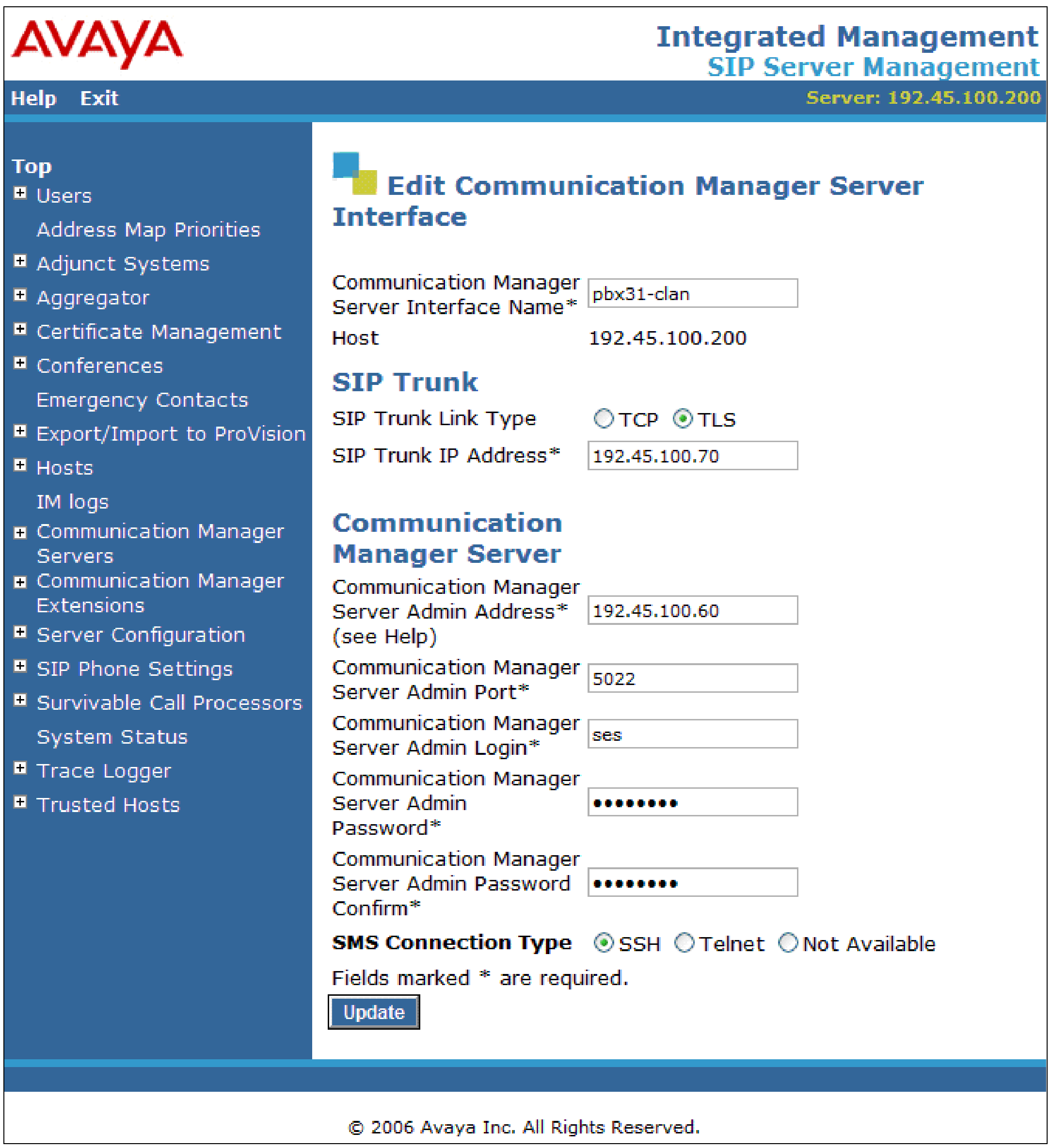


Figure 20: Communication Manager Server Interface

Calls originated from the DiVitas Clients (e.g., Nokia E71) registered to the DiVitas UC Server are routed through Avaya Communication Manager. **Communication Manager Maps** are required to route calls to the PSTN and other local stations on Avaya Communication Manager. For the map used to route calls to the PSTN shown in **Figure 21**, the leading digit '9' in the **Pattern** field maps to the ARS feature access code required for public network routing. This pattern allows 11-digit numbers starting with the prefix mark '1' and followed by the area code and 7-digit number to be routed to Avaya Communication Manager.

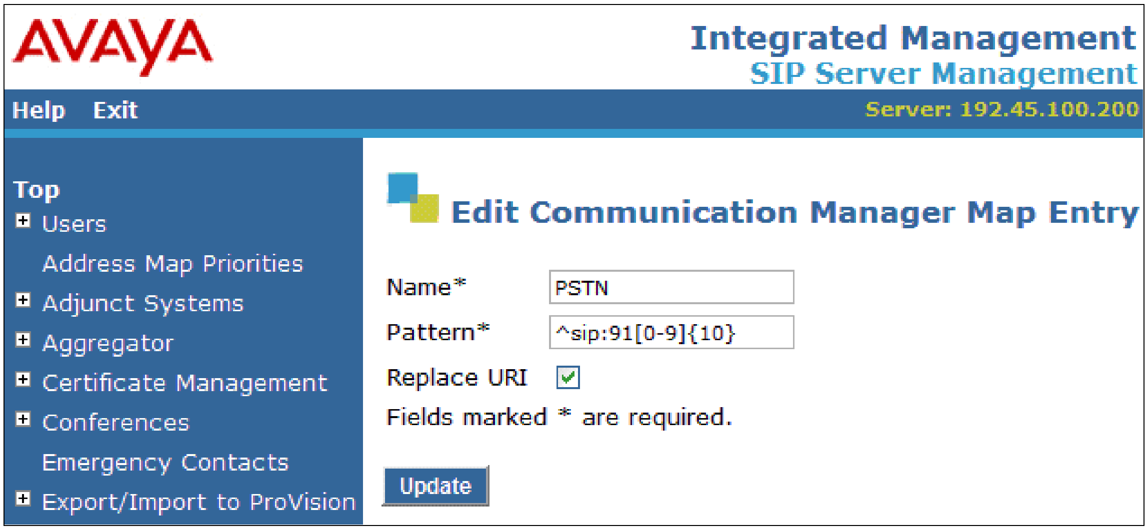


Figure 21: Communication Manager Map for PSTN Calls

The map shown in **Figure 22** allows calls to be routed to local stations that have a 5-digit extension beginning with '2'.

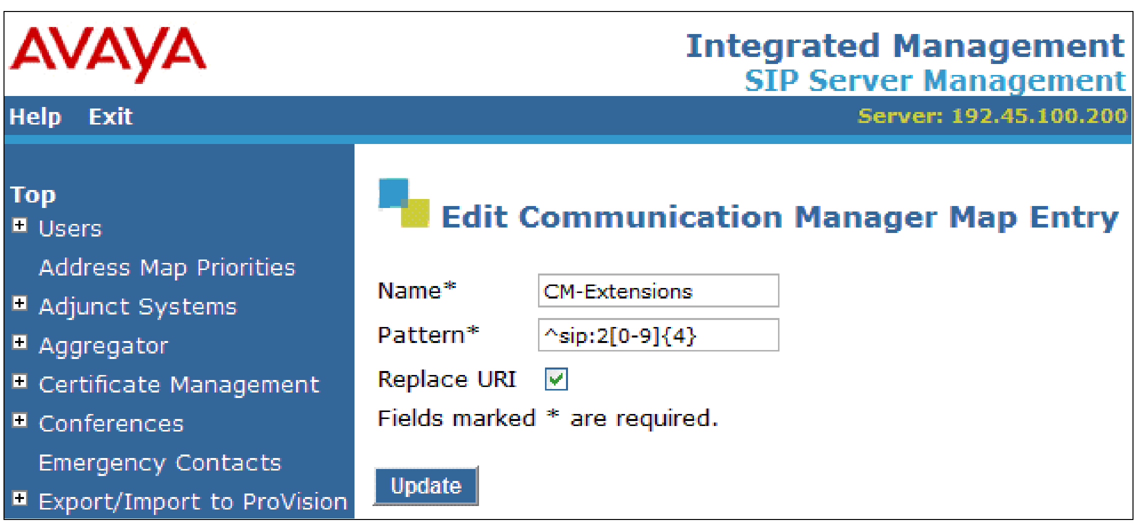


Figure 22: Communication Manager Map for Local Station Calls

The **Communication Manager Server Address Map** is listed in **Figure 23**.

Figure 23: List Communication Manager Server Address Map

A **Host Address Map** is required for routing calls to the DiVitas Clients (i.e., Nokia E71) on the DiVitas UC Server. Routing is based on the content of the SIP INVITE URI. Although the 5-digit extension of the office phone mapped to the DiVitas Client is dialed when calling the DiVitas Client, the dialed string is converted to the 7-digit extension configured on the DiVitas Mobile UC Server. This digit conversion is performed in the **Stations with Off-PBX Telephone Integration** form in **Figure 13**. **Figure 24** shows the **Host Map** for routing calls to DiVitas Clients using 7-digit extensions starting with ‘852’.

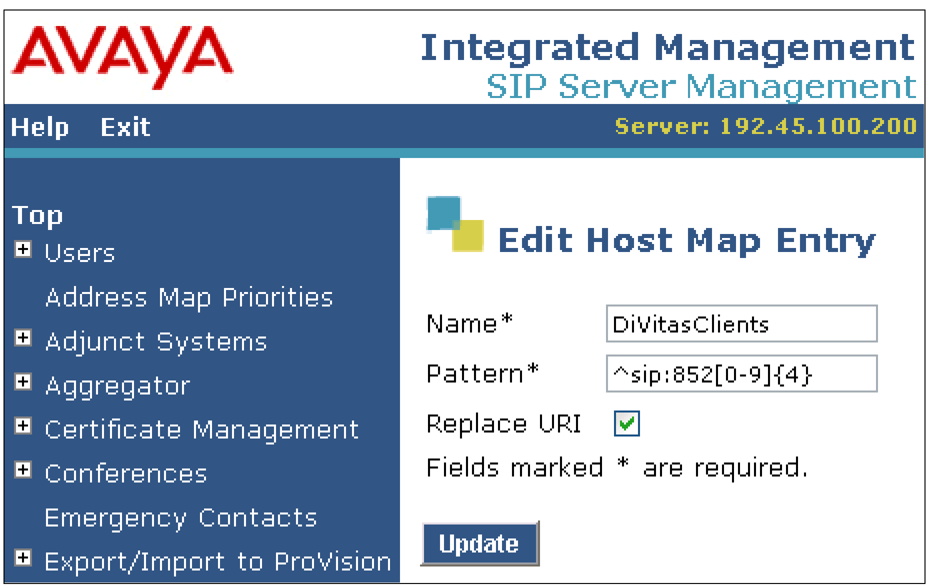


Figure 24: Host Address Map for DiVitas Clients

Figure 25 shows the **Host Map** used to establish the CVC from the DiVitas Clients in Cellular mode to the DiVitas Mobile UC Server. The request to establish the CVC originates from the PSTN by dialing a PSTN number or external number, which is then converted to a 5-digit internal number on Avaya Communication Manager (see Section 4.6.2). The CVC internal number is 24555.

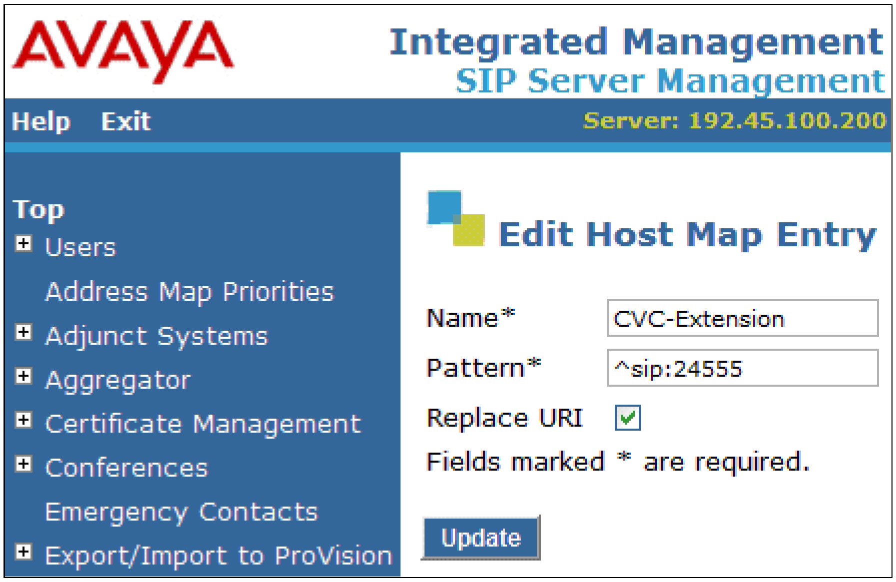


Figure 25: Host Address Map for CVC Extension

The **Host Contact** is shown in **Figure 26**. The **Contact** field includes the IP address of the DiVitas UC Server, the port, and the transport protocol. The **Contact** field in the **Host Contact** was set to `sip:$(user)@192.45.100.190:5060;transport=udp`.

Figure 26: Host Contact

The **Host Address Map** is listed in **Figure 27**.

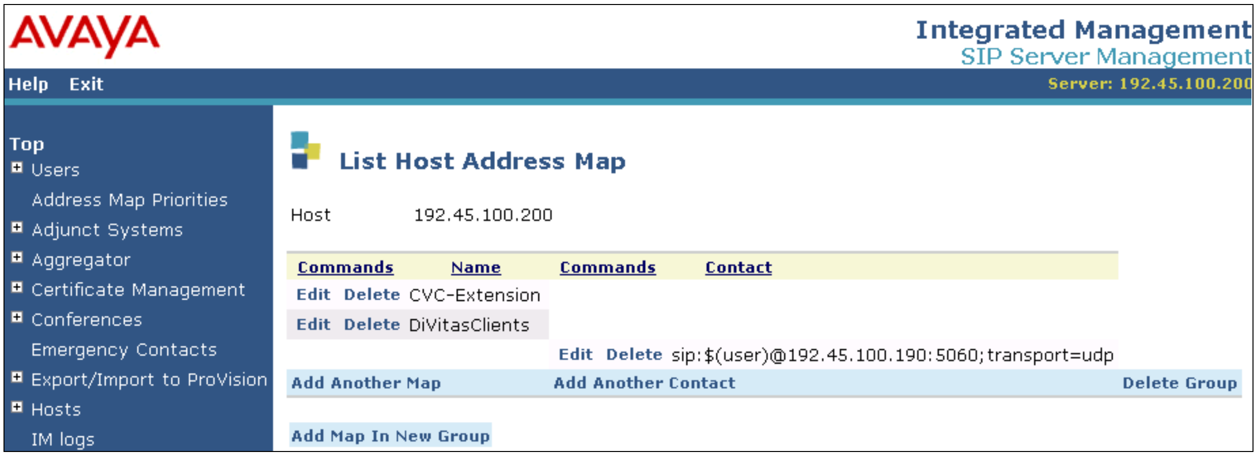


Figure 27: List Host Address Map

The IP address of the DiVitas UC Server must be configured as a trusted host on Avaya SES. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the DiVitas UC Server. The trusted host was configured as shown in **Figure 28**.

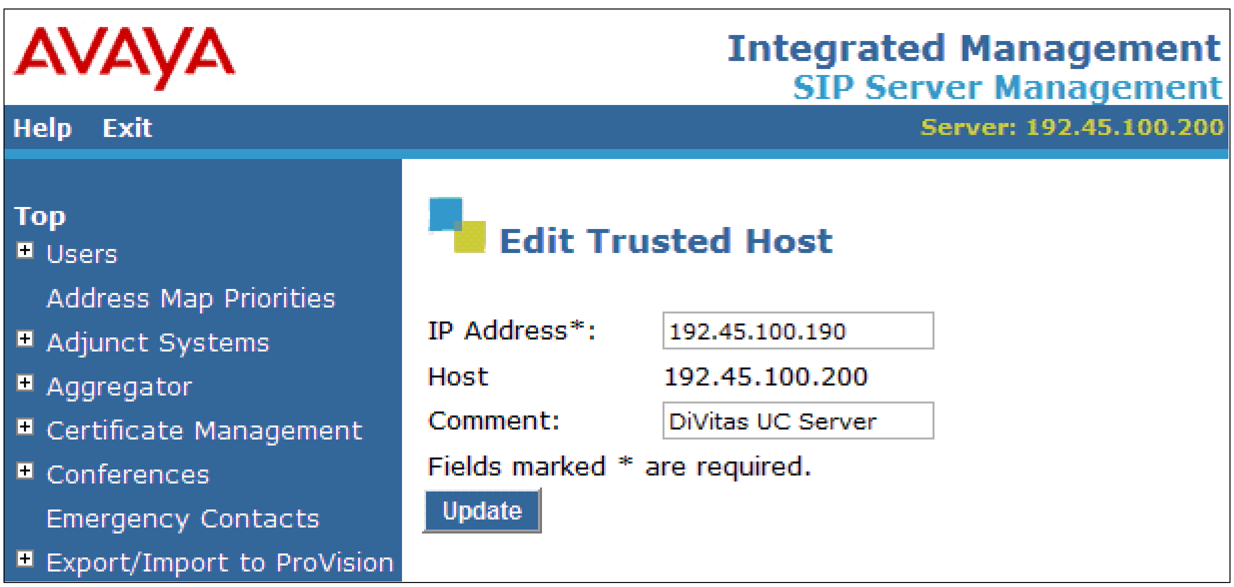


Figure 28: Trusted Host

6. Configure DiVitas Mobile Unified Communications

This section describes the steps for configuring the DiVitas Mobile Unified Communications Server (Mobile UC Server) which supports a variety of dual mode (WiFi/Cellular) telephones including Nokia E- and N-Series DiVitas Clients. Refer to [5] for additional configuration information.

All DiVitas Mobile UC Server configuration and management features are accessed from a Web-based interface. From an Internet browser, enter the IP address of the DiVitas Mobile UC Server in the URL field and log in using the appropriate credentials. The screen shown in **Figure 29** is displayed.

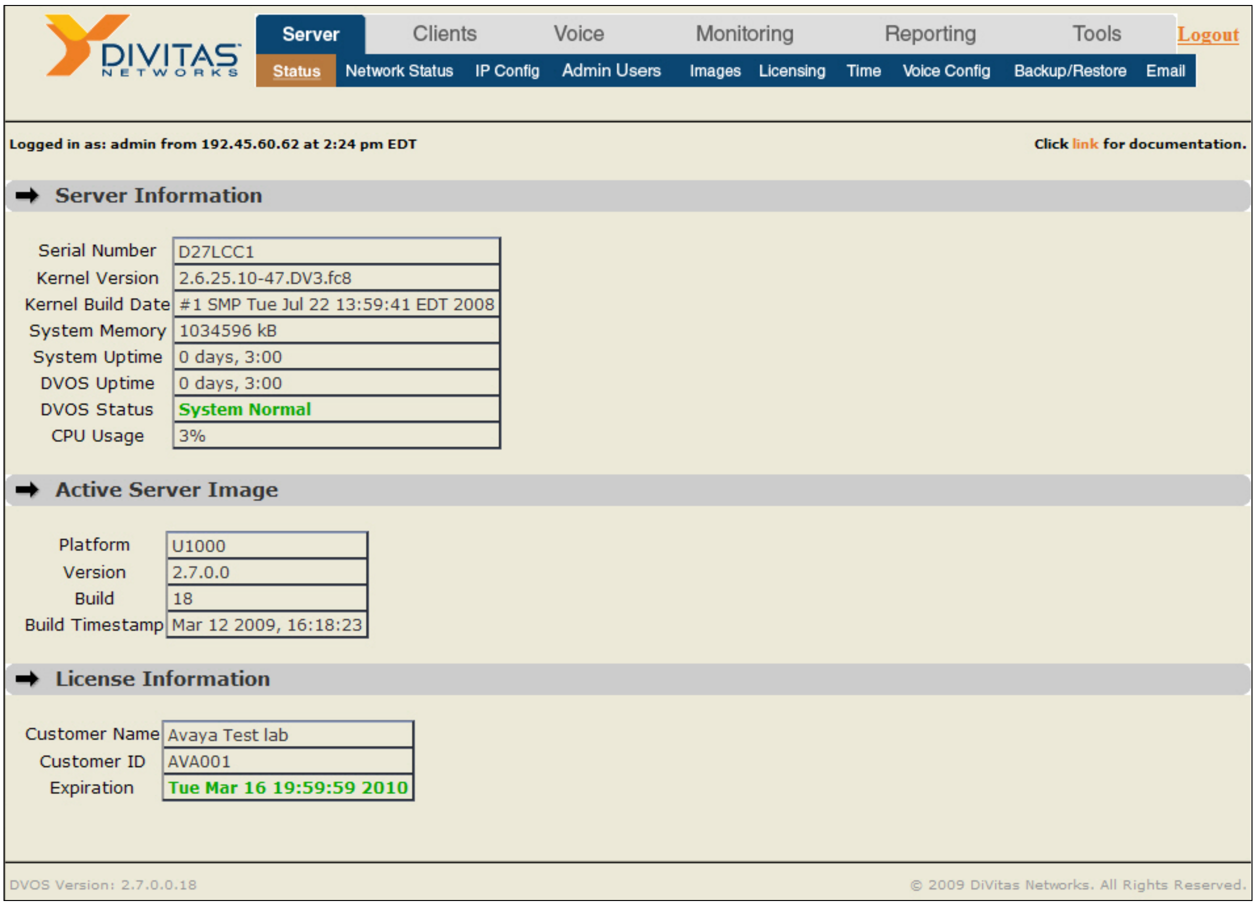


Figure 29: DiVitas Mobile UC Server Web Interface

In the **Server→IP Config** webpage, configure the IP network parameters of the DiVitas Mobile UC Server corresponding to the customer's network as shown in **Figure 30**. The remaining fields on this webpage (not shown) may be left at the default values. Click **Submit**.

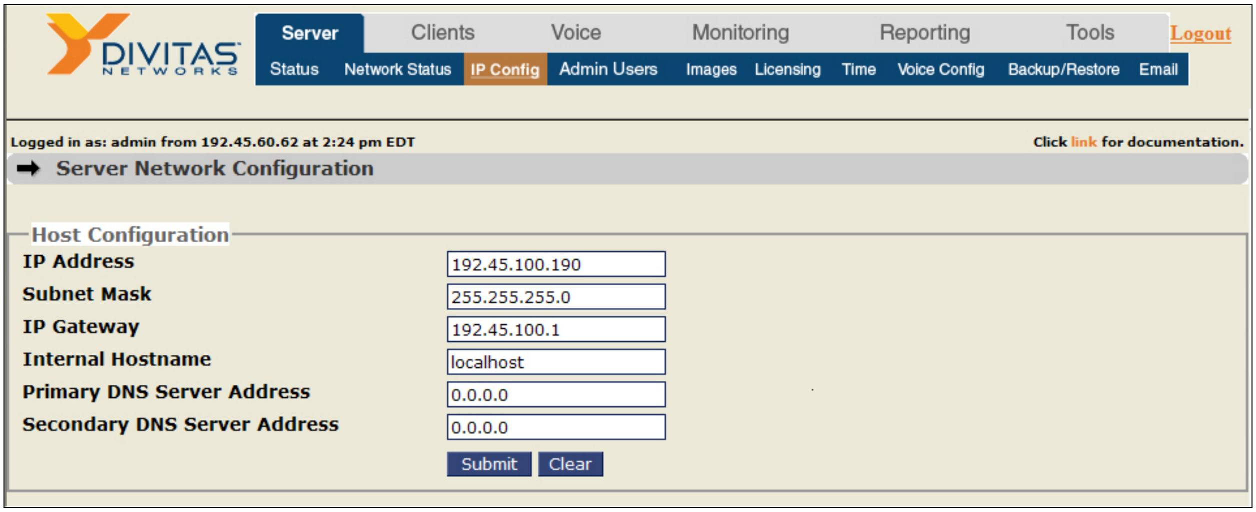


Figure 30: Server IP Configuration

The DiVitas Server and Avaya SES communicate over a SIP trunk. All calls from a DiVitas Client are routed over the SIP trunk to Avaya Communication Manager. The SIP trunk is also used to route calls to DiVitas Clients operating in Cellular mode. To configure the SIP trunk on the DiVitas Server, navigate to **Voice→Configuration** and then click on **Trunks→Add Trunk**. The following example shows the SIP trunk after it has been configured.

In the SIP trunk configuration, specify a descriptive name for the **Trunk Name** field. For the Dial Rules, specify the format of the dial patterns that are allowed to be routed over this SIP trunk. In this example, the DiVitas Clients are allowed to dial 5-digit numbers and 11-digit numbers preceded by a '9'. The '9' corresponds to the ARS feature access code on Avaya Communication Manager. **Figure 31** shows the first half of the SIP trunk configuration.

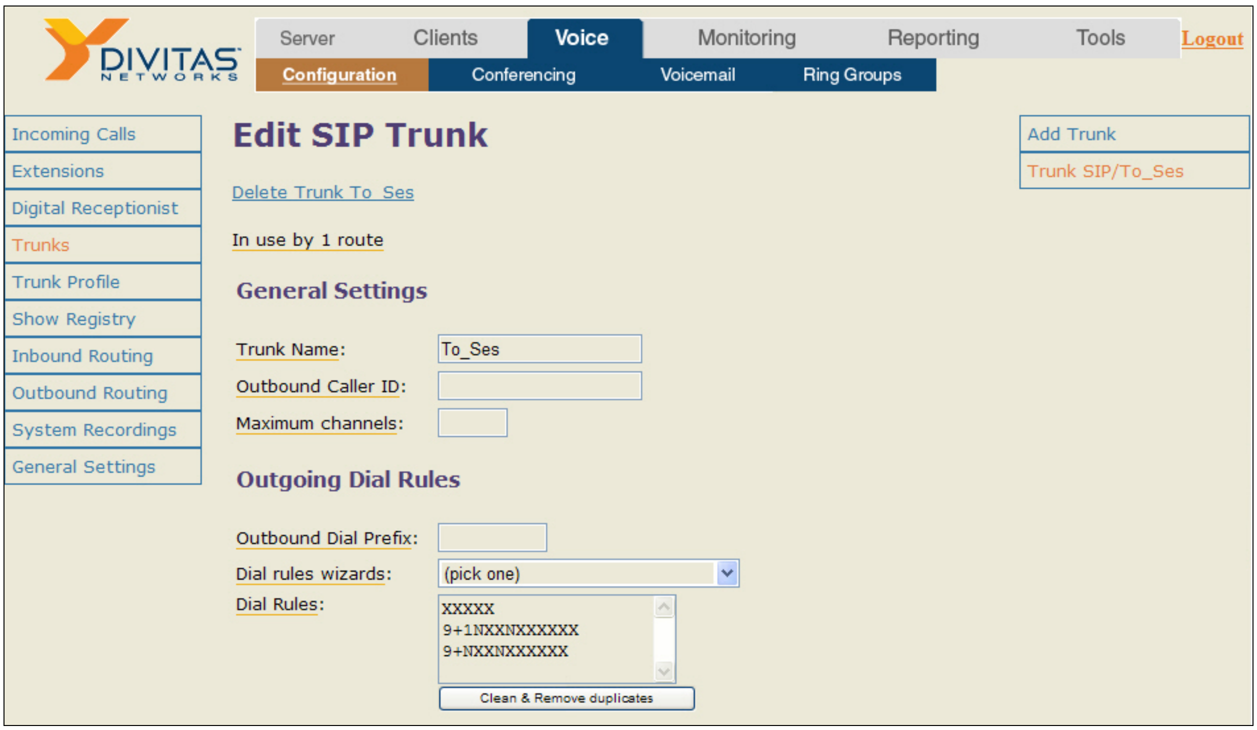


Figure 31: SIP Trunk - Top

Figure 32 displays the second half of the SIP trunk configuration webpage. The **Host Address** field should be set to the Avaya SES IP address, the **Port** field should be set to *5060* since UDP transport is used for the SIP trunk, and *rfc2833* should be used for the **DTMF Mode**. The other fields should be configured as shown below. Click **Submit Changes**.

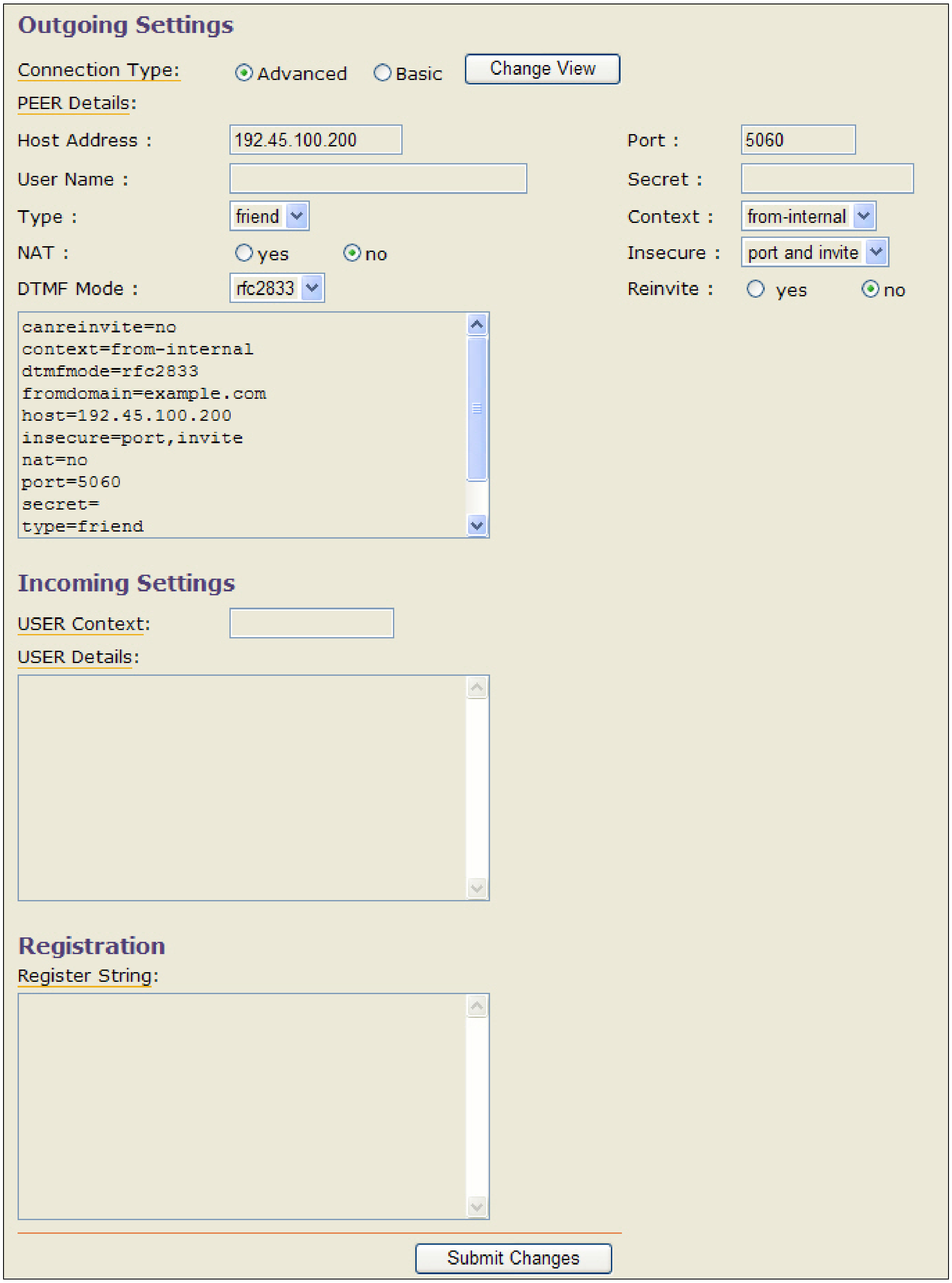


Figure 32: SIP Trunk – Bottom

To properly route calls to stations on Avaya Communication Manager or the PSTN, a generic route was defined. The **Route Name** was set to a descriptive name. The **Dial Pattern** was set to *X* which matches any digit and the **Insert** field is set to commonly used dial patterns. The **Trunk Sequence** specifies the trunk(s) for routing outbound calls. In this example, the calls are routed over the SIP trunk configured in **Figure 31** and **Figure 32**.

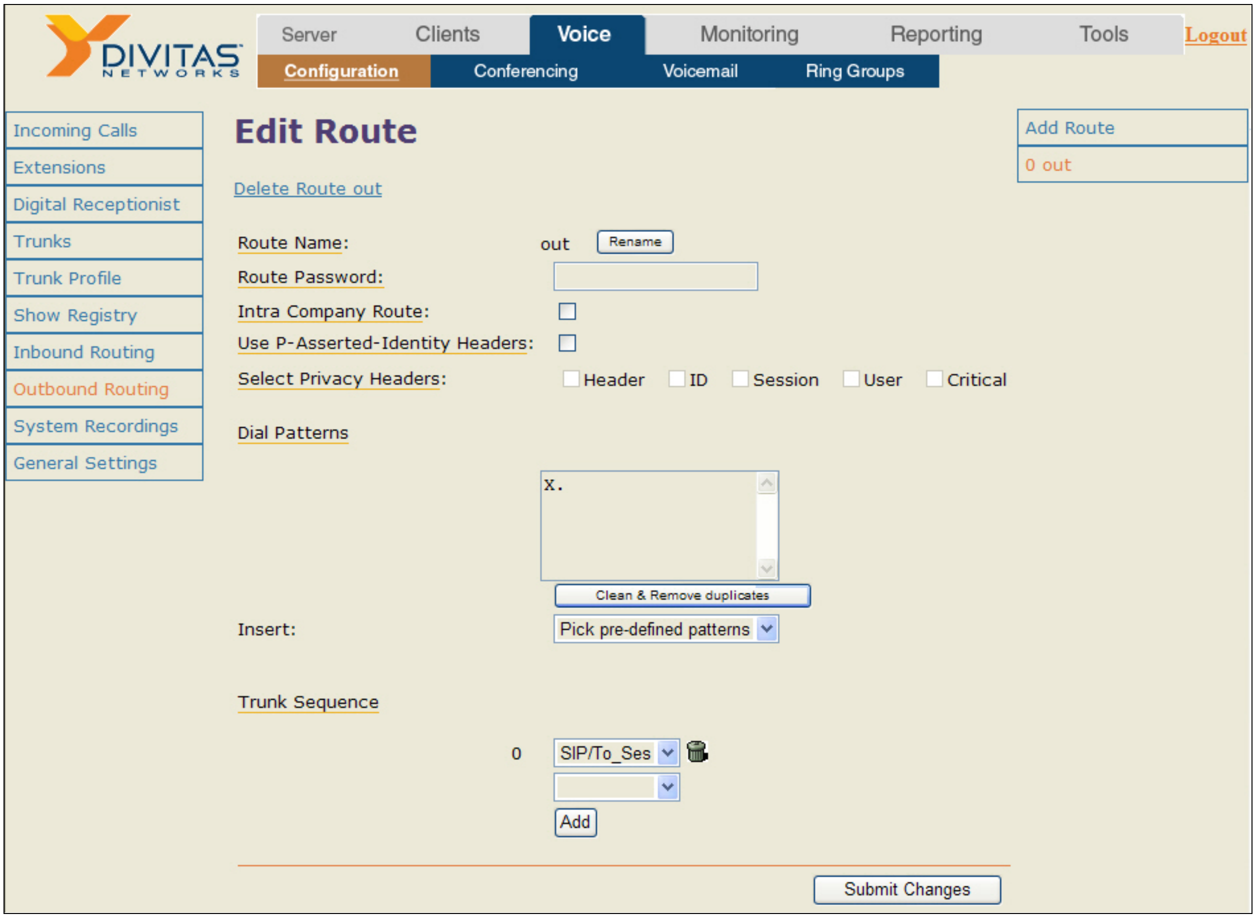


Figure 33: Outbound Routing

In the **Server →Voice Config** webpage, configure the CVC. In the **Server CVC Configuration** section, set the **CVC Operational Mode** field to *Allowed* and configure the external and internal CVC numbers. Click **Submit**.

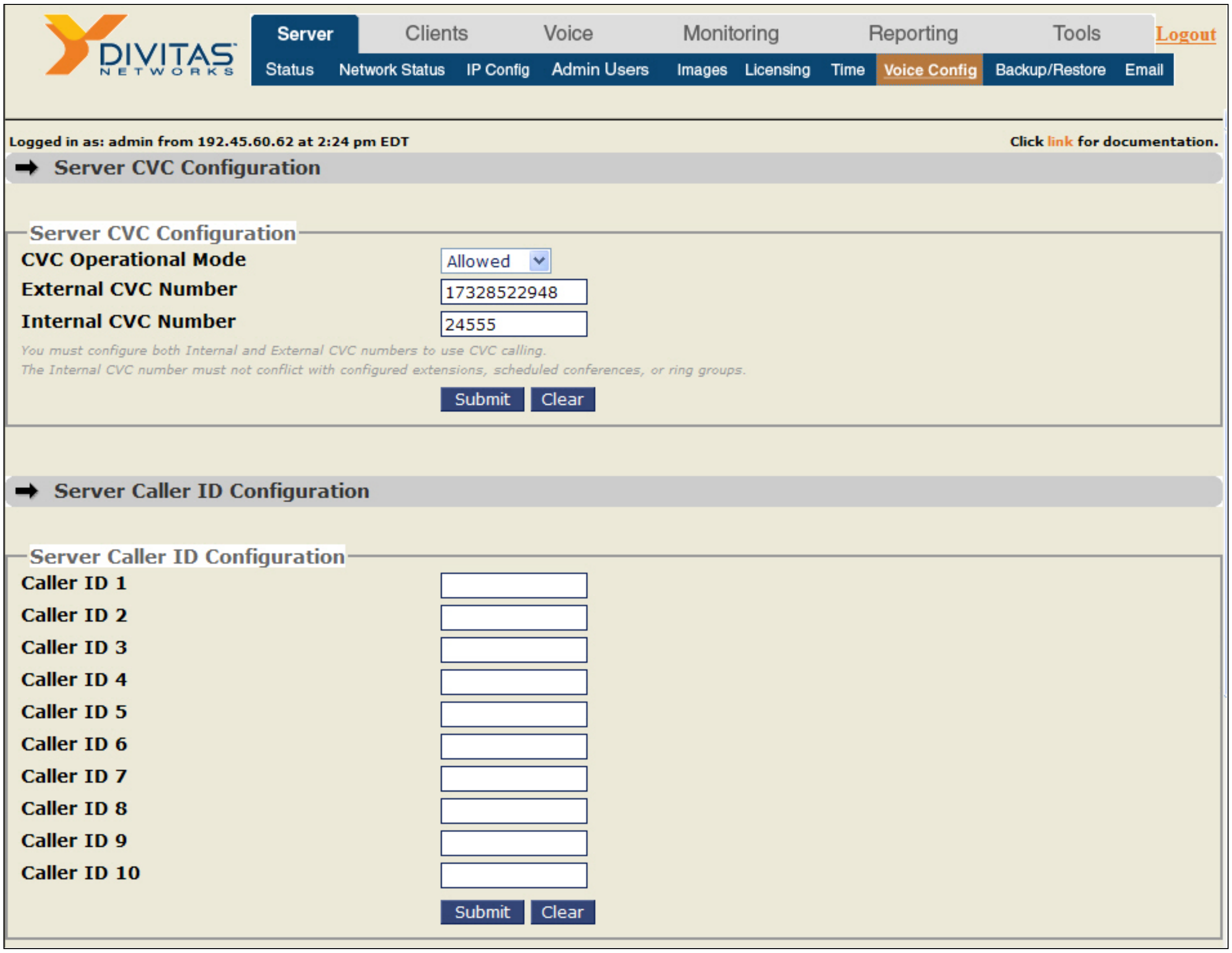


Figure 34: Server CVC Configuration

To view and add users to the DiVitas Server, navigate to **Clients→Users**. To add a **User**, click on the **Add** button under **Add User Account**. To view the details of a configured user account, select **Modify** in the **Action** field and click **Submit** under the **User Accounts** section.

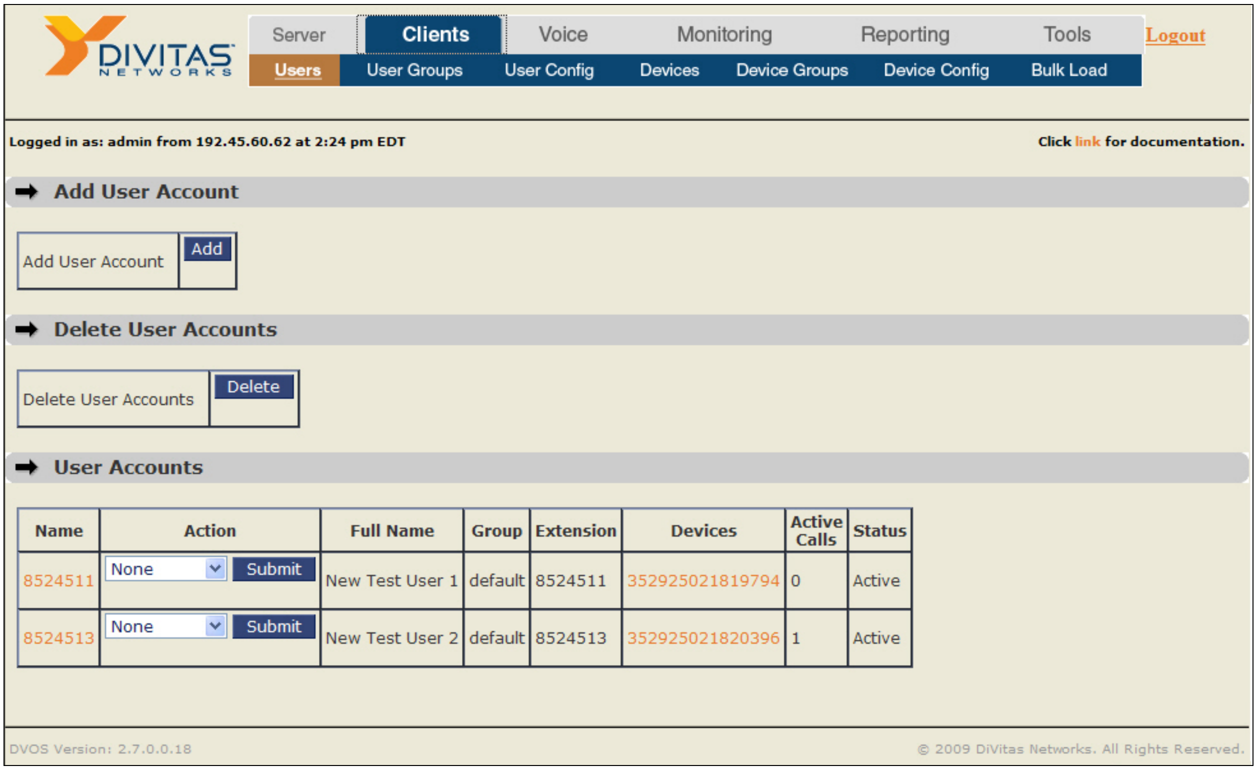


Figure 35: User Accounts

When adding a **User**, specify the user's **Full Name** and **Extension** as shown in **Figure 36**. The figure below shows the user account after it has been configured. The **Add User Account** webpage will appear slightly different, but contain similar fields.

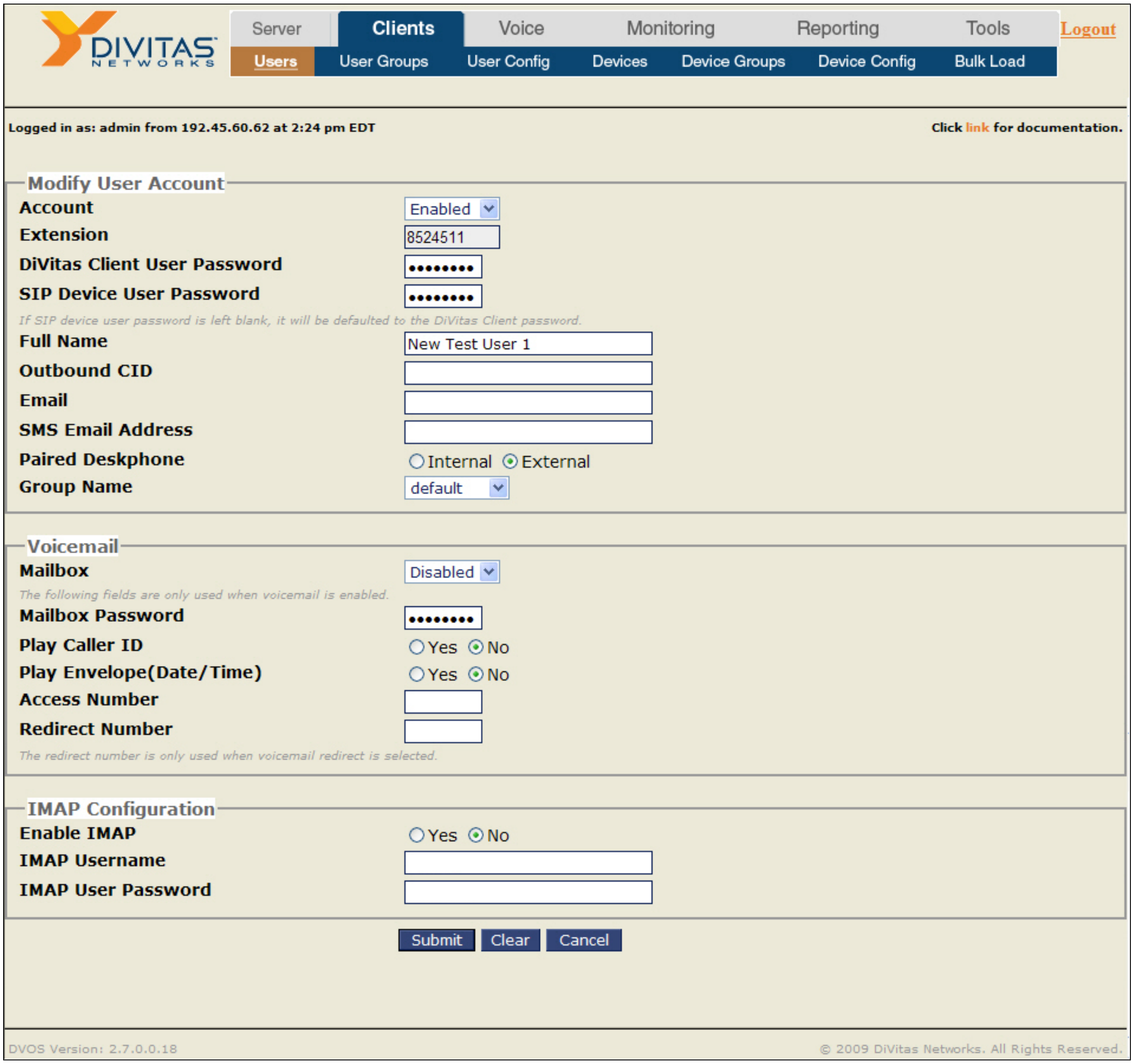


Figure 36: User

7. General Test Approach and Test Results

The focus of the interoperability compliance testing was to verify call establishment between the DiVitas Clients in WiFi and Cellular modes, the PSTN, and local stations on Avaya Communication Manager and Avaya SES. The DiVitas Clients were paired with an H.323 IP station on Avaya Communication Manager. In addition basic telephony features were exercised.

The serviceability testing focused on verifying the ability of the DiVitas Server to recover from adverse conditions, such as power failures and disconnecting cables to the IP network. In addition, the ability of the solution to recover from Avaya S8730 Server interchange and from cycling power on Avaya SES was also verified.

All tests passed successfully. However, to reiterate an important configuration guideline, the desktop phone on Avaya Communication Manager and the DiVitas Client configured on the DiVitas Server *must* have different extensions. Otherwise, if a desktop phone mapped to a DiVitas Client places a call to another DiVitas Client, only the desktop phone will ring.

8. Verification Steps

This section provides the verification steps that may be performed to verify that DiVitas Clients registered with the DiVitas Mobile UC server can establish calls to the PSTN or stations on Avaya Communication Manager.

1. From the Avaya Communication Manager SAT, verify that the SIP signaling group and trunk group are in-service using the **status signaling-group** and **status trunk** commands, respectively**.**

2. From the DiVitas web interface, navigate to the **Clients→Devices** webpage and verify that the DiVitas Clients are registered with the DiVitas Mobile UC Server and that their status is active.

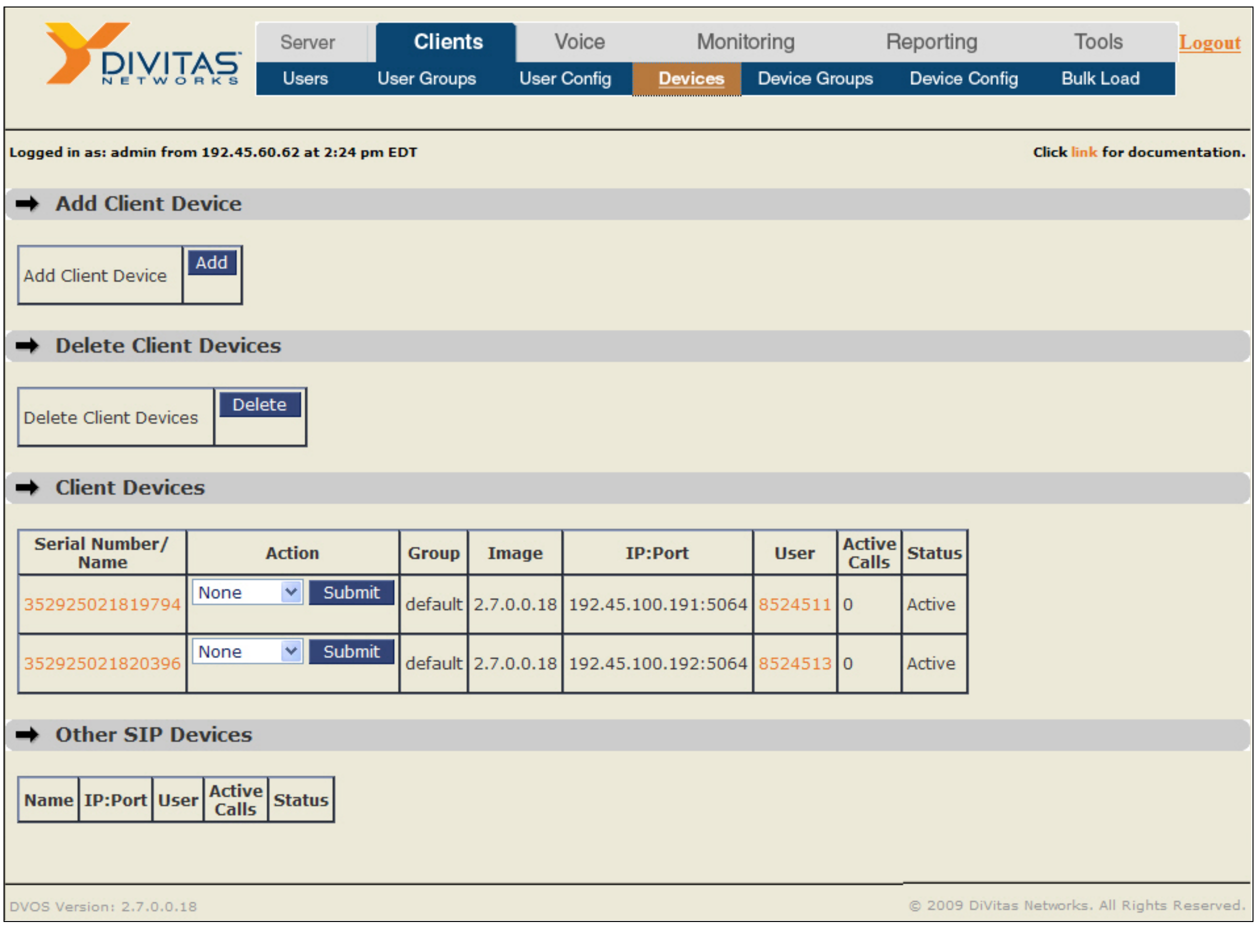
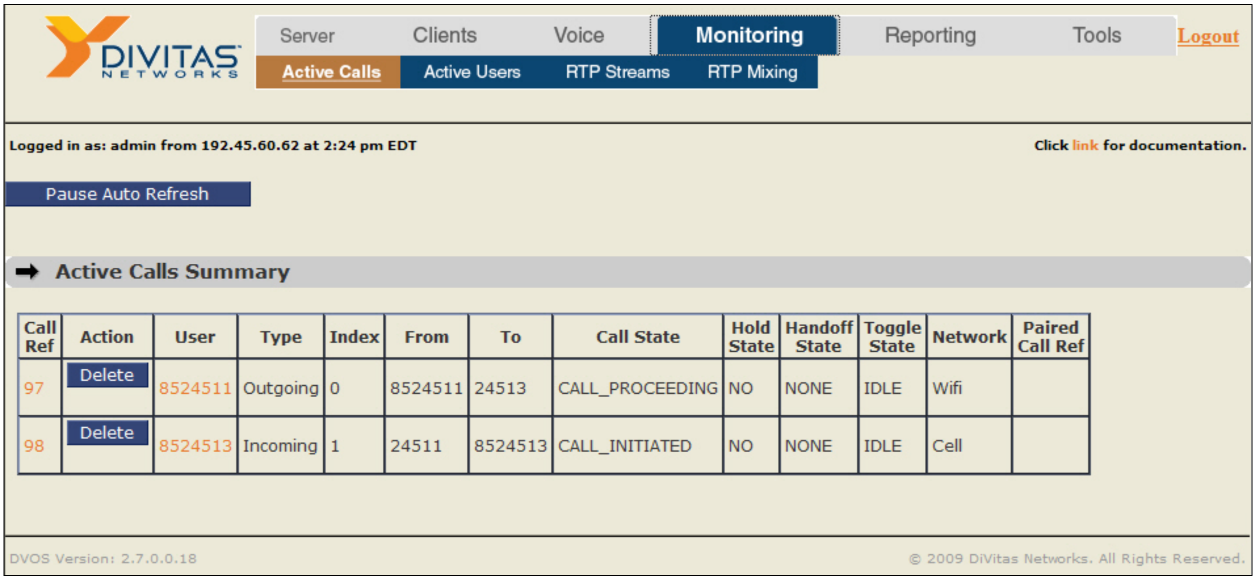


Figure 37: Status of Client Devices

3. Place a call between two DiVitas Clients routed through Avaya Communication Manager. Verify that the call completes successfully. From the DiVitas Web interface, navigate to **Monitoring→Active Calls** to view the call summary as shown in **Figure 38**.



Call Ref	Action	User	Type	Index	From	To	Call State	Hold State	Handoff State	Toggle State	Network	Paired Call Ref
97	Delete	8524511	Outgoing	0	8524511	24513	CALL_PROCEEDING	NO	NONE	IDLE	Wifi	
98	Delete	8524513	Incoming	1	24511	8524513	CALL_INITIATED	NO	NONE	IDLE	Cell	

Figure 38: Call Summary – Call between Two DiVitas Clients

4. While a call is active on a Nokia E71 with the DiVitas Client, the connected number is shown on the phone's display as shown in **Figure 39** and **Figure 40**.

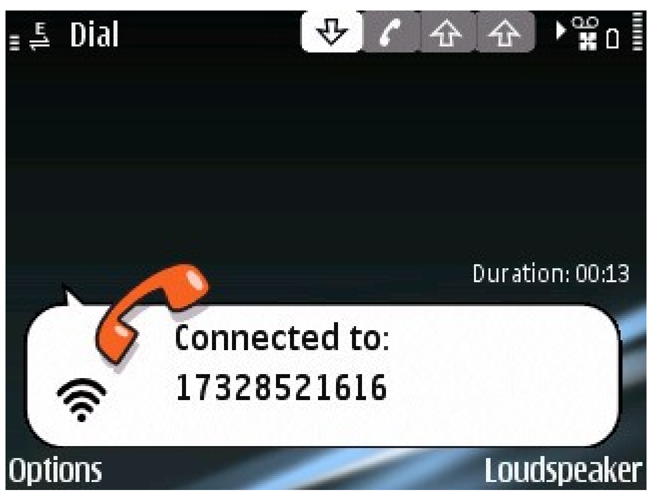


Figure 39: Active Call to DiVitas Client in WiFi Mode

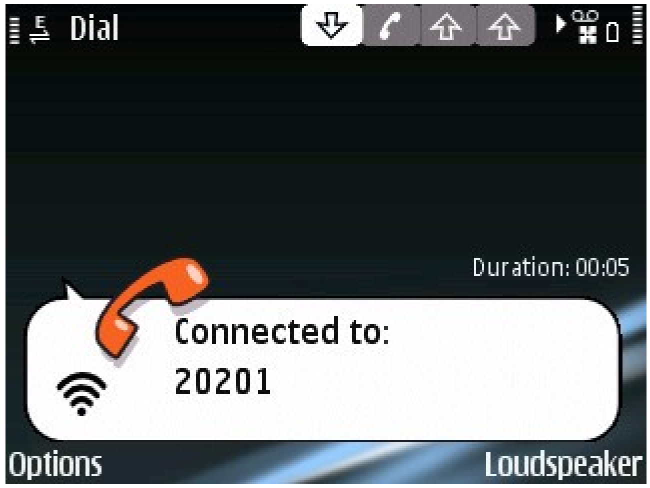


Figure 40: Active Call to an Avaya H.323 IP Telephone

After the call is completed, calls are maintained in the phone's call log as shown in **Figure 41.**

Figure 41: Call Log

9. Conclusion

These Application Notes describe the configuration steps required to integrate the DiVitas Mobile Unified Communications solution with Avaya Communication Manager and Avaya SIP Enablement Services. The DiVitas Clients were able to register with the DiVitas Server and originate and terminate calls to/from the PSTN and stations on Avaya Communication Manager.

10. Additional References

This section references the product documentation that is relevant to these Application Notes.

[1] *Administrator Guide for Avaya Communication Manager*, Document 03-300509, Issue 4, January 2008, available at http://support.avaya.com.

[2] *Feature Description and Implementation for Avaya Communication Manager*, Document 555-245-205, Issue 6, January 2008, available at http://support.avaya.com.

[3] *SIP Support in Avaya Communication Manager*, Issue 8, January 2008, Document Number 555-245-206, available at http://support.avaya.com.

[4] *Application Notes for DiVitas Mobile Unified Communications with Avaya Modular Messaging*, Issue 1.0, available at http://support.avaya.com.

[5] *DiVitas Server Administration Guide*, Version 2.7, Part Number: DOC-DVOS-AG-206.

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.