Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the compliance testing of the NovaLink NovaConf conference system with Avaya Communication Manager. These Application Notes contain an extensive description of the configurations for both NovaConf and Avaya Communication Manager.

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

1. Introduction

The purpose of this document is to describe the compliance testing done with 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 as 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 call 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 Integrated Services Digital Network (ISDN) Primary Rate (PRI) and Basic Rate (BRI) trunks described in these Application Notes.

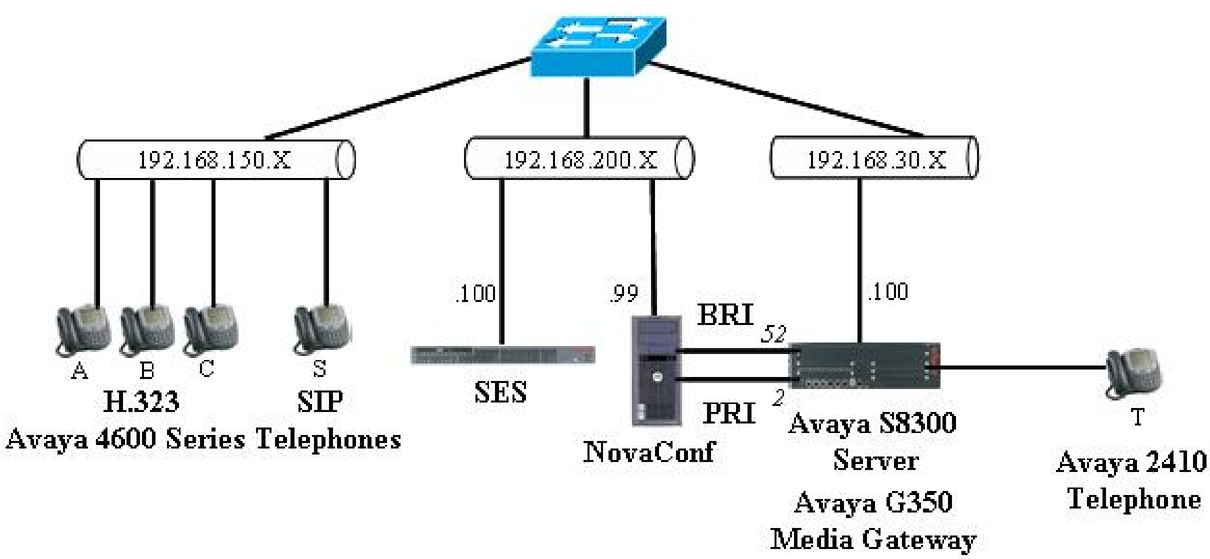

Figure 1: NovaConf Test Configuration

The numbers associated with the BRI (52) and PRI (2) trunks shown in the diagram are trunk numbers. 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 either a Basic Rate or Primary Rate ISDN interface between itself and the Avaya Media Gateway.
- 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 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
MM720AP BRI	HW05 FW007
MM712AP DCP	HW05 FW008
MM710AP DS1	HW05 FW018
Avaya 4600 series H.323 stations	2.8
Avaya 4600 series SIP stations	2.2.2
NovaLink NovaConf	7.5 SP 1A
Gerdes Primux 1S2M II / 4S0 II	3.6.4695
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
2000000	NovaConf via PRI
5200000	NovaConf via BRI

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 BRI and PRI interfaces that are 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.

```
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, the parameter must be set as show in the following table.

Parameter	Required Setting	Comment
IP Stations	y	This is required so that IP stations can be configured
Enhanced EC500	y	This is required enable the allocation of off-PBX SIP telephones
ISDN-BRI Trunks	y	This is required allow the allocation of the BRI trunk to be attached to NovaConf.
ISDN-PRI	y	This is required to allow the allocation of the PRI trunk to be attached to NovaConf.

Table 3: DS1 Parameters for PRI Interface

```
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 8, the "Value-Added (VALU)?" parameter must be set to "y" to enable QSIG features required by NovaConf.

```
display system-parameters customer-options                      Page   8 of  10
                             QSIG OPTIONAL FEATURES

                                         Basic Call Setup? y
                             Basic Supplementary Services? y
                                    Centralized Attendant? n
                                      Interworking with DCS? n
                    Supplementary Services with Rerouting? y
                            Transfer into QSIG Voice Mail? n
                                       Value-Added (VALU)? y
```

Figure 6: System-Parameters Customers-Options Form, Page 8

3.1.2. Configure Node Names

Use the **change node-names ip** command to configure the IP address of the NovaConf and the SES servers.

```
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 PRI Interface to the NovaConf Server

Use the **add ds1 <media module hardware address>** command to configure the DS1 interface card to serve as a Primary Rate ISDN interface. Assign those values for this command as shown in the following table.

Parameter	Usage
Bit Rate	Assign the bit rate to “2.048”, as required to connect to the NovaConf E1 interface card.
Line Coding	Assign the bit rate to "hdb3", as required to connect to the NovaConf E1 interface card.
Name	Assign a name to be used to identify the card.
Signaling Mode	Assign the signaling mode to “isdn-pri”.
Connect	Specify the connection is to a “pbx”
Interface	Specify the G350 is to serve as the “peer-master”.
Peer Protocol	Specify the QSIG protocol is to be used.
Interface Companding	Specify “a-law speech encoding is to be used.
CRC?	Specify a cyclic-redundancy-check sequence is to be sent with data frames to verify correct transmission.
Idle Code	Specify that an idle sequence of “11111111” is to be sent on the interface when no data is being transmitted.

Table 4: DS1 Parameters for PRI Interface

```
add ds1 1v5                                                   Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 001V5                              Name: QSIG-PRI
            Bit Rate: 2.048                       Line Coding: hdb3

      Signaling Mode: isdn-pri
             Connect: pbx                           Interface: peer-master
   TN-C7 Long Timers? n                         Peer Protocol: Q-SIG
Interworking Message: PROGress                           Side: a
Interface Companding: alaw                               CRC? y
           Idle Code: 11111111          Channel Numbering: sequential
                              DCP/Analog Bearer Capability: 3.1kHz

                                            T303 Timer(sec): 4

      Slip Detection? n                Near-end CSU Type: other
```

Figure 8: Ds1 Form for PRI Interface

Use the **add trunk-group** command to configure the Trunk Group to the NovaConf Server. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type	Specify the Group Type as “isdn”
Group Name	Select an appropriate name to identify the device.
TAC	Specify a trunk access code that can be used to provide dial access to the trunk.
Carrier Medium	Specify a Carrier Medium of “PRI/BRI”, as PRI will be used for this trunk.
Dial Access	Allow dial access to the trunk by dialing the trunk access code.
Service Type	Designate the trunk as a “tie” line to a peer system.
Supplementary Service Protocol	Specify a Supplementary Service Protocol of “b” for QSIG.
Digit Handling	Specify “overlap/overlap” to allow overlap sending of dialed digits.
Format (page 2)	Specify “unk-unk” to use unknown dialing plan for calls in both directions.
Send Name	Specify “y” so that the name of the caller is sent for outgoing calls.
Send Calling Number	Specify “y” so that the number of the caller is sent for outgoing calls.
Format (page 3)	Specify “unknown” to use unknown dialing plan for both for calls in both directions.
Send Connected Number	Specify “y” so that the number of the connected party is sent to the caller.
Group Member Assignments	Assign the interface ports on the MM710AP to the trunk group members. Note that port 16 is used for the D channel, which must be assigned to the signaling group associated with this trunk.

Table 5: Trunk-Group Parameters for PRI Interface

```
add trunk-group 2                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 2                   Group Type: isdn          CDR Reports: y
  Group Name: NOVA S2M QSIG              COR: 1        TN: 1        TAC: *02
   Direction: two-way        Outgoing Display? n         Carrier Medium: PRI/BRI
 Dial Access? y              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

Figure 9: Trunk-Group Form for PRI Interface, Page 1

```
add trunk-group 2                                               Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: b     Digit Handling (in/out): overlap/overlap
       Digit Treatment:                                        Digits:
            Trunk Hunt: ascend
                                                    Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                Format: unk-unk
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 0
          Administer Timers? n
```

Figure 10: Trunk-Group Form for PRI Interface, Page 2

```
add trunk-group 2                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n             Hop Dgt? n     Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: unkknown
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: y
                                                   Hold/Unhold Notifications? y
             Send UUI IE? y                    Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                          Ds1 Echo Cancellation? n

    Apply Local Ringback? n
 Show ANSWERED BY on Display? y
                             Network (Japan) Needs Connect Before Disconnect? n
```

Figure 11: Trunk-Group Form for PRI Interface, Page 3

```
add trunk-group 2                                               Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/30
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  29

       Port    Code Sfx Name        Night           Sig Grp
  1: 001V501  MM710                                   2
  2: 001V502  MM710                                   2
  3: 001V503  MM710                                   2
  4: 001V504  MM710                                   2
  5: 001V505  MM710                                   2
  6: 001V506  MM710                                   2
  7: 001V507  MM710                                   2
  8: 001V508  MM710                                   2
  9: 001V509  MM710                                   2
 10: 001V510  MM710                                   2
 11: 001V511  MM710                                   2
 12: 001V512  MM710                                   2
 13: 001V513  MM710                                   2
 14: 001V514  MM710                                   2
 15: 001V515  MM710                                   2
```

Figure 12: Trunk-Group Form for PRI Interface, Page 5

```
add trunk-group 2                                              Page   6 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/30
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  30

       Port    Code Sfx Name        Night           Sig Grp
 16: 001V517  MM710                                   2
 17: 001V518  MM710                                   2
 18: 001V519  MM710                                   2
 19: 001V520  MM710                                   2
 20: 001V521  MM710                                   2
 21: 001V522  MM710                                   2
 22: 001V523  MM710                                   2
 23: 001V524  MM710                                   2
 24: 001V525  MM710                                   2
 25: 001V526  MM710                                   2
 26: 001V527  MM710                                   2
 27: 001V528  MM710                                   2
 28: 001V529  MM710                                   2
 29: 001V530  MM710                                   2
 30: 001V531  MM710                                   2
```

Figure 13: Trunk-Troup Form for PRI Interface, Page 6

Use the **add signaling-group** command to allocate a signaling group to this trunk.

Parameter	Usage
Group Type	Specify "isdn-pri" for ISDN primary rate.
D-Channel	Assign port 16 of the DS1 interface as the D channel.
Trunk Group for Channel Selection	Specify "2" as the Trunk Group to be used for channel selection.
TSC Supplementary Service Protocol	Specify "b" to designate use of the QSIG protocol.
Max number of NCA TSC	Specify "4" to allow NovaConf to control the message waiting lamp of Avaya Telephones.

Table 6: Signaling-Group Parameters for PRI Interface

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

 Group Number: 2              Group Type: isdn-pri
                   Associated Signaling? y          Max number of NCA TSC: 4
                      Primary D-Channel: 001V516     Max number of CA TSC: 0
                                                  Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 2
      TSC Supplementary Service Protocol: b
```

Figure 14: Signaling-Group Form for PRI Interface

3.1.4. Configure Dial Plan and Call Routing

Use the **change dialplan analysis** command to specify that dialed strings which begin with "2"or "3" 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 "2". The dial string "*02" is used as a trunk access code to access the NovaConf trunk.

```
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
        2          7   ext
        3          7   ext
        52         7   ext
        *02        3   dac
        *52        3   dac
        *83        3   dac
```

Figure 15: Dialplan Analysis Form

Use the **change uniform-dialplan** command to designate extensions which begin with “2” or “52” as aar numbers.

```
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
  2                7   0                      aar  n
  52               7   0                      aar  n
```

Figure 16: Uniform-Dialplan Form

Use the **change aar analysis** command to select routing pattern 2 for numbers which have the leading dialed string “2”, as specified in the uniform dial plan shown in **Figure 16**.

```
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
    2                     7    7     2         aar           n
    52                    7    7     52        aar           n
```

Figure 17: Aar Analysis Form

Use the **change route-pattern** command to route numbers using Routing Pattern 2 via Trunk Group 2.

```
change route-pattern 2                                           Page   1 of   3
                    Pattern Number: 2   Pattern Name: NovaConf PRI
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 2    0                                                              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 18: Route-Pattern 2 Form

Use the **change route-pattern** command to dial numbers for routing pattern 52 via trunk 52, as shown in **Figure 19.**

```
change route-pattern 52                                          Page   1 of   3
                    Pattern Number: 52  Pattern Name: NovaConf BRI
                             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: 52    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 19: Route-Pattern From for BRI Interface

3.1.5. Configure BRI Interface to the NovaConf Server

Use the **add bri-trunk-board** command to configure port 1 of the MM720 interface card to serve as a basic rate interface. Assign those values for this command as shown in the following table.

Parameter	Usage
Termination Type	Set this to "NT".
Interface	Set this to "peer-master".
Max NCA TSC	Set this to 4 to provide control of the message waiting light from NovaConf.

Table 7: Parameters for BRI-Trunk-Board

```
add bri-trunk-board 1v2                                         Page   1 of   2
                          ISDN-BRI TRUNK CIRCUIT PACK

               Location: 001V2                            Name: S0 TRUNKS
   Interface Companding: a-law    DCP/Analog Bearer Capability: 3.1kHz
  T3 Timer Length (sec): 15                   Termination Type: NT
 Port  Interface   Side  Cntry/Peer  TEI                         Layer 1 Detect
                          Protocol                               Stable? Slips?
   1: peer-master   a       QSIG     0                              y      n
   2:                                0                              y      n
   3:                                0                              y      n
   4:                                0                              y      n
   5:                                0                              y      n
   6:                                0                              y      n
   7:                                0                              y      n
   8:                                0                              y      n
```

Figure 20: BRI-Trunk-Board Form for BRI Interface, page 1

```
add bri-trunk-board 1v2                                      Page   2 of   2
                          ISDN-BRI TRUNK CIRCUIT PACK

Port  Interwork  XID   Endpt      SPID       Endpt      SPID      Endpt    Max
       Message  Test?  Init?                  ID                   ID    NCA TSC
  1:  PROGress    n      n                                                  4
  2:  PROGress    n      n                                                  0
  3:  PROGress    n      n                                                  0
  4:  PROGress    n      n                                                  0
  5:  PROGress    n      n                                                  0
  6:  PROGress    n      n                                                  0
  7:  PROGress    n      n                                                  0
  8:  PROGress    n      n                                                  0

  Port Directory    Directory           Port Directory    Directory
         Number       Number                   Number       Number
    1:                                    5:
    2:                                    6:
    3:                                    7:
    4:                                    8:
```

Figure 21: BRI-Trunk-Board Form for BRI Interface, page 2

Use the **add trunk-group** command to configure the MM720AP interface card to serve as basic rate interface. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type	Specify the Group Type as “isdn”
Group Name	Select an appropriate name to identify the device.
TAC	Specify a trunk access code that can be used to provide dial access to the trunk. This dial string must be contained in the dial plan specified in **Figure 9**.
Carrier Medium	Specify a Carrier Medium of “PRI/BRI”, as BRI will be used for this trunk.
Dial Access	Allow dial access to the trunk by dialing the trunk access code.
Service Type	Designate the trunk as a “tie” line to a peer system.
Supplementary Service Protocol	Specify a Supplementary Service Protocol of “b” for QSIG.
Digit Handling (in/out)	Specify “overlap/overlap” to allow overlap sending of dialed digits.
Send Name	Specify “y” so that the name of the caller is sent for outgoing calls.
Send Caller Number	Specify “y” so that the number of the caller is sent for outgoing calls.
Format	Specify “unknown” to use unknown dialing plan for calls in both directions.
Send Connected Number	Specify “y” so that the number of the connected party is sent to the caller.
QSIG Value-Added	Specify “y” to allow NovaConf to control the message waiting lamp of the Avaya Telephones.
Group Member Assignments	Assign the interface ports on the MM720AP to the trunk group members.

Table 8: Parameters BRI Trunk Group

```
add trunk-group 52                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 52                   Group Type: isdn          CDR Reports: y
  Group Name: BRI QSIG                    COR: 1        TN: 1        TAC: *52
   Direction: two-way       Outgoing Display? n         Carrier Medium: PRI/BRI
 Dial Access? y             Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n           TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

Figure 22: Trunk-Group Form, Page 1

```
add trunk-group 52                                              Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: end-on-request
  Supplementary Service Protocol: b     Digit Handling (in/out): overlap/overlap
       Digit Treatment:                                        Digits:
            Trunk Hunt: cyclical
                                                    Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 0
          Administer Timers? n
```

Figure 23: Trunk-Group Form, Page 2

```
add trunk-group 52                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n             Hop Dgt? n     Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: unknown
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

           Decimal Point: period                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: y
                                                  Hold/Unhold Notifications? y
             Send UUI IE? n                    Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                         Ds1 Echo Cancellation? n

    Apply Local Ringback? n
 Show ANSWERED BY on Display? y
                             Network (Japan) Needs Connect Before Disconnect? n
```

Figure 24: Trunk-Group Form, Page 3

```
add trunk-group 52                                           Page   4 of  21
                       QSIG TRUNK GROUP OPTIONS




   TSC Method for Auto Callback: drop-if-possible
           Diversion by Reroute? y
               Path Replacement? y
Path Replacement with Retention? n
        Path Replacement Method: better-route
                            SBS? n
  Display Forwarding Party Name? y
    Character Set for QSIG Name: eurofont
               QSIG Value-Added? y
                Encoding Method: proprietary
```

Figure 25: Trunk-Group Form, Page 4

```
add trunk-group 52                                            Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/2
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   2

       Port    Code Sfx Name        Night           Sig Grp
  1: 001V201  MM720
  2: 001V217  MM720
  3:
  4:
  5:
  6:
  7:
  8:
  9:
 10:
 11:
 12:
 13:
 14:
 15:
```

Figure 26: Trunk-Group Form for BRI Interface, Page 5

3.1.6. 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 "3" for trunks "2" and "52" 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  3             2                         7           Maximum Entries: 240
 7  3             52                        7
```

Figure 27: Public-Unknown-Numbering Form

3.1.7. 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 phone, 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.

Table 9: 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 28: 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 29: Add Station Form, Page 4

3.1.8. Configure Interface to 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 phone.
Application	Enter “OPS”.
Phone Number	Enter the phone number assigned to the SIP phone.
Trunk Selection	Enter the number assigned to the SIP trunk in **Section 3.1.8**.
Call Limit	Enter “3” to allow the SIP phone to do call transfers.

Table 10: 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 30: 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 31: Off-PBX-Telephone Form, page 2

Use the **change off-pbx-telephone feature-name-extension** 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 phones, as shown in **Table 17**.

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 11: 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: 300-1805
          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 32: Off-PBX-Telephone Form, page 2

```
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 33: Off-PBX-Telephone Form, page 2

Use the **change feature-access-codes** command to assign feature codes required by SIP phones, 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 12: Parameters for Off-PBX-Telephone Feature-Name-Extension

```
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 34: Off-PBX-Telephone Form, page 2

```
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 35: Off-PBX-Telephone Form, page 2

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

Parameter	Usage
Authoritative Domain	Enter the name assigned to SES in **Figure 48**.
Name	Enter a descriptive name.

Table 13: 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 36: IP-Network-Region Form

Use the **change ip-codec-set** command to specify the codec to be used for Network Region assigned to SES. Specify that the G.711 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 37: IP-Codec-Set Form

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 SES in **Figure 7.**
Far-end Domain	Enter the domain name configured for SES in **Figure 48**.

Table 14: 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 38: SES Signaling-Group Form

Use the **add trunk-group** command to configure the MM720 interface card to serve as a primary rate interface. 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.
Service Type	Designate the trunk as a "tie" line to a peer system.
Signaling Group	Enter the number assigned to the SIP signaling group show in **Figure 38**.
Number of Members	Specify sufficient number of members to support the maximum simultaneous connections required.

Table 15: 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 39: Trunk-Group Screen Form

3.2. Configure Avaya IP Telephones

Configure **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 phones should register with the SES server to receive message waiting events.
SIPDOMAIN	Enter the name of the SIP domain.
ENHDIALSTAT	Set this parameter to "0" to indicate that enhanced dialing is not required.

Table 16: Parameters for Telephone Setting File

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

Figure 40: 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 11**.

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 17: Speed Dial Entry Assignments for Avaya SIP Telephones

3.3. Configure SIP Enablement Services

Avaya SES is needed in this configuration only if Avaya SIP IP Telephones are used. Log in to the SES Web-based Integrated Management tool by selecting the IP address of the SES server followed by "/admin" from the Web browser. After entering the login ID and password, select "Launch Administration Web Interface".

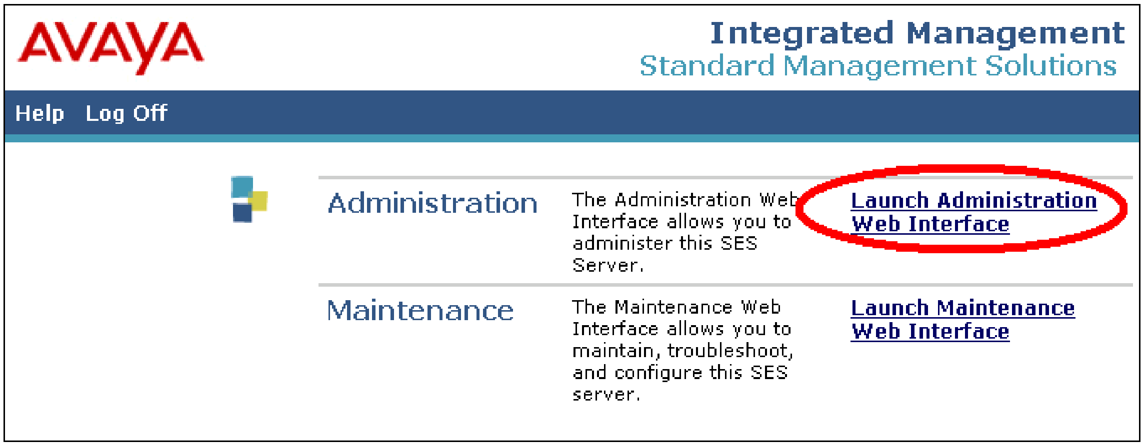

Figure 41: SES Initial Greeting Screen

From the top-level management screen, click "Manage Hosts" followed by "Add Host".

Figure 42: Host Management Selection from Top-Level Administration Screen

Enter the IP address of the SES Server, a database password, and a Profile Service Password that were allocated to the SES server when it was installed. Leave the other field assigned to their respective default values. Select the “Update” button.

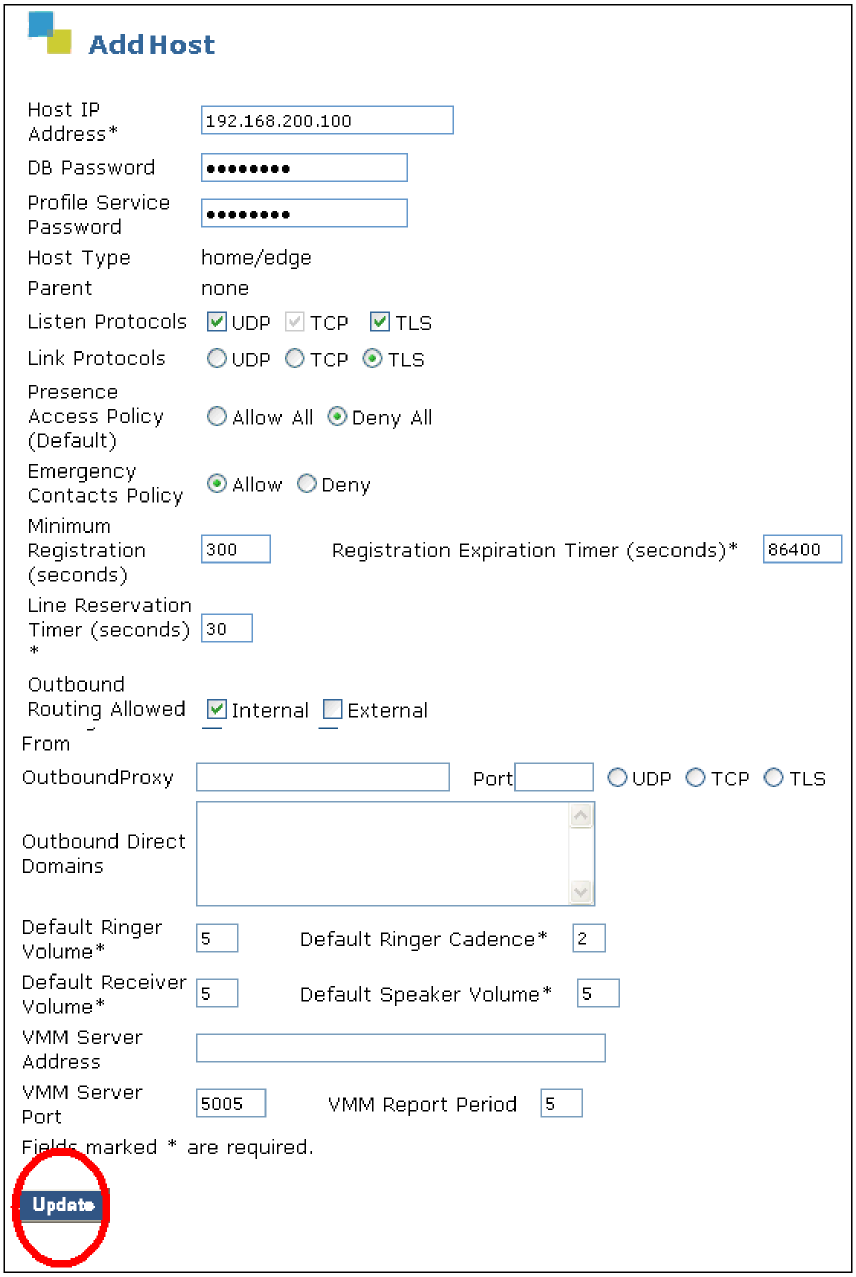

Figure 43: SES “Add Host” Screen

From the top-level management screen, select “Manage Media Servers”.

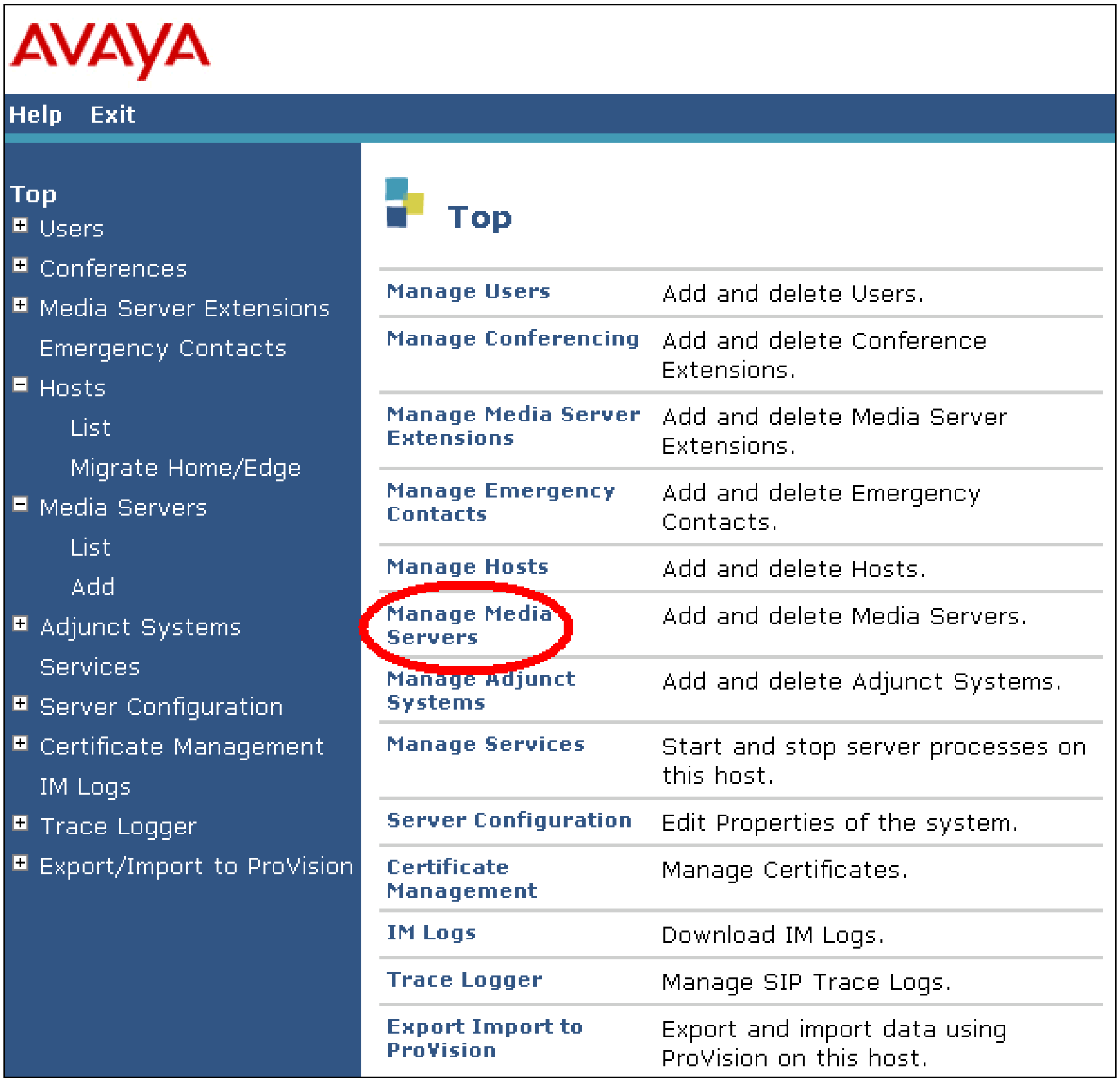

Figure 44: Media Server Management Selection from Top-Level Administration Screen

Assign a meaningful name to the "Media Server Interface Name". Select the IP address of the 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.

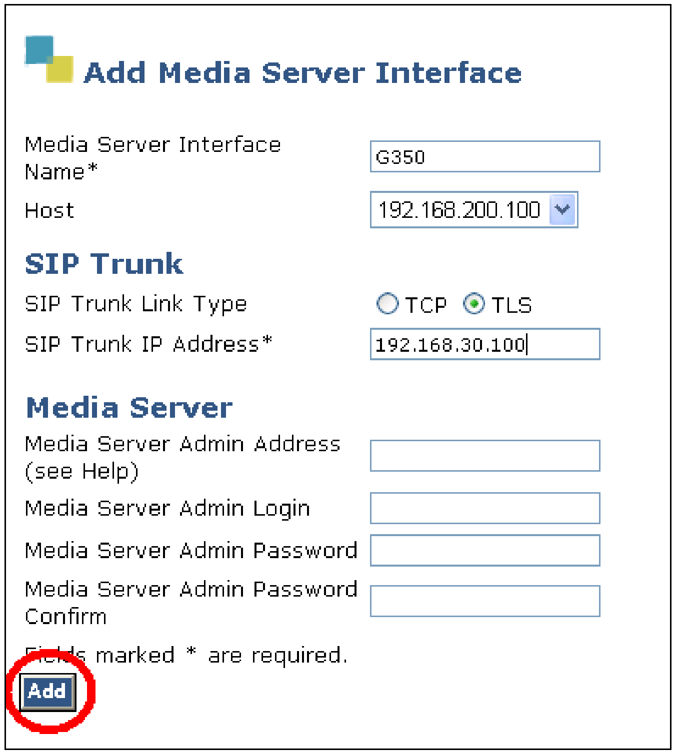

Figure 45: SES Add Media Server Interface Screen

From the top-level management screen, select “Server Configuration”.

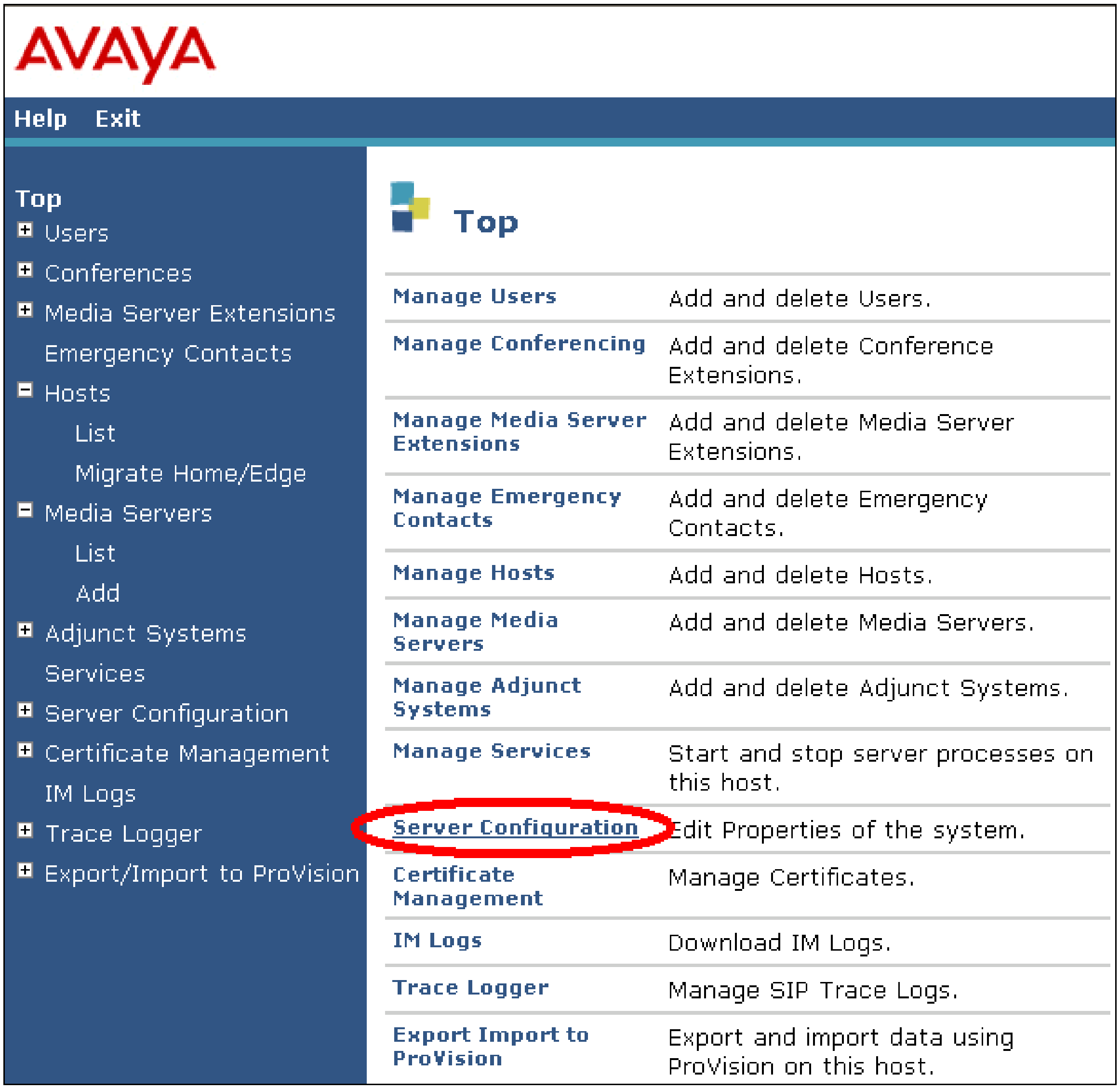

Figure 46: Server Configuration Selection from Top-Level Administration Screen

From the Server Configuration screen, select “System Properties”.

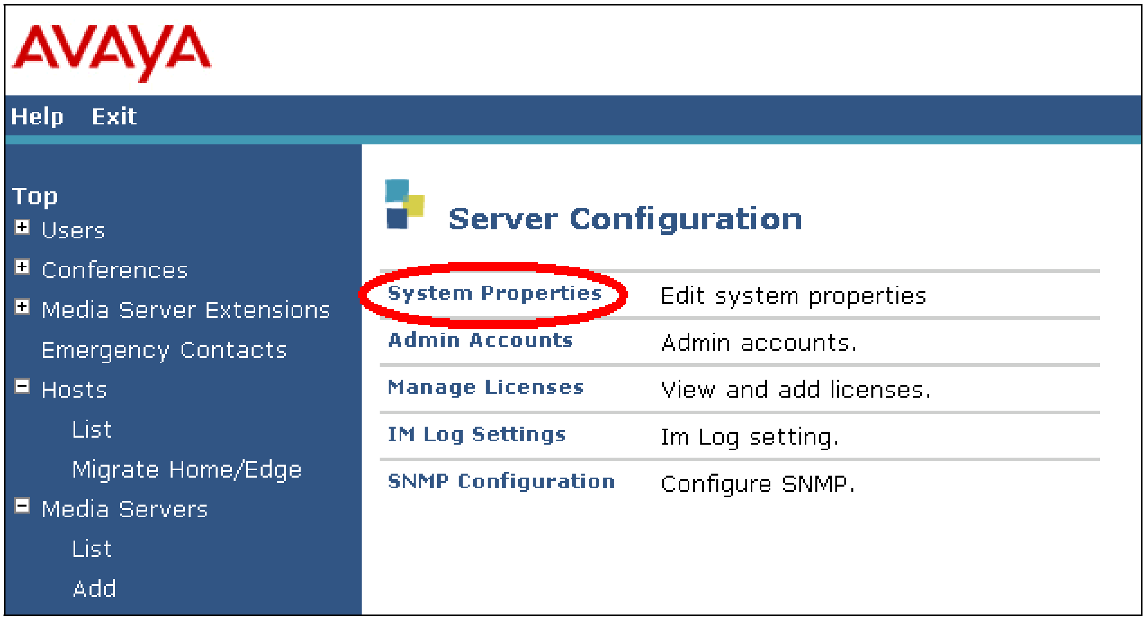

Figure 47: System Properties Selection from Server Configuration Screen

Enter the name to be assigned to the "SIP Domain". This must be the same name as is assigned in **Figure 36** and **Figure 38**. Select the "Update" button.

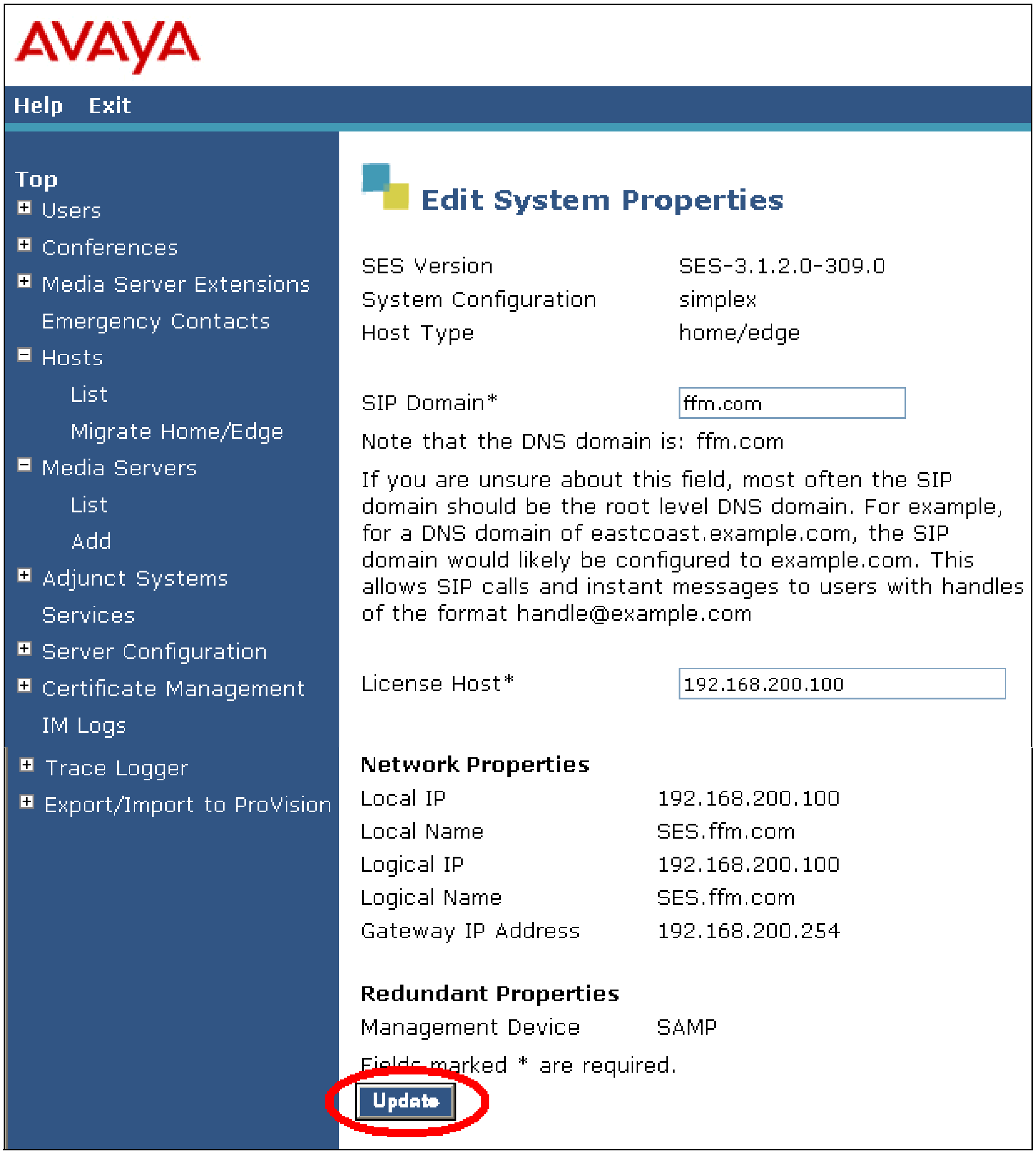

Figure 48: SES Edit System Properties Screen

From the top-level management screen, select “Manage Users”.

Figure 49: User Management Selection from Top-Level Administration Screen

Select “Add User”.

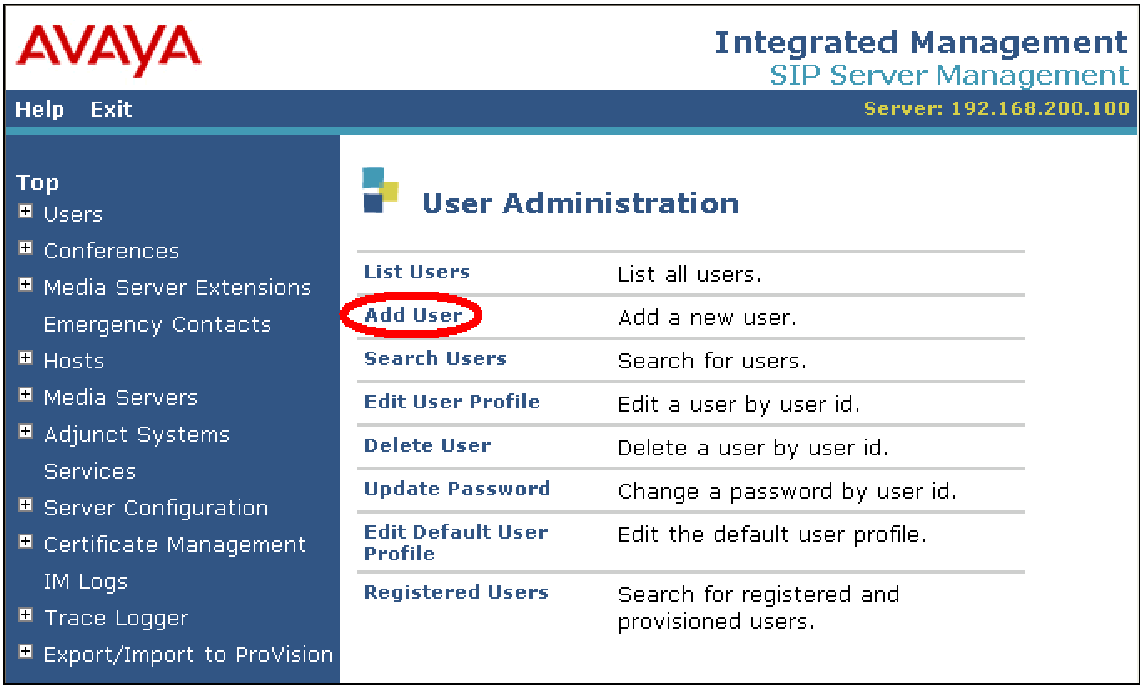

Figure 50: SES User Administration Screen

Enter the extension of the user to be added as the "Primary Handle". This is the same extension that was configured in **Section 3.1.8**. Enter a password and first/last name of the user, check the "Add Media Server Extension" box, and click "Add".

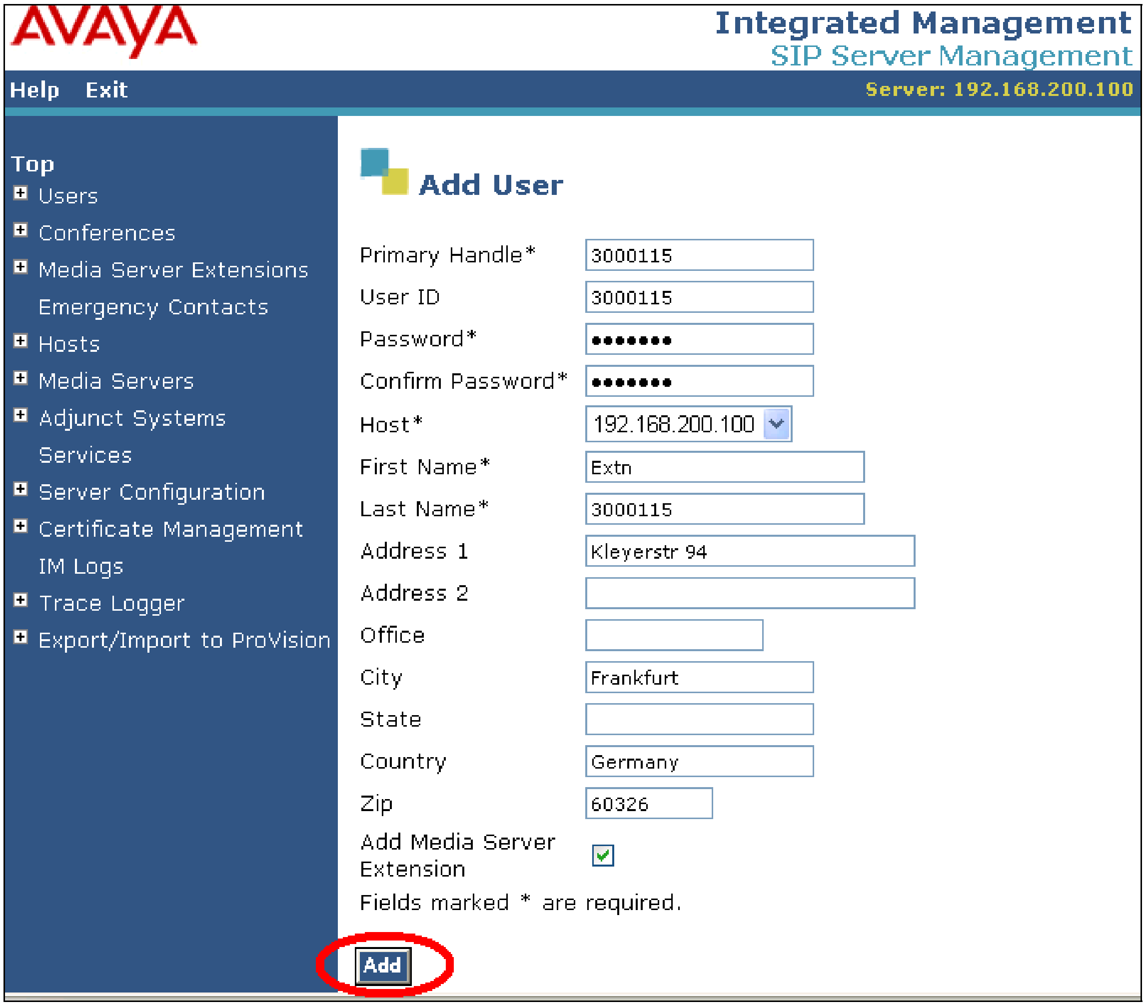

Figure 51: SES "Add User" Screen

Enter the Media Server Extension for the User ID 3000115 (the extension of the SIP telephone). Select the Media Server (refer to **Figure 45**) and drop down box and click “Add” to continue.

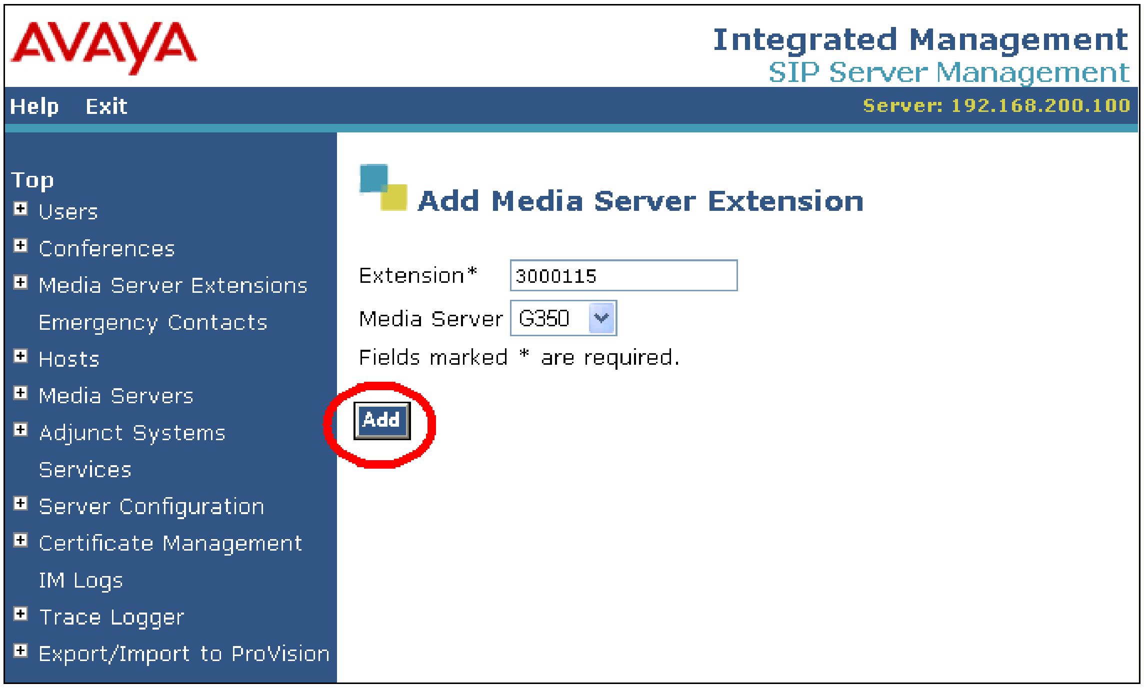

Figure 52: SES Add Media Server Extension Screen

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 "CardDriver" value must be set to 2 for CAPI cards.
The "Interface" value should be 2 for PRI and 3 for BRI.

The “DefaultCallingParty” and “TelNrLinie” number should be configured to lie within the dialing plan and be chosen such that calls originating from Avaya Communication Manager are routed to the trunk used to connect to NovaConf.

For PRI testing a value of “2000000” was used, and for BRI testing a value of “5200000”.

The other parameters in this file should be configured as shown.

```
[CallInfo]
CardDriver=2
Interface=2
GewählteNummer=1
MinDigits=0
AufschaltenAktiv=0
CallingPartyAktiv=1
DefaultCallingParty=2000000
DefaultLocalName=NovaConf
CNIPAktiv=1
QSIGStandard=2

[Watchdog]
TelNrLinie=2000000
```

Figure 53: NovaConf.ini Configuration File Content

3.4.2. Configure Interface to Avaya Communication Manager

Use the Windows “Start” button to select the program Primux ISDN / CAPI Configuration. If the BRI interface is used, the “PrimuX 450 II” icon should be selected. If the PRI interface is used, the “PrimuX 1S2M II” icon should be selected.

3.4.2.1 Configure PRI Interface

Set the parameters in the “General” tab as show in the following table.

Parameter	Usage
Switch Type	Specify “PBX, Q.SIG (experimental)”
Interface Type	Specify “Point-to-Point”
Inbound calls	Specify “No Phone Numbers”

Table 18: ISDN PRI Interface General Configuration Parameters

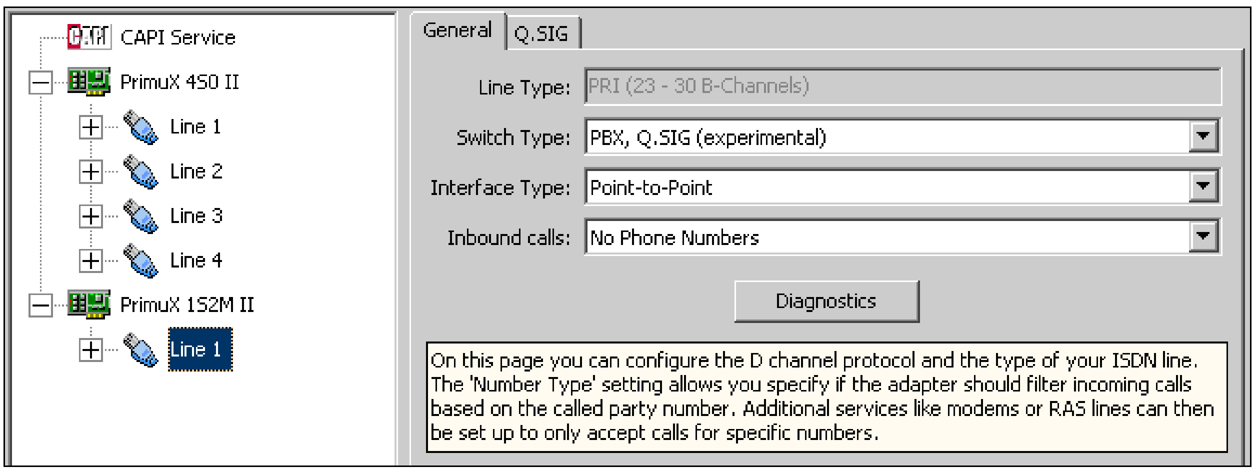

Figure 54: ISDN PRI Interface General Configuration Settings

Configure the parameters in the “Q.SIG” tab as shown in the following table.

Parameter	Usage
PBX type	Specify “Universal”
Q.SIG Standard	Specify “Automatic”
Length of CR Value	Specify “Default”
Length of Channel Info IE	Specify “Continuous Number”
Call Transfer Mode	Specify “Automatic”
Disconnect on PROGRESS	Specify “Off”
Process Interpretation APDU	Specify “Off”

Table 19: ISDN PRI Interface Q.SIG Configuration Parameters

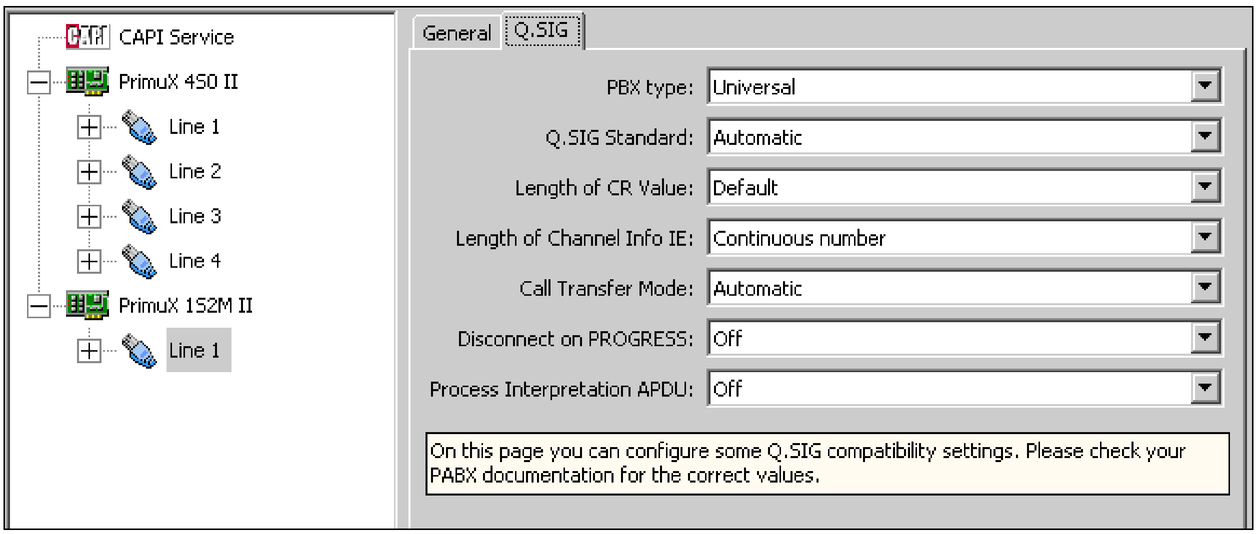

Figure 55: ISDN PRI Interface Q.SIG Configuration Settings

3.4.2.2 Configure BRI Interface

Set the parameters in the “General” tab as show in the following table.

Parameter	Usage
Switch Type	Specify “PBX, Q.SIG (experimental)”
Interface Type	Specify “Point-to-Point“
Inbound calls	Specify “No Phone Numbers”

Table 20: ISDN BRI Interface General Configuration Parameters

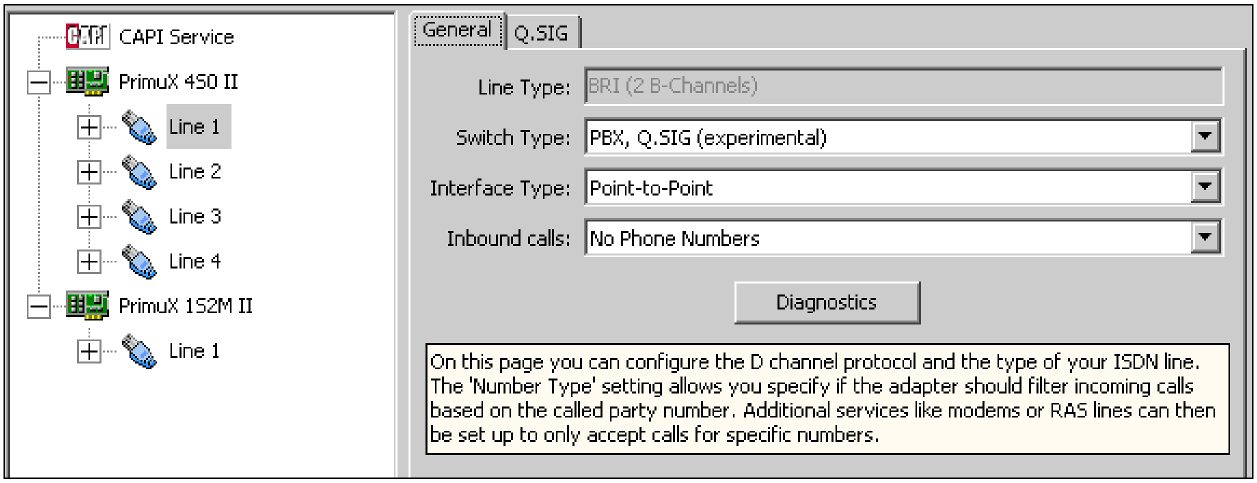

Figure 56: ISDN BRI Interface General Configuration Settings

Configure the parameters in the Q.SIG tab as shown in the following table.

Parameter	Usage
PBX type	Specify “Universal”
Q.SIG Standard	Specify “Automatic”
Length of CR Value	Specify “Default”
Length of Channel Info IE	Specify “Continuous Number”
Call transfer mode	Specify “Automatic”
Disconnect on PROGRESS	Specify “Off”
Process Interpretation APDU	Specify “Off”

Table 21: ISDN BRI Interface Q.SIG Configuration Parameters

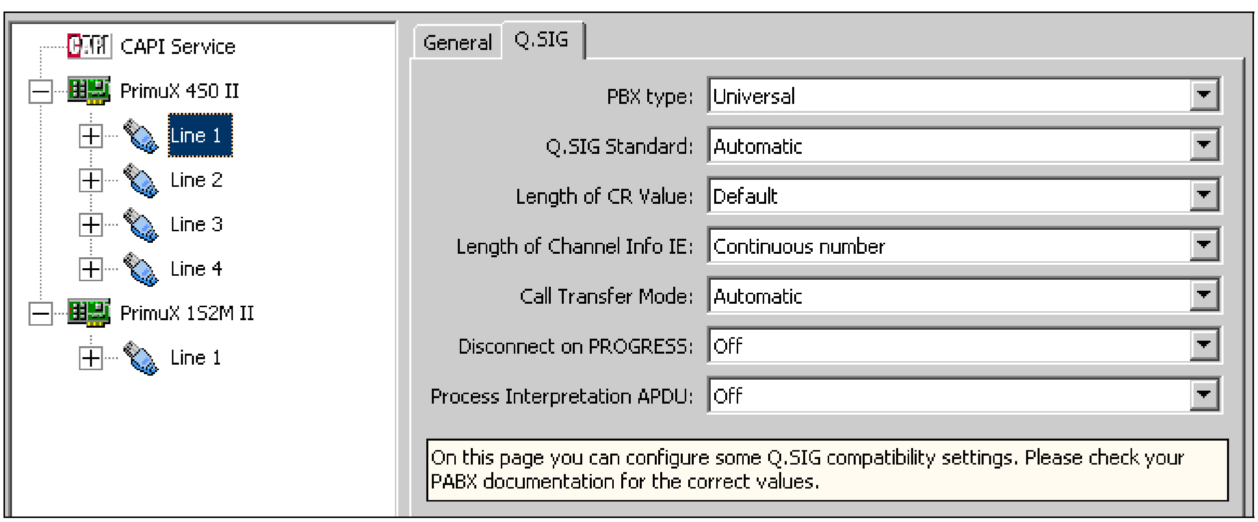

Figure 57: ISDN BRI Interface Q.SIG Configuration Settings

3.4.3. 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.

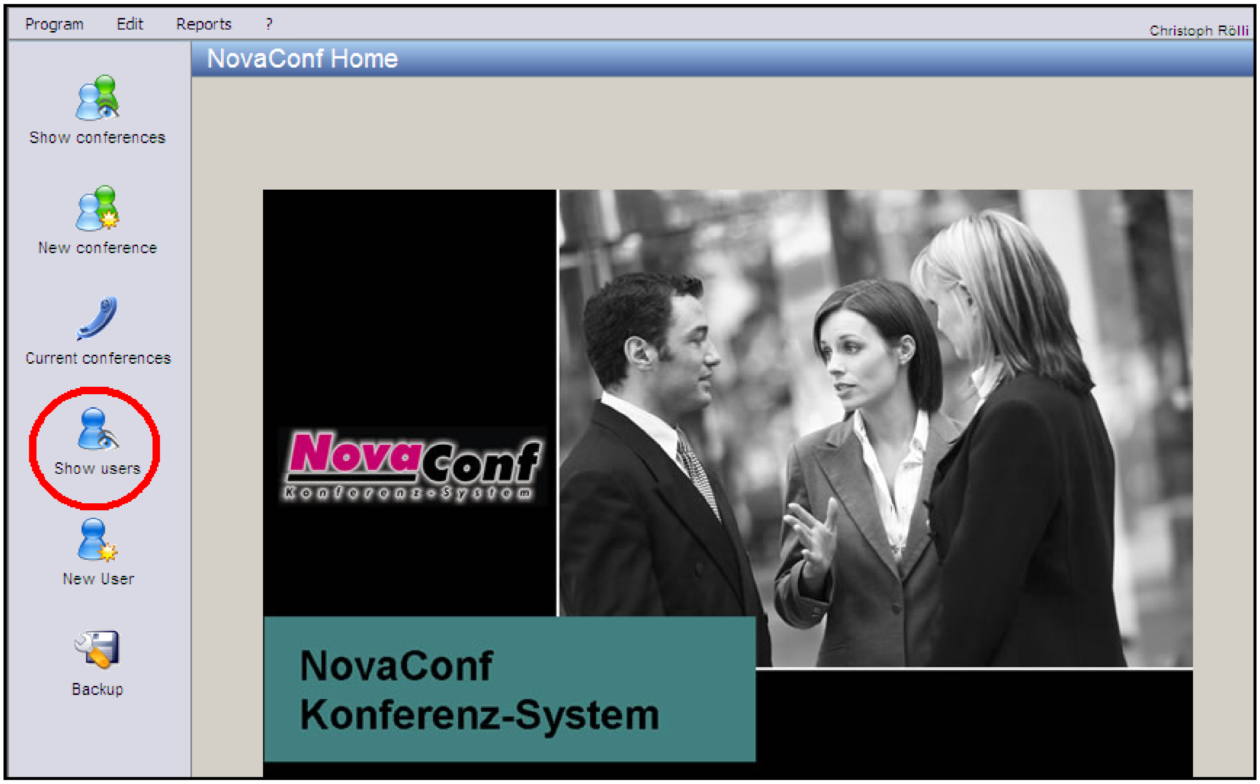

Figure 58: NovaConf Startup Screen

3.4.3.1 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**.

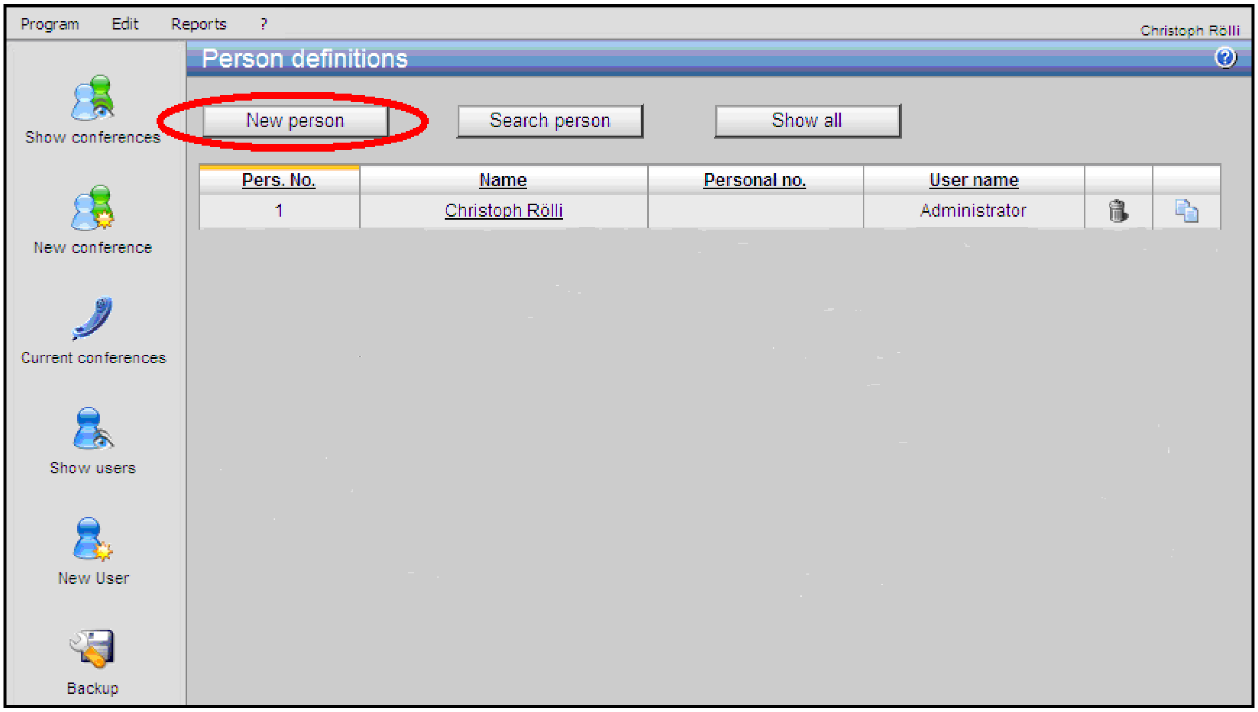

Figure 59: NovaConf User Configuration Screen

In the “Personal details” tab, enter the user’s name in the “Name” field (name not entered in **Figure 60**) 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.

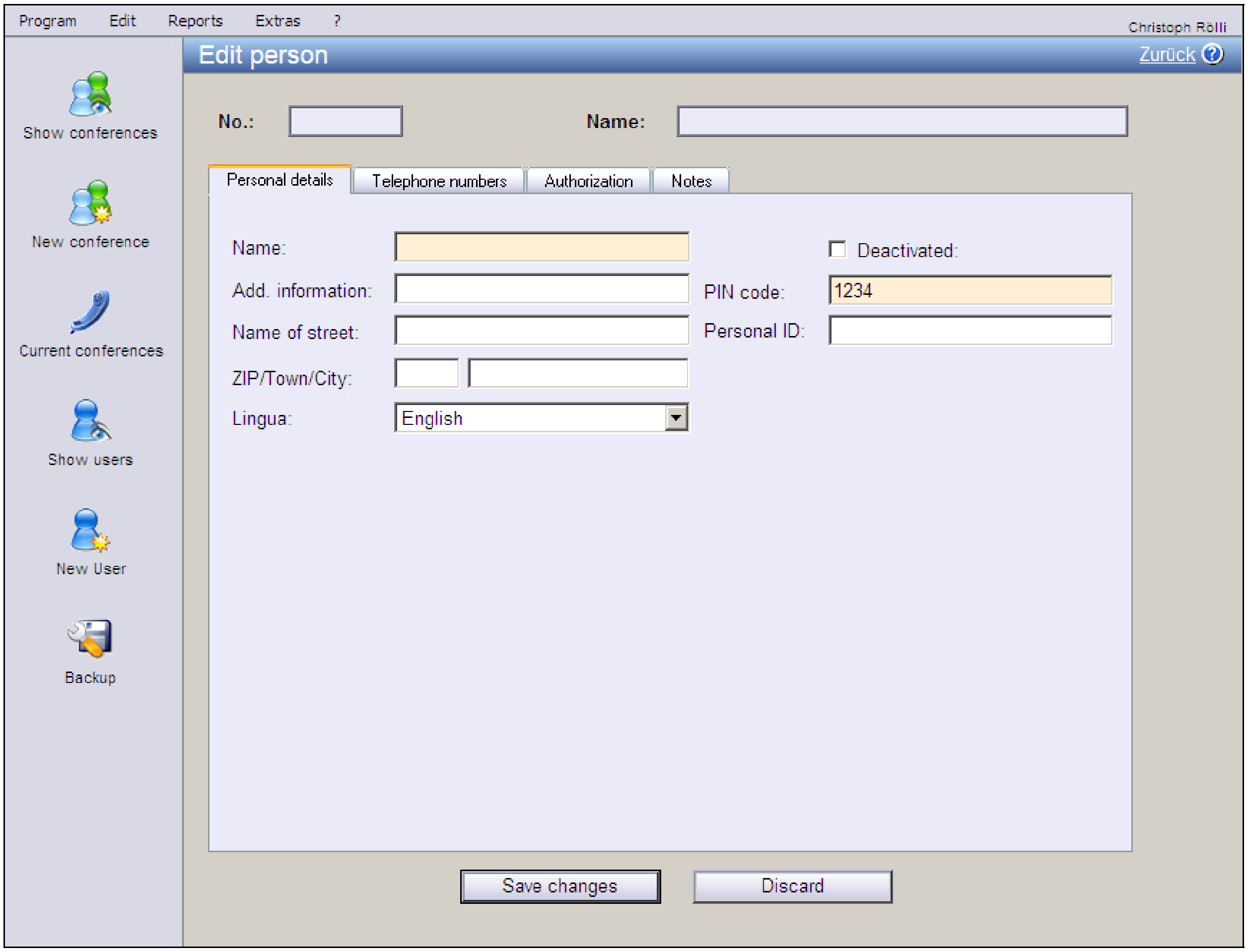

Figure 60: NovaConf Edit Personal Details Screen

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.

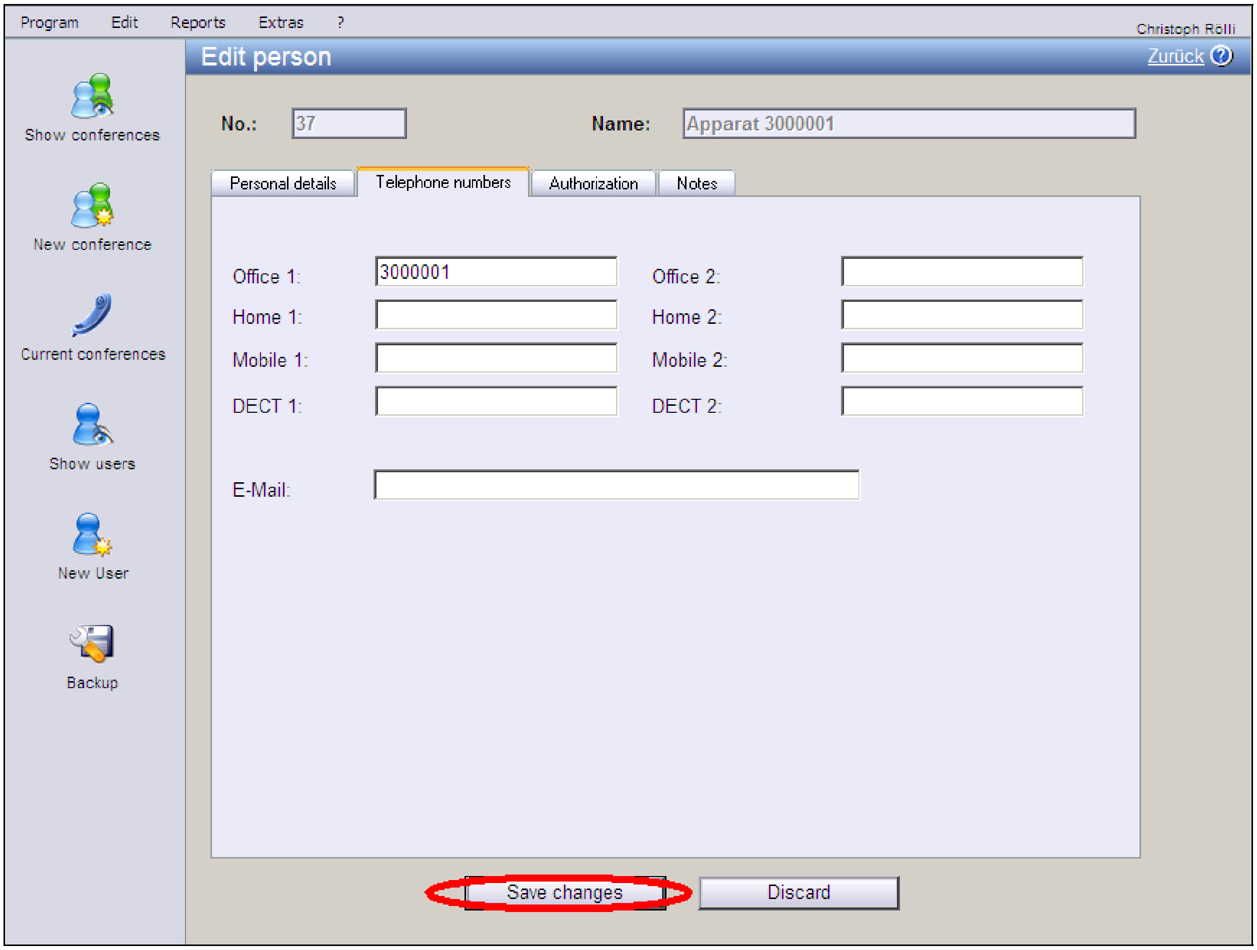

Figure 61: NovaConf Edit User Telephone Numbers Screen

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.

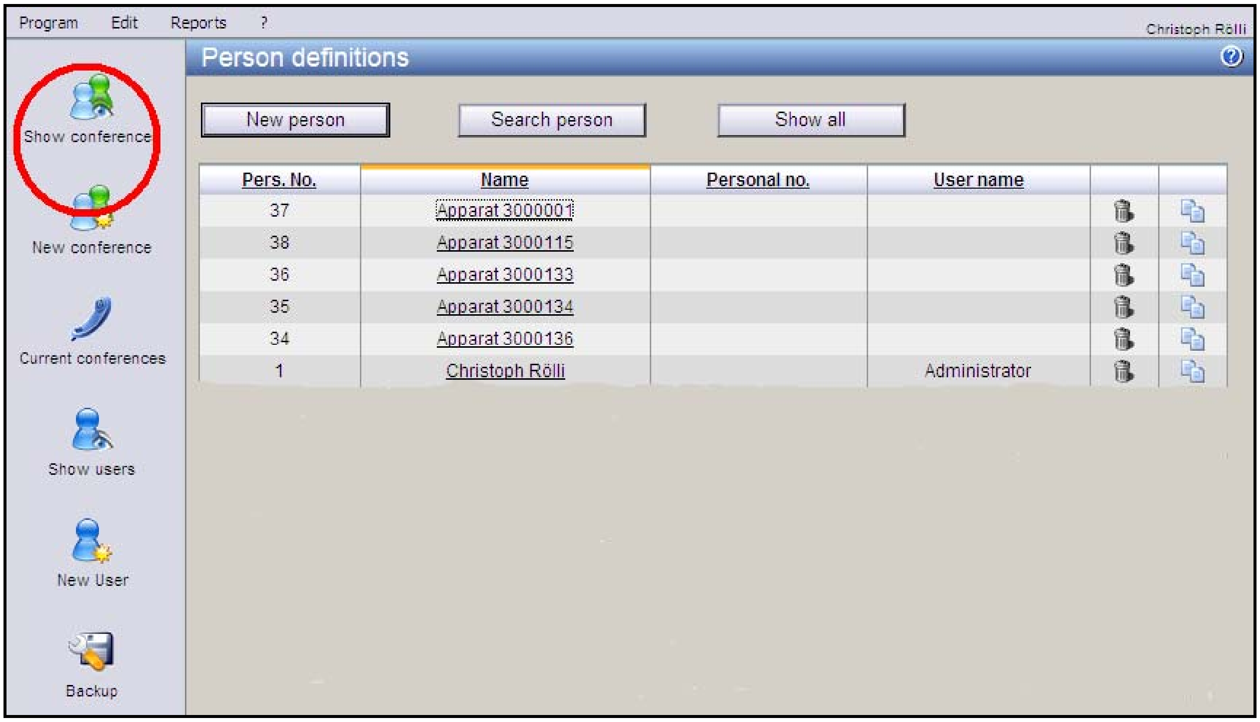

Figure 62: NovaConf Personal User Display Screen

3.4.3.2 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.

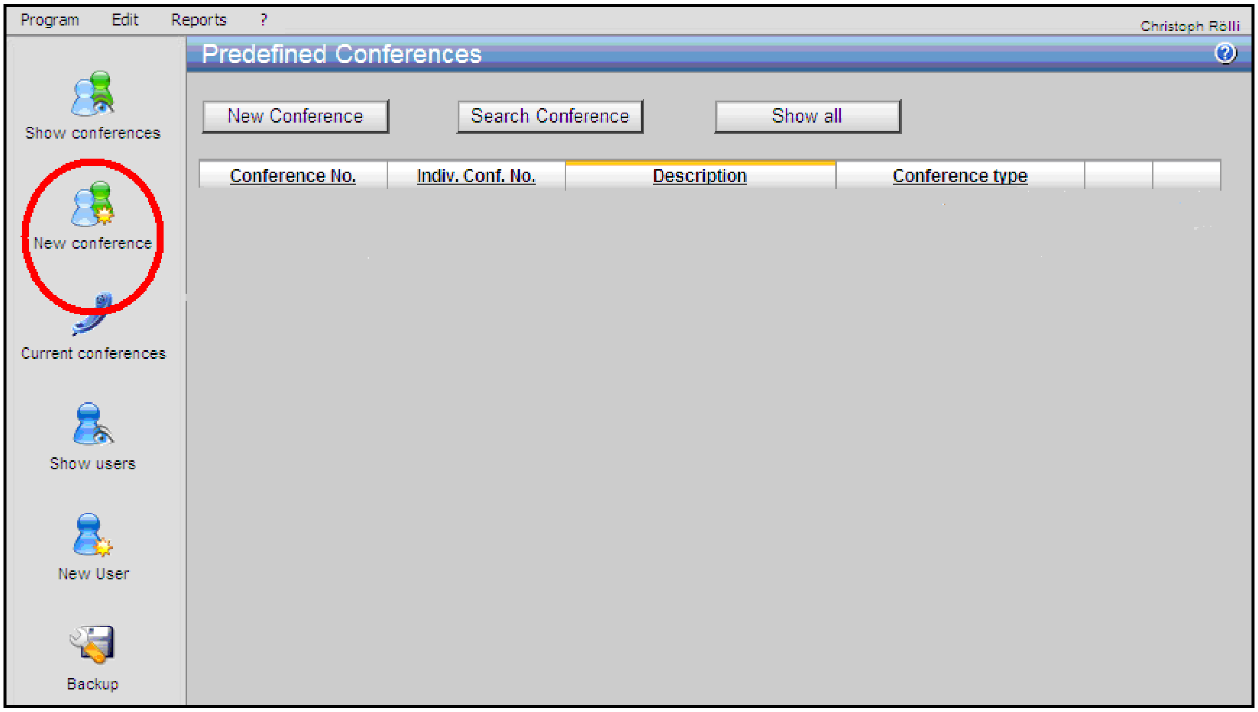

Figure 63: Predefined Conference List Screen

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 22: NovaConf Conference Common Configuration Parameters

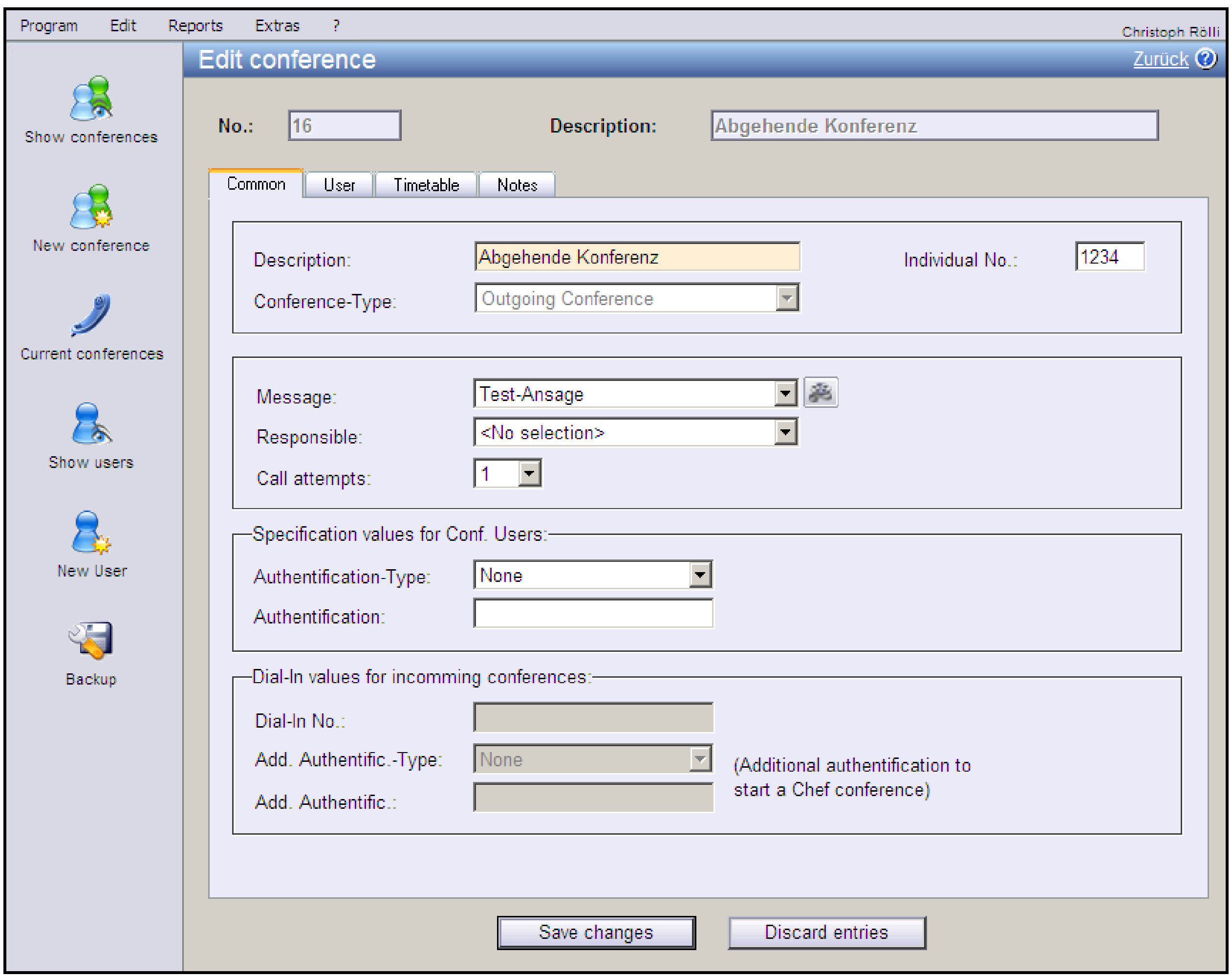

Figure 64: NovaConf New Outgoing Conference Screen

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

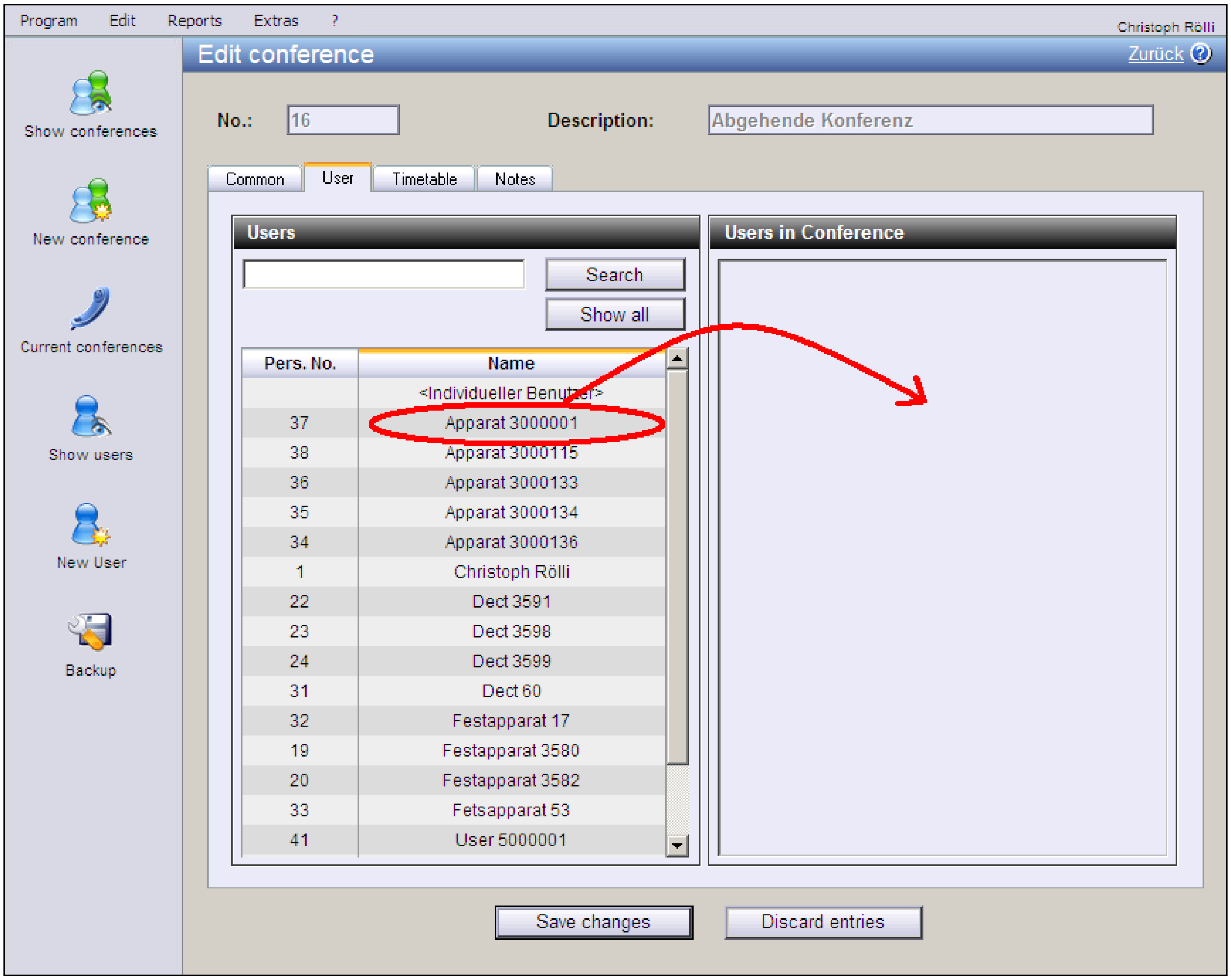

Figure 65: NovaConf User Allocation via Drag and Drop

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.

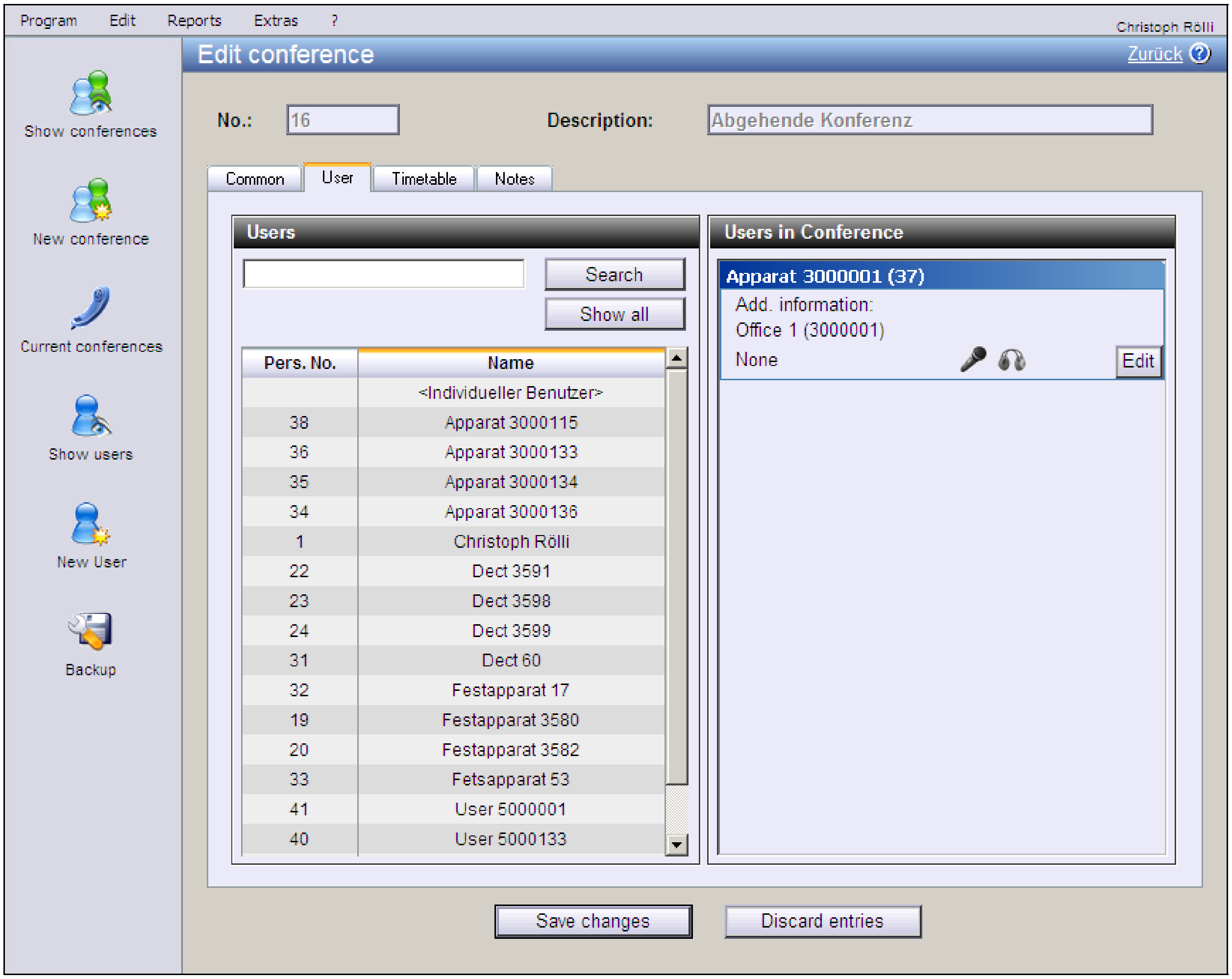

Figure 66: NovaConf Conference After Allocation of First Participant

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.
Inactiv	Check this box to stop conferences from being established.

Table 23: NovaConf Conference Timetable Configuration Parameters

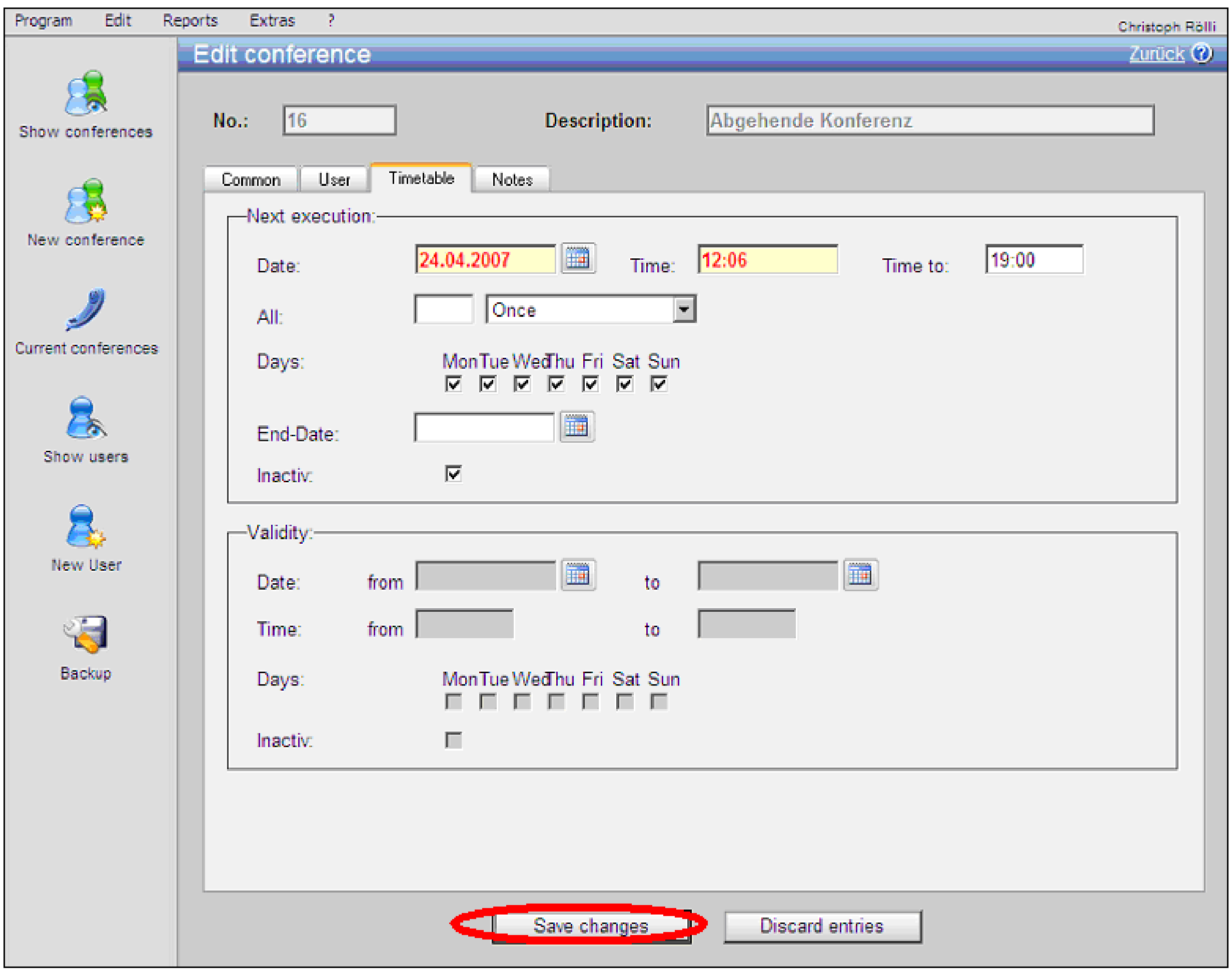

Figure 67: NovaConf Conference Timetable

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 NovaConf to receive overlap number transmission.
- 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 including:
  - The LAN connection between the NovaConf and the network
  - The BRI connection between NovaConf and the Avaya G350 Media Gateway
  - The PRI connection between NovaConf and the Avaya G350 Media Gateway
- 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 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 NovaConf to receive overlap numbers by using Avaya IP Telephones to place a call to NovaConf via its trunk access code to initiate a call setup, and subsequently dialing the NovaConf extension, which is sent as a series of individual digits via overlap sending, to establish the connection.
- 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.

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 Developer*Connection* Program at devconnect@avaya.com.