



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.

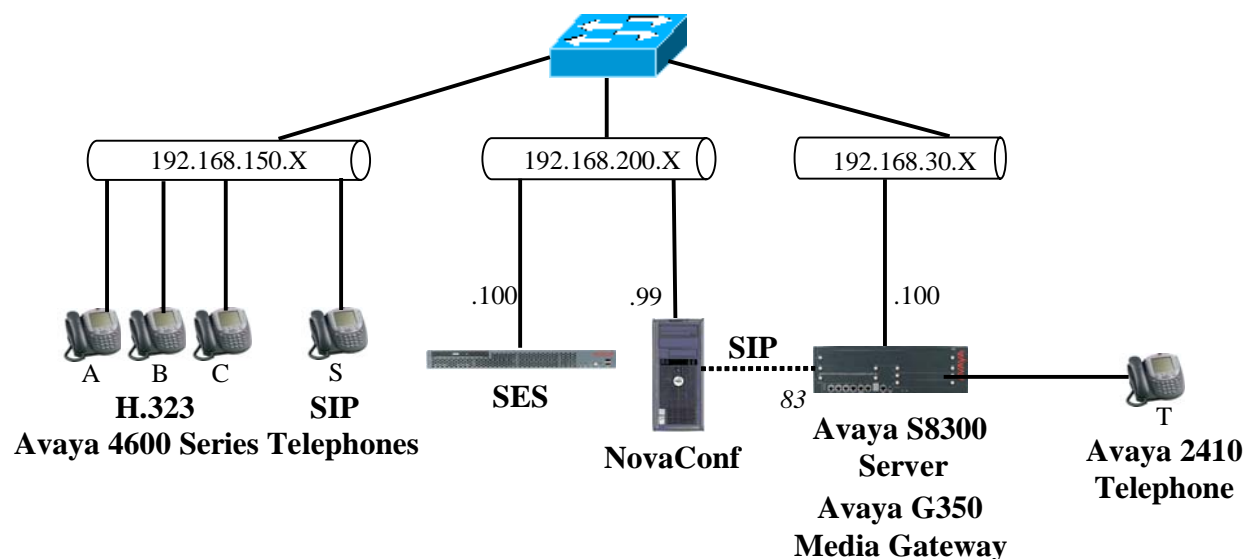


Figure 1: NovaConf Test Configuration

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
OPTIONAL FEATURES
Page 1 of 10

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

Platform Maximum Ports: 900
Maximum Stations: 450
Maximum XMOBILE Stations: 0
Maximum Off-PBX Telephones - EC500: 0
Maximum Off-PBX Telephones - OPS: 5
Maximum Off-PBX Telephones - PBFMC: 0
Maximum Off-PBX Telephones - PVFMC: 0
Maximum Off-PBX Telephones - SCCAN: 0
USED
76
7
0
0
2
0
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
OPTIONAL FEATURES
Page 2 of 10

IP PORT CAPACITIES
Maximum Administered H.323 Trunks: 30
Maximum Concurrently Registered IP Stations: 10
Maximum Administered Remote Office Trunks: 0
Maximum Concurrently Registered Remote Office Stations: 0
Maximum Concurrently Registered IP eCons: 0
Max Concur Registered Unauthenticated H.323 Stations: 0
Maximum Video Capable H.323 Stations: 0
Maximum Video Capable IP Softphones: 0
Maximum Administered SIP Trunks: 10
USED
5
3
0
0
0
0
0
0
3

Maximum Number of DS1 Boards with Echo Cancellation: 0
Maximum TN2501 VAL Boards: 0
Maximum Media Gateway VAL Sources: 0
Maximum TN2602 Boards with 80 VoIP Channels: 0
Maximum TN2602 Boards with 320 VoIP Channels: 0
Maximum Number of Expanded Meet-me Conference Ports: 0
0
0
0
0
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 be 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	ISDN Feature Plus? n	
Enhanced Conferencing? n	ISDN Network Call Redirection? n	
Enhanced EC500? y	ISDN-BRI Trunks? y	
Enterprise Survivable Server? n	ISDN-PRI? y	
Enterprise Wide Licensing? n	Local Survivable Processor? n	
ESS Administration? n	Malicious Call Trace? n	
Extended Cvg/Fwd Admin? n	Media Encryption Over IP? n	
External Device Alarm Admin? n	Mode Code for Centralized Voice Mail? n	
Five Port Networks Max Per MCC? n	Multifrequency Signaling? y	
Flexible Billing? n	Multimedia Call Handling (Basic)? n	
Forced Entry of Account Codes? n	Multimedia Call Handling (Enhanced)? n	
Global Call Classification? n		
Hospitality (Basic)? y		
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

    Personal Station Access (PSA)? n      System Management Data Transfer? n
    Posted Messages? n                    Tenant Partitioning? n
    PNC Duplication? n                    Terminal Trans. Init. (TTI)? n
    Port Network Support? n                Time of Day Routing? n
                                           Uniform Dialing Plan? y
    Processor and System MSP? n            Usage Allocation Enhancements? y
    Private Networking? n                  TN2501 VAL Maximum Capacity? y
    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 Domain: ffm.com       Far-end Network Region:

                                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

Night Service:

Dial Access? n

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								
DIAL PLAN ANALYSIS TABLE								
Page 1 of 12								
Percent Full: 3								
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call 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								
UNIFORM DIAL PLAN TABLE								
Page 1 of 2								
Percent Full: 0								
Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node 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								
AAR DIGIT ANALYSIS TABLE								
Page 1 of 2								
Percent Full: 3								
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI 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
                                     Intw
1: 83   0
2:
3:
4:
5:
6:

                                     n  user
                                     n  user
                                     n  user
                                     n  user
                                     n  user
                                     n  user

      BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature PARM  No. Numbering LAR
      0 1 2 M 4 W      Request      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

Ext  Ext      Trk      CPN      Total
Len  Code      Grp(s)   Prefix   Len
7   7          7        7        7
7   3          83       7        7

Total Administered: 2
Maximum Entries: 240

```

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
  Loss Group: 19                                       Time of Day Lock Table:
  Speakerphone: 2-way                                Personalized Ringing Pattern: 1
  Display Language: english                          Message Lamp Ext: 300-0136
  Survivable GK Node Name:                            Mute Button Enabled? y
  Survivable COR: internal                            Expansion Module? n
  Survivable Trunk Dest? y                            Media Complex Ext:
                                                         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
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:	
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
MWISVR	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 MWISVR	"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”.

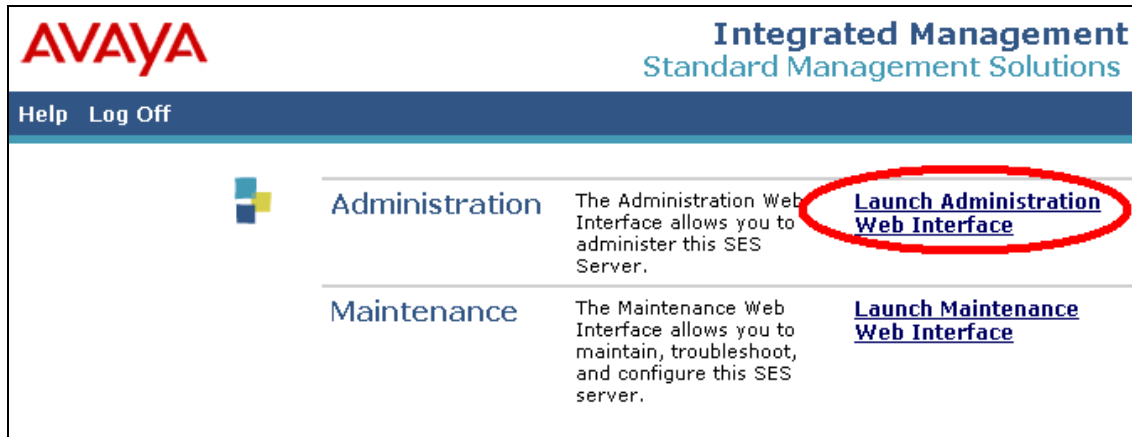



Figure 27: SES Initial Greeting Screen

The SES Integrated Management top level menu is then displayed.




Integrated Management
SIP Server Management

Help Exit

Server: 192.168.200.100

Top

- Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
- Media Servers
- Adjunct Systems
 - Services
- Server Configuration
- Certificate Management
- IM Logs
- Trace Logger
- Export/Import to ProVision

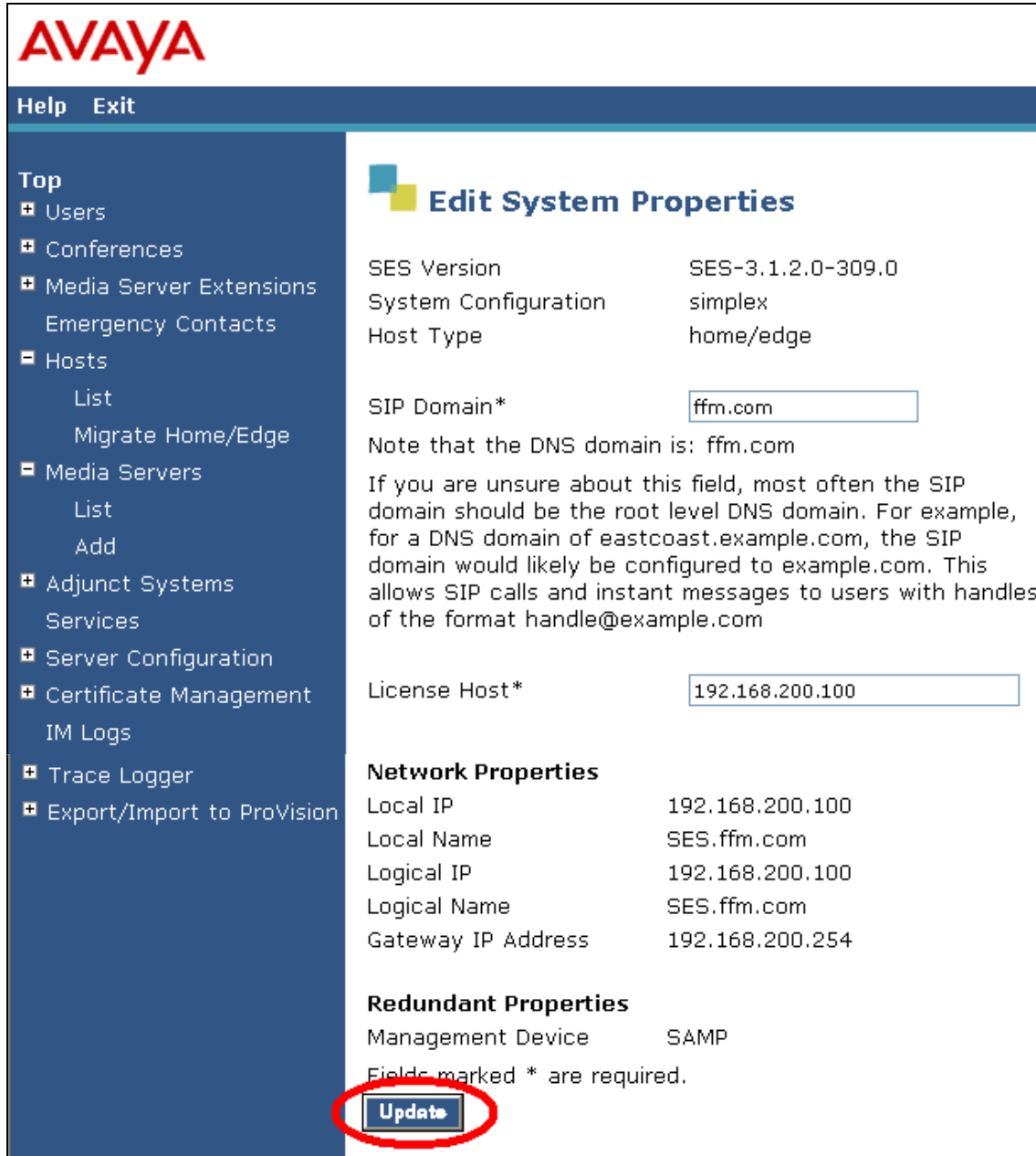

Top

Manage Users	Add and delete Users.
Manage Conferencing	Add and delete Conference Extensions.
Manage Media Server Extensions	Add and delete Media Server Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Manage Hosts	Add and delete Hosts.
Manage Media Servers	Add and delete Media Servers.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Services	Start and stop server processes on this host.
Server Configuration	Edit Properties of the system.
Certificate Management	Manage Certificates.
IM Logs	Download IM Logs.
Trace Logger	Manage SIP Trace Logs.
Export Import to ProVision	Export and import data using ProVision on this host.

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.



The screenshot shows the Avaya SES web interface. The top header features the Avaya logo and navigation links for 'Help' and 'Exit'. A left-hand sidebar contains a tree view of system management options, including 'Top', 'Users', 'Conferences', 'Media Server Extensions', 'Emergency Contacts', 'Hosts', 'Media Servers', 'Adjunct Systems', 'Server Configuration', 'Certificate Management', 'Trace Logger', and 'Export/Import to ProVision'. The main content area is titled 'Edit System Properties' and displays configuration details for the SIP system. It includes fields for 'SES Version' (SES-3.1.2.0-309.0), 'System Configuration' (simplex), and 'Host Type' (home/edge). The 'SIP Domain*' field is set to 'ffm.com', with a note explaining that the DNS domain is 'ffm.com' and providing guidance on domain configuration. Below this, the 'License Host*' field is set to '192.168.200.100'. A section titled 'Network Properties' lists 'Local IP', 'Local Name', 'Logical IP', 'Logical Name', and 'Gateway IP Address'. A 'Redundant Properties' section shows the 'Management Device' as 'SAMP'. A note at the bottom states 'Fields marked * are required.' The 'Update' button at the bottom left is circled in red.

Property	Value
SES Version	SES-3.1.2.0-309.0
System Configuration	simplex
Host Type	home/edge
SIP Domain*	ffm.com
License Host*	192.168.200.100
Local IP	192.168.200.100
Local Name	SES.ffmpeg.com
Logical IP	192.168.200.100
Logical Name	SES.ffmpeg.com
Gateway IP Address	192.168.200.254
Management Device	SAMP

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.

Add Host

Host IP Address*

DB Password

Profile Service Password

Host Type

Parent

Listen Protocols ☒ UDP ☒ TCP ☒ TLS

Link Protocols ☐ UDP ☐ TCP ☒ TLS

Presence Access Policy (Default) ☐ Allow All ☒ Deny All

Emergency Contacts Policy ☒ Allow ☐ Deny

Minimum Registration (seconds) Registration Expiration Timer (seconds)*

Line Reservation Timer (seconds)

Outbound Routing Allowed ☒ Internal ☐ External

From OutboundProxy Port ☐ UDP ☐ TCP ☐ TLS

Outbound Direct Domains

Default Ringer Volume* Default Ringer Cadence*

Default Receiver Volume* Default Speaker Volume*

VMM Server Address

VMM Server Port VMM Report Period

Fields marked * are required.

Update

Figure 30: Avaya SES “Add Host” Screen

3.3.2. Configure Interface to Avaya Communication Manager

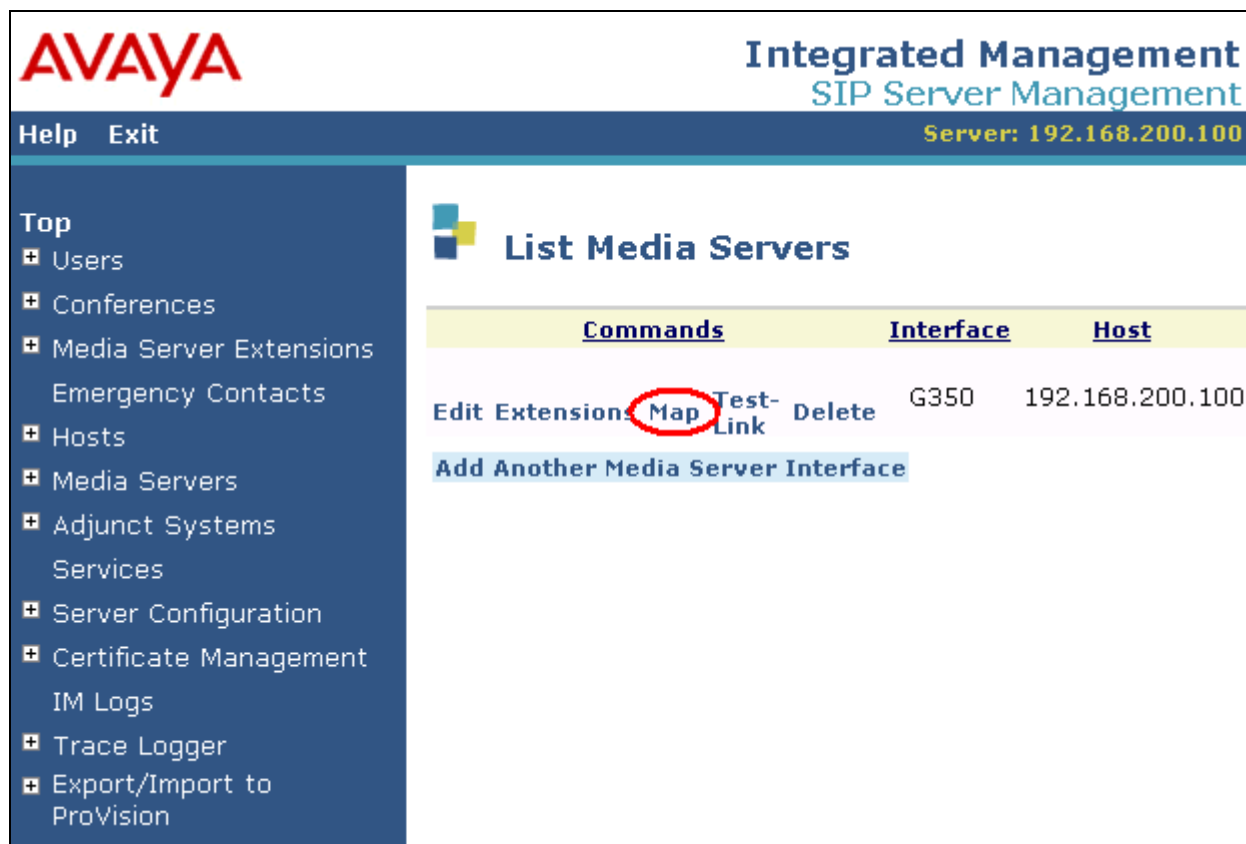
From the top-level management screen, select “Manage Media Servers”-> “Add Media Server”. Assign a meaningful name to the “Media Server Interface Name”. Select the IP address of the Avaya SES server as the “Host”. Enter the address of the Avaya S8300 Server as the SIP Trunk IP Address. Select the “Add”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

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top header features the Avaya logo on the left and the title 'Integrated Management SIP Server Management' on the right, with a server IP address '192.168.200.100' displayed below it. A navigation menu on the left lists various management options, including 'Media Servers'. The main content area is titled 'Add Media Server Interface' and contains several input fields and sections. The 'Media Server Interface Name*' field is populated with 'G350'. The 'Host' field is a dropdown menu showing '192.168.200.100'. Below this, the 'SIP Trunk' section has a 'SIP Trunk Link Type' with radio buttons for 'TCP' and 'TLS' (selected), and a 'SIP Trunk IP Address*' field populated with '192.168.30.100'. The 'Media Server' section includes fields for 'Media Server Admin Address (see Help)', 'Media Server Admin Login', 'Media Server Admin Password', and 'Media Server Admin Password Confirm'. A note at the bottom states 'Fields marked * are required.' The 'Add' button at the bottom left is circled in red.

Add Media Server Interface	
Media Server Interface Name*	G350
Host	192.168.200.100
SIP Trunk	
SIP Trunk Link Type	<input type="radio"/> TCP <input checked="" type="radio"/> TLS
SIP Trunk IP Address*	192.168.30.100
Media Server	
Media Server Admin Address (see Help)	
Media Server Admin Login	
Media Server Admin Password	
Media Server Admin Password Confirm	
Fields marked * are required.	
Add	

Figure 31: Avaya SES Add Media Server Interface Screen

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




The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, the title "Integrated Management SIP Server Management", and the server IP address "192.168.200.100". A left-hand navigation menu lists various system components, with "Media Servers" selected. The main content area, titled "List Media Servers", contains a table with columns for "Commands", "Interface", and "Host". A single row is visible, showing "Edit Extensions", "Map" (circled in red), "Test-Link", "Delete", "G350", and "192.168.200.100". Below the table is a button labeled "Add Another Media Server Interface".

Commands	Interface	Host
Edit Extensions	Map	Test-Link
Delete	G350	192.168.200.100

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.



The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top header features the Avaya logo and the title 'Integrated Management SIP Server Management' with the server IP '192.168.200.100'. A navigation menu on the left lists various system components, with 'Media Servers' highlighted. The main content area is titled 'Add Media Server Address Map' and contains a form with the following fields: 'Host' (G350), 'Name*' (G350map), 'Pattern*' (^sip:3.*), and 'Replace URI' (checked). A red circle highlights the 'Add' button at the bottom of the form. A note states 'Fields marked * are required.'

Host	G350
Name*	G350map
Pattern*	^sip:3.*
Replace URI	<input checked="" type="checkbox"/>

Fields marked * are required.

Add

Figure 33: Add Media Server Address Map



Integrated Management

SIP Server Management

Server: 192.168.200.100

[Help](#) [Exit](#)

Top

- + Users
- + Conferences
- + Media Server Extensions
Emergency Contacts
- + Hosts
 - List
 - Migrate Home/Edge
- + Media Servers
- + Adjunct Systems Services
- + Server Configuration
- + Certificate Management IM Logs
- + Trace Logger
- + Export/Import to ProVision



List Media Server Address Map

<u>Commands</u>	<u>Name</u>	<u>Commands</u>	<u>Contact</u>
Edit Delete	G350Map		
Edit Delete		sip:\${(user)}@192.168.30.100:5061;transport=tls	

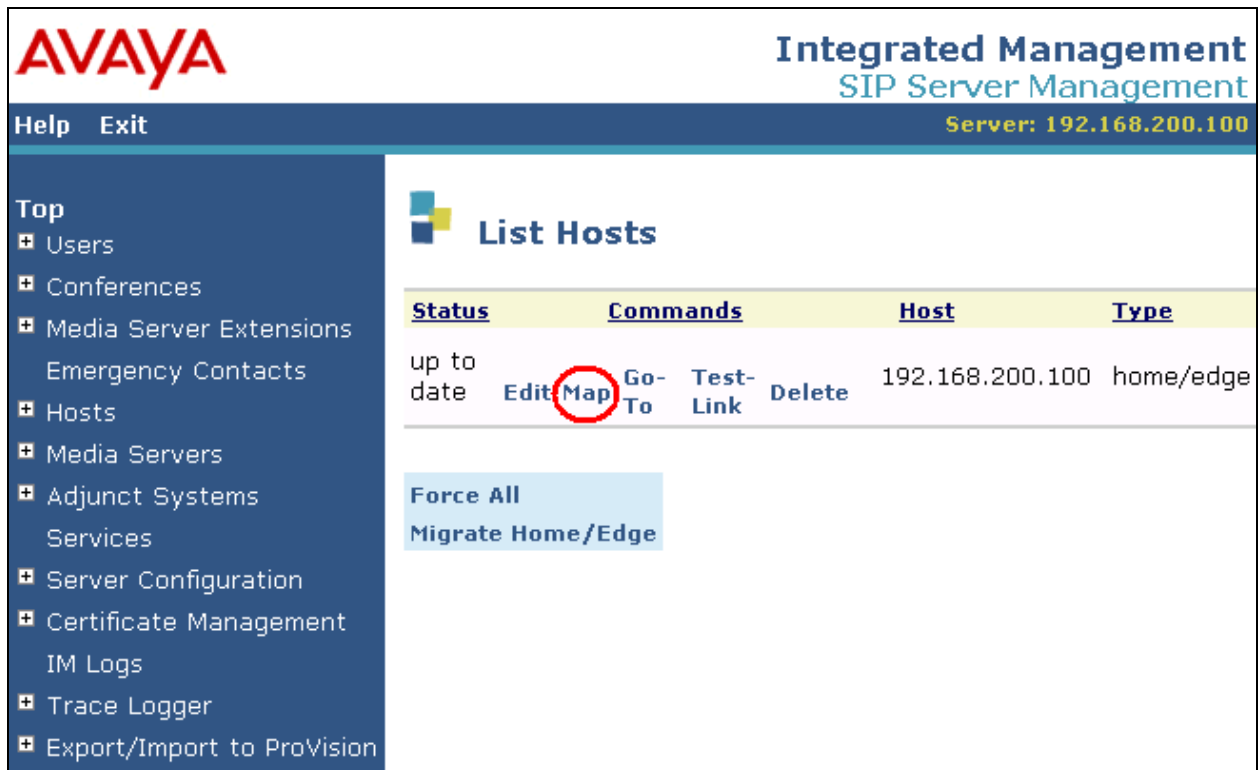
Add Another Map
Add Another Contact
Delete Group

Add Map In New Group

MRR; Reviewed:
SPOC 10/11/2007

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.



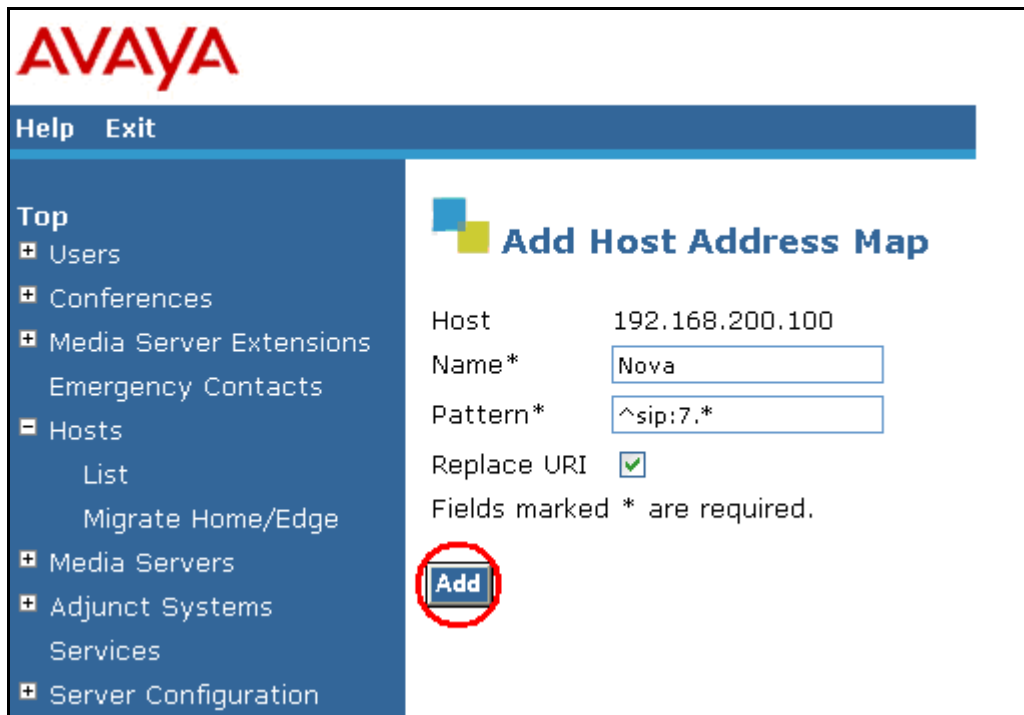
The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, the title "Integrated Management SIP Server Management", and the server IP "192.168.200.100". A left sidebar contains a menu with options like Users, Conferences, Media Server Extensions, Hosts, Media Servers, Adjunct Systems, Services, Server Configuration, Certificate Management, IM Logs, Trace Logger, and Export/Import to ProVision. The main content area is titled "List Hosts" and contains a table with columns: Status, Commands, Host, and Type. The table has one row with the status "up to date", a "Map" button circled in red, the host "192.168.200.100", and the type "home/edge". Below the table are buttons for "Force All" and "Migrate Home/Edge".

Status	Commands	Host	Type
up to date	Edit Map Go-To Test-Link Delete	192.168.200.100	home/edge

[Force All](#)
[Migrate Home/Edge](#)

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.



The screenshot shows the Avaya SES web interface. At the top left is the Avaya logo. Below it is a navigation menu with options: Help, Exit, Top, Users, Conferences, Media Server Extensions, Emergency Contacts, Hosts (with sub-options List and Migrate Home/Edge), Media Servers, Adjunct Systems, Services, and Server Configuration. The main content area is titled "Add Host Address Map" and contains the following fields: Host (192.168.200.100), Name* (Nova), Pattern* (^sip:7.*), and Replace URI (checked). A note states "Fields marked * are required." The "Add" button is circled in red.

Field	Value
Host	192.168.200.100
Name*	Nova
Pattern*	^sip:7.*
Replace URI	<input checked="" type="checkbox"/>

Fields marked * are required.

Add

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”.

The screenshot shows the Avaya SES web interface. At the top is the Avaya logo. Below it is a navigation bar with 'Help' and 'Exit' links. A sidebar on the left contains a 'Top' section with expandable/collapsible items: Users, Conferences, Media Server Extensions, Emergency Contacts, Hosts (expanded), Media Servers, and Adjunct Systems. Under 'Hosts', there are links for 'List' and 'Migrate Home/Edge'. The main content area is titled 'Add Host Contact' and contains the following fields:

Host	192.168.200.100
Handle	Nova
Contact*	<input type="text" value="sip:\$(user)@192.168.200.99:5060;transport=udp"/>

Fields marked * are required.

An 'Add' button is located at the bottom of the form, circled in red.

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”.

AVAYA Integrated Management
SIP Server Management
Server: 192.168.200.100

Help Exit

Add User

Primary Handle* 3000115
User ID 3000115
Password*
Confirm Password*
Host* 192.168.200.100
First Name* Extn
Last Name* 3000115
Address 1 Kleyerstr 94
Address 2
Office
City Frankfurt
State
Country Germany
Zip 60326
Add Media Server Extension ☒
Fields marked * are required.

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.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, the title 'Integrated Management SIP Server Management', and the server IP '192.168.200.100'. A left-hand navigation menu lists various system components. The main content area is titled 'Add Media Server Extension' and contains two input fields: 'Extension*' with the value '3000115' and 'Media Server' with a dropdown menu showing 'G350'. A note below the fields states 'Fields marked * are required.' The 'Add' button is highlighted with a red circle.

Menu Item	Field Label	Field Value
Top	Extension*	3000115
Users	Media Server	G350
Conferences		
Media Server Extensions		
Emergency Contacts		
Hosts		
Media Servers		
Adjunct Systems		
Services		
Server Configuration		
Certificate Management		
IM Logs		
Trace Logger		
Export/Import to ProVision		

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.

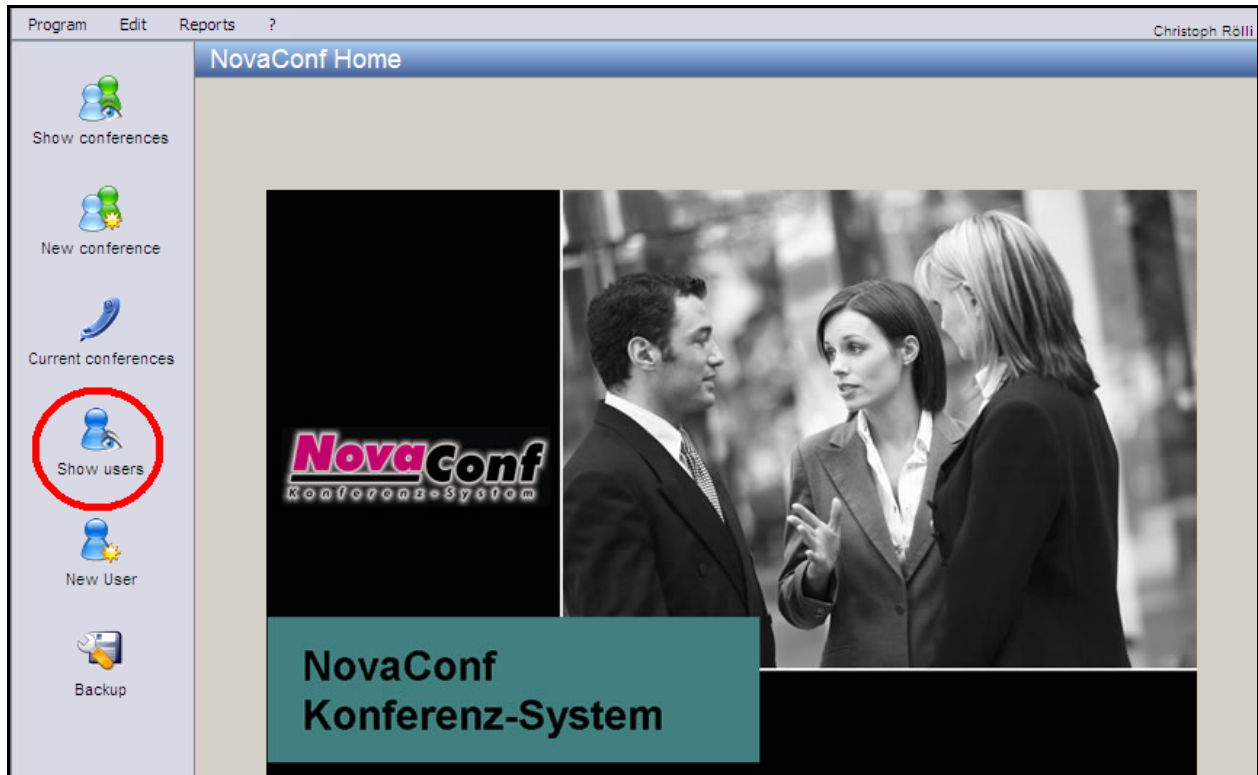


Figure 41: NovaConf Startup Screen

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**.

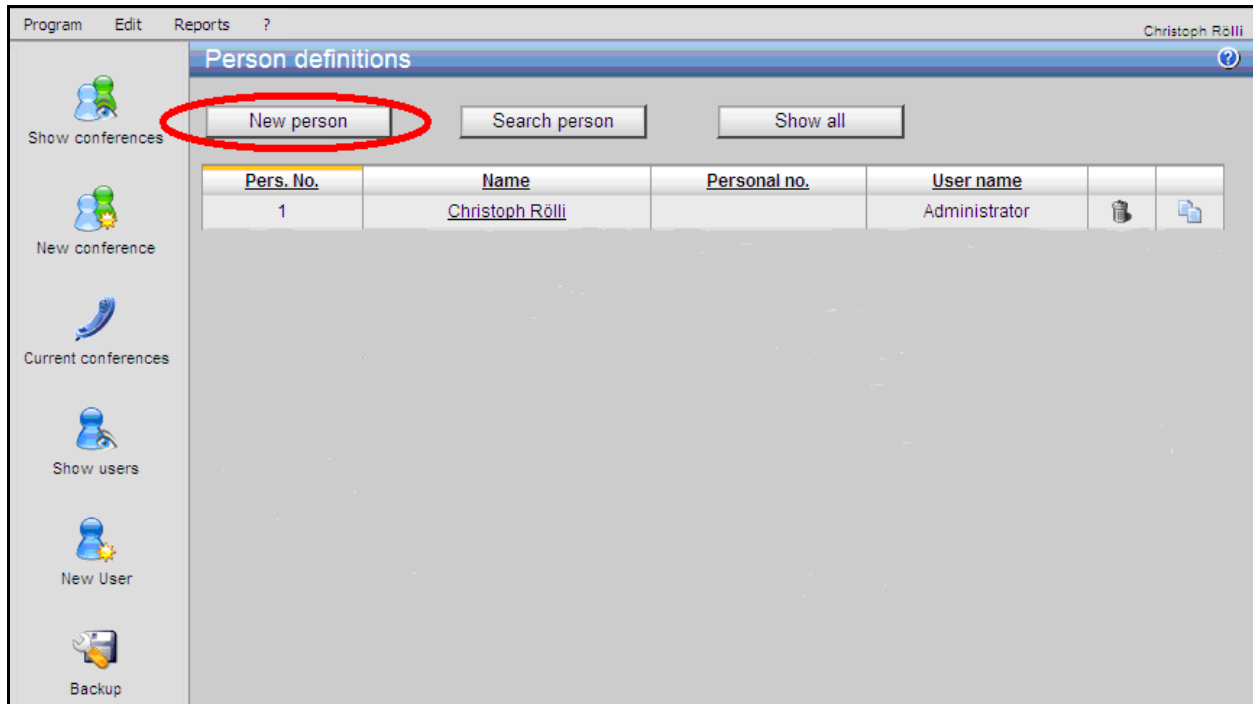


Figure 42: NovaConf User Configuration Screen

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.

The screenshot shows the 'Edit person' window in NovaConf. The window has a menu bar with 'Program', 'Edit', 'Reports', 'Extras', and '?'. The title bar says 'Edit person' and includes the user 'Christoph Rölli', a 'Zurück' button, and a help icon. On the left is a sidebar with icons for 'Show conferences', 'New conference', 'Current conferences', 'Show users', 'New User', and 'Backup'. The main content area has four tabs: 'Personal details', 'Telephone numbers', 'Authorization', and 'Notes'. The 'Personal details' tab is selected. It contains the following fields:

- 'No.:' followed by an empty text box.
- 'Name:' followed by an empty text box.
- 'Name:' followed by an empty text box.
- 'Add. information:' followed by an empty text box.
- 'Name of street:' followed by an empty text box.
- 'ZIP/Town/City:' followed by two empty text boxes.
- 'Lingua:' followed by a dropdown menu showing 'English'.
- 'PIN code:' followed by a text box containing '1234'.
- 'Personal ID:' followed by an empty text box.
- A checkbox labeled 'Deactivated:'.

At the bottom of the window are two buttons: 'Save changes' and 'Discard'.

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.

The screenshot shows the 'Edit person' window in the NovaConf application. The window has a menu bar at the top with 'Program', 'Edit', 'Reports', 'Extras', and '?'. The user's name 'Christoph Rölli' is in the top right corner. On the left is a sidebar with icons and labels: 'Show conferences', 'New conference', 'Current conferences', 'Show users', 'New User', and 'Backup'. The main area is titled 'Edit person' and has a 'Zurück' button with a question mark. Below the title bar, there are two input fields: 'No.: 37' and 'Name: Apparat 3000001'. Below these are four tabs: 'Personal details', 'Telephone numbers' (which is selected), 'Authorization', and 'Notes'. The 'Telephone numbers' tab contains several input fields arranged in two columns. The left column has 'Office 1: 3000001', 'Home 1:', 'Mobile 1:', and 'DECT 1:'. The right column has 'Office 2:', 'Home 2:', 'Mobile 2:', and 'DECT 2:'. Below these columns is an 'E-Mail:' field. At the bottom of the window, there are two buttons: 'Save changes' (which is circled in red) and 'Discard'.

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.

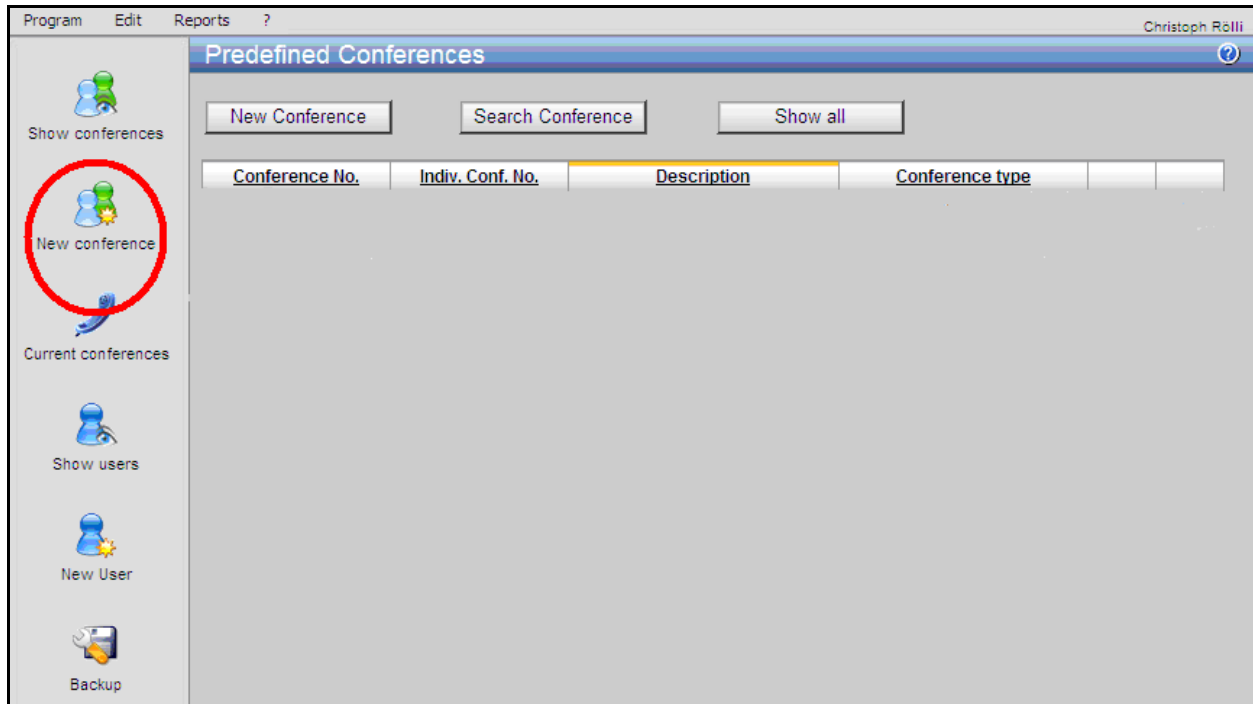


Figure 46: Predefined Conference List Screen

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

The screenshot displays the 'Edit conference' window with the 'Common' tab selected. The left sidebar contains navigation icons for 'Show conferences', 'New conference', 'Current conferences', 'Show users', 'New User', and 'Backup'. The main area contains the following fields and controls:

- No.:** 16
- Description:** Abgehende Konferenz
- Individual No.:** 1234
- Conference-Type:** Outgoing Conference (dropdown)
- Message:** Test-Ansage (dropdown with a record button)
- Responsible:** <No selection> (dropdown)
- Call attempts:** 1 (dropdown)
- Specification values for Conf. Users:**
 - Authentication-Type:** None (dropdown)
 - Authentication:** (text field)
- Dial-In values for incoming conferences:**
 - Dial-In No.:** (text field)
 - Add. Authentic.-Type:** None (dropdown)
 - Add. Authentic.:** (text field)
 - (Additional authentication to start a Chef conference)
- Buttons:** Save changes, Discard entries

Figure 47: NovaConf New Outgoing Conference Screen

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

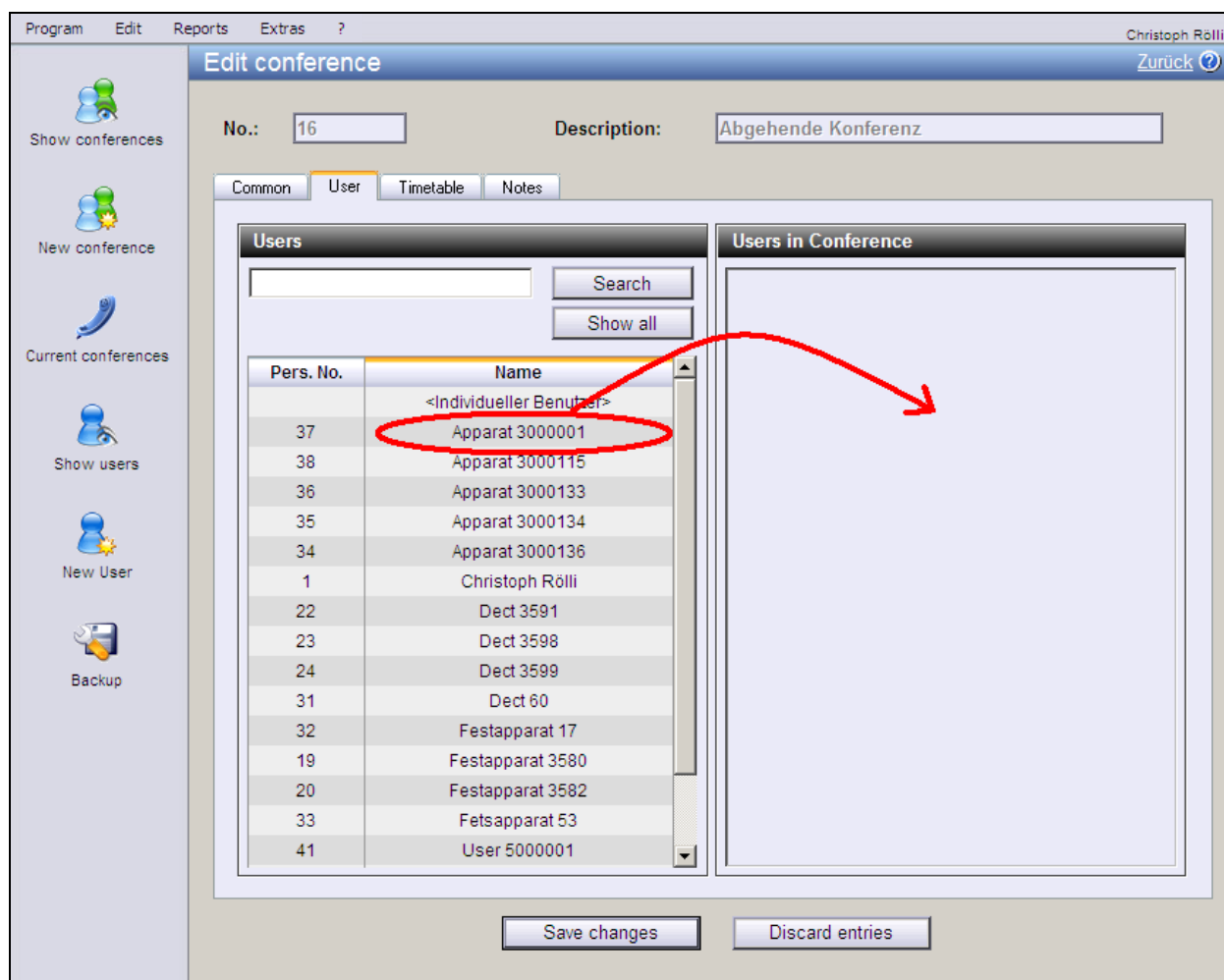


Figure 48: NovaConf User Allocation via Drag and Drop

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.

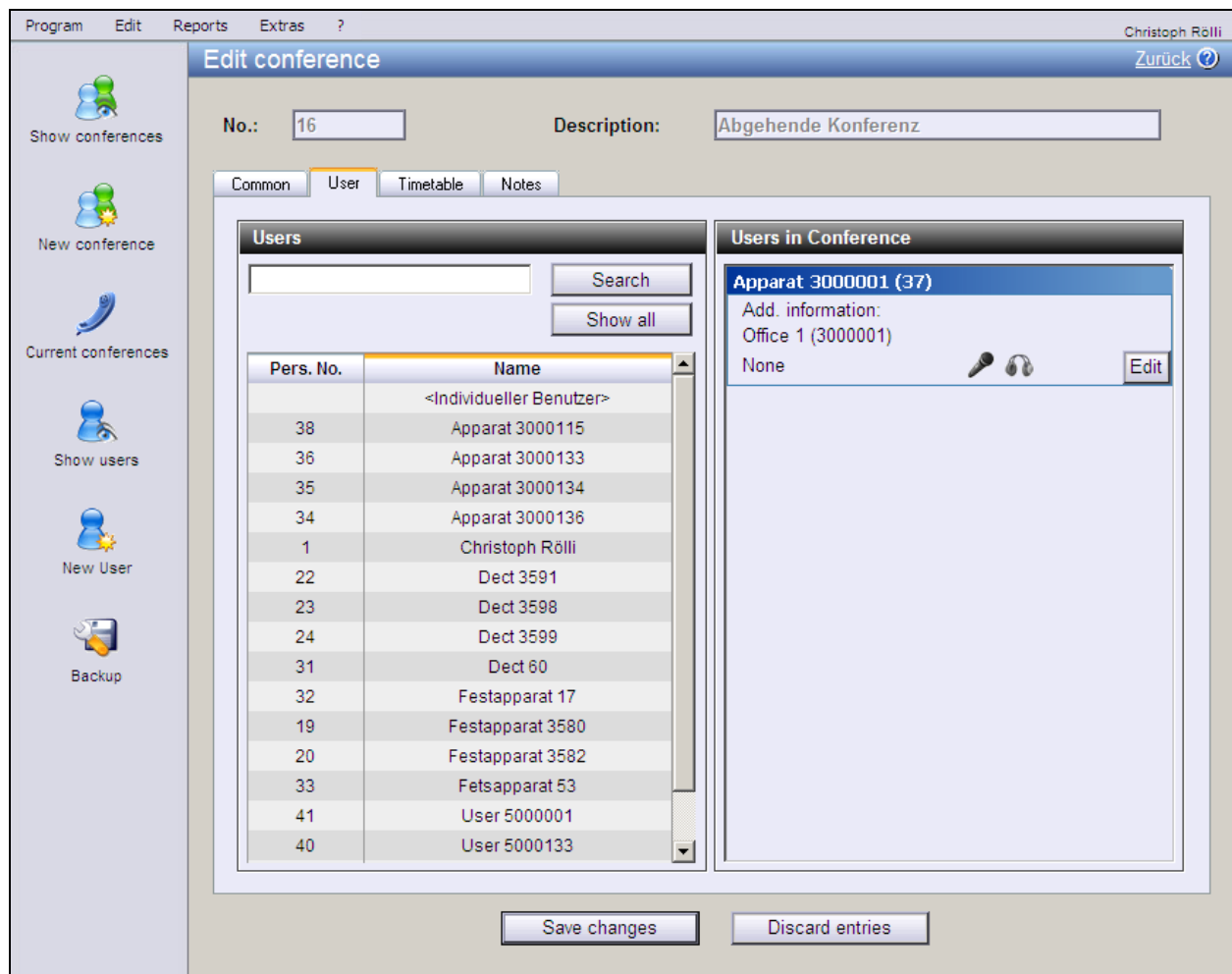


Figure 49: NovaConf Conference After Allocation of First Participant

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

The screenshot shows the 'Edit conference' window in NovaConf. The 'Timetable' tab is selected. The 'Next execution' section is configured with the following values:

- No.: 16
- Description: Abgehende Konferenz
- Date: 24.04.2007
- Time: 15:45
- Time to: 17:00
- All: 1
- Day(s): Mon, Tue, Wed, Thu, Fri (checked)
- End-Date: 30.04.2007
- Inactiv: ☐

The 'Validity' section is currently empty. At the bottom, the 'Save changes' button is circled in red, and the 'Discard entries' button is also visible.

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

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