Avaya Solution & Interoperability Test Lab

Application Notes for TAS FEP-IP Flush-mounting Telephone with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the conformance testing of the TAS FEP-IP Flush-mounting SIP telephone with Avaya Communication Manager. These Application Notes contain an extensive description of the configurations for both FEP-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.

Table of Contents

1. Introduction

The FEP-IP is a SIP telephone for use in a high-availability industrial environment. It consists of a control unit, handset, optical incoming call indicator, optical missed call indicator, optional keypad, and optional display. The FEP-IP which was used for testing had a keypad, but no display. The FEP-IP can optionally be connected to a BRI trunk which is used to make calls if the SIP connection is unavailable for some reason. In this case, a warning LED on the FEP-IP flashes until the SIP connection becomes available again.

The FEP-IP can do speed dialing via configurable shortcodes, and has a callback function which can be used to dial the last missed call. The FEP-IP supports in-band DTMF dialing.

The FEP-IP without keypad calls a preconfigured number when it goes off-hook. This configuration was not tested.

The TAS FEP-IP Flush-mounting SIP telephone is attached to the Avaya S8720 Server on which Avaya Communication Manager resides via a SIP trunk to Avaya SIP Enablement Services (SES) which is attached via a SIP trunk to the Avaya G650 Media Gateway. The Avaya G650 Media Gateway uses a PRI connection to complete calls to the PSTN which originate from the TAS FEP-IP SIP interface. Should the FEP-IP SIP interface for some reason become unavailable, the FEP-IP uses its direct Basic Rate Interface (BRI) interface to complete calls to the PSTN until the SIP interface becomes available again. Note that the Avaya SES used for testing was configured as a combined home/edge server.

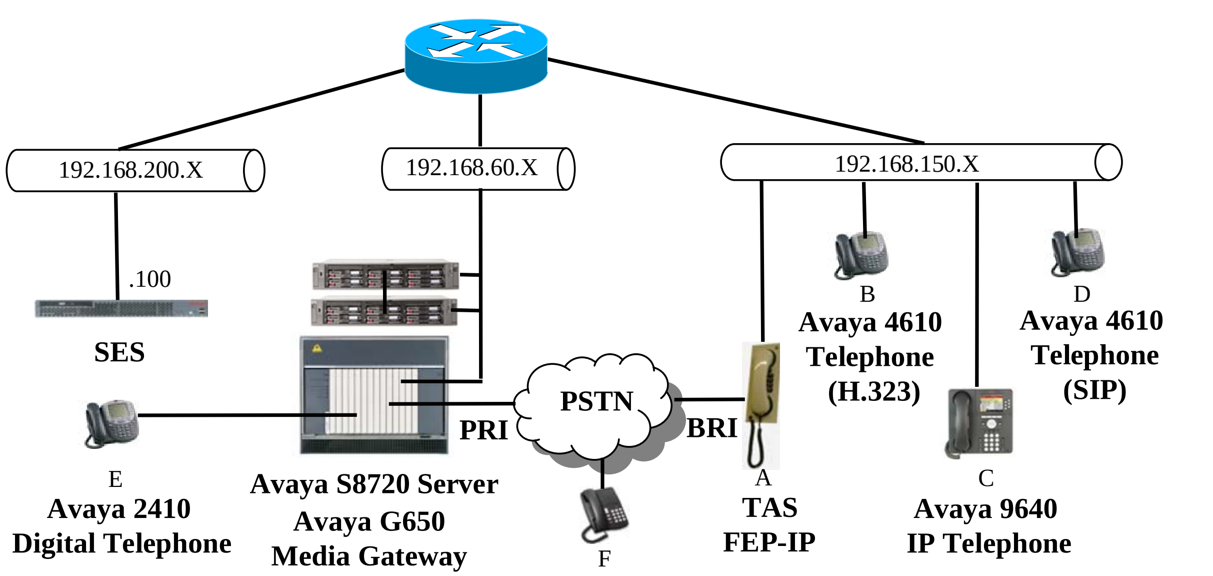

Figure 1: FEP-IP Test Configuration

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 FEP-IP
B	60116	069 9073 9887 60116	Avaya 4610 Telephone (H.323)
C	60093	069 9073 9887 60093	Avaya 9640 one-X Deskphone
D	60113	069 9073 9887 60113	Avaya 4610 Telephone (SIP)
E	60007	069 9073 9887 60007	Avaya 2410 Digital Telephone
F		069 7505 6176	ISDN telephone

Table 1: Extensions Used for Testing

2. Equipment and Software Validated

Equipment	Software Version
Avaya S8720 Server with Avaya Communication Manager	4.0.1 (R014x.00.0.731.2)
Avaya SIP Enablement Services Server	4.0 (SES-4.0.0.0-33.6)
Avaya 2410 Digital Telephone	5.0
Avaya 4610 H.323 Telephone	2.8.3
Avaya 4610 SIP Telephone	2.2.2
Avaya 9640 H.323 Telephone	1.5
TAS FEP-IP	3.3

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).

The configuration of the PRI trunk to the PSTN is outside the scope of these Application Notes.

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 FEP-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” as 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                 192.168.100.200                       .   .   .
```

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 S8720 .
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.
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. Note that the FEP-IP cannot establish a direct IP-IP connection to an Avaya SIP telephone. However, this works correctly between the FEP-IP and Avaya H.323 telephones. Thus, this parameter can be set to "yes" to enable direct IP-IP connections to Avaya H.323 telephones and to "no" for configurations which only have Avaya SIP telephones. For a mixed configuration of Avaya H.323 and SIP telephones, this parameter should be set to "no".

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 FEP-IP

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

Parameter	Usage
Type	Enter the model identification of the phone to be used as shown in **Table 1.** The FEP-IP was configured as an Avaya 4610 telephone, which has sufficient features to provide correction operation of the FEP-IP.
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

```
add 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 FEP-IP

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

Parameter	Usage
Station Extension	Enter the extension FEP-IP from **Table 1**.
Application	Enter "OPS".
Phone Number	Enter the extension FEP-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".

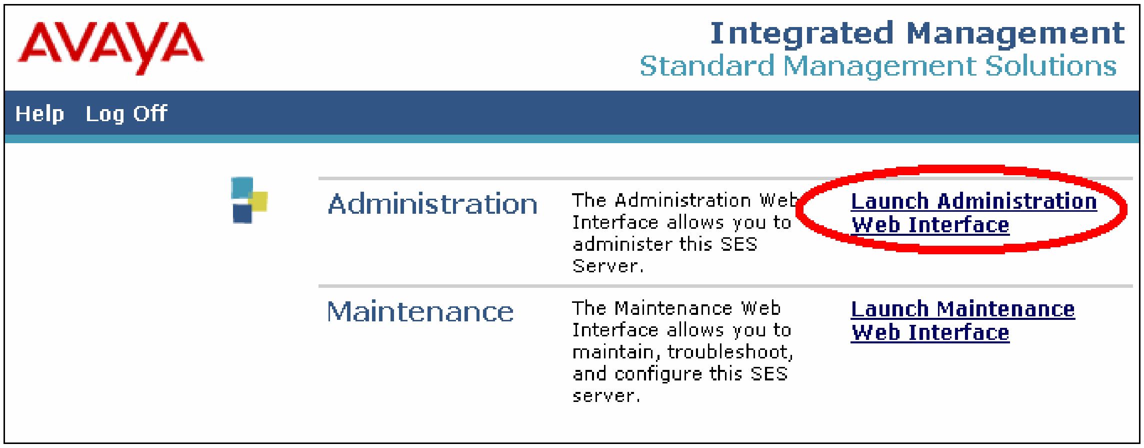

Figure 11: SES Initial Greeting Screen

The SES Integrated Management top level menu is then displayed.

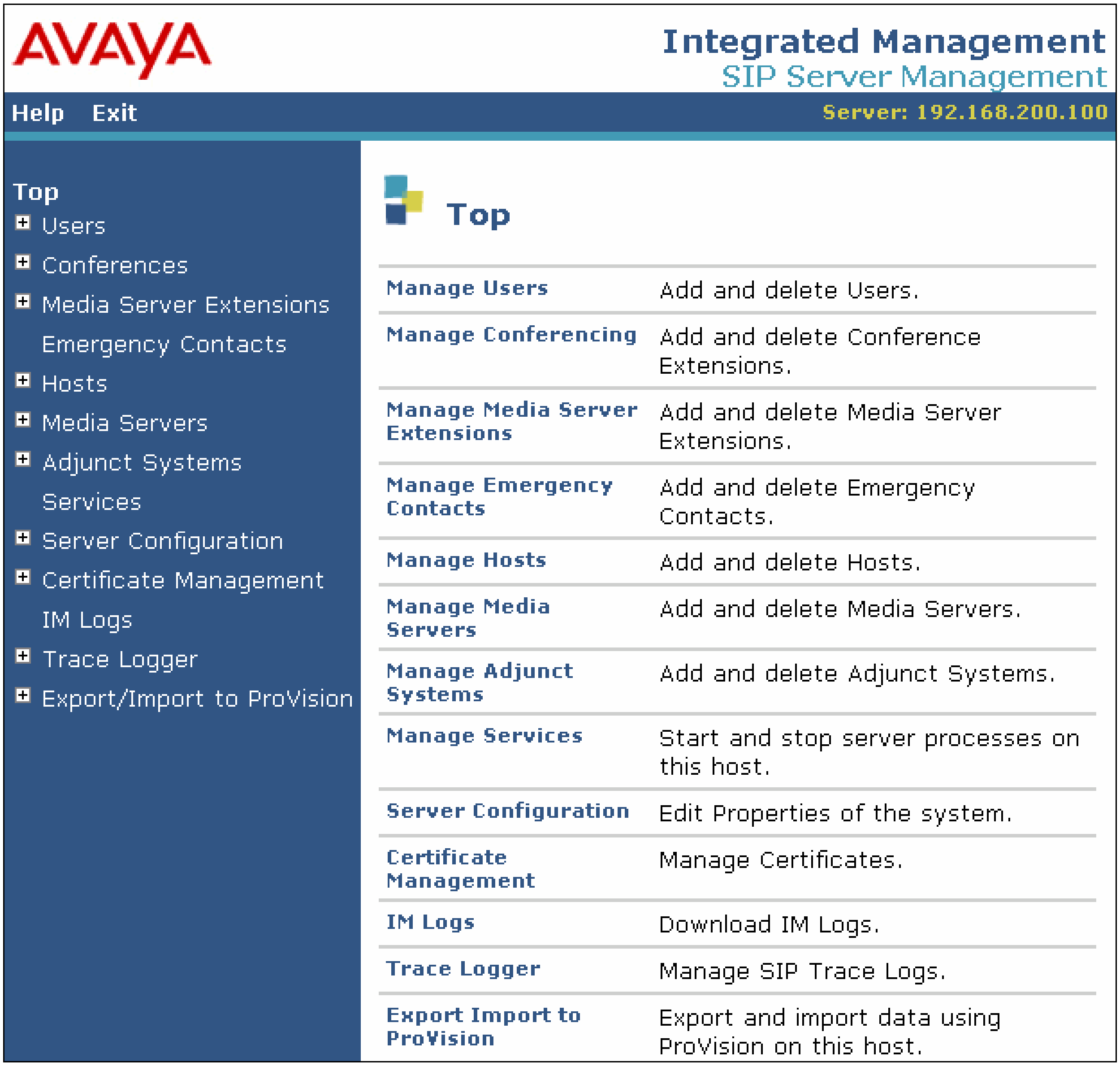

Figure 12: SES Integrated Management Top Level Menu

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.

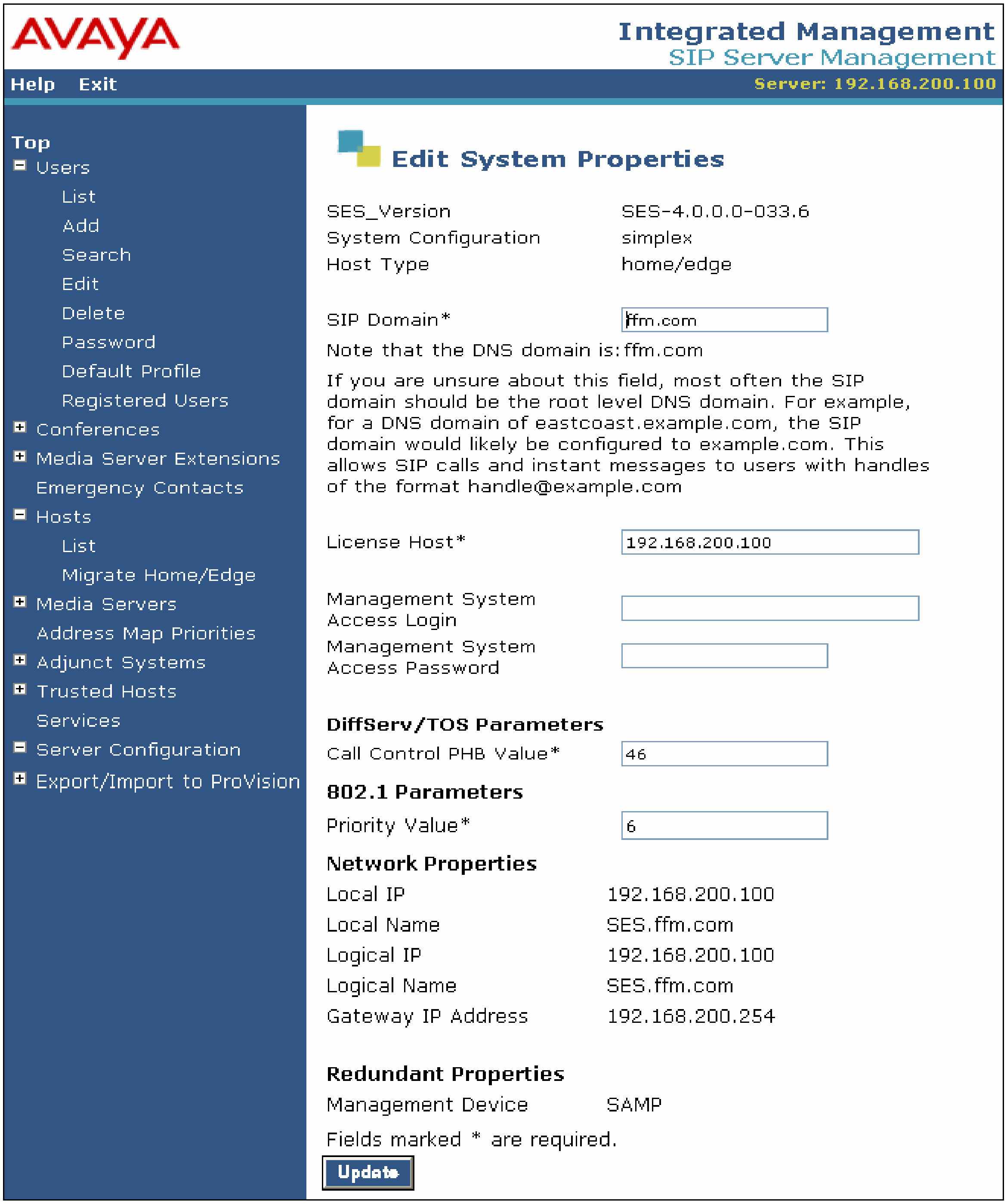

Figure 13: Avaya SES Edit System Properties Screen

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 the Avaya SES used for testing was configured as a combined home/edge server.

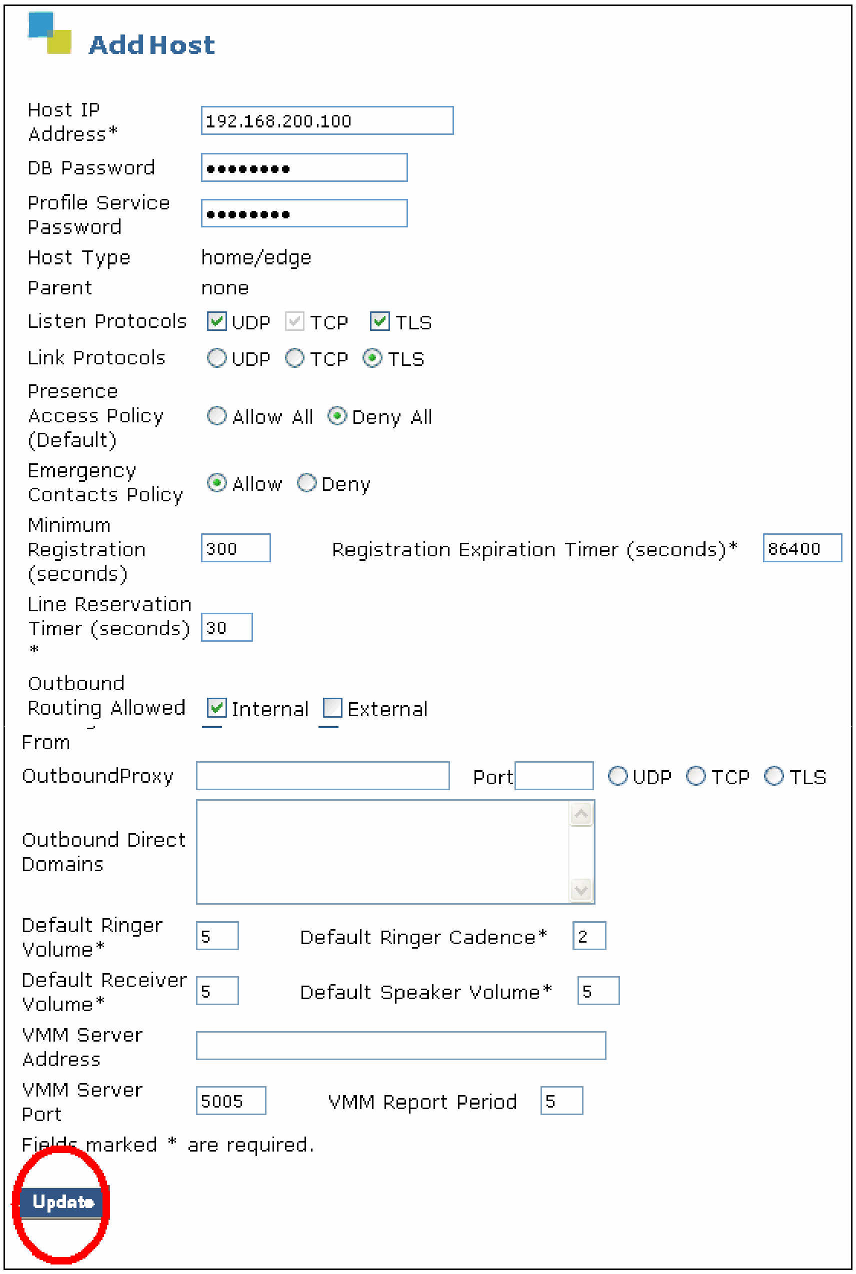

Figure 14: Avaya SES "Add Host" Screen

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.

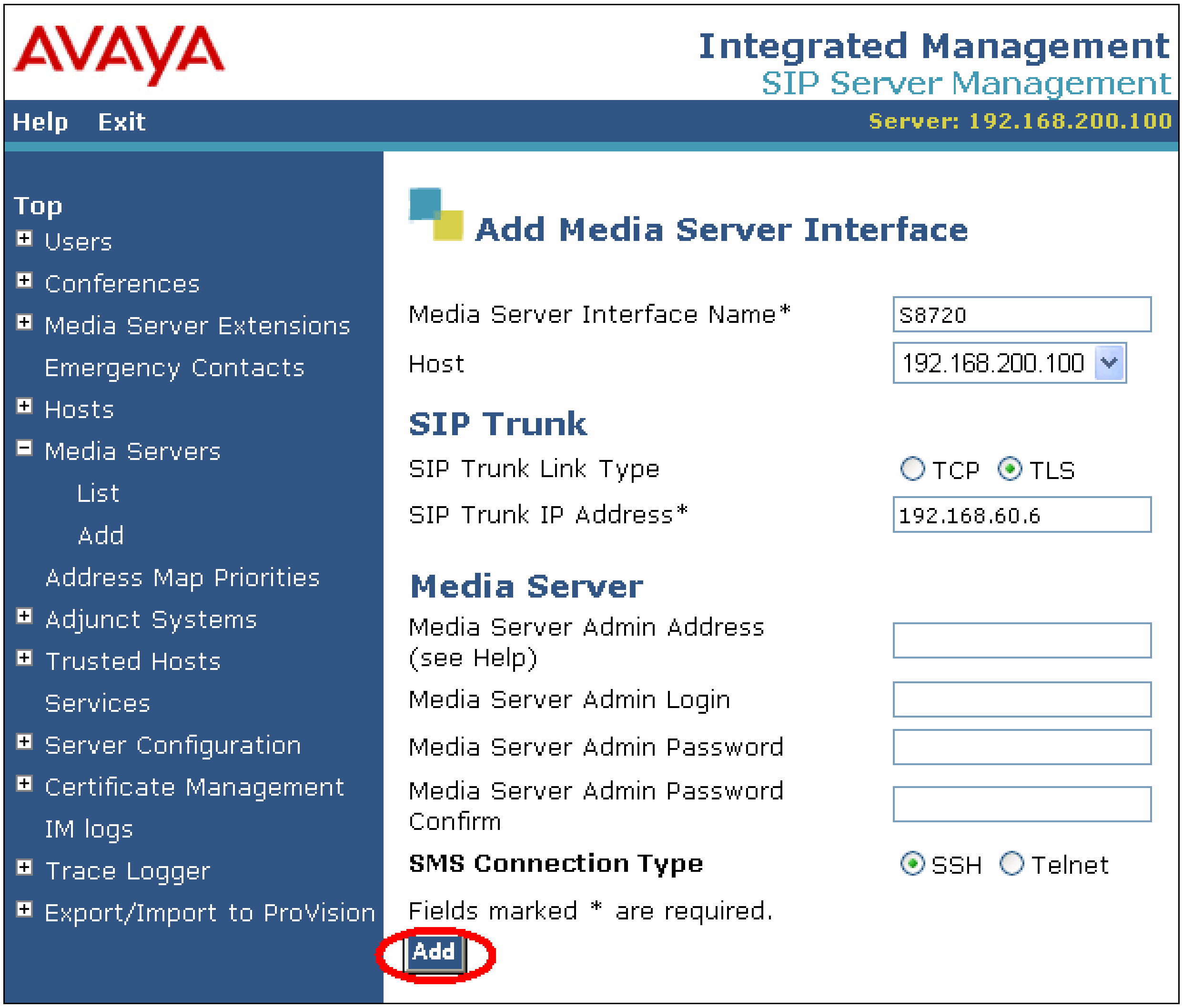

Figure 15: Avaya SES Add Media Server Interface Screen

3.2.3. Configure SIP Endpoint for FEP-IP

From the top level menu, select the "Manage Users" -> "Add User" menu entries. Enter the extension for FEP-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 SIP stations configured in **Section 3.1.5**. Enter a **Password** and **First/Last name** of the user, check the "Add Media Server Extension" box, and click "Add".

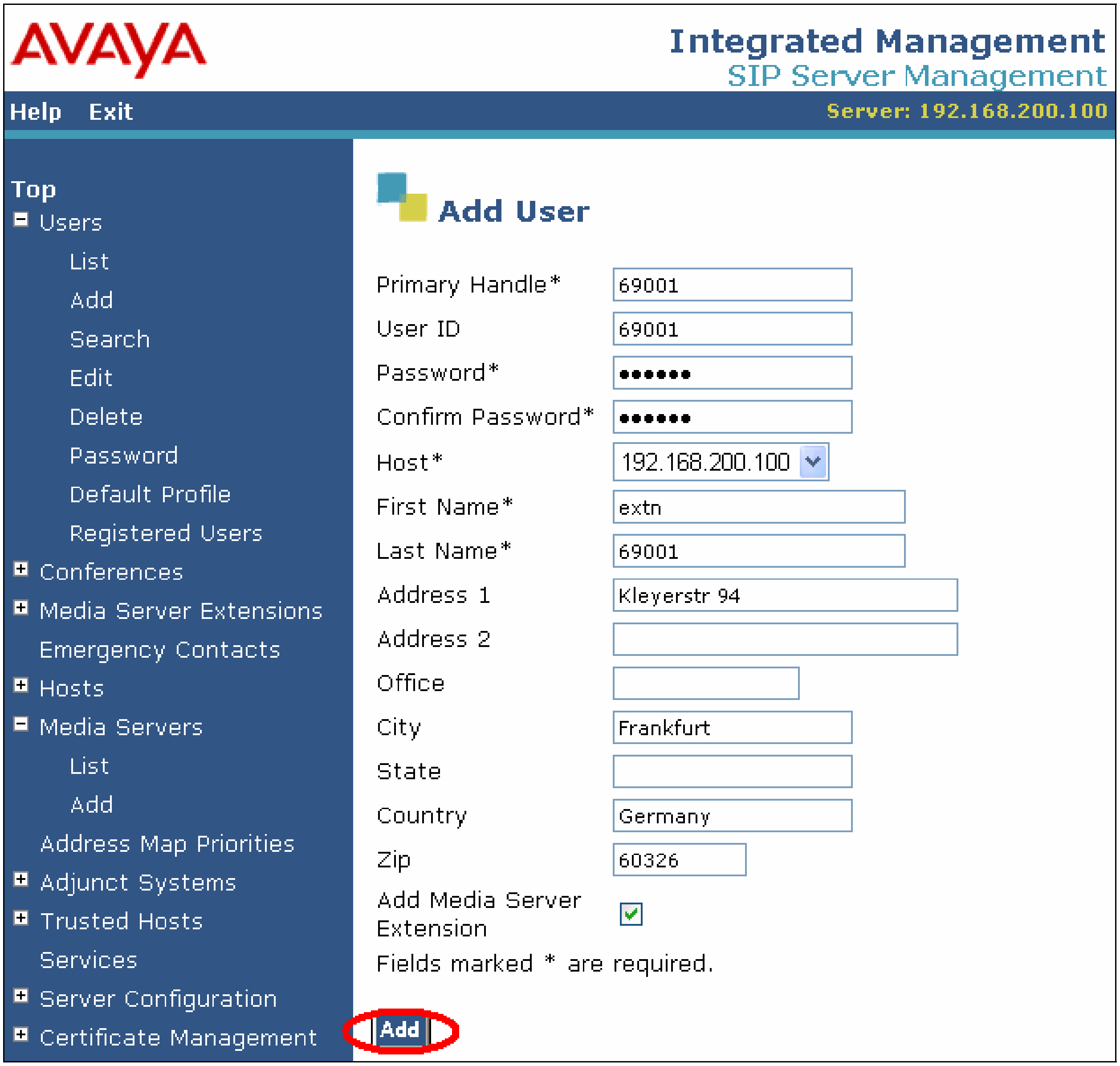

Figure 16: Avaya SES "Add User" Screen

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

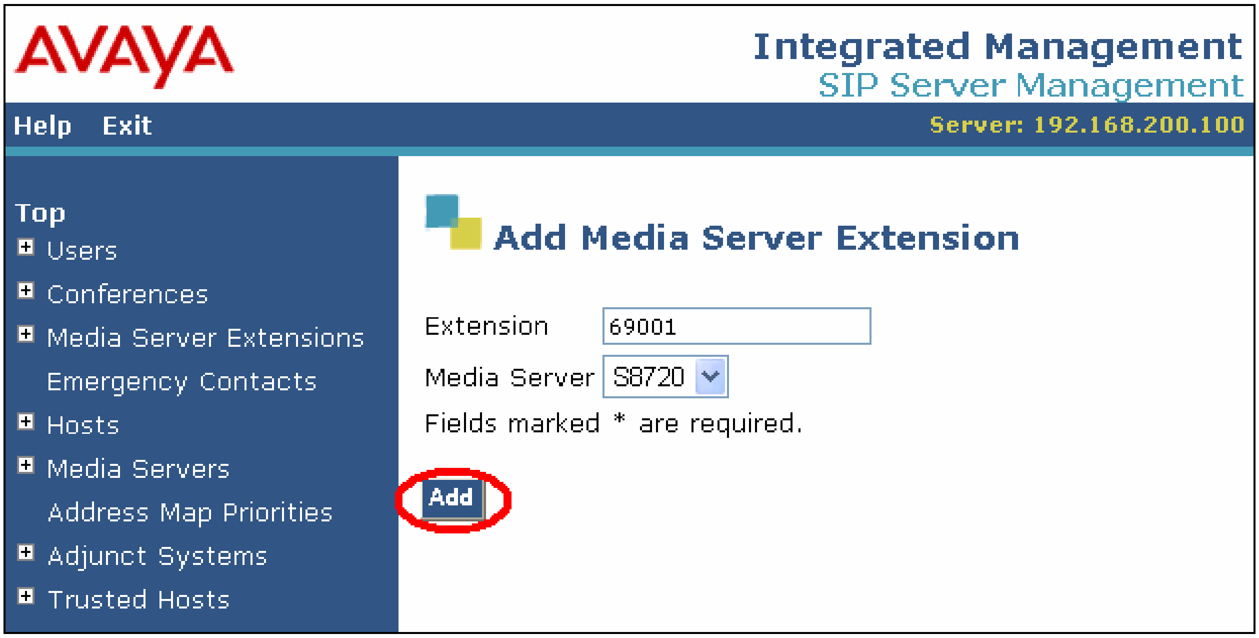

Figure 17: Avaya SES Add Media Server Extension Screen

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

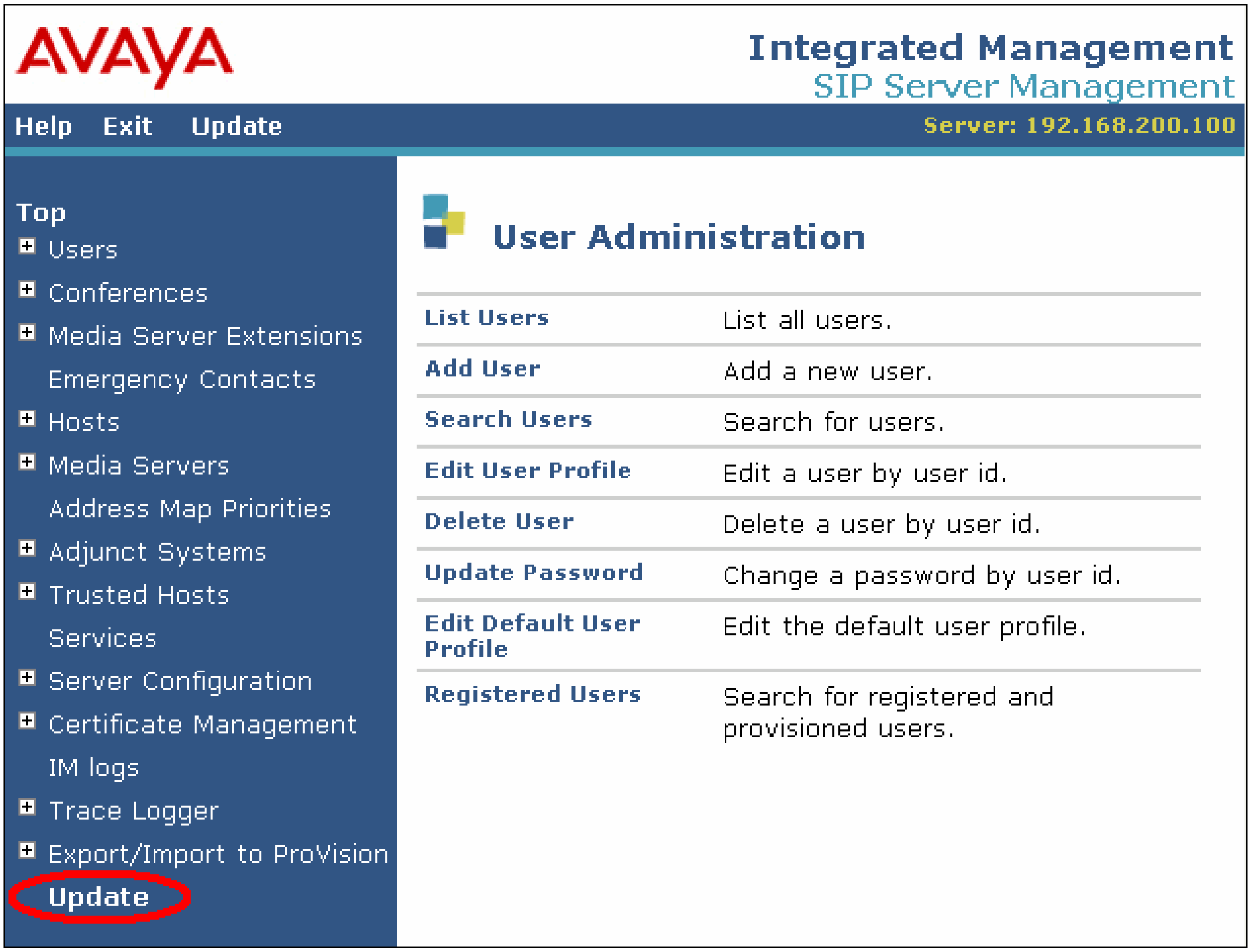

Figure 18: Update from Top SES Screen

3.3. Configure FEP-IP

The TAS FEP-IP can be configured via one of the following interfaces:

- Telephone keypad
- Console attached to serial interface
- Telnet console
- Console attached remotely via ISDN interface

The telephone keypad and serial interface are both active when the device is in its initial un-configured state, and either the telephone keypad or a terminal attached to the serial interface can be used to activate the Telnet or remote ISDN interface. However, the telnet interface should be disabled at completion of the configuration session for security reasons. The operation of the ISDN remote console interface is described in **Section 7 [4]**.

The configuration of the FEP-IP SIP interface to SES is described within these Application Notes. Additional configuration steps required to configure the ISDN telephone interface and other features of FEP-IP are described within **Section 7 [4]**.

3.3.1. Keypad Input Simulation from Telnet or Serial Console

Input from a serial or telnet console can be used to simulate telephone activity, such as on/off-hook actions or keypad input. The layout of the FEP-IP keypad is show in **Figure 19**.

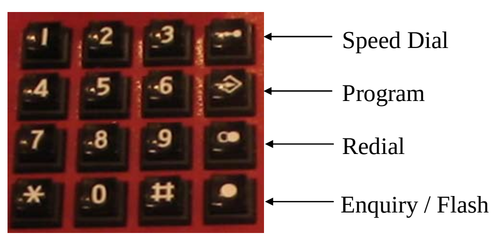

Figure 19: FEP-IP Keypad Layout

Console keys are mapped to telephone actions and keypad input as shown in **Table 9**.

Console Key	Telephone Key or Action
"0" – "9"	Dial keys for digits "0" – "9"
"*"	Dial key "*"
"r"	Enquiry / Flash (Ground key)
"a"	Lift handset
"e"	Replace handset
"s"	Speed dial key
"p"	Program key
"w"	Redial key
"S"	SIP Menu

Table 9: Console Keyboard Mapping

Input from either the telephone keypad or console keyboard can be used to configure the FEP-IP, although some configuration steps can only be performed via a console.

3.3.2. Configuration of the SIP Interface via Telephone Keypad

The FEP-IP can have up to eight ISDN Multiple Subscriber Numbers (MSNs). For the purpose of configuration, the SIP interface is treated as if it were a ninth MSN. The following four configuration settings can be made via the telephone keypad or via one of the console interfaces:

- IP address
- Subnet mask
- Default router
- Telnet access

Thus, it is possible to change the above parameters affecting the IP interface via the telephone keypad so the telnet interface can be used for subsequent changes. If changes need to be made to configuration parameters other than one of the above, one of the console interfaces must be used for this purpose.

Use the procedure described in **Table 10** to set one of these parameters. Note that step 6 of this table requires that one of the procedures described in **Table 11** to **Table 14** must be performed dependent on which configuration parameter is to be set. Any configuration changes will only take affect after the FEP-IP has been restarted.

Step	Action	LED	Display (optional)
1	Lift the handset		
2	Push the “program” key.	RED	“Programme Nr.”
3	Push the digit key “1”.		“Setup MSN – Enter PIN ####”
4	Enter PIN “1590” using the keypad.	GREEN	
5	Enter “9”, to select SIP.	RED	
6	Perform one of the actions described in **Table 11** to **Table 14,** dependent on which configuration parameter is to be set.		
7	Replace the handset.		

Table 10: Procedure for Setting Parameters via Telephone Keypad

Step	Action	LED	Display (optional)
1	Enter “1”.		Previously configured IP address
2	Enter the IP address as a 12-digit number, including any leading zeroes.		
3	Push the “program” key.	GREEN	

Table 11: Procedure for Setting IP Address

Step	Action	LED	Display (optional)
1	Enter “2”.		Previously configured IP mask
2	Enter the IP mask as a 12-digit number, including any leading zeroes.		
3	Push the “program” key.	GREEN	

Table 12: Procedure for Setting Network Mask

Step	Action	LED	Display (optional)
1	Enter “3”.		Previously configured default gateway address.
2	Enter the default gateway address as a 12-digit number, including any leading zeroes.		
3	Push the “program” key.	GREEN	

Table 13: Procedure for Setting Default Gateway

Step	Action	LED	Display (optional)
1	Enter “4”.		Previously configured telnet access status: “0” or “1”
2	Enter “0” to disable or “1” to enable telnet access.		
3	Push the “program” key.	GREEN	

Table 14: Procedure for Configuring Telnet Access

3.3.3. Configuration of the SIP Interface Console

After initiation of a console session, the screen shown in **Figure 20** is presented after the "S" key is depressed. Use the key codes shown in **Table 15** 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 21**. Note that the "I", "M", and "R" parameters can also be changed via the telephone keypad as described in **section 3.3.2**, so that the IP interface can be configured so that telnet can be used for subsequent configuration operations.

```
SIP Konfiguration
Ret -Hauptmenü
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
r   -Regist. Time
A   -Option Adresse
D   -Option Destination
T   -Option Repeat Time
c   -Show Config
u   -Update Url
I   -IP Adress
M   -IP MASK
R   -Default Router
t   -TCP Zugangslock toggeln
P   -Sip Parameter (Refer,Autoanswer)
l   -Logout, Reset Parameter Update
```

Figure 20: SIP Configuration Commands for FEP-IP

Key	Designation	Value
I	IP Ad.	Enter the IP address to be assigned to FEP-IP
M	IP Mask.	Enter the IP mask used by the network to which the FEP-IP is attached, in this case “255.255.255.0”.
R	Def Router Adr.	Enter the default gateway used by the network to which the FEP-IP is attached, in this case “192.168.150.254”.
3	SIP User	Enter the extension which is to be used by FEP-IP.
4	SIP Domain	Enter the domain which is to be used by FEP-IP. This must be the same value as used in **Figure 5** and **Figure 7** .
5	Auth. User	Enter the extension which is to be used by FEP-IP.
6	Auth. Password	Enter the password assigned in **Figure 9** and **Figure 16.**
7	SIP Proxy	Enter “192.168.200.100” the address of the Avaya SES server, as shown in **Figure 4** and **Figure 14**.
9	Displ. Name	Enter a descriptive name to be used for calls made by FreiTel-IP.
c	Show config	Display the current values of the configuration parameters, as shown in **Figure 21**.
r	Registry Time	Enter the registry – repeat time in sec. (>=300)

Table 15: FEP-IP SIP Configuration Parameters

Enter “c”:

```
Rtp First Port:
Rtp Last Port:
SIP User:       69001
SIP Domain:      ffm.com
Auth. User:      69001
Auth. Password:  100960
SIP Proxy:       192.168.200.100
Stun Server:
Displ.Name:      69001
Sip Reg. Time:   300
DNSServer:
IP  Adr.         192.168.150.202
IP Mask.         255.255.255.0
Def Router Adr.  192.168.150.254
Opt. Adr:
Opt.Destin.:

Opt.Rep. Time:   0 Minuten
SipVermittelung  0 AutoAnswer 2
TCP Lock 0

Update URL:
LAN State = 1, SIP Register State = 0  SIP RegisterSend 0 Sip OptionResp: 0
```

Figure 21: SIP Configuration Status for FEP-IP

4. Interoperability Compliance Testing

The objective of the compliance testing performed on the TAS FEP-IP product was to verify that it is compatible with Avaya Communication Manager. This includes verifying that the essential FEP-IP features function properly when used with Avaya Communication Manager, and that Avaya Communication Manager features are not hindered by the interaction with FEP-IP. Furthermore, FEP-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 FEP-IP were tested by manually making calls to and from the unit.
- FEP-IP's robustness was tested by verifying its ability to recover from interruptions its external LAN.
- FEP-IP's robustness was further tested by verifying the ability to recover from power interruptions to the FEP-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 FEP-IP were tested for proper interoperability 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 an Avaya SIP telephone and the FEP-IP which were configured to shuffle did not do so. This is a minor problem which does not affect the ability of FEP-IP to interoperate with Avaya Communication Manager. Calls between Avaya H.323 telephones and the FEP-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 FEP-IP:

- Verify that the Avaya SES and FEP-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 FEP-IP speakerphone call button.
- Verify that that FEP-IP can be configured to accept incoming calls both automatically and manually.

6. Support

Support for FEP-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] *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] "FEP-IP User / Setup Manual", Version 3

8. Conclusion

These Application Notes describe the conformance testing of the TAS FEP-IP handsfree telephone with Avaya Communication Manager and Avaya SES. The various features of the FEP-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 FEP-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.