Avaya Solution & Interoperability Test Lab

Application Notes for Configuring NovaLink NovaConf with Avaya Communication Manager SIP – Issue 1.0

Abstract

These Application Notes describe the compliance testing of the NovaLink NovaConf conference system connected to Avaya Communication Manager via a SIP trunk. These Application Notes contain an extensive description of the configurations for both NovaLink NovaConf and Avaya Communication Manager.

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

1. Introduction

The purpose of this document is to describe the compliance testing done with NovaLink NovaConf and Avaya Communication Manager, including a description of the configuration of each, a description of the tests that were performed, and a summary of the results of those tests.

The NovaConf server includes a web-based administration facility that allows remote administration of users and conferences from a web browser. Various types of conferences can be configured, dependent on conference participant needs:

> **Incoming Conferences** allow users to "dial in" to conferences held at specific times.
>
> **Outgoing Conferences** can be configured to call a pre-defined list of conference participants at a specific time.
>
> **Ad-hoc conferences** can be created to meet an immediate need.
>
> **Chief conferences** are started by calling a pre-defined telephone number, and calling a pre-defined list of conference participants at that time.

Conference participants can optionally be assigned a PIN code with which they are required to authenticate themselves.

NovaConf supports multiple interfaces, including the SIP trunk described in these Application Notes.

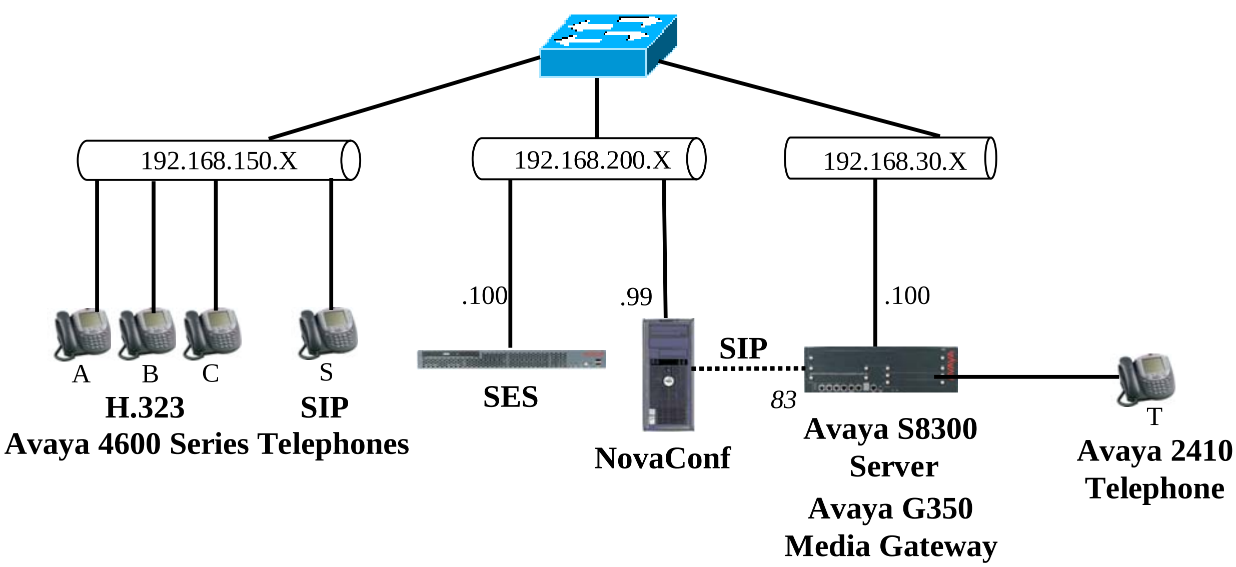

Figure 1: NovaConf Test Configuration

The SIP trunk connecting Avaya Communication Manager to the NovaConf server was configured as trunk group 83, as shown in the diagram. The function of each of the components in **Figure 1** is as follows:

- The NovaConf server initiates conferences among telephones attached to Avaya Communication Manager via the SIP trunk.
- Avaya Communication Manager runs on the Avaya S8300 Server and communicates with the NovaConf server and Avaya Telephones via the Avaya G350 Media Gateway.
- The Avaya SIP Enablement Services (SES) server is the interface between Avaya Communication Manager and the NovaConf SIP trunk, as well as the Avaya SIP Telephones.

2. Equipment and Software Validated

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

Equipment	Software Version
Avaya S8300 Server	Avaya Communications Manager 4.0 (R014x.00.0.730.5) Service Pack 00.0.730.5-13566
Avaya SIP Enablement Services Server	SES-3.1.2.0-309.0
Avaya G350 Media Gateway	26.31.0
MM712AP DCP	HW05 FW008
Avaya 4600 series H.323 stations	2.8
Avaya 4600 series SIP stations	2.2.2
NovaLink NovaConf	7.5 SP 1A
Microsoft Windows Server 2003 SE	SP2

Table 1: Version Numbers of Equipment and Software

3. Configuration

The following table contains the extensions that are used for testing. The capital letter designations correspond to the telephones shown in **Figure 1**.

Extension	Designation
3000136	A
3000134	B
3000133	C
3000115	S
3000001	T
7111111	NovaConf via SIP

Table 2: Extensions Used for Testing

3.1. Configure Avaya Communication Manager

The configuration and verification operations illustrated in this section were all performed using the Avaya Communication Manager SAT terminal via SSH port 5022.

The information provided in this section describes the configuration of Avaya Communication Manager for this solution. For all other provisioning information such as installation and configuration, please refer to the product documentation in reference [**1**].

The configuration operations described in this section can be summarized as follows:

- Verify that the licenses allocated to the system are sufficient to support the required configuration.
- Configure the dial plan and call routing required for the NovaConf configuration.
- Configure the SIP interface that is used to connect to the NovaConf server.
- Configure the telephone stations that are to be used for testing.
- Configure Avaya Communication Manager as required to interface to the Avaya SIP Enablement Services server.

3.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer-options** command to verify that Avaya Communication Manager is licensed to meet the minimum requirements to interoperate with the NovaConf server. Those items shown in bold indicate required values or minimum capacity requirements. If these are not met in the configuration, please contact an Avaya representative for further assistance.

On page 1 of this form, verify that the “Maximum Off-PBX Telephones – OPS” is sufficient for the number of Avaya SIP Telephones to be used.

```
display system-parameters customer-options                     Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V14
       Location: 2                          RFA System ID (SID): 1
       Platform: 13                         RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 900   76
                                      Maximum Stations: 450   7
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 0     0
                    Maximum Off-PBX Telephones -   OPS: 5     2
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

Figure 2: System-Parameters Customers-Options Form, Page 1

On page 2, the value configured for “Maximum Concurrently Registered IP Stations” must be sufficient to support the total number of IP stations used.

The number “Maximum Administered SIP Trunks” must be sufficient to support the maximum number of members assigned to all SIP trunks. This is the sum of the number of SIP telephones plus the SIP trunk to the NovaConf Server.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 30    5
           Maximum Concurrently Registered IP Stations: 10    3
             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: 10   3

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 0     0
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 3: System-Parameters Customers-Options Form, Page 2

On page 3 of this form, the “Cvg Of Calls Redirected Off-net” parameter must by set to “y” to allow redirection of calls to NovaConf.

```
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? y
          ARS/AAR Dialing without FAC? y                      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
```

Figure 4: System-Parameters Customers-Options Form, Page 3

On page 4 and 5, the parameters must be set as shown in **Table 3**.

Parameter	Required Setting	Comment
IP Stations	y	This is required so that IP stations can be configured.
Enhanced EC500	y	This is required to enable the allocation of off-PBX SIP telephones.
IP Trunks	y	This is required to allow the allocation of the H.323 trunks to be attached to NovaConf.
Uniform Dialing Plan	y	This is required to support the call routing scheme chosen for testing.

Table 3: System-Parameters Customers-Options Form, Page 4

```
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? n                          ISDN Feature Plus? n
                  Enhanced EC500? y            ISDN Network Call Redirection? n
    Enterprise Survivable Server? n                            ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                   ISDN-PRI? y
              ESS Administration? n               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? 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
                       IP Trunks? y

           IP Attendant Consoles? n
```

Figure 5: System-Parameters Customers-Options Form, Page 4

On page 5, the “Uniform Dialing Plan” parameter must be set to “y”.

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

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

Figure 6: System-Parameters Customers-Options Form, Page 5

3.1.2. Configure Node Names

Use the **change node-names ip** command to configure the IP addresses of the NovaConf and the Avaya SES servers. The value assigned to SES must be the same value that was assigned in **Figure 8**, **Figure 24**, **Figure 26**, and **Figure 29**.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
NovaConf            192.168.200.99
procr               192.168.30.100
ses                 192.168.200.100
```

Figure 7: Node-Names IP Form

3.1.3. Configure SIP Interface to the NovaConf Server

Use the **add signaling-group** command to configure the Signaling Group parameters for the SIP trunk group. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type	Enter the Group Type as "sip".
Far-end Node Name	Enter node name assigned to the Avaya SES in **Figure 7.**
Near-end Listen Port	Accept the default value of 5061. This must be the same value which is assigned to the SES contact shown in **Figure 34**.
Far-end Listen Port	Specify "5061" as the far end listen port.
Far-end Domain	Specify "ffm.com" as the far end node name. This must be the same value that was assigned in **Figure 8**, **Figure 24**, **Figure 26**, **Figure 29**, and **Figure 40**.

Table 4: Signaling-Group Parameters for SIP Interface

```
add signaling-group 83                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 83             Group Type: sip
                        Transport Method: tls







   Near-end Node Name: procr                  Far-end Node Name: ses
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region:
       Far-end Domain: ffm.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
```

Figure 8: Avaya SES Signaling-Group Form

Use the **add trunk-group** command to configure the SIP interface to Avaya SES. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type	Specify the Group Type as "sip"
Group Name	Select an appropriate name to identify the device.
TAC	Specify a trunk access code which can be used to provide dial access to the trunk group. This code must be defined in **Figure 11**.
Service Type	Designate the trunk as a "tie" line to a peer system.
Signaling Group	Enter the number assigned to the SIP signaling group shown in **Figure 8**.
Number of Members	Specify sufficient number of members to support the maximum simultaneous connections required.

Table 5: Trunk-Group Parameters for the SIP Interface

```
add trunk-group 83                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 83                   Group Type: sip           CDR Reports: y
  Group Name: SIP                         COR: 1        TN: 1        TAC: *83
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 83
                                                     Number of Members: 5
```

Figure 9: Trunk-Group Screen Form

Login to the SES server as user sroot and assign the IP address of the NovaConf server as a trusted host so that SES will not request authorization from NovaConf.

```
 admin@SES> trustedhost -a  192.168.200.99 -n 192.168.200.100 -c NovaLink
192.168.200.99 is added to trusted host list.
```

Figure 10: Trusted Host Assignment for SIP Interface

3.1.4. Configure Dial Plan and Call Routing

Use the **change dialplan analysis** command to specify that dialed strings which begin with “3”or “7” are extensions. The extensions local to this PBX are all seven digit numbers which begin with a “3”. The extensions assigned to the NovaConf are all seven digit numbers which begin with “7”. The dial string “*83” is used as the TAC for the SIP trunk define in **Figure 9**.

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

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type

        3          7   ext
        7          7   ext
        *83        3   dac
```

Figure 11: Dialplan Analysis Form

Use the **change uniform-dialplan** command to designate extensions which begin with “7” and are seven digits in length to use the Automatic Alternate Routing (AAR) table.

```
change uniform-dialplan 0                                       Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching                    Insert             Node
  Pattern       Len Del       Digits      Net Conv Num
  7               7   0                   aar   n
```

Figure 12: Uniform-Dialplan Form

Use the **change aar analysis** command to select routing pattern “7” for numbers which have the leading dialed string “7”, as specified in the uniform dial plan shown in **Figure 12**.

```
change aar analysis 0                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    3

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    7                     7    7     7        aar          n
```

Figure 13: AAR Analysis Form

Use the **change route-pattern** command to route numbers using Routing Pattern 7 via Trunk Group 83.

```
change route-pattern 7                                          Page   1 of   3
                    Pattern Number: 2   Pattern Name: NovaConf SIP
                             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: 83   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                                      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 14: Route-Pattern 6 Form

3.1.5. Configure Public-Unknown-Numbering Format

Use the **change public-unknown-numbering** command to designate how telephone numbers are to be displayed on stations that have displays. Specify that seven digit numbers starting with “7” from trunk group “7” and “3” from trunk group “83” should not be modified.

```
change public-unknown-numbering 7                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                     Total Administered: 2
 7  7              7                        7           Maximum Entries: 240
 7  3             83                        7
```

Figure 15: Public-Unknown-Numbering Form

3.1.6. Configure Telephone Stations

Use the **add station** command to configure all of the telephones shown in **Table 2**. The settings for Avaya 2400 Telephones are the same as those required for the Avaya 4621 Telephone, except that the "Type" designation must be set to match the telephone type.

Parameter	Usage
Type	Enter the type of station that is to be configured.
Security Code	Enter a numeric security code.
Name	Enter a descriptive name for the user of the station.
BUTTON ASSIGNMENTS	Assign "send-calls" and "call-fwd" buttons to the stations, as required to test call coverage and call forwarding with NovaConf. This not required for SIP telephones.

Table 6: Station Parameters

```
add station 3000136                                             Page   1 of   5
                                     STATION

Extension: 300-0136                      Lock Messages? n             BCC: 0
     Type: 4621                          Security Code: 6310003        TN: 1
     Port: S00006                      Coverage Path 1:               COR: 1
     Name: extn 3000136                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: 300-0136
            Speakerphone: 2-way                 Mute Button Enabled? y
        Display Language: english                  Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal                Media Complex Ext:
   Survivable Trunk Dest? y                            IP SoftPhone? n



                                            Customizable Labels? y
```

Figure 16: Add Station Form, Page 1

```
add station 3000136                                          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: call-fwd   Ext:
 2: call-appr                           6:
 3: call-appr                           7:
 4: send-calls Ext:                     8:
```

Figure 17: Add Station Form, Page 4

3.1.7. Configure Interface to Avaya SES and Integration for SIP Telephones

Use the **change off-pbx-telephone station-mapping** command to configure SIP telephones. Assign values for this command as shown in the following table.

Parameter	Usage
Station Extension	Enter the extension of the SIP telephone.
Application	Enter “OPS”.
Phone Number	Enter the phone number assigned to the SIP telephone.
Trunk Selection	Enter the number assigned to the SIP trunk group in **Figure 9**.
Call Limit	Enter “3” to allow the SIP telephone to do call transfers.

Table 7: Parameters for Off-PBX-Telephone Station-Mapping

```
change off-pbx-telephone station-mapping 3000115             Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 300-0115          OPS          -     3000115             83         1
```

Figure 18: Off-PBX-Telephone Form, Page 1

```
change off-pbx-telephone station-mapping 3000115                Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 300-0115       3       both           all         both
```

Figure 19: Off-PBX-Telephone Form, Page 2

Use the **change feature-access-codes** command to assign feature codes required by SIP telephones, as shown in the following table:

Parameter	Usage
Call Forwarding Activation Busy/DA and Deactivation	Assign unused feature access codes that are within the local dial plan to activate/deactivate call forwarding.
Send All Calls Activation and Deactivation	Assign unused feature access codes that are within the local dial plan to activate/deactivate call sending all calls to coverage.

Table 8: Parameters for the Feature Access Codes

```
change feature-access-codes                                     Page   1 of   5
                               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:       Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *75    All: *73    Deactivation: *74
   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:
                   Contact Closure   Open Code:          Close Code:
```

Figure 20: Feature Access Codes Form, Page 1

```
change feature-access-codes                                     Page   3 of   5
                               FEATURE ACCESS CODE (FAC)
             Leave Word Calling Send A Message:
           Leave Word Calling Cancel A Message:
   Limit Number of Concurrent Calls Activation:        Deactivation:
               Malicious Call Trace Activation:        Deactivation:
         Meet-me Conference Access Code Change:

 PASTE (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code:
      Per Call CPN Unblocking Code Access Code:

                  Priority Calling Access Code:
                           Program Access Code:

       Refresh Terminal Parameters Access Code:
              Remote Send All Calls Activation:        Deactivation:
               Self Station Display Activation:
                     Send All Calls Activation: *71    Deactivation: *72
         Station Firmware Download Access Code:
```

Figure 21: Feature Access Code Form, Page 3

Use the **change off-pbx-telephone feature-name-extensions** command to assign extensions to features required by SIP telephones, as shown in the following table below. Note that the extensions used here are assigned to speed dial entries for SIP telephones, as shown in **Table 12**.

Parameter	Usage
Call Forward All	Assign an unused extension within the local dial plan to the “Call Forward All” feature.
Call Forward Cancel	Assign an unused extension within the local dial plan to the “Call Forward Cancel” feature.
Send All Calls	Assign an unused extension within the local dial plan to the “Send All Calls” feature.
Send All Calls Cancel	Assign an unused extension within the local dial plan to the “Send All Calls Cancel” feature.

Table 9: Parameters for Off-PBX-Telephone Feature-Name-Extension

```
change off-pbx-telephone feature-name-extensions              Page   1 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

     Active Appearance Select:
          Automatic Call Back:
   Automatic Call-Back Cancel:
             Call Forward All: 300-1804
  Call Forward Busy/No Answer:
          Call Forward Cancel: 300-1806
                    Call Park:
        Call Park Answer Back:
                 Call Pick-Up:
         Calling Number Block:
       Calling Number Unblock:
         Conference on Answer:
        Directed Call Pick-Up:
        Drop Last Added Party:
    Exclusion (Toggle On/Off):
   Extended Group Call Pickup:
       Held Appearance Select:
```

Figure 22: Off-PBX-Telephone Feature Name Extensions Form, Page 1

```
change off-pbx-telephone feature-name-extensions              Page   2 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

       Idle Appearance Select:
           Last Number Dialed:
         Malicious Call Trace:
  Malicious Call Trace Cancel:
          Off-Pbx Call Enable:
         Off-Pbx Call Disable:
                Priority Call:
               Send All Calls: 300-1825
        Send All Calls Cancel: 300-1826
          Transfer On Hang-Up:
       Transfer to Voice Mail:
      Whisper Page Activation:
```

Figure 23: Off-PBX-Telephone Feature Name Extensions Form, Page 2

Use the **change ip-network-region** command to configure the network region used by Avaya SES. Assign values for this command as shown in the following table.

Parameter	Usage
Authoritative Domain	Enter the domain name assigned to Avaya SES. This must be the same value which was assigned in **Figure 8**, **Figure 26**, **Figure 29**, and **Figure 40**.
Name	Enter a descriptive name.

Table 10: Parameters for IP-Network-Region 1

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: ffm.com
    Name: FFM
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? y
   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 24: IP-Network-Region Form

Use the **change ip-codec-set** command to specify the codec to be used for the Network Region assigned to Avaya SES. Specify that the G.711A codec is to be used.

```
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.711A            n         2        20
 2:
 3:
 4:
 5:
 6:
 7:
```

Figure 25: IP-Codec-Set Form

3.2. Configure Avaya IP Telephones

Configure the **46xxsettings.txt** text file to be used by Avaya IP Telephones. The parameters that are required to be configured in this file are shown in the following table. This is a "flat" ASCII file that must reside in the directory of the TFTP server accessible by the Avaya IP Telephones. Avaya IP Telephones must be configured so that the "FileSv" parameter is set to the address of the TFTP server that contains this configuration file, which is re-read each time the phone is restarted.

Parameter	Usage
MWISRVR	The value "SES_IP_address" indicates that Avaya SIP telephones should register with the Avaya SES server to receive message waiting events.
SIPDOMAIN	Enter the name of the SIP domain. This must be the same value which was assigned in **Figure 8**, **Figure 24**, **Figure 29**, and **Figure 40**.
ENHDIALSTAT	Set this parameter to "0" to indicate that enhanced dialing is not required.

Table 11: Parameters for Telephone Setting File

```
SET MWISRVR          "SES_IP_address"
SET SIPDOMAIN        "ffm.com"
SET ENHDIALSTAT      0
```

Figure 26: Telephone Settings File Content

In addition to these settings, Avaya SIP Telephones must be configured manually to add speed dial entries to activate/deactivate Call Forwarding and Send All Calls features, by assigning the extensions that were assigned to the features shown in the following table to speed dial entries. These extensions are those that were assigned to using the **Off-Pbx-Telephone Feature-Name-Extensions** command described in **Table 9**.

Parameter	Extension	Usage
CallFwd On	3001804	Activate Call Forwarding
CallFwd Off	3001806	Deactivate Call Forwarding
SendAll On	3001825	Activate Send All Calls
SendAll Off	3001826	Deactivate Send All Calls

Table 12: Speed Dial Entry Assignments for Avaya SIP Telephones

3.3. Configure Avaya SIP Enablement Services

Log in to the Avaya SES Web-based Integrated Management tool by selecting the IP address of the Avaya SES server followed by "/admin" from the Web browser. After entering the login ID and password, select "Launch Administration Web Interface".

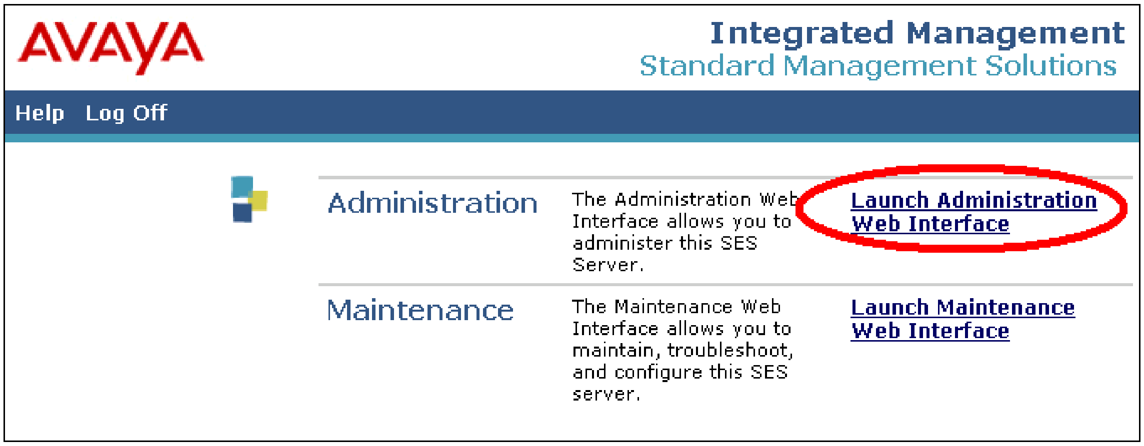

Figure 27: SES Initial Greeting Screen

The SES Integrated Management top level menu is then displayed.

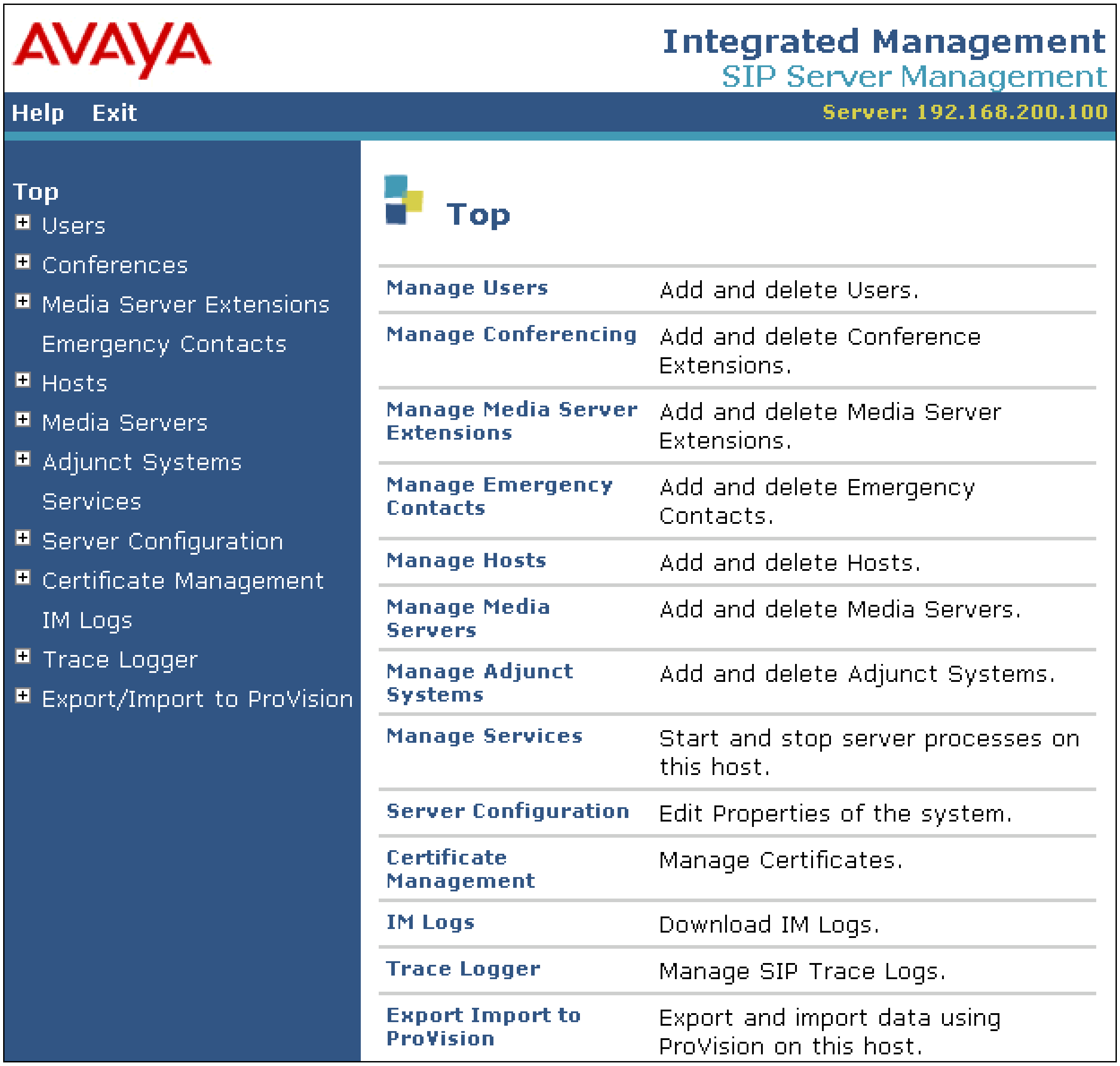

Figure 28: SES Integrated Management Top Level Menu

3.3.1. Configure Basic SES Parameters

From the top-level management screen, select "Server Configuration" -> "System Properties". Enter the name to be assigned to the "SIP Domain". This must be the same value which was assigned in **Figure 8**, **Figure 24**, **Figure 26**, and **Figure 40**. Select the "Update" button.

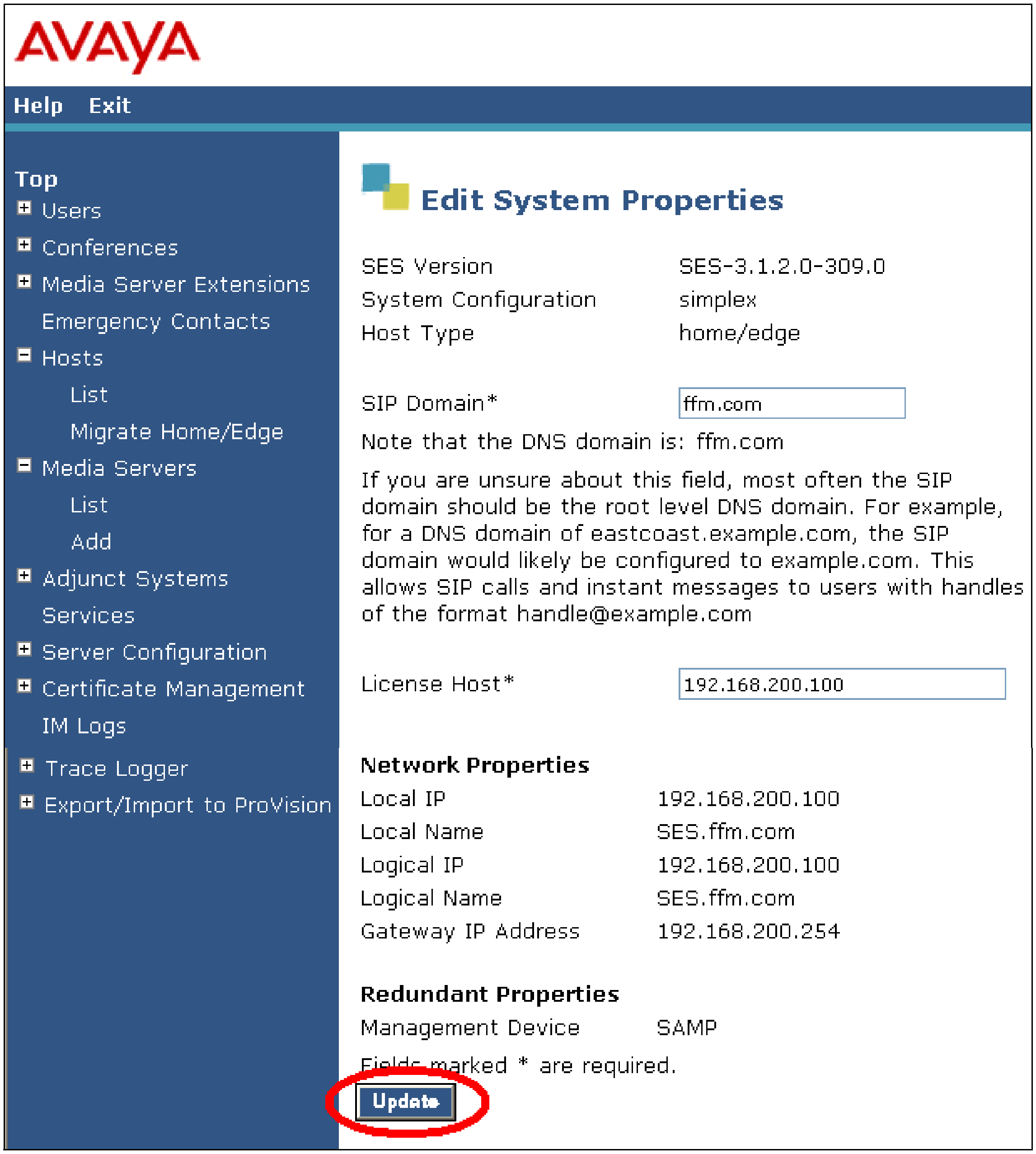

Figure 29: Avaya SES Edit System Properties Screen

From the top-level management screen, click “Manage Hosts” → “Add Host”.
Enter the IP address of the Avaya SES Server, a **DB password**, and a **Profile Service Password** that were allocated to the Avaya SES server when it was installed. Leave the other field assigned to their respective default values. Select the “Update” button.

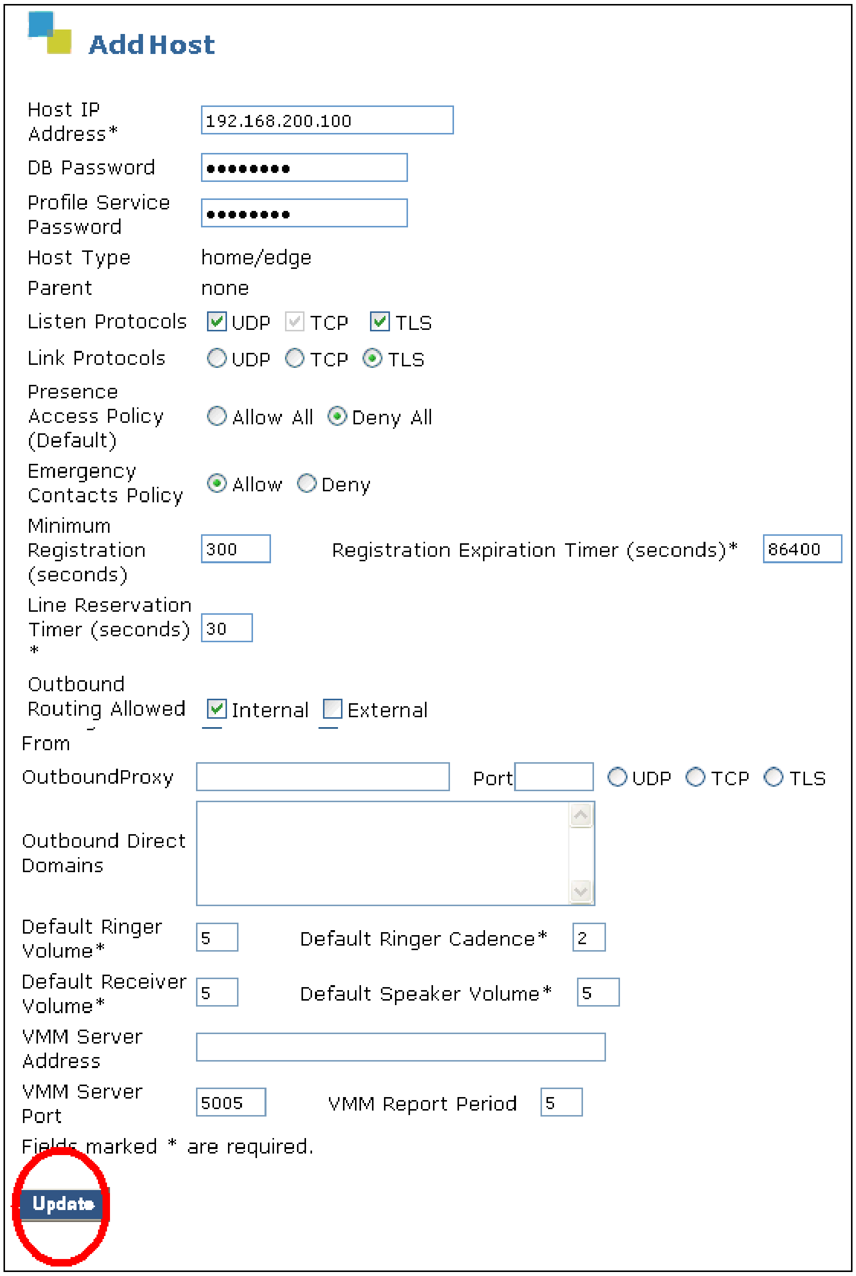

Figure 30: Avaya SES “Add Host” Screen

3.3.2. Configure Interface to Avaya Communication Manager

From the top-level management screen, select "Manage Media Servers"-> "Add Media Server". Assign a meaningful name to the "Media Server Interface Name". Select the IP address of the Avaya SES server as the "Host". Enter the address of the Avaya S8300 Server as the SIP Trunk IP Address. Select the "Add" button when these parameters have been entered.

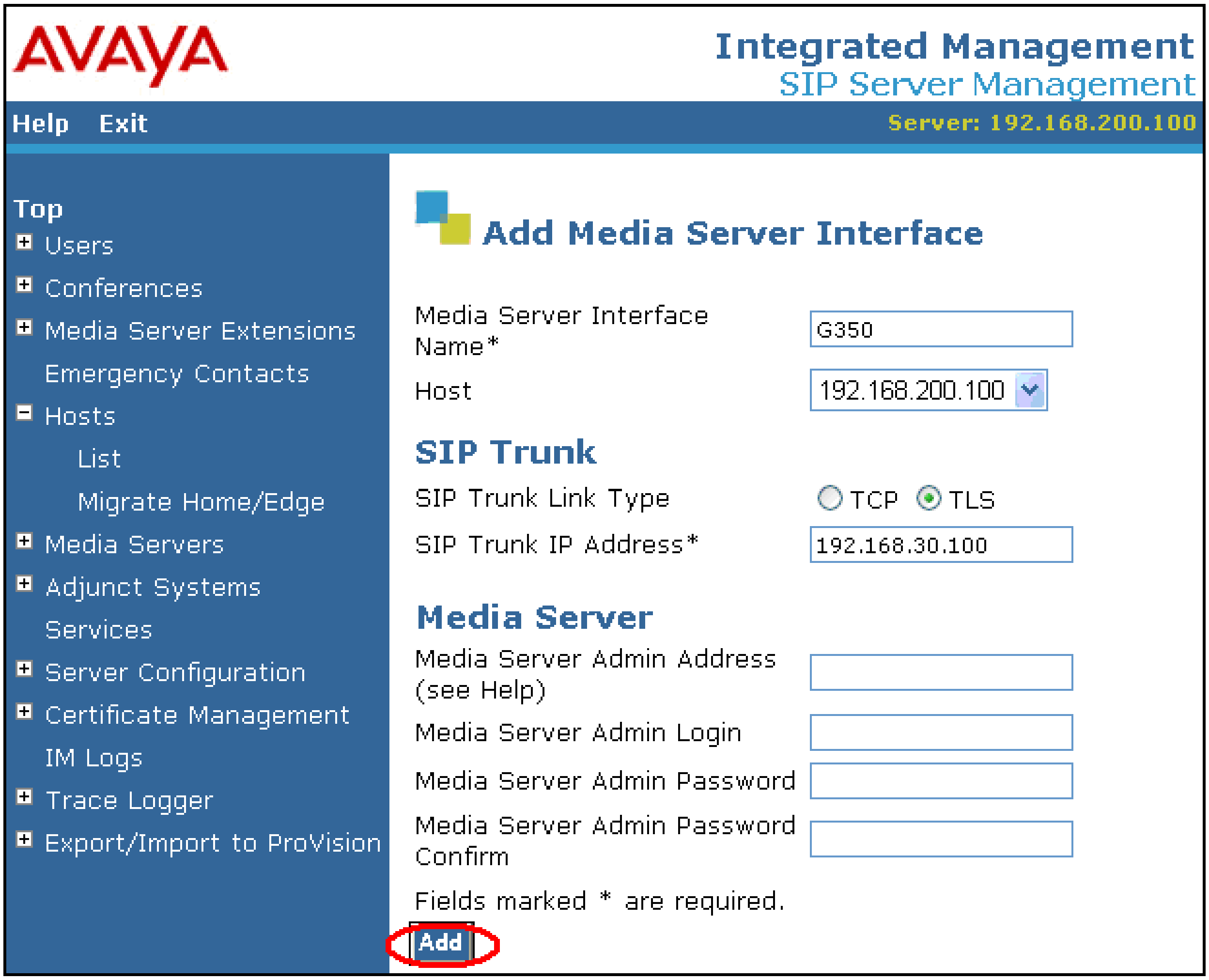

Figure 31: Avaya SES Add Media Server Interface Screen

From the Media Server List screen select “Map” -> “Add Map In New Group”.

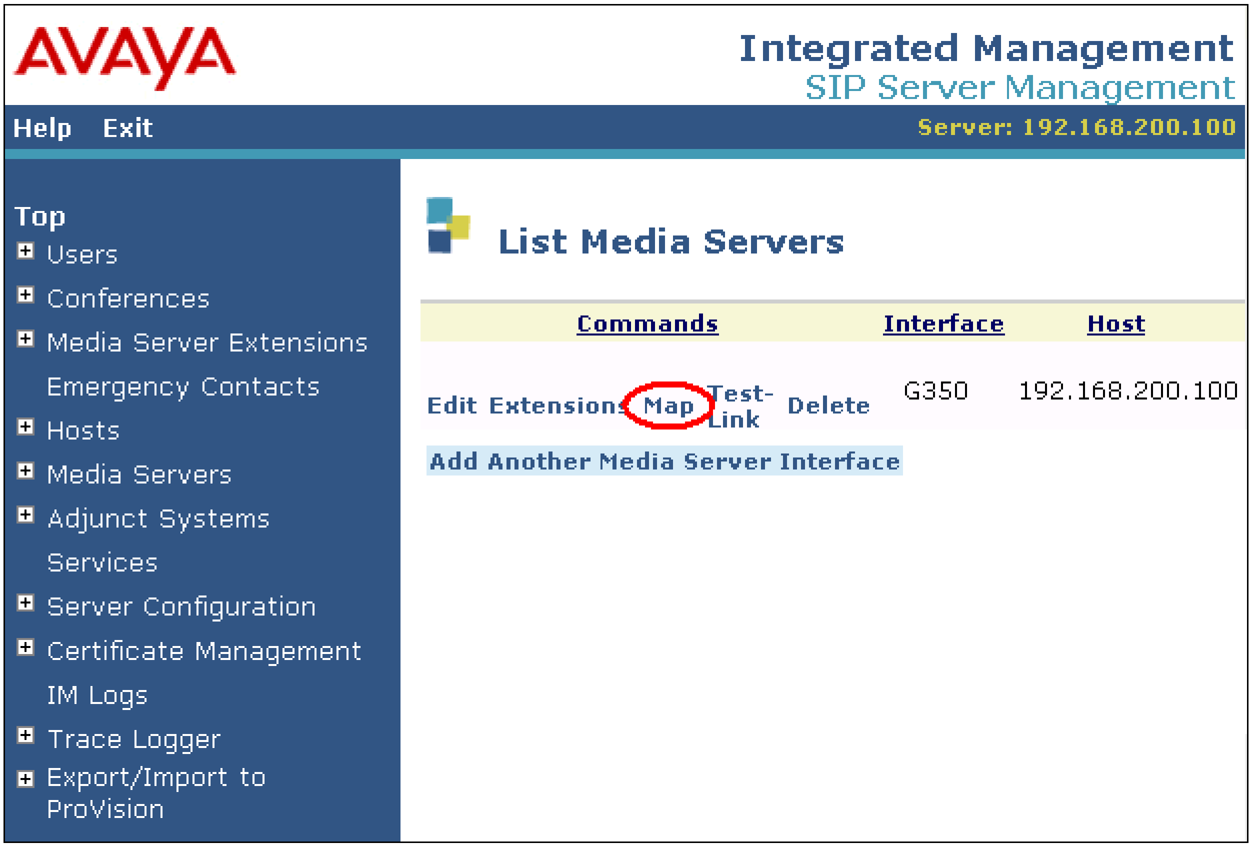

Figure 32: Media Server List Screen

Enter a meaningful name for the address map. Enter the "^sip:3.*" to cause all calls to numbers beginning with "3" to be routed to the G350.

Click the **Add** button upon completion.

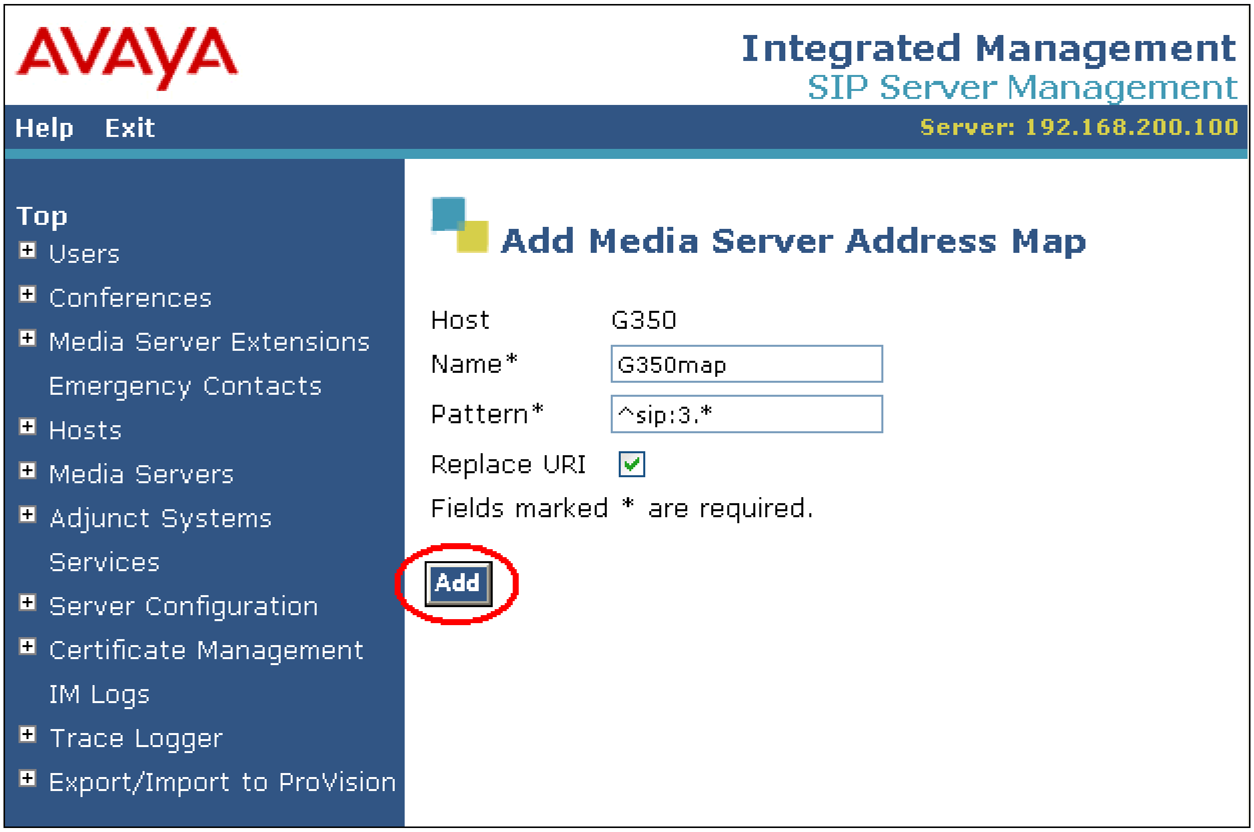

Figure 33: Add Media Server Address Map

A default “contact” is added automatically using the default TLS port. This must be the same port address assigned to the SES signaling group as the “Near-end Listen Port” in **Figure 8**.

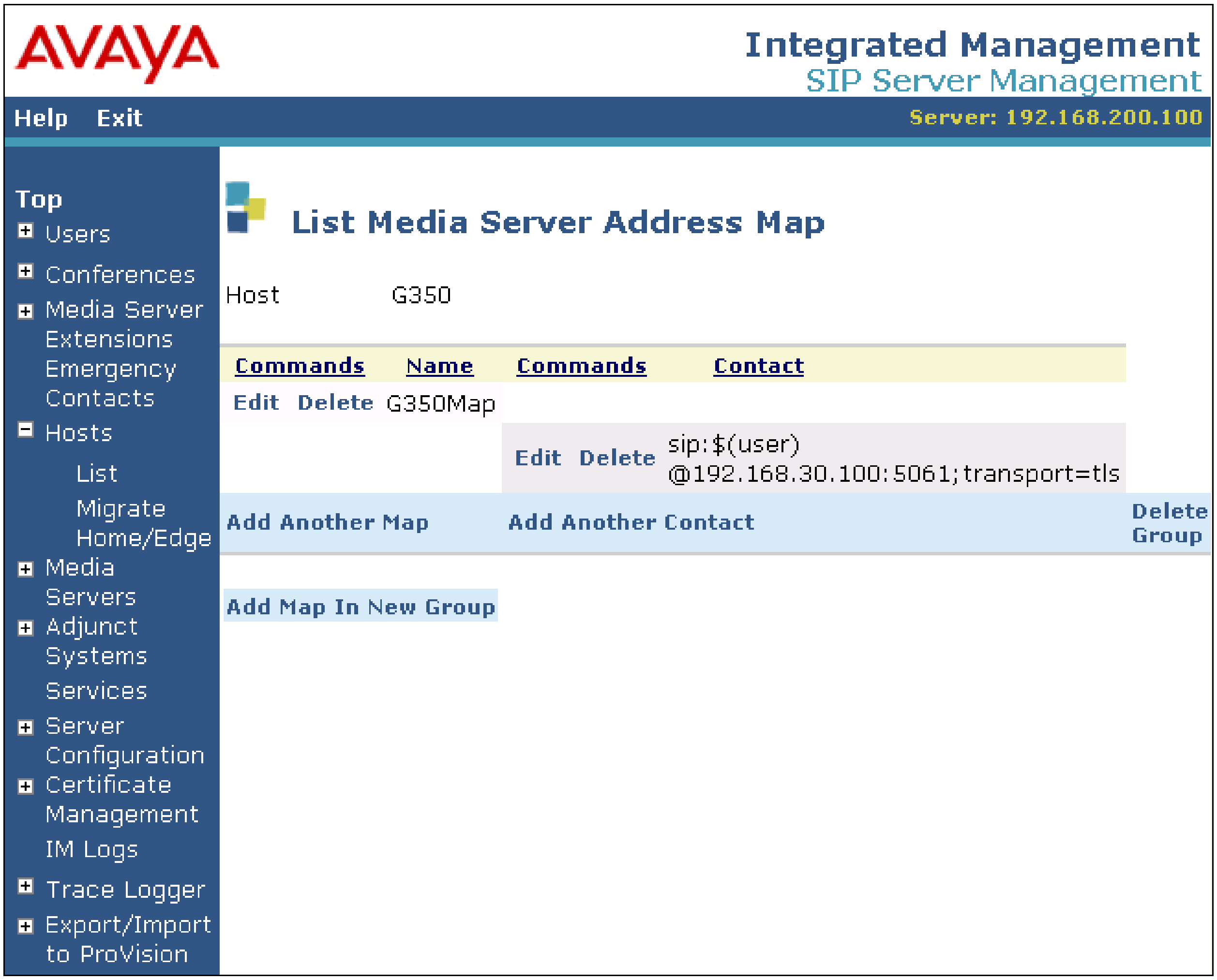

Figure 34: Media Server with Default Contact

3.3.3. Configure SIP Interface to NovaConf

From the top level menu select “Manage Hosts” -> “List Hosts”. Select the “Map” -> “Add Map In New Group” menu point to add a host map for NovaConf.

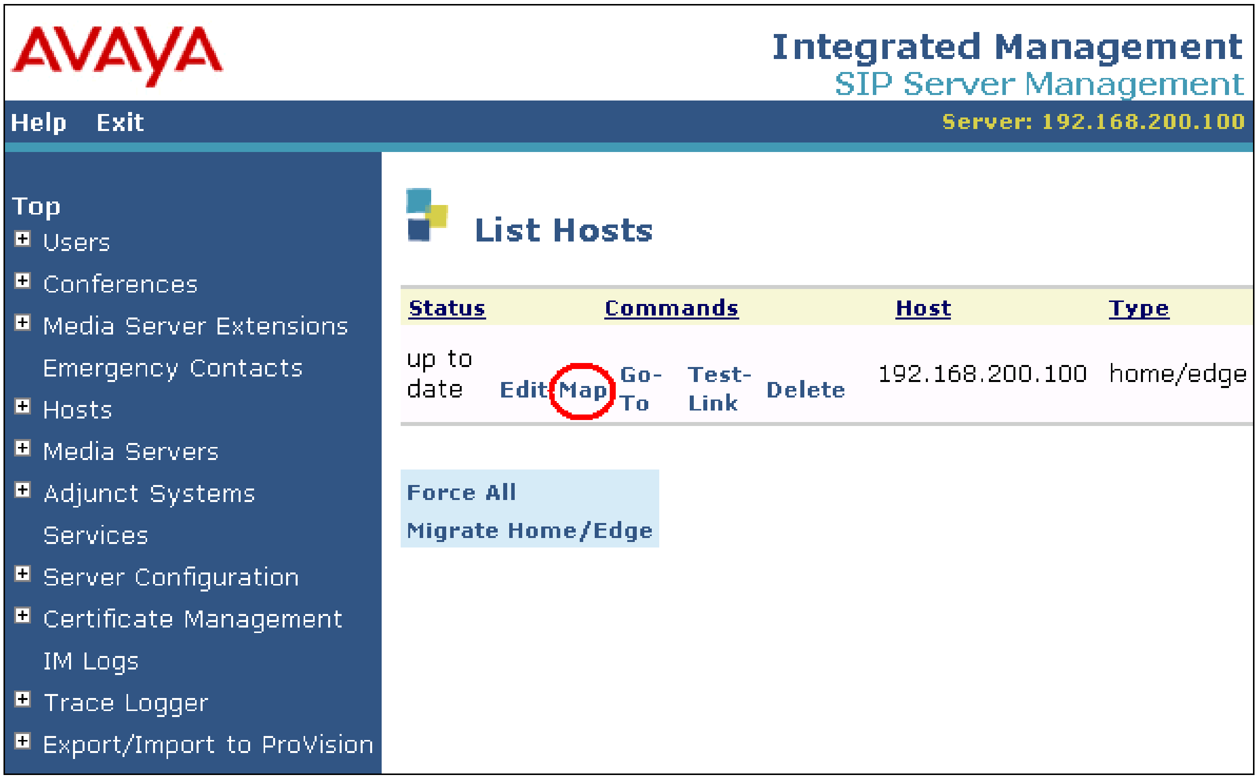

Figure 35: SES Host List

Enter a meaningful name for the host map and patent which matches the telephone extensions assigned to NovaConf. The pattern "^sip:7.*" matches all SIP telephone numbers assigned to NovaConf (numbers beginning with "7"). Click the "Add" button upon completion.

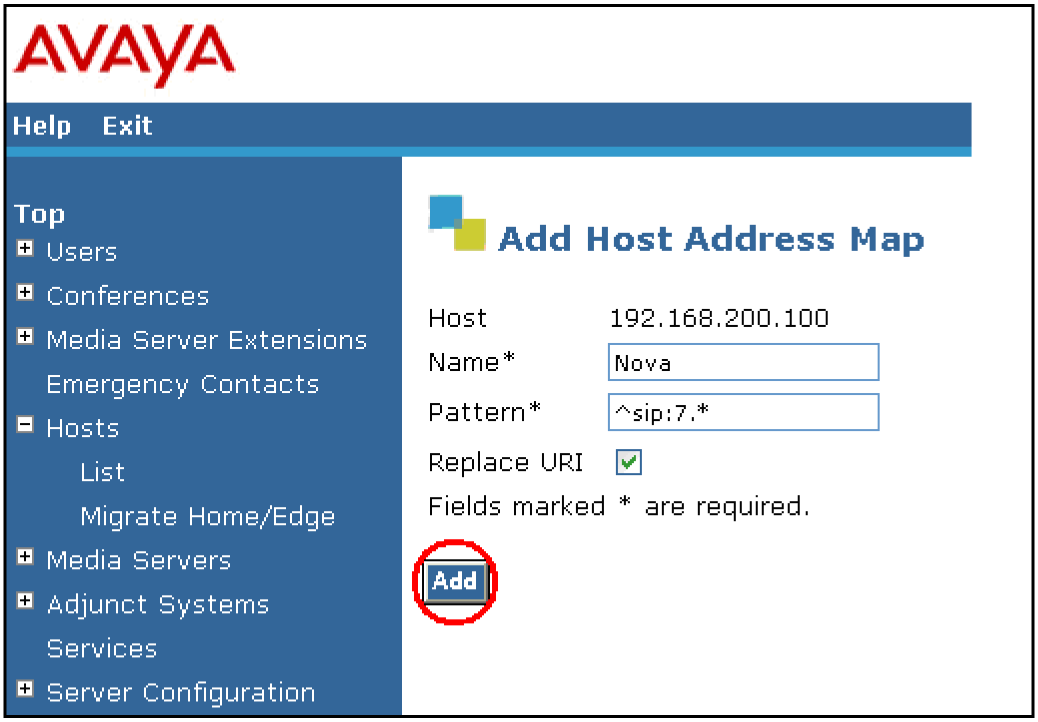

Figure 36: SES Host Address Map for NovaConf

When the Host Address Map list is displayed, select “Add Contact” to create a contact for NovaConf. Enter “sip:$(user)@”<NovaConf IP address>”;”<NovaConf port>”;transport=udp”. Upon completion of this form, click “Add”.

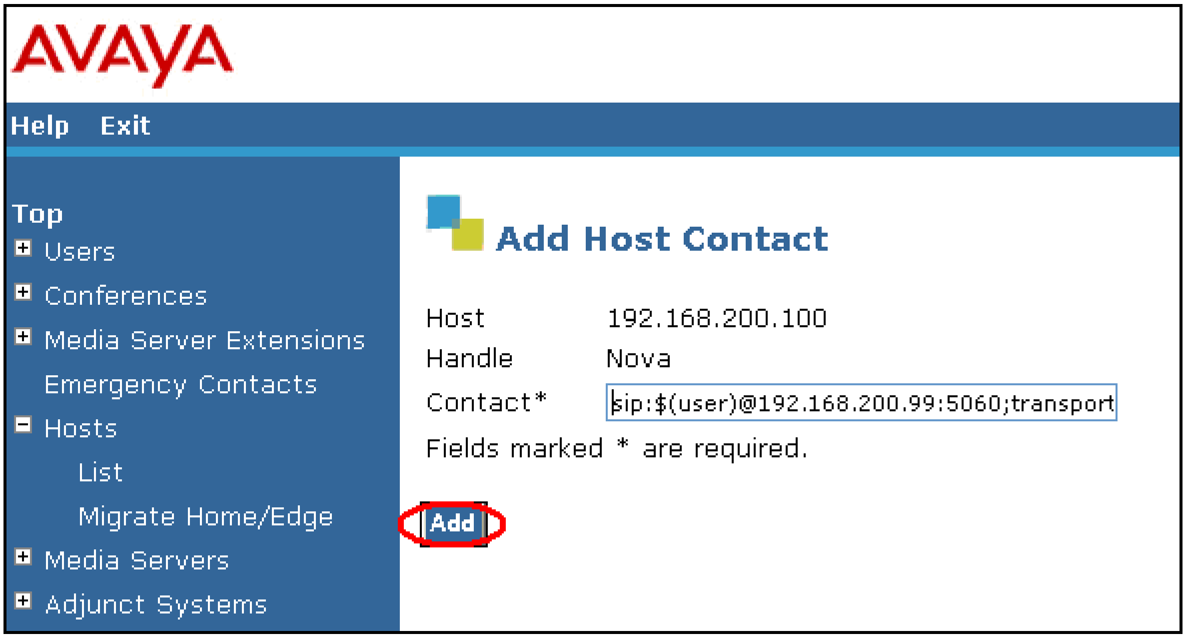

Figure 37: SES Host Contact for NovaConf

3.3.4. Configure Users for SIP Endpoints

This step can be omitted if no SIP endpoints are present in the configuration. From the top level menu, select the "Manage Users" -> "Add User" menu entries. Enter the extension of the user to be added as the "Primary Handle". This is the same extension that was configured in **Section 3.1.7**. Enter a Password and First/Last name of the user, check the "Add Media Server Extension" box, and click "Add".

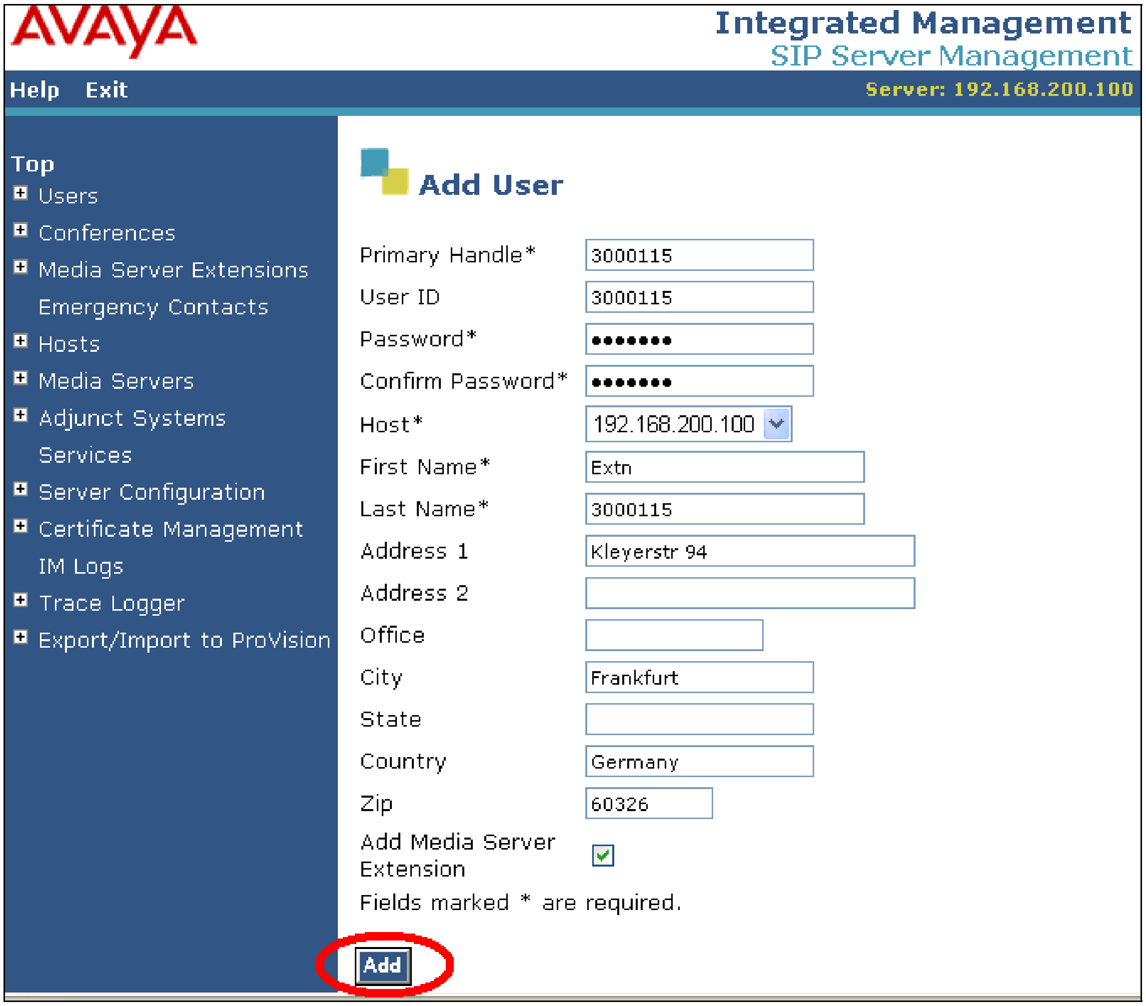

Figure 38: Avaya SES "Add User" Screen

Enter the Media Server Extension for the User ID 3000115 (the extension of the Avaya SIP telephone). Select the Media Server (refer to **Figure 32**) and from the drop down box and click "Add" to continue.

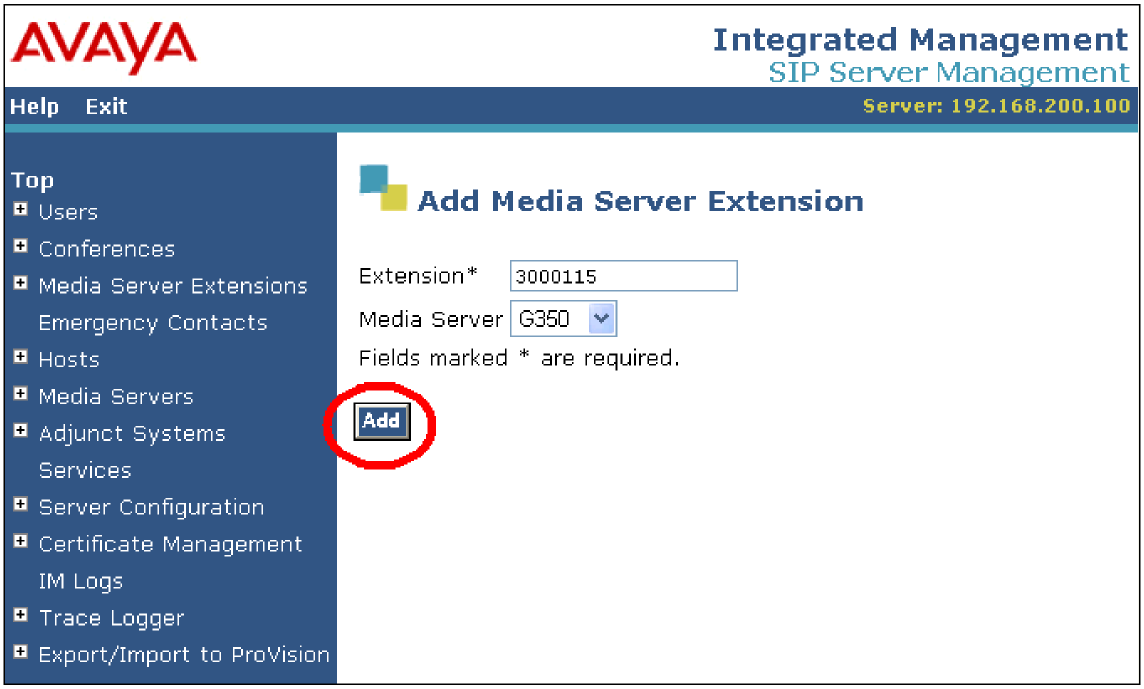

Figure 39: Avaya SES Add Media Server Extension Screen

3.4. Configure NovaConf

3.4.1. Configuration file NovaConf.ini

The NovaConf.ini configuration file is a "flat" ASCII file that can be edited with a text editor. This file is contained in the main installation directory on the NovaConf server (e.g. C:\Program Files\NovaConf). The values within this file must be set as shown in **Figure 40**. The values for those items shown in bold may vary, depending on the configuration of external components. The values to be used for these entries are described in the following table.

Parameter	Usage
CardDriver	Enter "3" to specify the SIP driver.
DefaultCallingParty	Enter the extension chosen for NovaConf.
DriverPref	Enter "3" to specify the SIP driver.
LocalUserName	Enter the extension chosen for NovaConf.
SIP_Gateway	Enter the Authoritative Domain. This must be the same value which was assigned in **Figure 8**, **Figure 24**, **Figure 26**, and **Figure 29**. This is followed by the IP address for was the Avaya SES server, as defined in the "ses" entry in the Node-Names form shown in **Figure 7**.
Rufnummer	This is the extension assigned to NovaConf. A value of "7111111" was used for these tests.

Table 13: Parameters for Telephone Setting File

```
[CallInfo]
CardDriver=3
DefaultCallingParty=7111111

[VoIP]
DriverPref=3
LocalUserName=7111111
SIP_Gateway=ffm.com,192.168.200.100

[NovaConf]
Rufnummer=7111111
```

Figure 40: NovaConf.ini Configuration File Content

3.4.2. Configure NovaConf Application

Use the Windows "Start" button to select the program "NovaConf Webclient". After entering the user name and password, the NovaConf startup screen is displayed. Click the "Show users" icon to show potential conference participants.

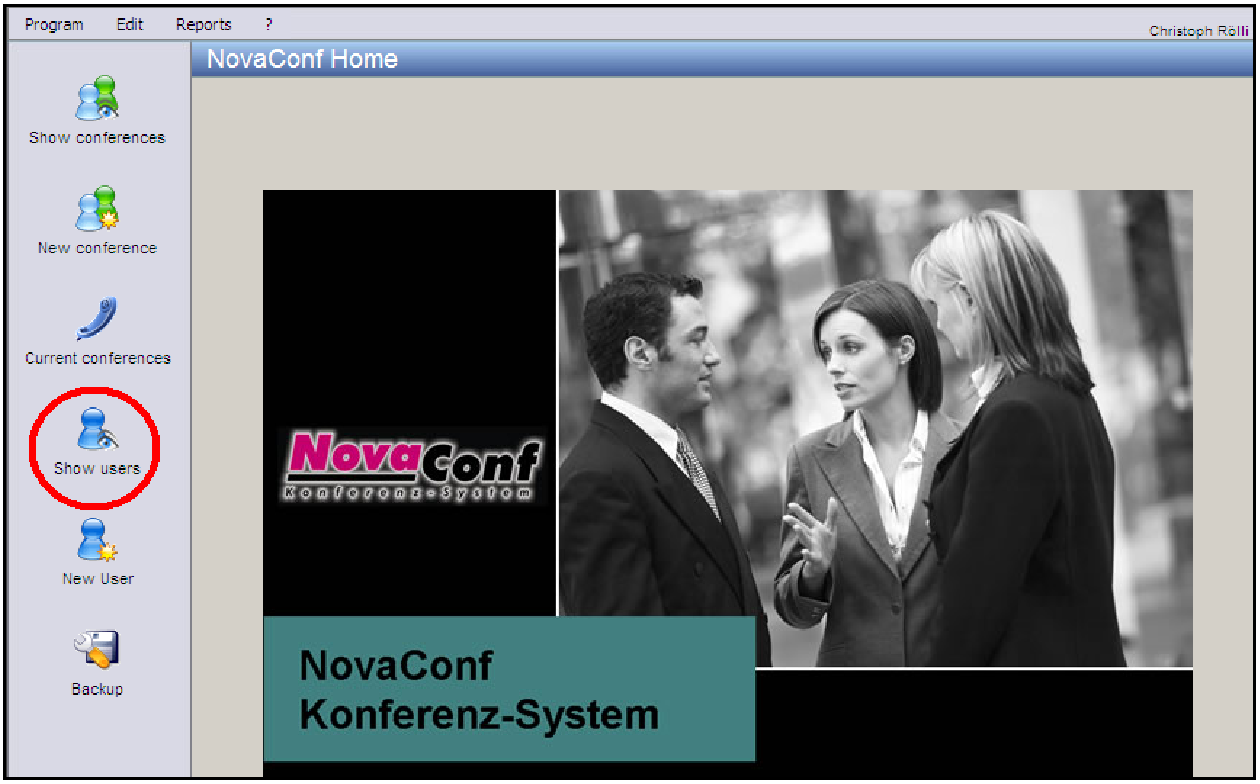

Figure 41: NovaConf Startup Screen

3.4.2.1 Configure Users

Assuming that no other users have been defined, the user designated as administrator is displayed. The configuration of the administrator is beyond the scope of these Application Notes. See reference [4] for additional information. Click the “New person” icon to add a potential conference participant. A conference user should be configured for each of the telephone extensions shown in **Table 2**.

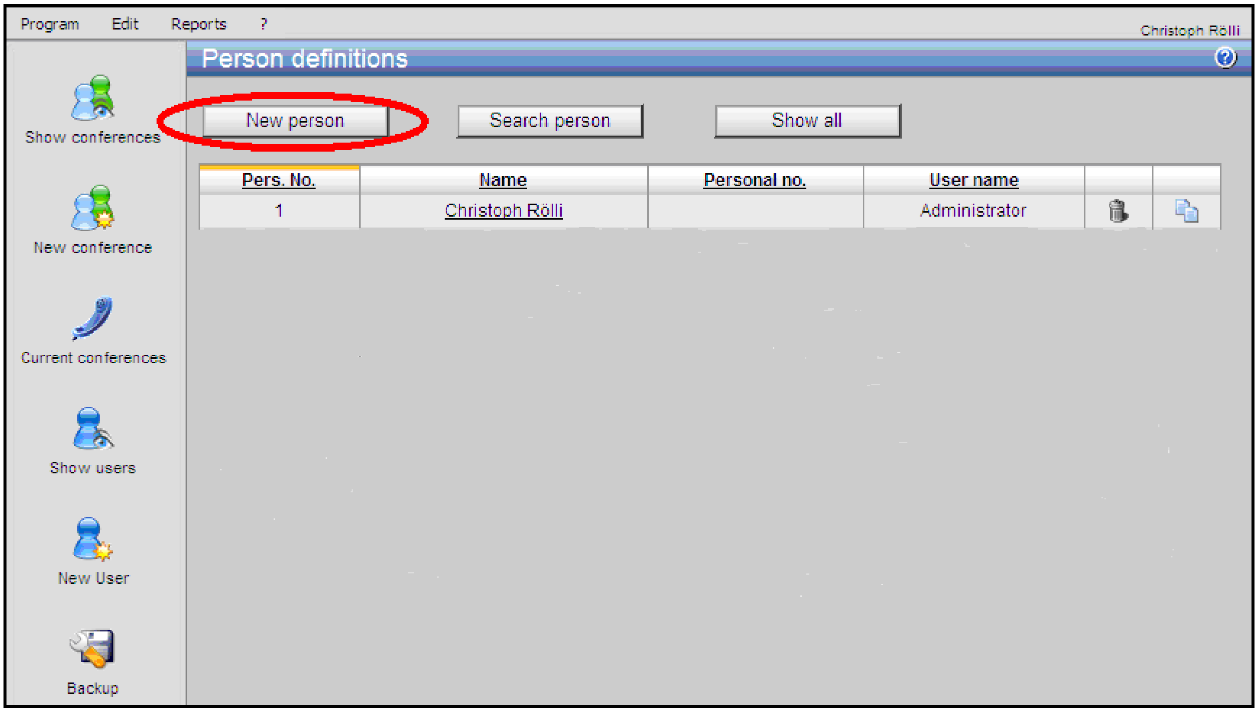

Figure 42: NovaConf User Configuration Screen

In the “Personal details” tab, enter the user’s name in the “Name” field (name not entered in **Figure 46**) and a numeric PIN code to be assigned to the user in the “PIN code” field. The user will use this PIN code when an authorization sequence for a conference operation is required.

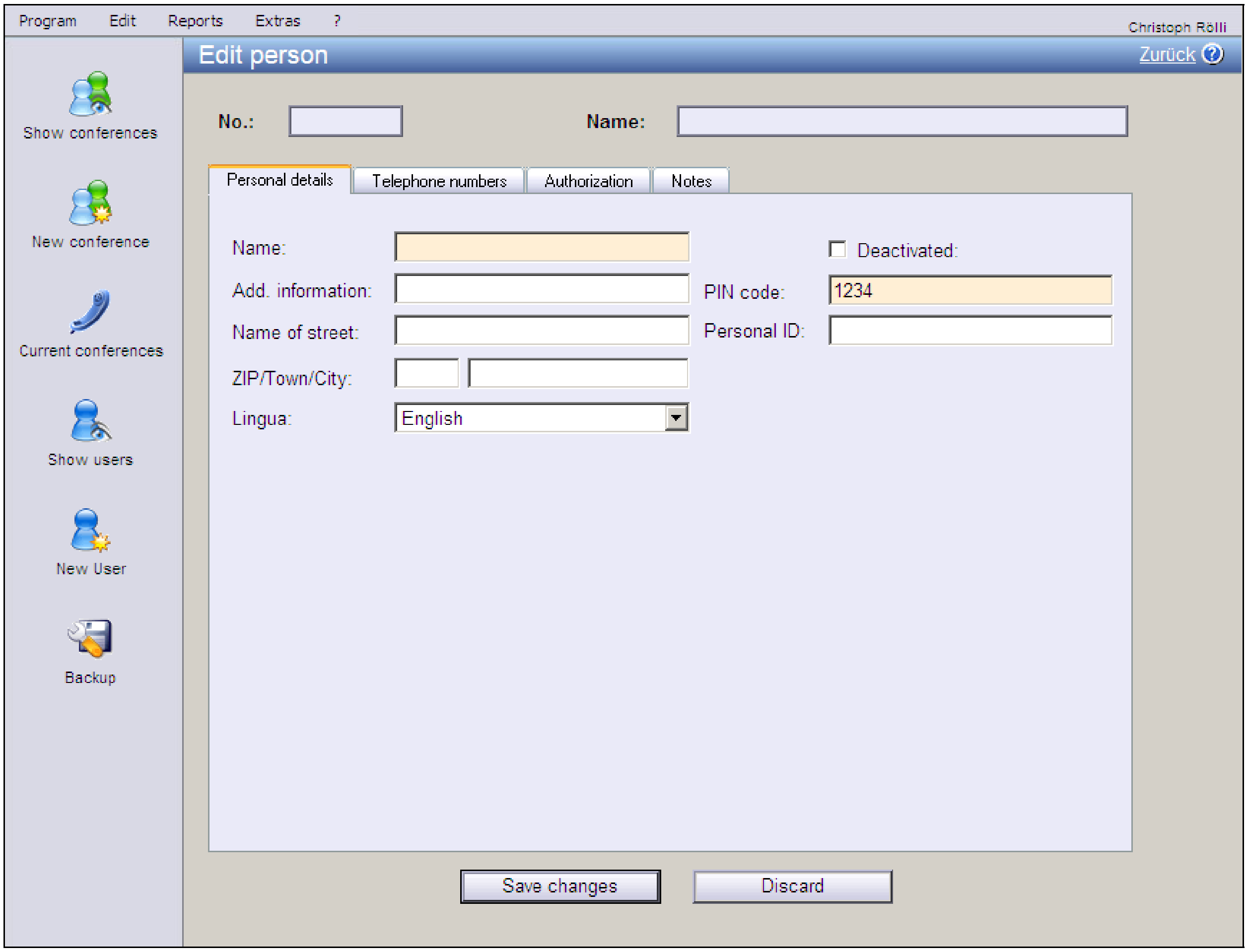

Figure 43: NovaConf Edit Personal Details Screen

Select the “Telephone numbers” tab to enter the telephone number to be assigned to the user. For testing purposes, it is sufficient to configure one telephone extension, which can be entered into the “Office 1” field. Click the “Save changes” button to save the user’s configuration and return to the “Person definitions” screen.

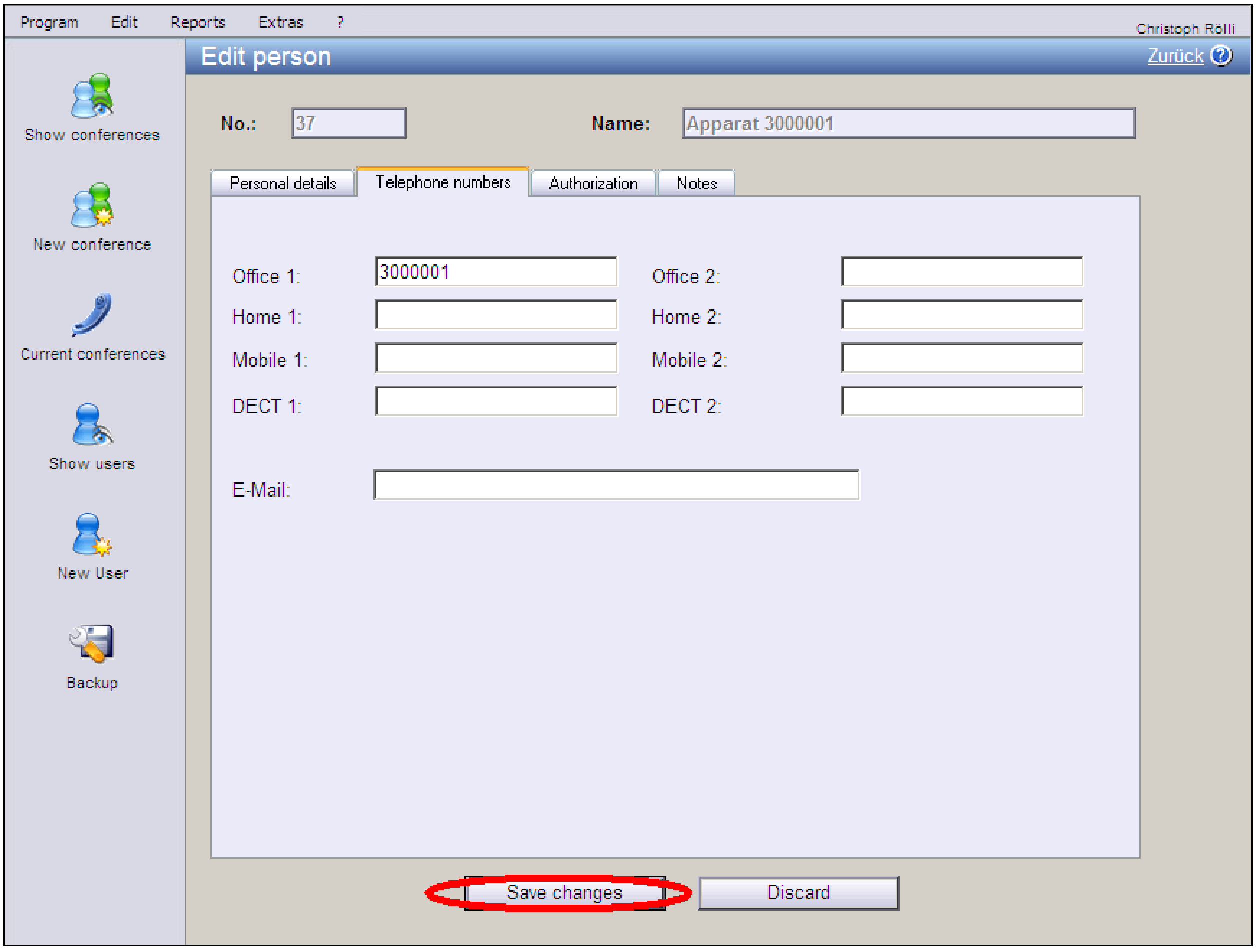

Figure 44: NovaConf Edit User Telephone Numbers Screen

When users have been allocated for each of the extensions in **Table 2**, the newly configured users are now listed in the “Person definitions” screen. Click the “Show conference” icon to continue.

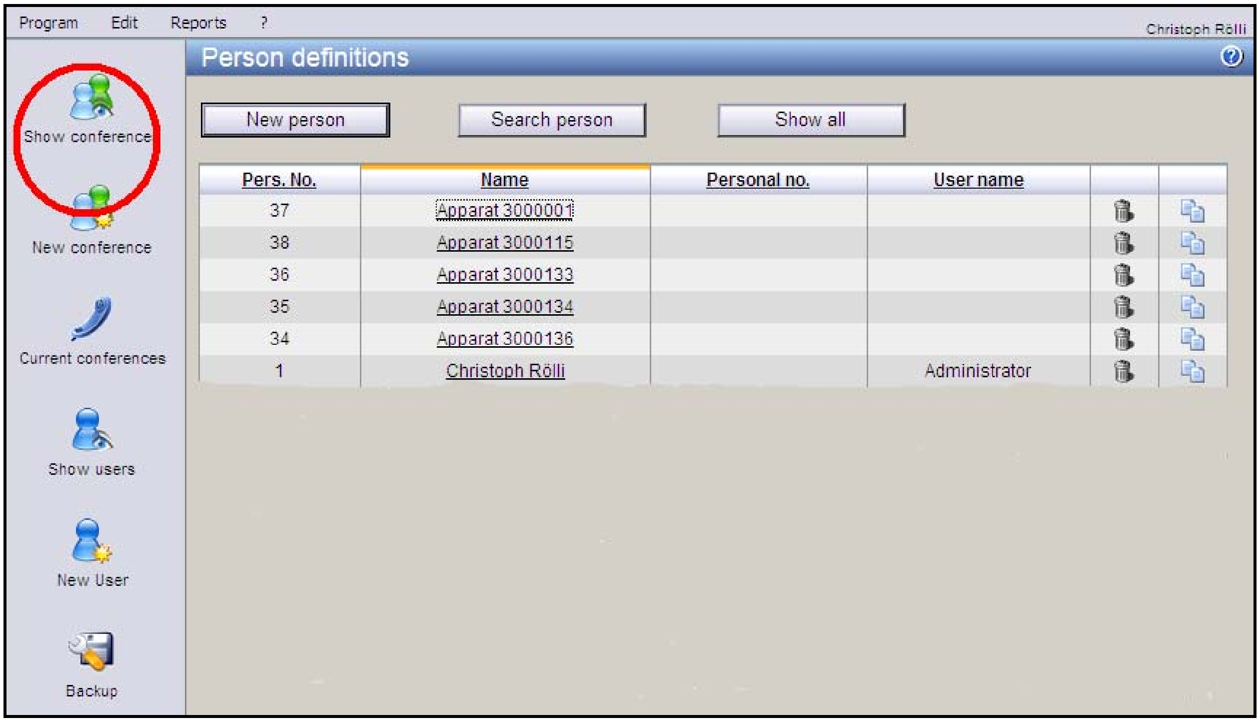

Figure 45: NovaConf Personal User Display Screen

3.4.2.2 Configure Conferences

From the “Predefined Conferences” screen, click the “New Conference” button to create a new conference. This operation is performed once for each of the three conference types used by the tests described in these Application Notes: incoming conference, outgoing conference, and ad-hoc conference.

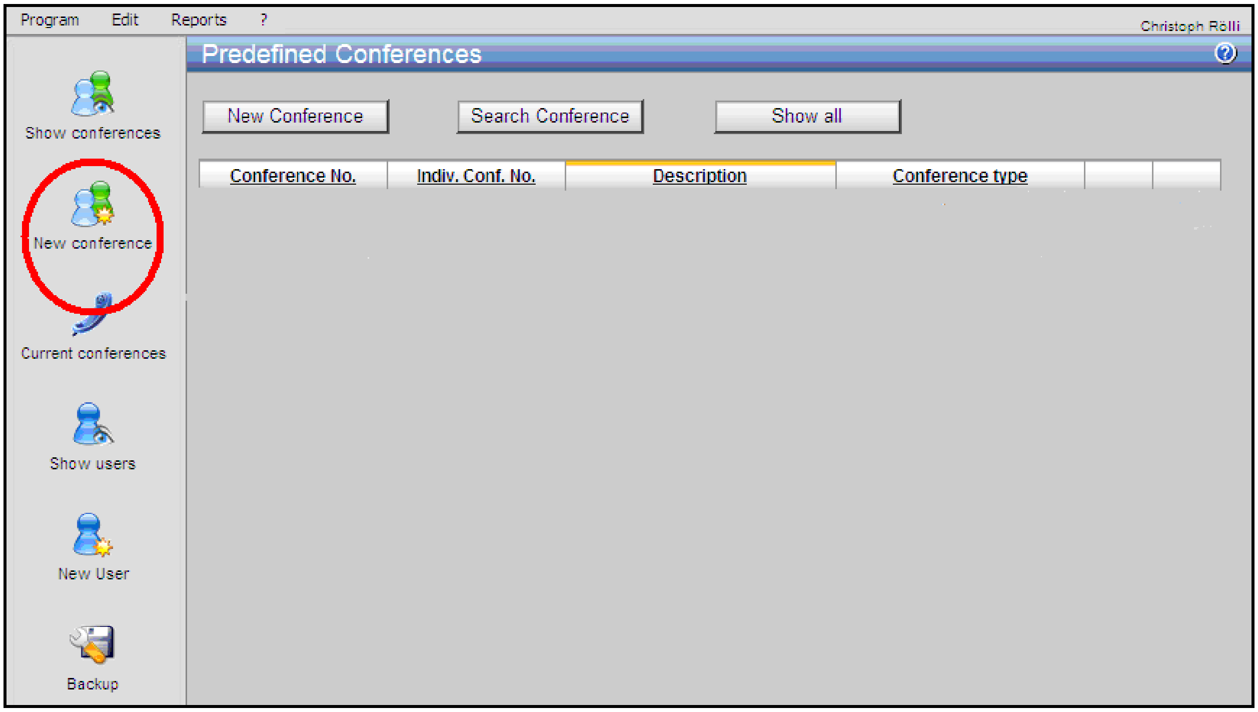

Figure 46: Predefined Conference List Screen

The “Common” tab of the “Edit conference” screen allows the creation of various conference types. Enter the parameters for the conference to be configured as shown in the table below.

Parameter	Usage
Description	Assign a descriptive name to the conference.
Individual No.	Assign an unused conference number to be used as an identifier for this conference.
Conference-Type	Select “Outgoing Conference”, “Incoming Conference”, or “Ad-hoc Conference” from this drop-down box, dependent on the type of conference which it to be created.
Message	Select an existing message from the list of files contained within this drop-down box, or click the button to the right to record a new message.

Table 14: NovaConf Conference Common Configuration Parameters

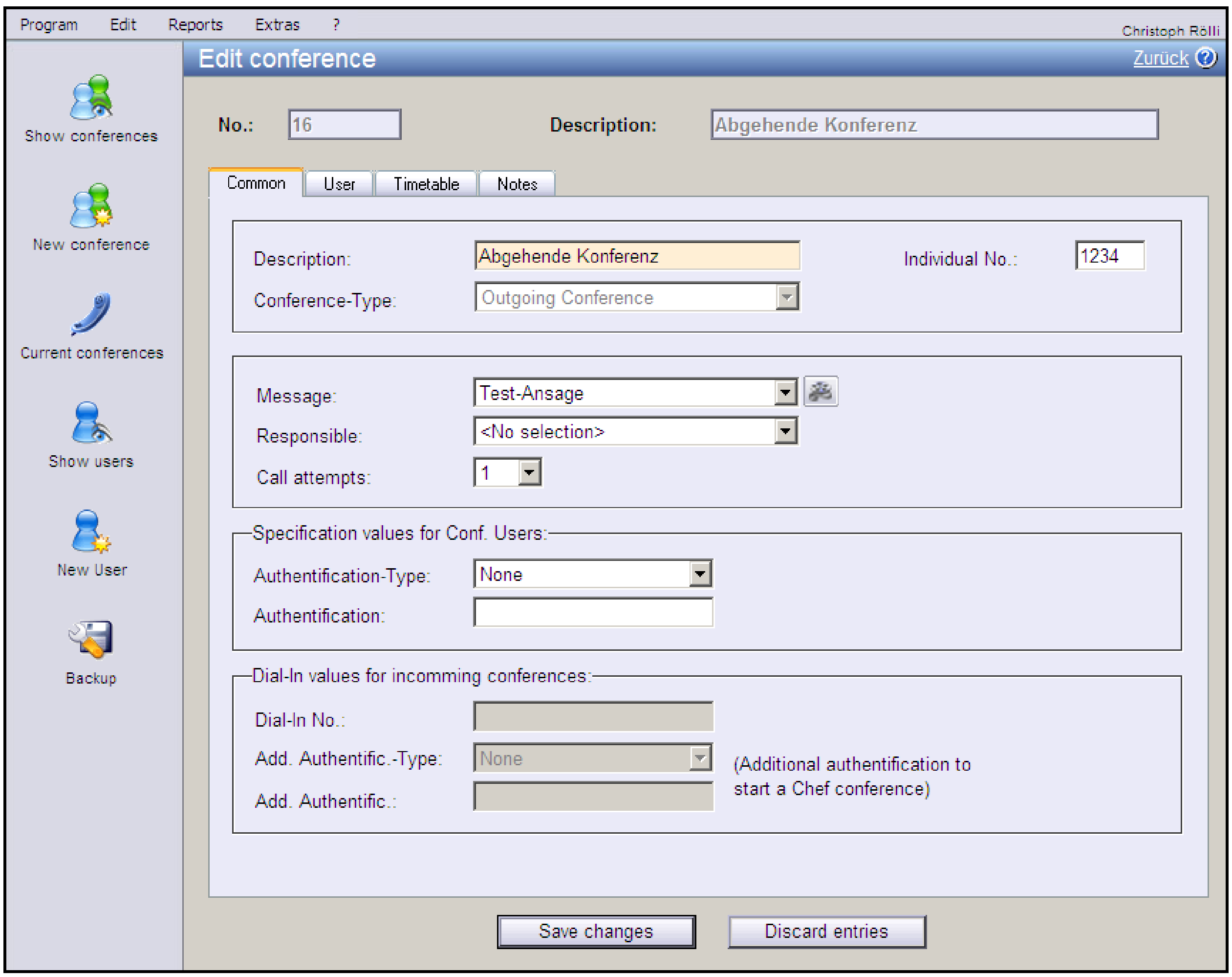

Figure 47: NovaConf New Outgoing Conference Screen

Select the “User” tab and allocate users to the conference using “drag and drop” operations.

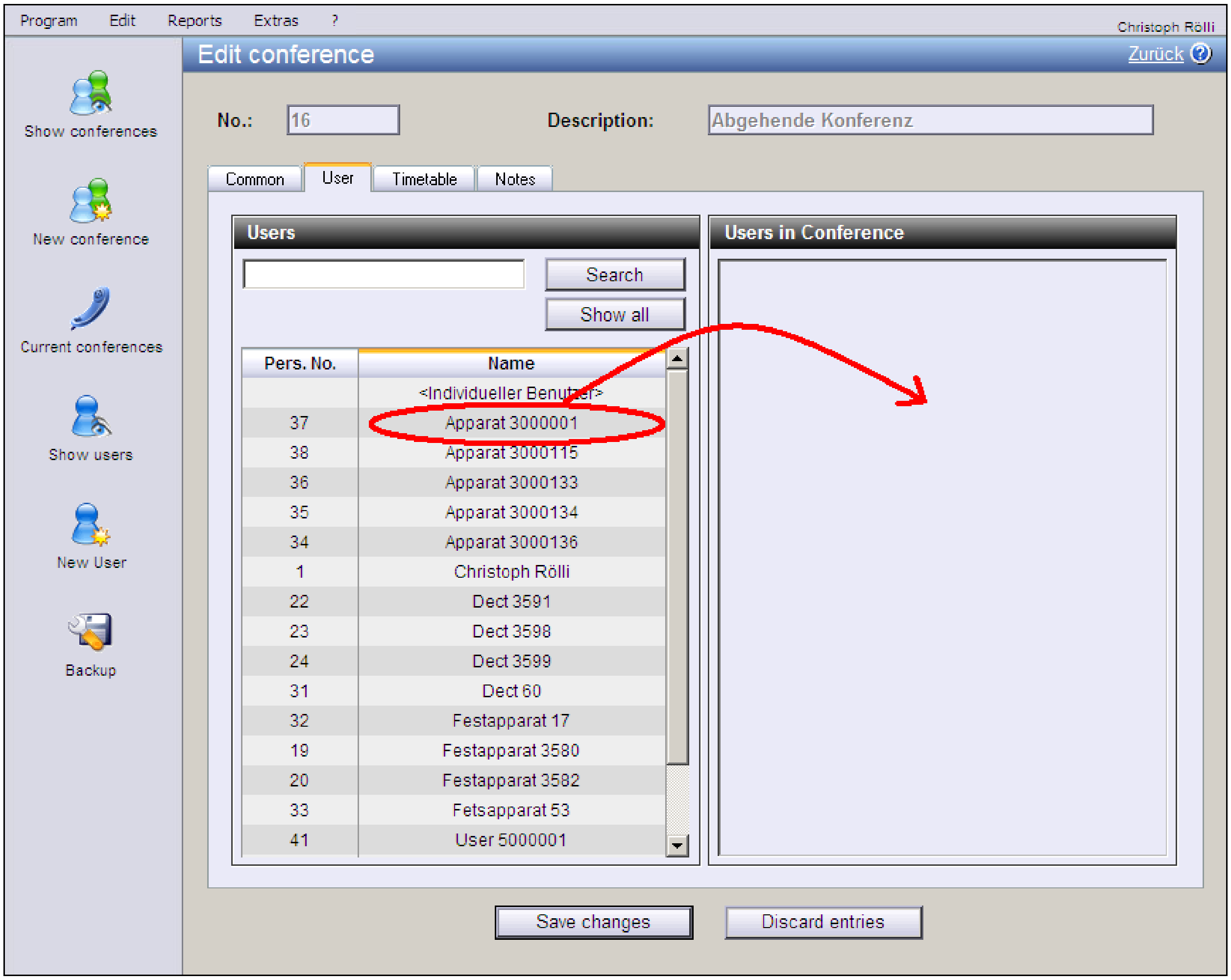

Figure 48: NovaConf User Allocation via Drag and Drop

A newly selected conference participant is removed from the list of “Users” and added to the list of “Users in Conference”. Repeat this operation for all users who are to participate in the conference.

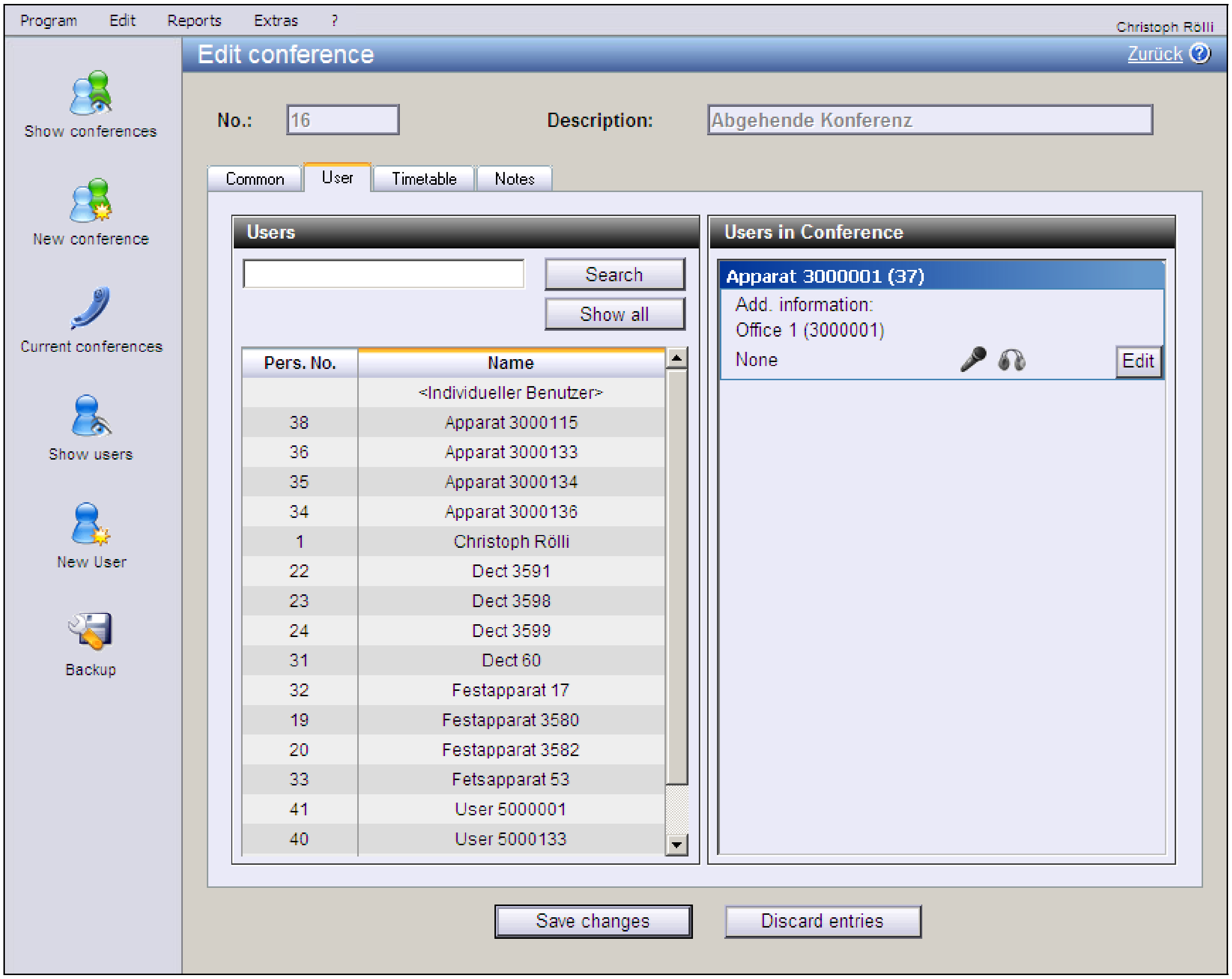

Figure 49: NovaConf Conference After Allocation of First Participant

Select the "Timetable" tab and configure the parameters in the "Next execution" area of this screen as shown in the following table. Click the "Save changes" button after completion.

Parameter	Usage
Date	Use the calendar icon to the right of this field to select a starting date for the conference.
Time	Enter the starting time for the conference.
Time to	Enter the ending time for the conference.
All	Select whether the conference shall be started repeatedly (e.g. Daily, Weekly and so on) in the drop down box on the right.
Days	Select the days on which the conference is to be established.
End-Date	Select the last date on which the conference is to be established. An empty parameter designates a single instance conference.

Table 15: NovaConf Conference Timetable Configuration Parameters

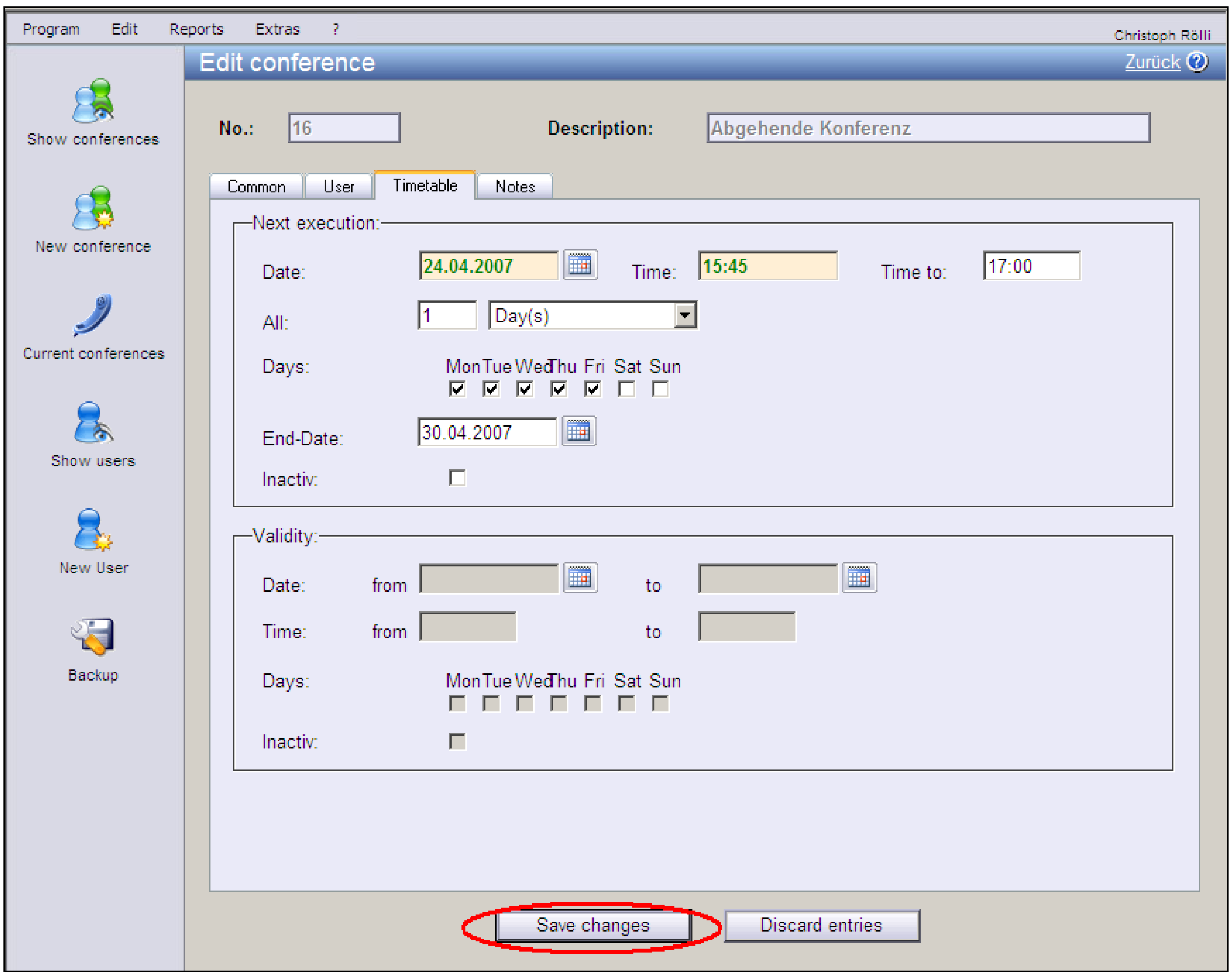

Figure 50: NovaConf Conference Timetable

4. Interoperability Compliance Testing

The interoperability compliance tests included feature and serviceability testing.

The feature testing focused on testing scenarios that involve interaction between the NovaConf server and Avaya products, including various sequences involving the following:

- Verification of the ability to establish conferences initiated by various Avaya telephones calling the NovaConf server.
- Verification of the ability of the NovaConf server to establish conferences by calling various Avaya telephones.
- Verification of the ability of the NovaConf server to establish conferences with parties that have activated call diversion. The conference should be established with the diverted-to station.
- Verification of the ability of NovaConf to recognize DTMF tones.
- Verification of the ability of Avaya telephones to correctly log unanswered conference calls.

The serviceability testing focused on verifying that the NovaConf product components can recover from interruption to interface connections that can occur during routine maintenance activities. The NovaConf server was also tested for recovery from unexpected power interruption.

4.1. General Test Approach

The test method employed can be described as follows:

- Correct interoperation between the NovaConf server and Avaya Communication Manager was verified by confirming that the various telephony operations that can be invoked by conferencing activity all function properly.
- NovaConf server robustness was tested by verifying its ability to recover from interruptions to its external connections via the LAN between the NovaConf and the network.
- Verifying the ability to recover from power interruptions to the NovaConf server further tested its robustness.

All testing was performed manually. The tests were all functional in nature, and no performance testing was done.

4.2. Test Results

The following was observed during testing:

- It is not possible for NovaConf to detect that an Avaya 4600 Series H.323 phone is disconnected, as Avaya Communication Manager does not report this status to the caller.

5. Verification Steps

The following steps can be performed to verify the basic operation of the various system components:

- Verify that Avaya Communication Manager and the NovaConf server can ping each other. The "ping" command can be executed from the NovaConf server by executing the "cmd" component via the run facility from the Windows "Start" control and entering "ping" followed by the IP address to which the ping message is to be sent. The "ping" command can be executed from Avaya Communication Manager via an SSH login session.
- Verify that the Avaya IP Telephones can call each other.
- From the Avaya Communication Manager SAT terminal, use the "status trunk" command to verify that the ports for the SIP trunk connected to NovaConf are in the "in-service/idle" state.
- Verify that each of the Avaya Telephones can call the extension allocated to NovaConf to participate in an incoming conference.
- Verify that it is possible for NovaConf to call each of the Avaya IP Telephones to participate in an outgoing conference.
- Verify that it is possible to navigate the NovaConf voice menu from each of the Avaya Telephones by calling the NovaConf extension, and entering key sequences in response to prompting requests from NovaConf.
- Verify the ability of Avaya Telephones to correctly log unanswered calls by initiating an unanswered conference call from NovaConf to each of the Avaya Telephones, verifying the name and number in the log of the telephone, and subsequently dialing the caller from the telephone log.
- From the Avaya SES Maintenance Web Interface, select the "Status Summary" screen and verify that the server is in "Active" mode, no alarms are being generated, the "Server Hardware" is "okay", and that server "Processes" are "okay".
- Verify that it is possible to place calls between SIP and H.323 telephones.

6. Support

Technical support from NovaLink can be obtained through the following:

NovaLink GmbH
Businesstower
Zuercherstrasse 310
8500 Frauenfeld
Switzerland
helpdesk@novalink.ch
Phone: +41 52 762 66 77
Fax: +41 52 762 66 99

7. Conclusion

These Application Notes describe the conformance testing of the NovaConf with Avaya Communication Manager. The various features of the NovaConf that involve its telephone interface were tested. A detailed description of the configuration required for both the Avaya and the NovaLink equipment is documented within these Application Notes. NovaConf passed all of the tests performed, which included both functional and robustness tests.

8. Additional References

[1] *Administrator Guide for Avaya Communication Manager*, February 2007, Issue 3, Document Number 03-300509

[2] *Feature Description and Implementation for Avaya Communication Manager*, February 2007, Issue 5, Document Number 555-245-205

[3] *Installing and Administering SIP Enablement Services,* March 2007, Issue 2.1, Document Number 03-600768

[4] *NovaConf 7.5 manual*, May 2007

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.