Avaya Solution & Interoperability Test Lab

Application Notes for TAS FreiTel-IP Handsfree Telephone with Avaya Communication Manager Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the configuration of TAS FreiTel-IP handsfree telephone for operation with Avaya Communication Manager. TAS FreiTel-IP is a SIP telephone intended for use in environments where a handset and keypad are not required, such as elevators. These Application Notes contain an extensive description of the configurations for both FreiTel-IP and Avaya Communication Manager which were used for testing.

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

1. Introduction

The FreiTel-IP handsfree telephone system consists of a SIP endpoint which can control up to three speakerphones. This product is intended for use in environments where a handset and keypad are not required, such as elevators.

The speakerphone is a surface-mounted unit which has a user interface which consists of a button, two LEDs (green and yellow), and a microphone/speaker. Pressing the speakerphone button causes a call to be made to a preconfigured telephone number, which can be terminated only by the called party. The green LED on the speakerphone remains on while a call is active.

The FreiTel-IP can be configured to accept calls automatically or manually. If configured for automatic call acceptance, every incoming call is accepted automatically without delay, and remains active (indicated by the green LED being on) until terminated by the calling party. If the unit is configured for manual call acceptance, an incoming call is signaled by the yellow light and can be answered by pushing and holding the speakerphone button, which turns the yellow LED off and the green LED on. A manually accepted call remains active until terminated by releasing the speakerphone button, which turns off the green LED.

FreiTel-IP can be configured to filter incoming calls so unwanted calls are ignored.

The control unit has an Ethernet connector and a single multicolored LED. If attached to an Ethernet network, the LED turns red until the controller registers with a SIP proxy, at which time the LED turns green.

The configuration used for testing consisted of a TAS FreiTel-IP control unit and a single speakerphone together with a pair of Avaya S8720 Server running Avaya Communication Manager, Avaya G650 Media Gateway, Avaya SIP Enablement Services (SES) server, and various Avaya telephones. Note that the SES server is a combined home/edge configuration

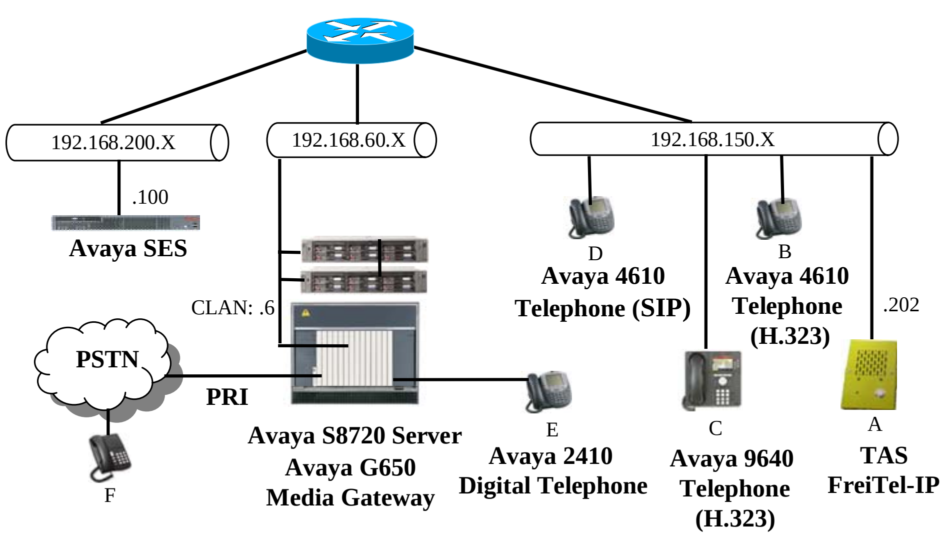

Figure 1: FreiTel-IP Test Configuration

The TAS FreiTel-IP handsfree telephone is attached to Avaya Communication Manager via SIP trunk by way of the Avaya SIP Enablement Services (SES).

The following table contains additional information about each of the telephones contained in the above diagram:

Diagram	Ext	PSTN Number	Endpoint
A	69001	069 9073 9887 69001	TAS FreiTel-IP
B	60116	069 9073 9887 60116	Avaya 4610 IP
C	60093	069 9073 9887 60093	Avaya 9640 one-X Deskphone
D	60113	069 9073 9887 60113	Avaya 4610 SIP
E	60007	069 9073 9887 60007	Avaya 2410 Digital
F		069 7505 6176	ISDN telephone

Table 1: Extensions Used for Testing

2. Equipment and Software Validated

Equipment		Software Version
Avaya S8720 Server		Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)
Avaya G650 Media Gateway		
	TN2312BP IP Server Interface TN799DP CLAN Interface TN2302AP IP Media Processor	HW11 FW040 HW01 FW024 HW20 FW117
Avaya SIP Enablement Services Server		SES-4.0.0.0-0.33.6.
Avaya 2410 Digital Telephone		5.0
Avaya 4610 H.323 Telephone		2.8
Avaya 4610 SIP Telephone		2.2.2
Avaya 9640 H.323 Telephone		1.5
TAS FreiTel-IP		V01.00

Table 2: Hardware/Software Component Versions

3. Configuration

3.1. Configure Avaya Communication Manager

The configuration and verification operations illustrated in this section were performed using the Avaya Communication Manager System Administration Terminal (SAT) terminal.

3.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer-options** command to verify that Avaya Communication Manager is configured to meet the minimum requirements to support the configuration used for these tests. 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.

Verify that the number of SIP trunks supported by the system is sufficient for the combination of trunks to the TAS FreiTel-IP and optional SIP endpoints to be supported.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 10    0
           Maximum Concurrently Registered IP Stations: 50    10
             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: 200   20

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 1     0
                     Maximum G250/G350/G700 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 2: System-Parameters Customer-Options Form

3.1.2. Configure Dial Plan

3.1.2.1 Configure Dial Plan Analysis

Use the **change dialplan analysis** command to specify that dialed strings which begin with "6" are extensions. Include the string "*83" to be used as trunk access code for the SIP trunk as described in **section 3.1.3.3.**

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

      Dialed  Total  Call       Dialed  Total  Call       Dialed  Total  Call
      String  Length Type       String  Length Type       String  Length Type
        6       5    ext
        *83     3    dac
```

Figure 3: Dialplan Analysis Form

3.1.3. Configure Interface to SES

3.1.3.1 Specify IP node names

Use the **change node-names ip** command to define the address of the "clan" interface and the Avaya SIP Enablement Services server.

```
change node-names ip                                           Page   1 of   1
                                  IP NODE NAMES
    Name              IP Address           Name              IP Address
clan                192.168.60.6                           .   .   .
default             0 .0 .0 .0                             .   .   .
ipsi                192.168.60.5                           .   .   .
medpro              192.168.60.7                           .   .   .
procr                 .   .   .                            .   .   .
ses                 192.168.200.100                        .   .   .
```

Figure 4: Node-Names IP Form

3.1.3.2 Configure Signaling Group for the SIP Trunk Interface to SES

Use the **add signaling-group <x>** command, where <x> is a free signaling group number, to create a signaling group which is to be used to connect to the SES. Accept defaults for parameters, except for those which are highlighted.

Parameter	Usage
Group Type	Enter "sip" to specify a SIP trunk.
Transport Method	Enter "tls" to specify that Transport Layer Security should be used to encode data information flow on this signaling group.
Near-end Node Name	Enter "clan" to use the CLAN interface on the S8500, which was assigned in **Figure 4**
Near-end Listen Port	Accept the default of "5061" to specify the standard TLS listen port.
Far-end Node Name	Enter "ses" to specify the SES server name assigned in **Figure 4**.
Far-end Listen Port	Accept the default of "5061" to specify the standard TLS listen port.
Far-end Domain	Enter the domain name which is configured for SES, configured in **Figure 13**.
Direct IP-IP Audio Connections	Enter "y" to specify that direct IP-IP audio connections should be used.

Table 3: Configuration Signaling Group for SIP Interface to SES

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

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







   Near-end Node Name: clan                   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: in-band-g711           Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

Figure 5: SIP Signaling-Group Form

3.1.3.3 Configure Interface to SIP Trunk

Use the **add trunk-group <x>** command, where <x> is a free trunk group number, to create a trunk group which is to be used to connect to the Avaya SES. Accept defaults for parameters, except for those which are highlighted.

Parameter	Usage
Group Type	Specify a type of “sip”.
TAC	Set the Trunk Access Code to “*83”.
Group Name	Specify “SIP” to identify this trunk. Any identifier can be used.
Service Type	Specify the trunk is used as a “tie” line to another PBX.
Signaling Group	Specify the signaling group which was configured for the sip trunk in **Figure 5**.
Number of Members	Specify a value sufficient for the maximum number of IP connections to be allowed via this trunk.

Table 4: Configuration Parameters for Trunk Interface to SIP Trunk

```
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 6: SIP Trunk-Group Form

3.1.3.4 Configure Network Region

Use the **change network-region <x>** command, where <x> is the network region used by the SIP trunk. Enter the following parameters:

Parameter	Usage
Location	Use a location of "1", in this example.
Authoritative Domain	Use a domain of "ffm.com", as configured for Avaya SES in **Figure 13**.
Name	Assign a name for identification purposes.
Intra-region IP-IP Direct Audio	Specify "y" to allow direct connections between IP endpoints.

Table 5: Configuration Parameters for Network Region

```
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: no
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3327
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 7: IP-Network-Region Form

3.1.3.5 Configure Codec Set

Use the **change ip-codec-set <x>** command, where <x> is the codec set assigned to the network region used by the SIP trunk. Enter the following parameters:

Parameter	Usage
Audio Codec	Enter "G.711A" to specify the use of the G711 A-Law codec.

Table 6: Configuration Parameters for Trunk Interface to SES

```
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 8: IP-Codec-Set Form

3.1.4. Configure Station for FreiTel-IP

Use the **add station <x>** command to allocate a station for FreiTel-IP, where <x> is the extension for FreiTel-IP shown in **Table 1**.

Parameter	Usage
Type	Enter the model identification of the phone to be used as shown in **Table 1.** Note that the TAS FreiTel telephone is configured as an Avaya 4610 IP Telephone.
Name	Enter the name of the user which is to be associated with the phone.
Security Code	Enter the security code assigned to the extension.

Table 7: Configuration Parameters IP Telephones

```
change station 69001                                           Page   1 of   4
                                     STATION

Extension: 69001                           Lock Messages? n          BCC: 0
     Type: 4610                            Security Code: xxxxxxx     TN: 1
     Port: S00009                        Coverage Path 1:            COR: 1
     Name: ext 69001                     Coverage Path 2:            COS: 1
                                         Hunt-to Station:

STATION OPTIONS
              Loss Group: 19             Personalized Ringing Pattern: 1
                                                     Message Lamp Ext: 69001
            Speakerphone: 2-way                   Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal                  Media Complex Ext:
   Survivable Trunk Dest? y                              IP SoftPhone? n




                                                  Customizable Labels? y
```

Figure 9: Station Form

3.1.5. Configure Interface to Avaya SES for FreiTel-IP

Use the **change off-pbx-telephone station-mapping** command to configure an interface to SES for the FreiTel-IP. Assign values for this command as shown in the following table.

Parameter	Usage
Station Extension	Enter the extension FreiTel-IP from **Table 1**.
Application	Enter "OPS".
Phone Number	Enter the extension FreiTel-IP from **Table 1**.
Trunk Selection	Enter the number assigned to the SIP trunk group in **Figure 6**.

Table 8: Parameters for Off-PBX-Telephone Station-Mapping

```
change off-pbx-telephone station-mapping 69001                Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk        Config
 Extension                  Prefix                       Selection    Set
 69001            OPS          -       69001             83           1
```

Figure 10: Off-PBX-Telephone Form, Page 1

3.2. 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".

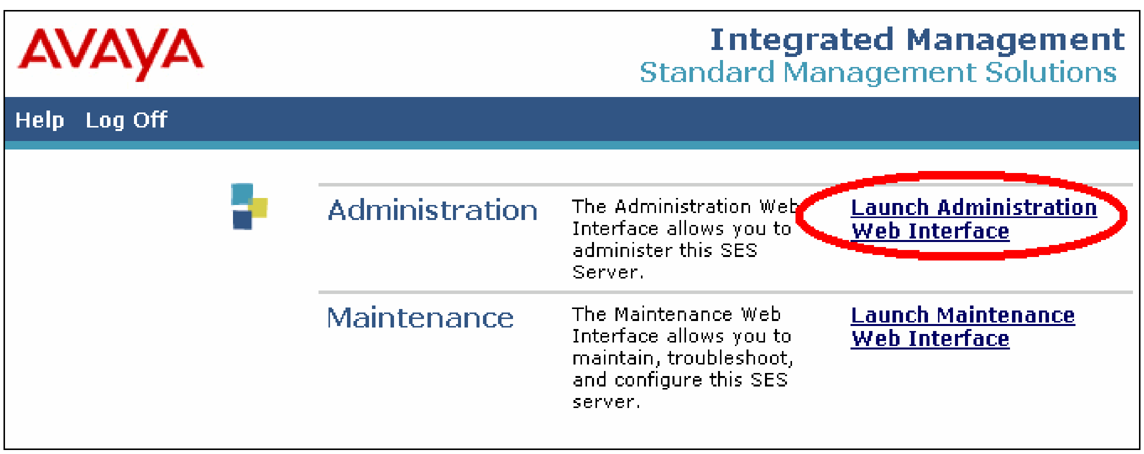

Figure 11: SES Initial Greeting Screen

The SES Integrated Management top level menu is then displayed.

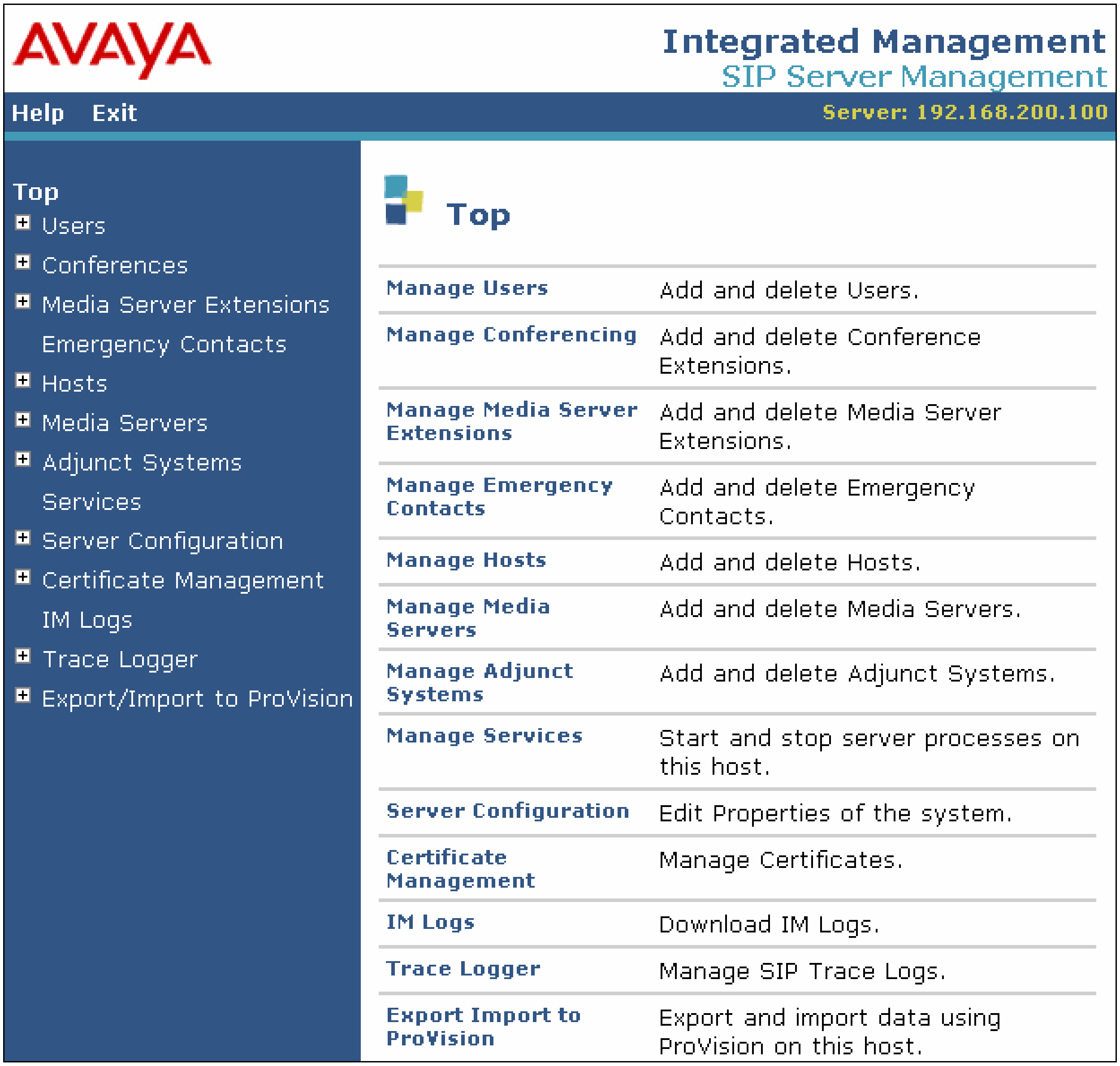

Figure 12: SES Integrated Management Top Level Menu

3.2.1. 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” that was assigned in **Figure 5**, and the IP address of the SES server which was assigned in **Figure 4** as the IP address of the “License Host”. Select the “Update” button.

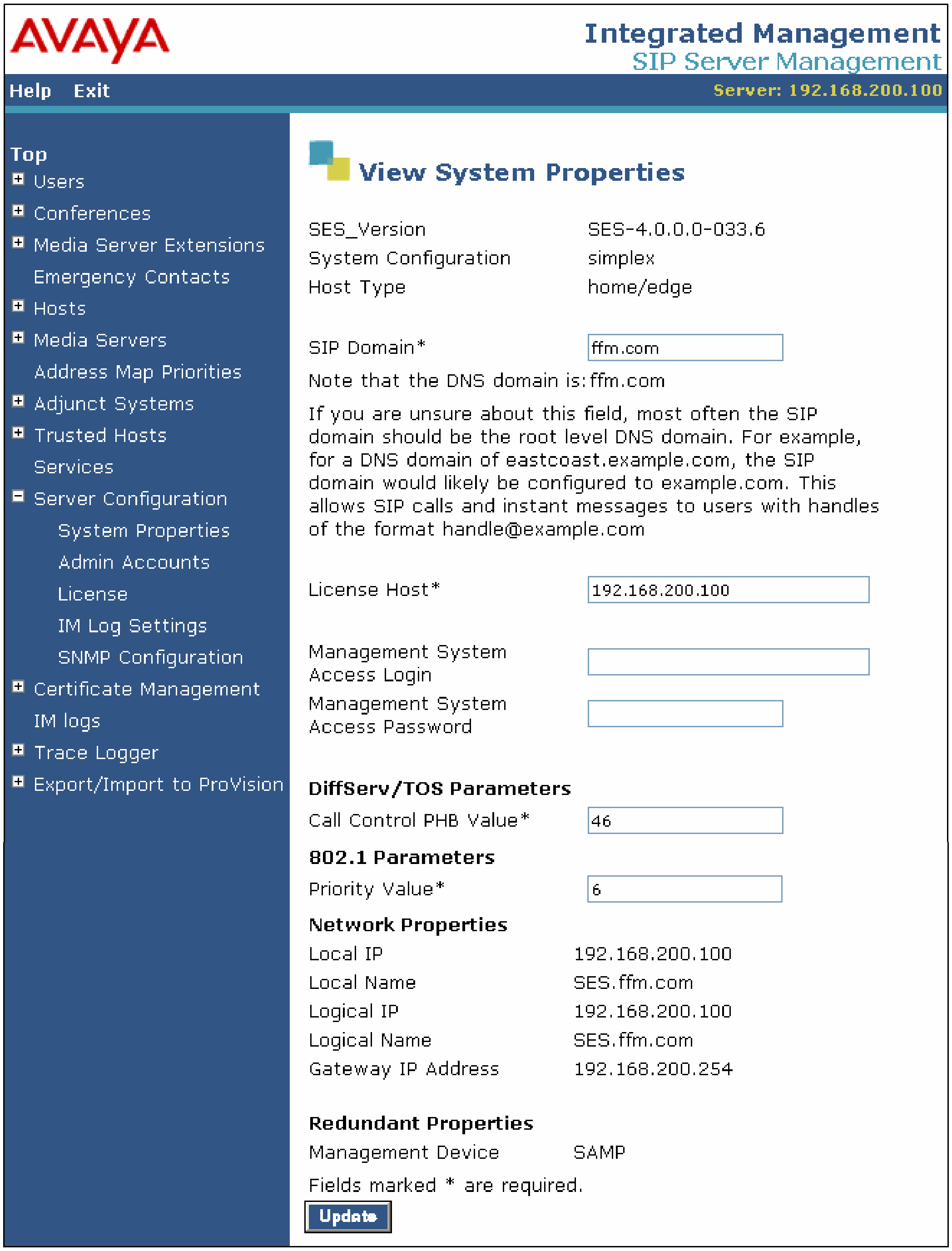

Figure 13: Avaya SES Edit System Properties Screen

From the top-level management screen, click "Manage Hosts" -> "Add Host".
Enter the **Host 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 fields assigned to their respective default values. Select the "Update" button. Note that this is a combined home/edge configuration.

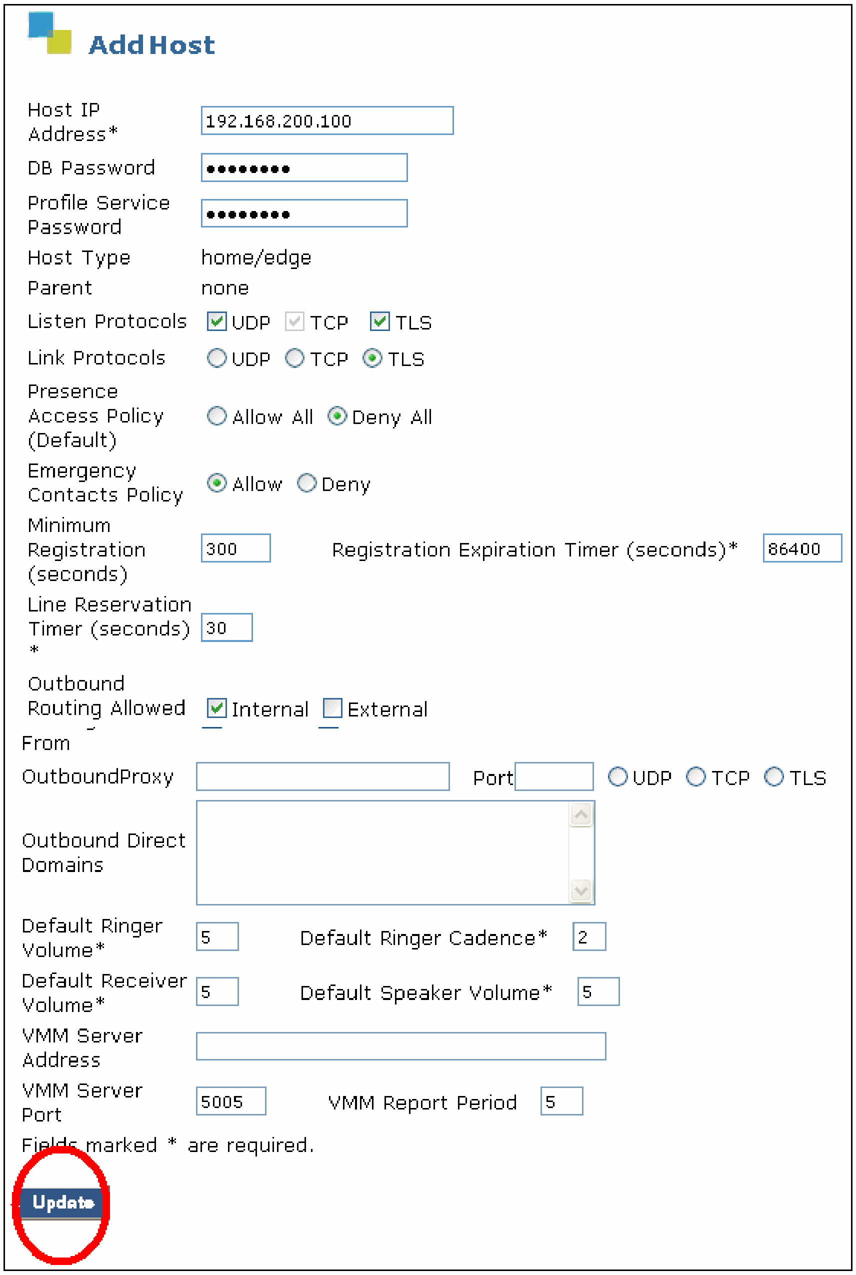

Figure 14: Avaya SES "Add Host" Screen

3.2.2. 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 from the "Host" drop-down box. Enter the address of the Avaya S8720 CLAN interface as the SIP Trunk IP Address. Select the "Add" button when these parameters have been entered.

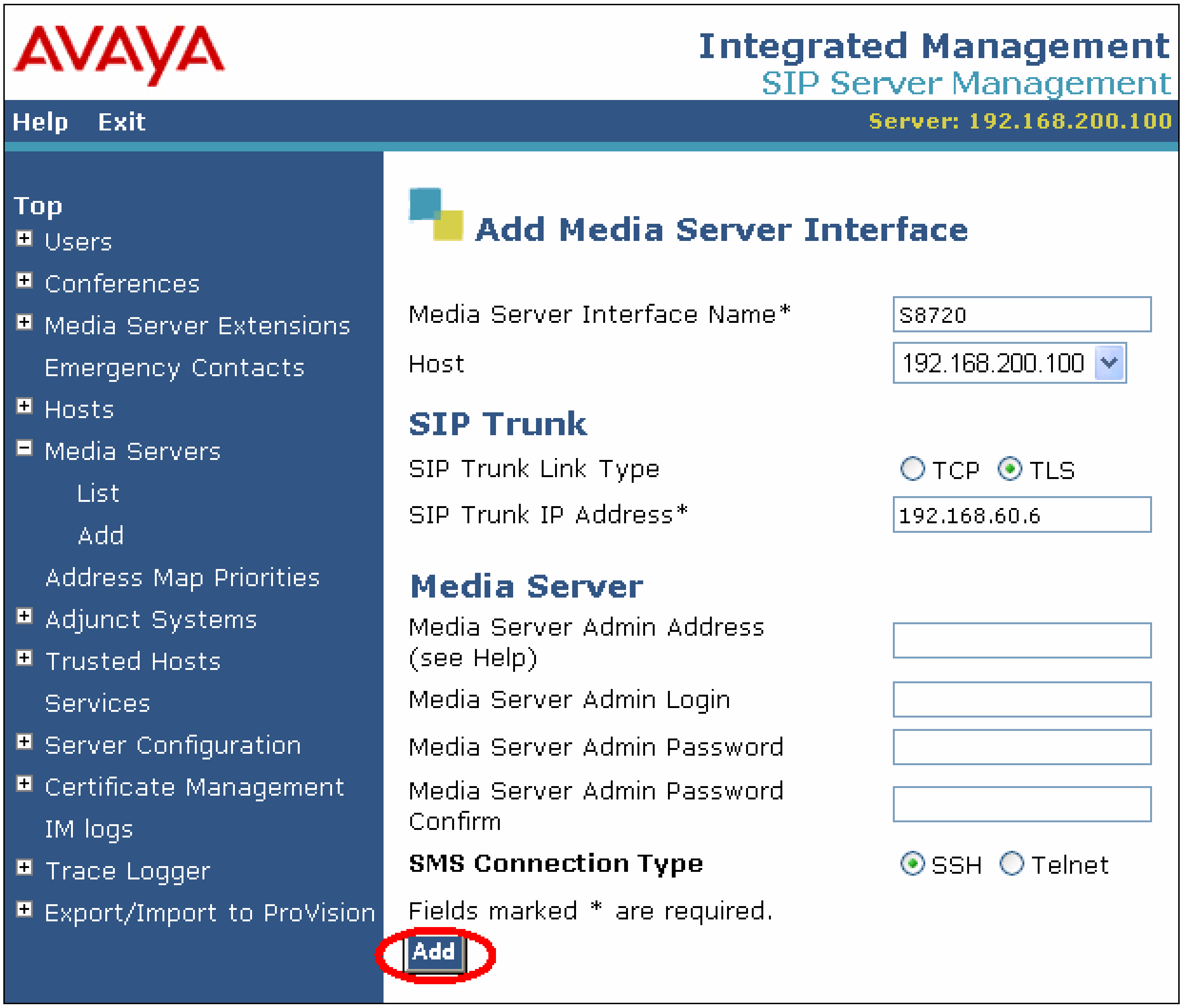

Figure 15: Avaya SES Add Media Server Interface Screen

3.2.3. Configure SIP Endpoint for FreiTel-IP

From the top level menu, select the “Manage Users” -> “Add User” menu entries. Enter the extension for FreiTel-IP as both the “Primary Handle” and the “User ID”. This is the same extension that was configured for the station in **Figure 9** and for the **off-pbx-telephone station-mapping** in **Figure 10**. Enter a **Password** and **First/Last name** of the user, check the “Add Media Server Extension” box, and click “Add”.

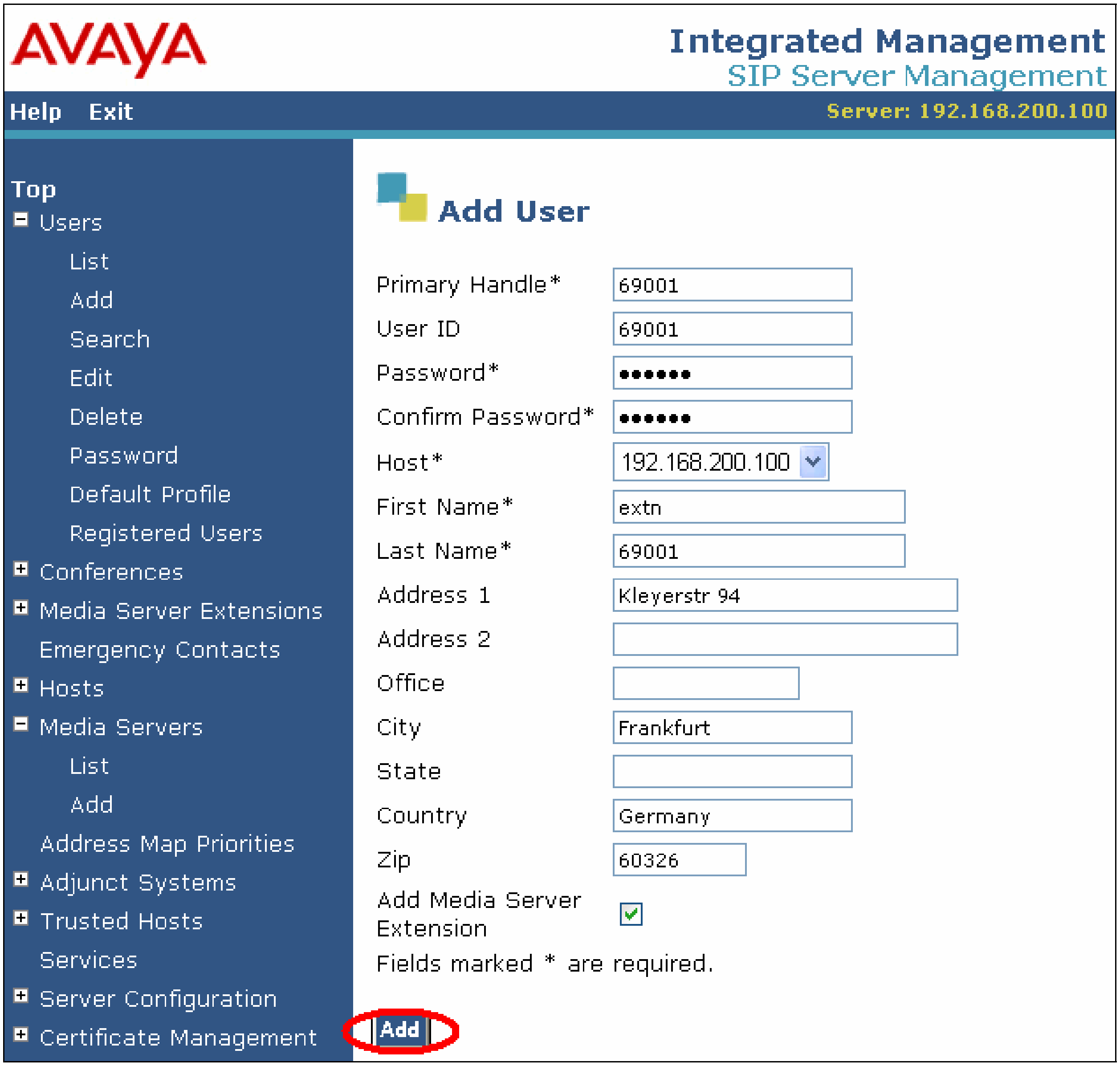

Figure 16: Avaya SES “Add User” Screen

Enter the Media Server Extension for the User ID 69001 (the extension of the Avaya SIP telephone). Select the Media Server (refer to **Figure 15**) from the drop-down box and click "Add" to continue.

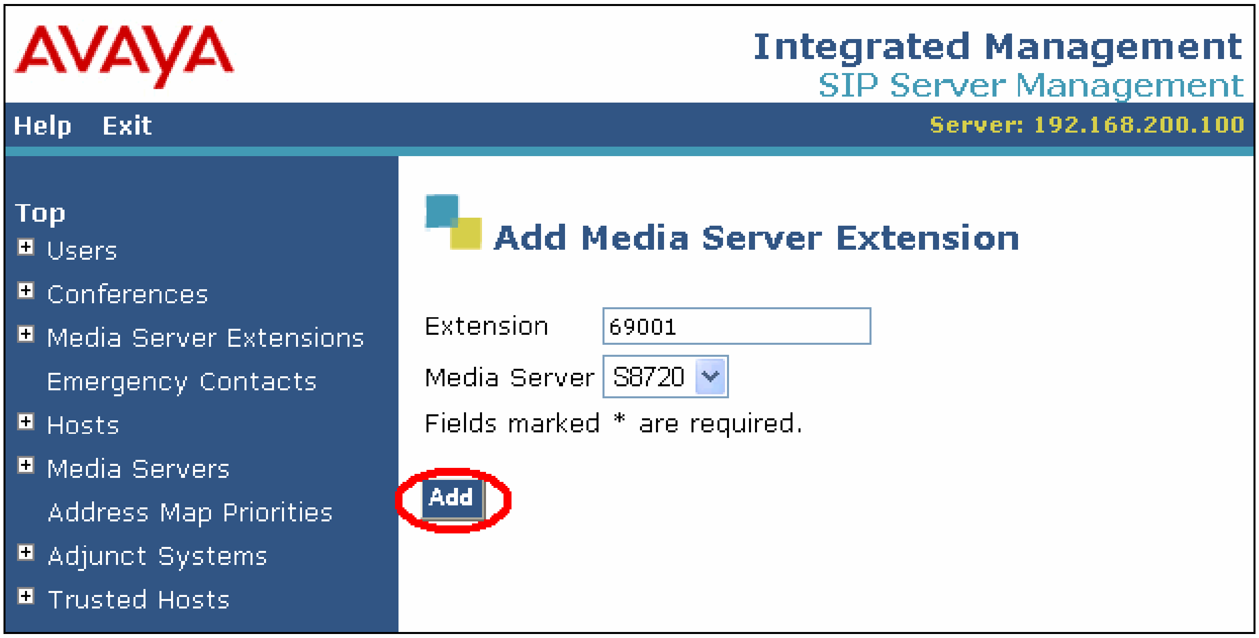

Figure 17: Avaya SES Add Media Server Extension Screen

From the main menu, click the “Update” control in the left frame to commit the changes made.

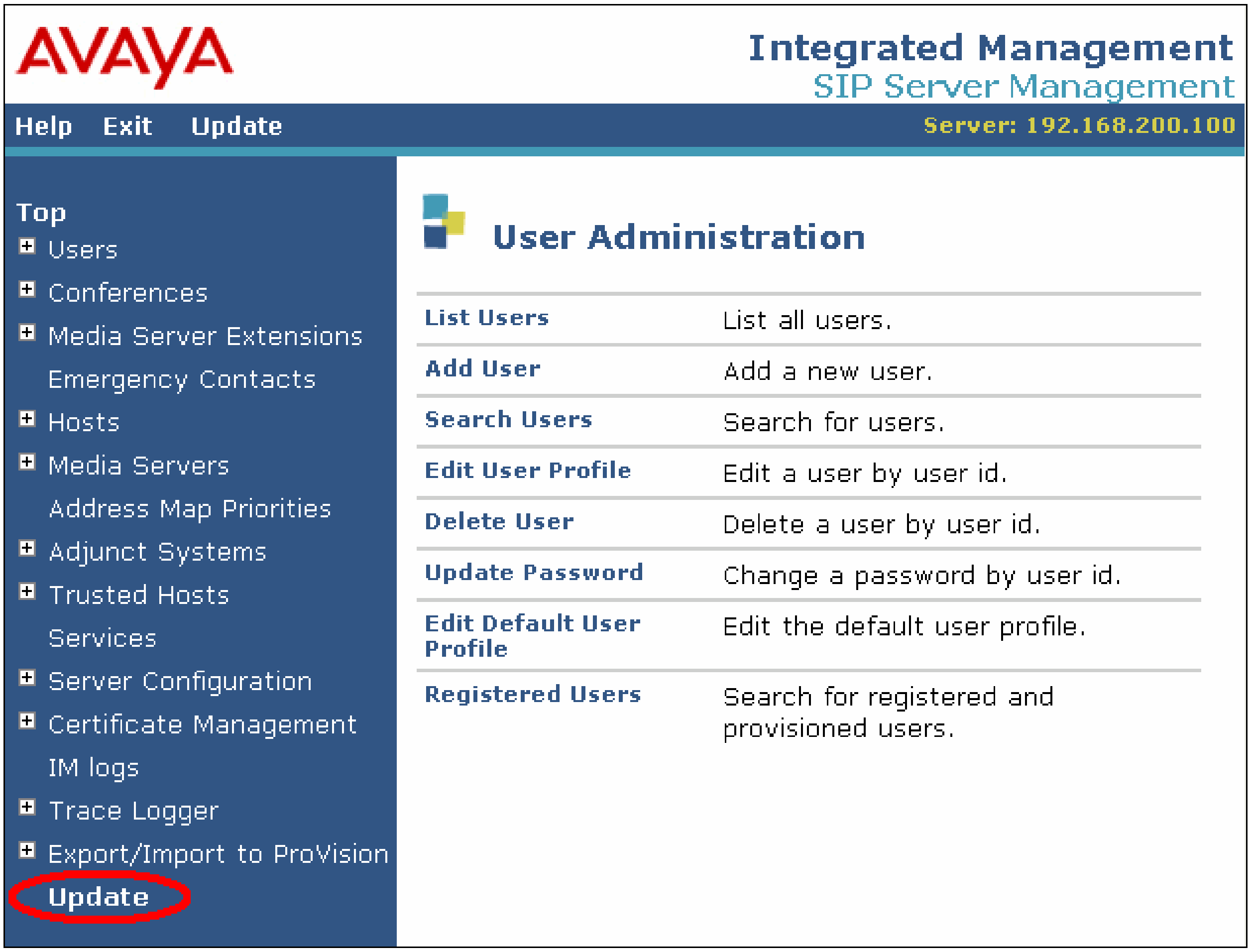

Figure 18: Update from Top SES Screen

3.3. Configure FreiTel-IP

The TAS FreiTel-IP is configured via a terminal attached to the unit's serial maintenance interface. From this terminal, it is also possible to temporarily enable a telnet interface, so that further configuration steps can be performed via a telnet session. However, the telnet interface should be disabled at completion of the configuration session for security reasons.

The screen shown below is presented after initiation of a telnet session to FreiTel-IP. Use the key codes shown in the follow table to configure the parameters which must be changed from the default values. The "Designation" column shows the designation for each of the parameters shown in **Figure 20**.

Key	Usage	Designation	Value
I	IP Address	IP Ad.	Enter the IP address to be assigned to FreiTel-IP, in this case 192.168.150.202
M	IP Mask	IP Mask.	Enter the IP mask used by the network to which the FreiTel-IP is attached, in this case "255.255.255.0".
R	Default router	Def Router Adr.	Enter the default gateway used by the network to which the FreiTel-IP is attached, in this case "192.168.150.254".
a	Call destination	RufNr. Zeil-Taste 1	Enter one of the telephone numbers from **Table 1** to designate the destination which is to be called when the "call" button on the speakerphone is pressed.
3	SIP user	SIP User	Enter the extension which is to be used by FreiTel-IP.
4	SIP domain	SIP Domain	Enter the domain which is to be used by FreiTel-IP. This must be the same value as used in **Figure 5** and **Figure 7** .
5	Authorized user	Auth. User	Enter the extension which is to be used by FreiTel-IP.
6	Authorized password	Auth. Password	Enter the password assigned in **Figure 9** and **Figure 16.**
7	SIP proxy	SIP Proxy	Enter "192.168.200.100" the address of the Avaya SES server, as shown in **Figure 4** and **Figure 14**.
9	Display name	Displ. Name	Enter a descriptive name to be used for calls made by FreiTel-IP.

Table 9: FreiTel-IP Configuration Parameters

Upon completion, click "k" to show to the configuration screen so that the parameters which were changed can be verified.

```
I   -IP Adress
M   -IP MASK
R   -Default Router

a - Zieltaste 1 Rufnummer: 600007
b - Zieltaste 2 Rufnummer:
c - Zieltaste 3 Rufnummer:

O - Option = 0 ( 1=Trenn-Taste, 2=Annahme-Taste)
o - Option Alarmoutput = $10
A - Audio-Modus (Echounterdrueckung)=0
P - Pegelschwelle EIN fuer Audiomode 7
p - Pegelschwelle Aus fuer Audiomode 7
E - Echounterdrⁿckungsfaktor
t - Audio Test Lock
h - Lauth÷rer 1-3  Vol.m - Mic. 1-3 Vol.

1   -Rtp First Port
2   -Rtp Last Port
3   -SIP User
4   -SIP Domain
5   -Auth. User
6   -Auth. Password
7   -SIP Proxy
8   -Stun Server
9   -Displ Name
0   -DNS Server


f   -CLI Filter-Nr. je Sp. setzen
F   -CLI Filter setzen

k   -Show Konfig.
s   -Status
D   -Debug
T  - Temperatur Init
w  - Audio Test manuell
u   -Update Url
l   -Logout
```

Figure 19: FreiTel-IP Introductory Screen

```
----------------- IP ----------------------
IP  Adr.         192.168.150.202
IP Mask.         255.255.255.0
Def Router Adr.  192.168.150.254

Update URL:      it
 ---------------- SIP ----------------------
Rtp First Port:
Rtp Last Port:
SIP User:        69001
SIP Domain:      ffm.com
Auth. User:      69001
Auth. Password:  xxxxx
SIP Proxy:       192.168.200.100
Stun Server:
Displ.Name:      69001
DNSServer:
 ---------------- Applikation --------------
CLI Filter[1]:
CLI Filter[2]:
CLI Filter[3]:

Filter Nr[1]:
Filter Nr[2]:
Filter Nr[3]:
Filter Nr[4]:
Filter Nr[5]:
Filter Nr[6]:
Filter Nr[7]:
Filter Nr[8]:
RufNr.Ziel-Taste 1  = 60007
RufNr.Ziel-Taste 2  =
RufNr.Ziel-Taste 3  = 0k
 ---------------- Audio --------------------
Lautspr.1 Vol (0-63)= 60
Lautspr.2 Vol (0-63) = 60
Lautspr.3 Vol (0-63) = 60
Mic.1 Vol (0-63)    = 63
Mic.2 Vol (0-63)    = 59
Mic.3 Vol (0-63)    = 59
Audio Test Lock 1,2,4
Audio Mode 0
Echo Absenkfaktor 0 0 0
Echo Pegelschwelle-Ein in Mode 7: 0 0 0
Echo Pegelschwelle-Aus in Mode 7: 0 0 0
Option: 3 Alarm Option: 10
```

Figure 20: FreiTel-IP Configuration Screen

4. Interoperability Compliance Testing

The objective of the compliance testing performed on the TAS FreiTel-IP product was to verify that it is compatible with Avaya Communication Manager. This includes verifying that the essential FreiTel-IP features function properly when used with Avaya Communication Manager and that Avaya Communication Manager features are not hindered by the interaction with FreiTel-IP. Furthermore, FreiTel-IP's robustness was verified.

4.1. General Test Approach

The test method employed can be described as follows:

- Avaya Communication Manager was configured to support various local telephones and the PSTN.
- The individual features of the FreiTel-IP were tested by manually making calls to and from the unit.
- FreiTel-IP's robustness was tested by verifying its ability to recover from interruptions its external LAN.
- FreiTel-IP's robustness was further tested by verifying the ability to recover from power interruptions to the FreiTel-IP endpoint.

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

4.2. Test Results

The following capabilities of the FreiTel-IP were tested for proper interoperation with Avaya Communication Manager:

- Incoming call with and without media stream shuffle
- Outgoing call with and without media stream shuffle
- Manual call acceptance
- Incoming call filtering

The only problem which was encountered during testing was that calls between the Avaya SIP telephone and the FreiTel-IP which were configured to shuffle did not do so. This is a minor problem which does not affect the ability of FreiTel-IP to interoperate with Avaya Communication Manager. Calls between Avaya H.323 telephones and the FreiTel-IP which were configured to shuffle did so correctly.

5. Verification Steps

The following steps can be performed to verify the correct installation and configuration of FreiTel-IP:

- Verify that the Avaya SES and FreiTel-IP systems can ping each other.
- Verify that the various telephones can call each other.
- Verify that it is possible to initiate calls from the FreiTel-IP speakerphone call button.
- Verify that that FreiTel-IP can be configured to accept incoming calls both automatically and manually.

6. Support

Support for FreiTel-IP is available at:

TAS GmbH & Co.KG
Langmaar 25
41238 Mönchengladbach
Phone: +49 2166 858 0
Fax: +49 2166 858 150
Email: info@tas.de
http://www.tas.de

7. References

[1] "Feature Description and Implementation for Avaya Communication Manager", 555-245-205, Issue 3, June 2005

[2] "Administrator Guide for Avaya Communication Manager", 03-300509, Issue 1, June 2005

[3] "Installing and Administering SIP Enablement Services R3.1.1", 03-600768, Issue 2.0, August 2006

[4] "SIP Support in Release 3.1 of Avaya Communication Manager", 555-245-206, Issue 6, February 2006

[1] "FreiTel-IP Konfiguration", July 25, 2006, Version 1.1 (German)

8. Conclusion

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

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