Avaya Solution & Interoperability Test Lab

Application Notes for iscoord is-phone for IBM Lotus Notes with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the compliance testing of the iscoord is-phone for IBM Lotus Notes (is-phone) with Avaya Communication Manager. The testing which was performed tested the major functions of the is-phone product.

Information in these Application Notes has been obtained through 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 iscoord is-phone for IBM Lotus Notes (is-phone) is a SIP softphone client which works together with IBM Lotus Notes, providing telephone features to users of IBM Lotus Notes. It provides multiple call appearances and offers enhanced SIP calling features.

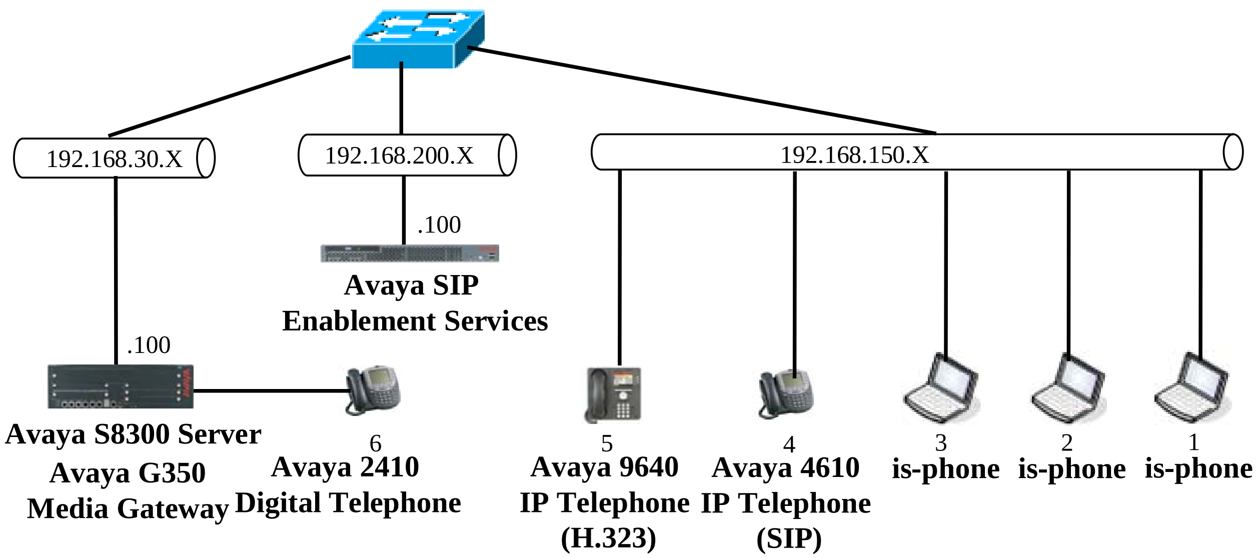


Figure 1: iscoord is-phone for IBM Lotus Notes Test Configuration

The is-phone is attached to Avaya Communication manager via SIP trunk by way of the Avaya SIP Enablement Services server.

The following table contains additional information about how each of the telephones contained in the above diagram are configured in Avaya Communication Manager:

Endpoint	Ext	Station Type
audix	3000000	
1	3800001	4610
2	3800002	4610
3	3800003	4610
4	3800004	4610
5	3000094	9640
6	3000001	2410

Table 1: Extensions Used for Testing

2. Equipment and Software Validated

Equipment	Software Version
Avaya S8300 Server / Avaya Communication Manager	R014x.00.1.731.2
Avaya G350 Media Gateway	26.36.0
Avaya SIP Enablement Services	4.0.0.0-033.6
Avaya 2410 Digital Telephone	5.0
Avaya 4610 IP Telephone	2.2.2 (SIP)
Avaya 9640 IP Telephone	1.5 (H.323)
iscoord is-phone for IBM Lotus Notes	7.0.2.108
IBM Lotus Notes	8.0
Workstations	MS XP PRO Version 5.1 SP 2

Table 2: Version Numbers of Equipment and Software

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).

Although the Avaya Audix voicemail system was used for testing, the configuration of this component is outside the scope of this document.

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 number of SIP endpoints attached to the system.

```
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: 20    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 system-parameters features

Use the **change system-parameters features** command to configure the features required to support the is-phone. If the Directed Call Pickup feature is to be used by the is-phone, this feature must be set to "y".

```
change system-parameters features                               Page   4 of  17
                        FEATURE-RELATED SYSTEM PARAMETERS
          Reserved Slots for Attendant Priority Queue: 5
                           Time before Off-hook Alert: 10
               Emergency Access Redirection Extension:
 Number of Emergency Calls Allowed in Attendant Queue: 5
 Maximum Number of Digits for Directed Group Call Pickup:4
               Call Pickup on Intercom Calls? y         Call Pickup Alerting? n
 Temporary Bridged Appearance on Call Pickup? y         Directed Call Pickup? y
                  Extended Group Call Pickup: none

  Deluxe Paging and Call Park Timeout to Originator? n
 Controlled Outward Restriction Intercept Treatment: tone
Controlled Termination Restriction (Do Not Disturb): tone
          Controlled Station to Station Restriction: tone
AUTHORIZATION CODE PARAMETERS        Authorization Codes Enabled? n




               Controlled Toll Restriction Replaces: none
```

Figure 3: System-Parameters Features Form

3.1.3. Configure Dial Plan

Use the **change dialplan analysis** command to specify that dialed strings which begin with "3" are extensions. Include the string "*83" to be used for a trunk access code for the SIP trunk as described in **section 3.1.4.3.** The "*7" entry is used by the feature codes described in **Table 12**.

```
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
        3       7    ext
        *83     3    dac
        *7      4    fac
```

Figure 4: Dialplan Analysis Form

3.1.4. Configure Interface to SES

3.1.4.1 Specify IP node names

Use the **change node-names ip** command to define the address of the "procr" interface and the Avaya SIP Enablement Services server. An address for "audix" must also be configured if the Avaya voicemail system is to be included in the configuration.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
audix               192.168.30.10
default             0.0.0.0
procr               192.168.30.100
ses                 192.168.200.100
```

Figure 5: Node-Names IP Form

3.1.4.2 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 "procr" to use the processor interface on the S8300.
Near-end Listen Port	Accept the default of "5061" to specify the standard TLS listening port.
Far-end Node Name	Enter "ses" to specify the SES server name assigned in **Figure 5**.
Far-end Listen Port	Accept the default of "5061" to specify the standard TLS listening port.
Far-end Domain	Enter the domain name which is configured for SES, configured in **Figure 25**.
DTMF over IP	Enter "rtp-payload" to use RTP payload events.
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: procr                  Far-end Node Name: ses
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region:
       Far-end Domain: ffm.com

                                              Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
                                                         IP Audio Hairpinning? y
 Enable Layer 3 Test? n
 Session Establishment Timer(min): 3
```

Figure 6: SIP Signaling-Group Form

3.1.4.3 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 7: SIP Trunk-Group Form

3.1.4.4 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 25**.
Name	Assign a name for identification purposes.
Intra-region IP-IP Direct Audio	Specify “yes” 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 8: IP-Network-Region Form

3.1.4.5 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 9: IP-Codec-Set Form

3.1.5. Configure Telephones

3.1.5.1 Configure Stations

Use the **add station <x>** command to allocate a station for is-phone, where <x> is the extension for iscoord is-phone 1 shown in **Table 1**. Repeat this for the other endpoints shown in **Table 1**.

Parameter	Usage
Type	Enter the station type of the phone to be used as shown in **Table 1.**
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.
Coverage Path 1	Enter the coverage path number assigned to audix in **Figure 13**.

Table 7: Configuration Parameters IP Telephones

```
change station 3800001                                          Page   1 of   4
                                     STATION

Extension: 3800001                          Lock Messages? n          BCC: 0
     Type: 4610                           Security Code: 1000083       TN: 1
     Port: S00009                       Coverage Path 1: 99           COR: 1
     Name: ext 3800001                    Coverage Path 2:            COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 19            Personalized Ringing Pattern: 1
                                                    Message Lamp Ext: 3800001
            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 10: Station Form

3.1.5.2 Configure off-pbx-telephone station-mapping

Use the **change off-pbx-telephone station-mapping** command to configure an interface to SES for the is-phones (1-3 in **Figure 1)** and the Avaya SIP Telephone shown in **Table 1**. Assign values for this command as shown **Table 8**.

Parameter	Usage
Station Extension (p. 1)	Enter the extension is-phone from **Table 1**.
Application (p. 1)	Enter “OPS”.
Phone Number (p. 1)	Enter the telephone extension from **Table 1**.
Trunk Selection (p. 1)	Enter the number “83” assigned to the SIP trunk group in **Figure 7**.
Call Limit (p. 2)	Enter the number of simultaneous calls which stations can have. A value of “3” was used for testing.

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

```
change off-pbx-telephone station-mapping 3800001              Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 3800001            OPS          -     3800001            83         1
```

Figure 11: 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
 3800001       3          both         all         both
```

Figure 12: Off-PBX-Telephone Form, Page 2

3.1.5.3 Add Coverage

Use the **add coverage path <x>**, where x is the coverage path number, to forward unanswered calls to the Avaya IA 770 INTUITY AUDIX Messaging Application (Audix) The coverage path number should be the same as was assigned to the stations in **Figure 10**. The values to be assigned to these parameters are shown in **Table 9**.

Parameter	Usage
Number of Rings	Enter the number of times that the phone will ring before being transferred to the coverage point.
Point1	Enter “h99” to use the Audix hunt group defined for coverage in **Figure 14** and **Figure 15**.

Table 9: Parameters for Off-PBX-Telephone Station-Mapping

```
add coverage path 99                                          Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 99
                                                  Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?             n              n
               Busy?             y              y
        Don't Answer?            y              y         Number of Rings: 3
                All?             n              n
 DND/SAC/Goto Cover?             y              y
   Holiday Coverage?             n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h99             Rng:    Point2:
  Point3:                         Point4:
  Point5:                         Point6:
```

Figure 13: Coverage Path Form

Use the **add hunt-group** command to create a hunt group for calls to the Avaya Audix voicemail system. The number of the hunt group should be the same as was configured for the coverage point configured in **Figure 13**. Assign values to the parameters in this form as shown in **Table 10**.

Parameter	Usage
Group Name (p. 1)	Assign a name by which the hunt group can be recognized.
Group Extension (p. 1)	Assign the extension assigned to Audix.
Group Type (p. 1)	Enter “ucd-mia” (Uniform Call Distribution-Most Idle Agent) as required by Audix.
Message Center (p. 2)	Enter “qsig-mwi” (QSIG Message Waiting Indication) as required by Audix.
Send Reroute Request (p. 2)	Enter “y”.
Voice Mail Number (p. 2)	Assign the extension assigned to Audix.
Routing Digits (p. 2)	Enter the access code assigned to AAR in **Figure 17**.

Table 10: Audix Hunt-Group Parameters

```
add hunt-group 99                                               Page   1 of  60
                                  HUNT GROUP

            Group Number: 99                                ACD? n
              Group Name: audix                           Queue? n
         Group Extension: 3000000                        Vector? n
              Group Type: ucd-mia                  Coverage Path:
                      TN: 1         Night Service Destination:
                     COR: 1                  MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display:
```

Figure 14: Audix Hunt-Group Form, Page 1

```
add hunt-group 99                                               Page   2 of  60
                                  HUNT GROUP

                       LWC Reception: none           AUDIX Name:



                      Message Center: qsig-mwi
                     Send Reroute Request: y
                   Voice Mail Number: 3000000
Routing Digits (e.g. AAR/ARS Access Code): *708        Provide Ringback? n
                TSC per MWI Interrogation? n
```

Figure 15: Audix Hunt-Group Form, Page 2

3.1.5.4 Configure Class of Restriction

Use the **change cor** command to confingure Class of Restriction (COR) 1 with parameters required to use the call pickup feature of the is-phone.

Parameter	Usage
Can Be Picked Up By Directed Call Pickup?	Enter "y" to allow calls to stations assigned to this COR to be answered via directed call pickup.
Use Directed Call Pickup?	Enter "y" to allow the stations assigned to this COR to answer other telephones via directed call pickup.

Table 11: Parameters for Off-PBX-Telephone Station-Mapping

```
change cor 1                                                Page   1 of  22
                              CLASS OF RESTRICTION

               COR Number: 1
          COR Description:

                      FRL: 0                                APLT? y
  Can Be Service Observed? n           Calling Party Restriction: none
Can Be A Service Observer? n            Called Party Restriction: none
 Partitioned Group Number: 1       Forced Entry of Account Codes? n
         Priority Queuing? n                Direct Agent Calling? n
     Restriction Override: none       Facility Access Trunk Test? n
     Restricted Call List? n                 Can Change Coverage? n

            Access to MCT? y            Fully Restricted Service? n
Group II Category For MFC: 7
         Send ANI for MFE? n
            MF ANI Prefix:              Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                        Can Be Picked Up By Directed Call Pickup? y
                                    Can Use Directed Call Pickup? y
                                    Group Controlled Restriction: inactive
```

Figure 16: Off-PBX-Telephone Form, Page 2

3.1.6. Configure Access to Extended Features

Use the **change feature-access-codes** command to assign unused feature codes to those features used by the is-phone, as shown in the following **Table 12**. Note the "*7" entry for the dial plan shown in **Figure 4** is used by these entries.

Parameter	Usage
Auto Alternate Routing (AAR) Access Code (p.1)	Allow alternate routing as required by Audix, as defined in **Figure 15.**
Call Forwarding Activation Busy/DA, All, Deactivation (p. 1)	Activate/deactivate call forwarding. This is required by the Feature Name Extensions (FNE) for call forwarding defined in **Figure 21**.
Directed Call Pickup Access Code (p. 2)	This is required by the Directed Call Pickup FNE defined in **Figure 21**.
Last Number Dialed Access Code (p. 2)	This is required by the Last Number Dialed FNE defined in **Figure 22**.
Priority Calling Access Code (p. 3)	This is required by the Priority Call FNE defined in **Figure 22**
Send All Calls Activation (p. 3)	This is required by the Send All Calls FNE defined in **Figure 22**
Send All Calls Deactivation (p. 3)	This is required by the Send All Calls Cancel FNE defined in **Figure 22**
Transfer to Voice Mail Access Code (p. 4)	This is required by the Transfer to Voice Mail FNE defined in **Figure 22**

Table 12: Parameters for the Feature Access Codes

```
change feature-access-codes                                     Page   1 of   5
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *701
         Abbreviated Dialing List2 Access Code: *702
         Abbreviated Dialing List3 Access Code: *703
Abbreviated Dial - Prgm Group List Access Code: *704
                      Announcement Access Code: *705
                       Answer Back Access Code: *706
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: *708
    Auto Route Selection (ARS) - Access Code 1: *709  Access Code 2:
                 Automatic Callback Activation: *710   Deactivation: *711
Call Forwarding Activation Busy/DA: *712   All: *713   Deactivation: *714
   Call Forwarding Enhanced Status: *715   Act: *716   Deactivation: *717
                         Call Park Access Code: *718
                       Call Pickup Access Code: *719
CAS Remote Hold/Answer Hold-Unhold Access Code: *720
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code: *723
                   Contact Closure   Open Code: *724     Close Code: *725
```

Figure 17: Feature Access Codes Form, Page 1

```
change feature-access-codes                                     Page   2 of   5
                               FEATURE ACCESS CODE (FAC)
                   Contact Closure  Pulse Code: *726
                  Data Origination Access Code: *727
                      Data Privacy Access Code: *728
              Directed Call Pickup Access Code: *729
      Directed Group Call Pickup  Access Code: *730
     Emergency Access to Attendant Access Code: *731
         EC500 Self-Administration Access Code: *732
                     Enhanced EC500 Activation: *733   Deactivation: *734
           Enterprise Mobility User Activation: *735   Deactivation: *736
  Extended Call Fwd Activate Busy D/A *737 All: *738   Deactivation: *739
        Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code: *741
                             Flash Access Code: *742
             Group Control Restrict Activation: *743   Deactivation: *744
                    Hunt Group Busy Activation: *745   Deactivation: *746
                              ISDN Access Code:
               Last Number Dialed Access Code: *748
     Leave Word Calling Message Retrieval Lock: *749
   Leave Word Calling Message Retrieval Unlock: *750
```

Figure 18: Feature Access Code Form, Page 2

```
change feature-access-codes                                     Page   3 of   5
                               FEATURE ACCESS CODE (FAC)
             Leave Word Calling Send A Message: *751
           Leave Word Calling Cancel A Message: *752
   Limit Number of Concurrent Calls Activation: *753   Deactivation: *754
               Malicious Call Trace Activation:        Deactivation:
         Meet-me Conference Access Code Change: *757

 PASTE (Display PBX data on Phone) Access Code: *758
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code: *761
      Per Call CPN Unblocking Code Access Code: *762

                  Priority Calling Access Code: *763
                           Program Access Code: *764

       Refresh Terminal Parameters Access Code: *765
              Remote Send All Calls Activation: *766   Deactivation: *767
               Self Station Display Activation:
                     Send All Calls Activation: *769   Deactivation: *770
         Station Firmware Download Access Code: *771
```

Figure 19: Feature Access Codes Form, Page 3

```
change feature-access-codes                                     Page   4 of   5
                               FEATURE ACCESS CODE (FAC)
                       Station Lock Activation: *772   Deactivation: *773
      Station Security Code Change Access Code: *774
              Station User Admin of FBI Assign:                   Remove:
  Station User Button Ring Control Access Code:
             Terminal Dial-Up Test Access Code: *778
Terminal Translation Initialization Merge Code:          Separation Code:
            Transfer to Voice Mail Access Code: *781
          Trunk Answer Any Station Access Code: *782
              User Control Restrict Activation: *783   Deactivation: *784
  Voice Coverage Message Retrieval Access Code: *785
 Voice Principal Message Retrieval Access Code: *786
          Whisper Page Activation Access Code: *787
```

Figure 20: Feature Access Codes Form, Page 4

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 the is-phone.

Parameter	Usage
Call Forward All (p. 1)	Assign an unused extension within the local dial plan to the “Call Forward All” feature.
Call Forward Cancel (p. 1)	Assign an unused extension within the local dial plan to the “Call Forward Cancel” feature.
Directed Call Pick-Up (p. 1)	Assign an unused extension within the local dial plan to the “Directed Call Pick-Up” feature.
Last Number Dialed (p. 2)	Assign an unused extension within the local dial plan to the “Last Number Dialed” feature.
Priority Call (p. 2)	Assign an unused extension within the local dial plan to the “Priority Call” feature.
Send All Calls (p. 2)	Assign an unused extension within the local dial plan to the “Send All Calls” feature to activate coverage to Audix.
Send All Calls Cancel (p. 2)	Assign an unused extension within the local dial plan to the “Send All Calls Cancel” feature.
Transfer to Voice Mail (p. 2)	Assign an unused extension within the local dial plan to the “Transfer to Voice Mail” feature.

Table 13: Parameters for Off-PBX-Telephone Feature-Name-Extensions

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

     Active Appearance Select: 3001801
          Automatic Call Back: 3001802
   Automatic Call-Back Cancel: 3001803
             Call Forward All: 3001804
  Call Forward Busy/No Answer: 3001805
          Call Forward Cancel: 3001806
                    Call Park: 3001807
        Call Park Answer Back: 3001808
                 Call Pick-Up: 3001809
        Calling Number Block: 3001810
      Calling Number Unblock: 3001811
        Conference on Answer: 3001812
       Directed Call Pick-Up: 3001813
       Drop Last Added Party: 3001814
    Exclusion (Toggle On/Off): 3001815
   Extended Group Call Pickup:
       Held Appearance Select: 3001817
```

Figure 21: 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: 3001818
           Last Number Dialed: 3001819
         Malicious Call Trace:
  Malicious Call Trace Cancel:
          Off-Pbx Call Enable: 3001822
         Off-Pbx Call Disable: 3001823
                Priority Call: 3001824
               Send All Calls: 3001825
        Send All Calls Cancel: 3001826
          Transfer On Hang-Up: 3001827
       Transfer to Voice Mail: 3001828
      Whisper Page Activation: 3001829
```

Figure 22: Off-PBX-Telephone Feature Name Extensions Form, Page 2

3.2. Configure Avaya SIP Enablement Services

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

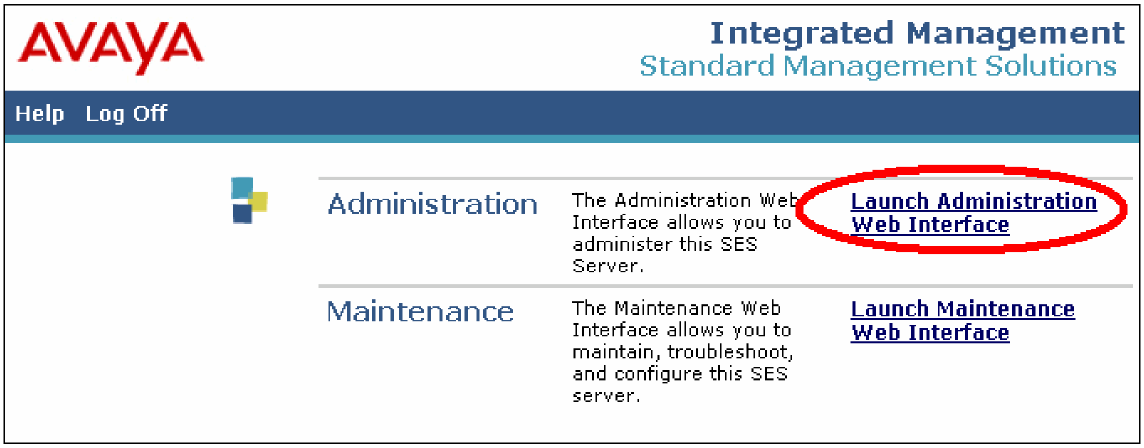


Figure 23: SES Initial Greeting Screen

The SES Integrated Management top level menu is then displayed.

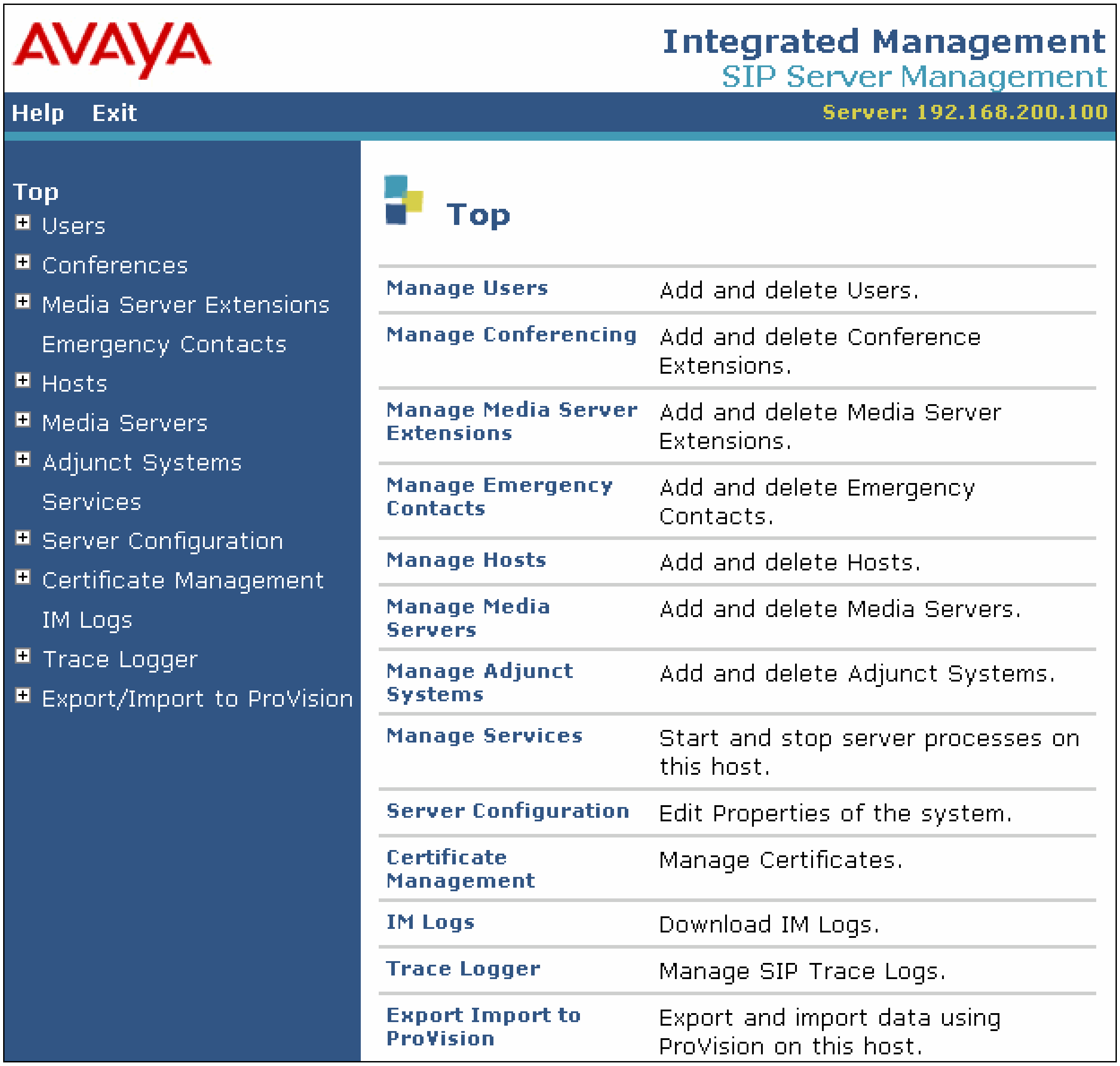


Figure 24: SES Integrated Management Top Level Menu

3.2.1. Configure Basic Avaya SIP Enablement Services 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 6**, and the IP address of the SES server which was assigned in **Figure 5** as the IP address of the “License Host”. Select the “Update” button.

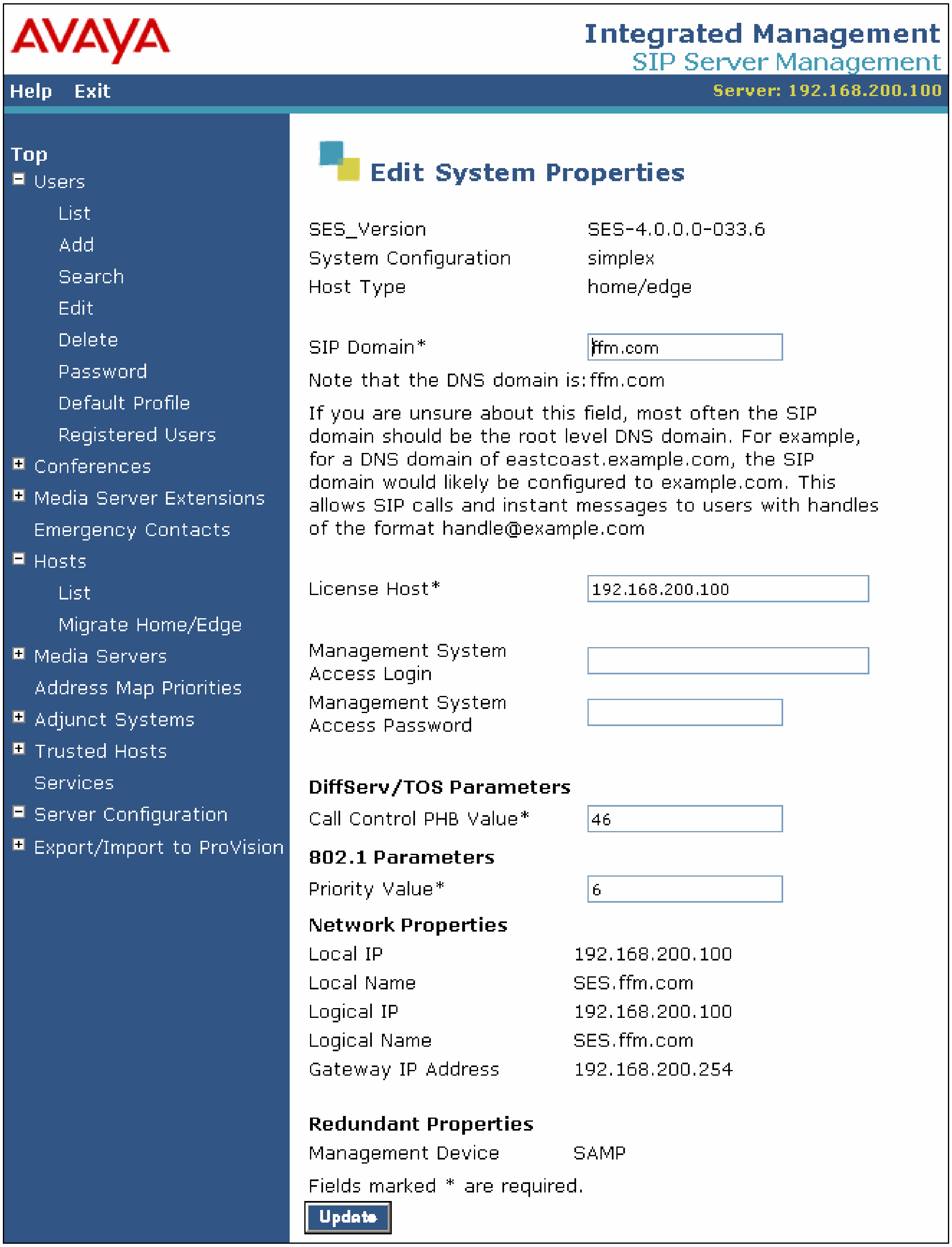


Figure 25: Avaya SIP Enablement Services Edit System Properties Screen

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.

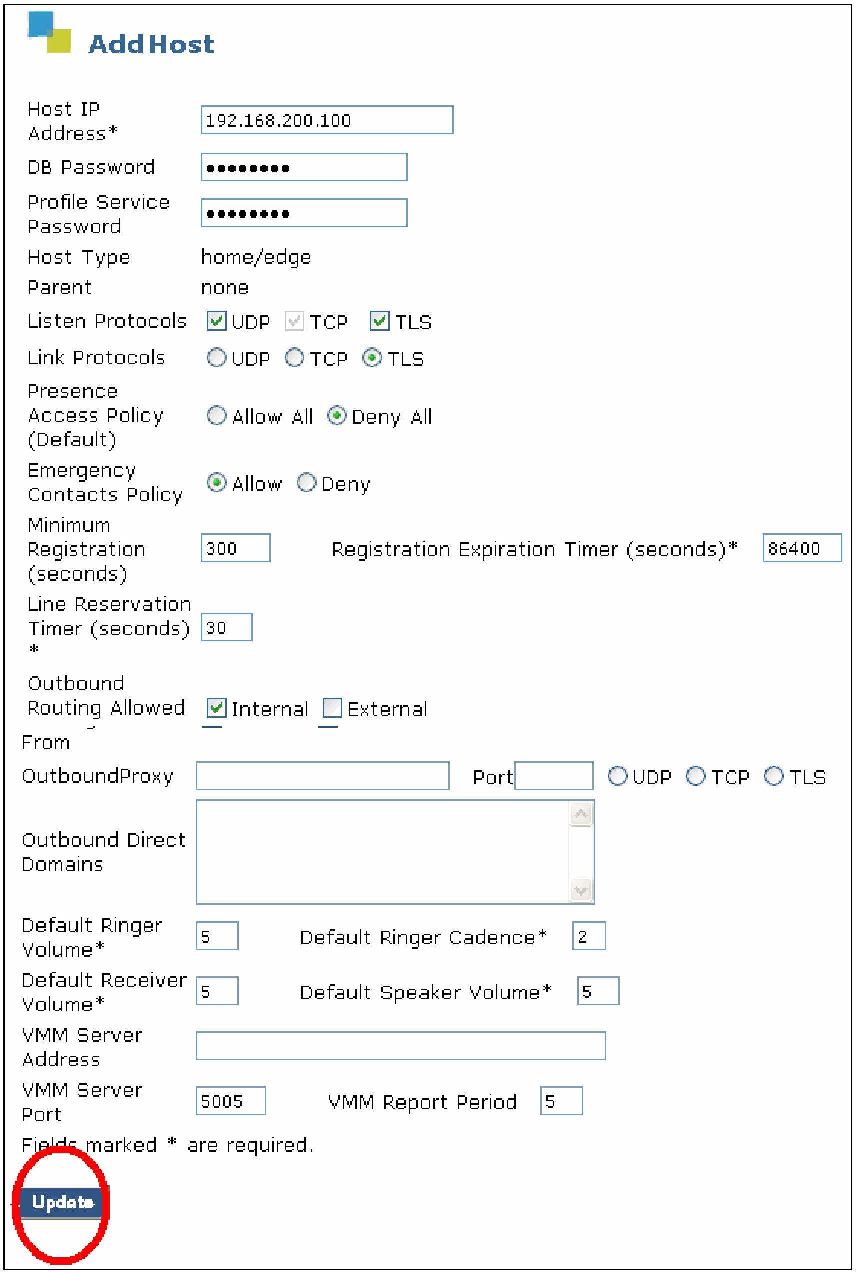


Figure 26: Avaya SES “Add Host” Screen

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.

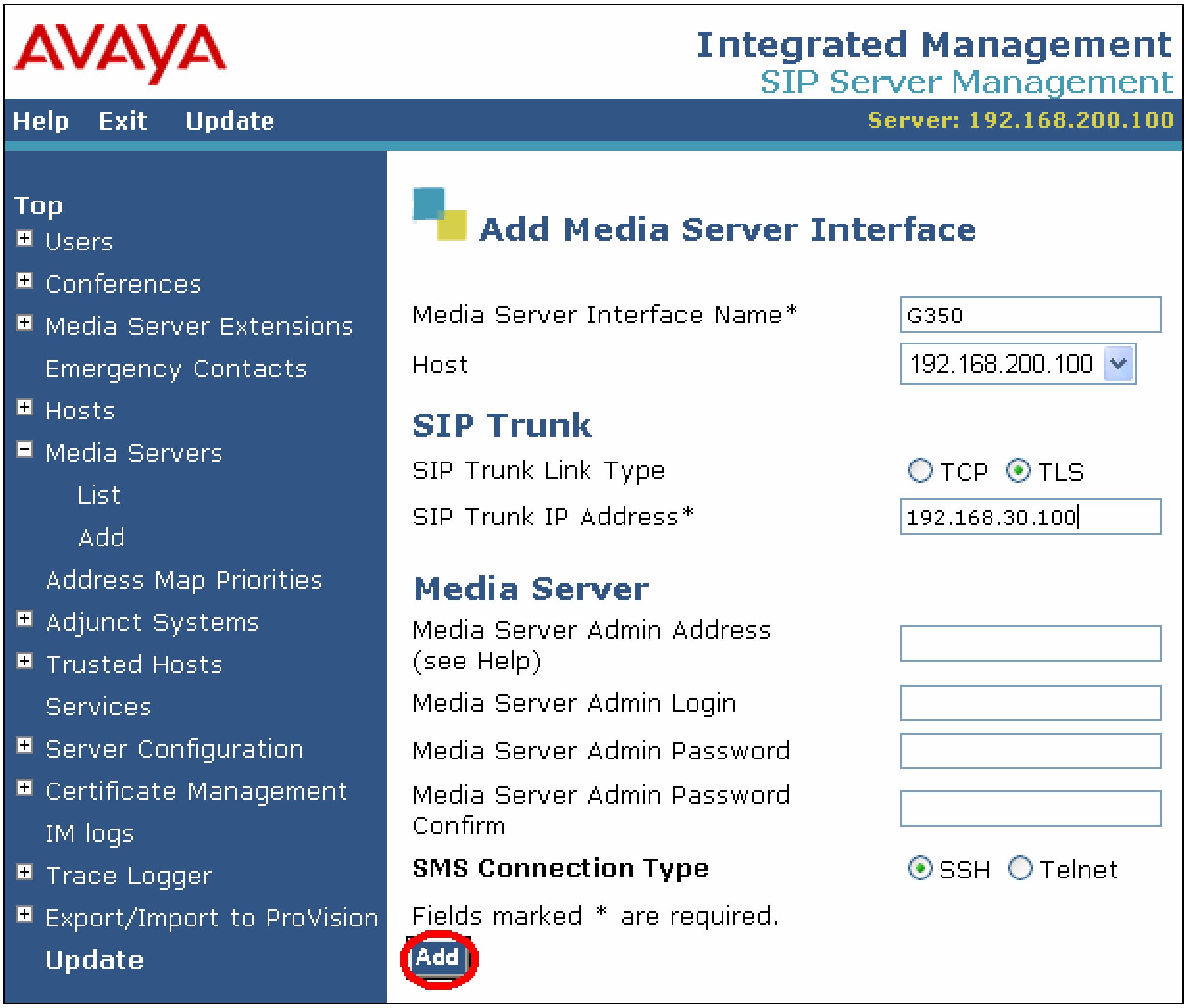


Figure 27: Avaya SES Add Media Server Interface Screen

3.2.3. Configure SIP softphone user for iscoord is-phone for IBM Lotus Notes

From the top level menu, select the “Manage Users” -> “Add User” menu entries. Enter the extension for an is-phone as both the “Primary Handle” and the “User ID”. This is the same extension that was configured for the station in **Figure 10** and for Off-PBX-Telephone Station-Mapping shown in **Figure 11**. Enter a **Password** and **First/Last name** of the user, check the “Add Media Server Extension” box, and click “Add”.

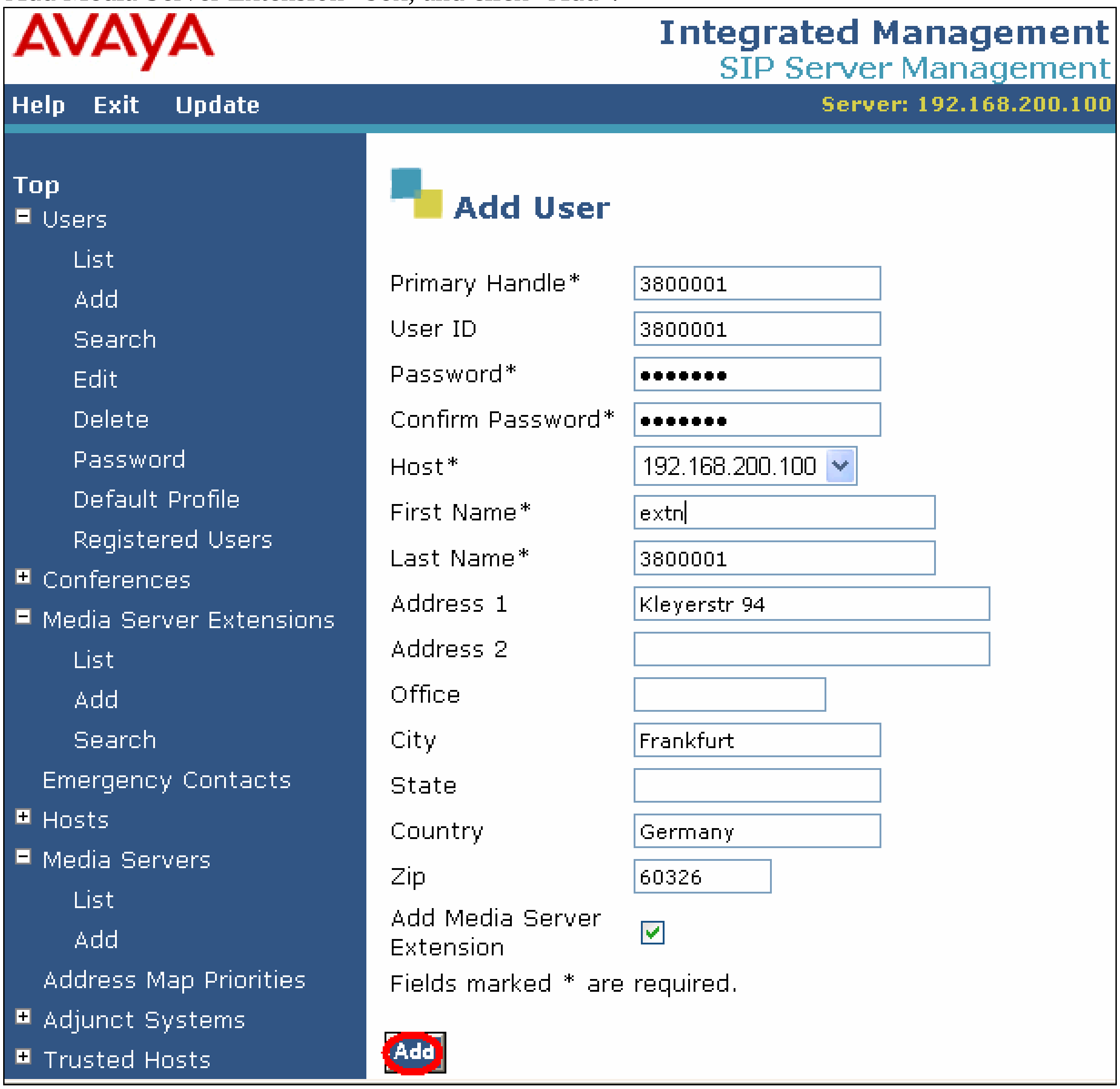


Figure 28: Avaya SES “Add User” Screen

Enter the Media Server Extension for each of the is-phone extensions and the Avaya SIP phone shown in **Table 1**. Select the Media Server (refer to **Figure 27**) from the drop-down box and click "Add" to continue.

Repeat this for each of the is-phone extensions and the Avaya SIP phone shown in **Table 1**.

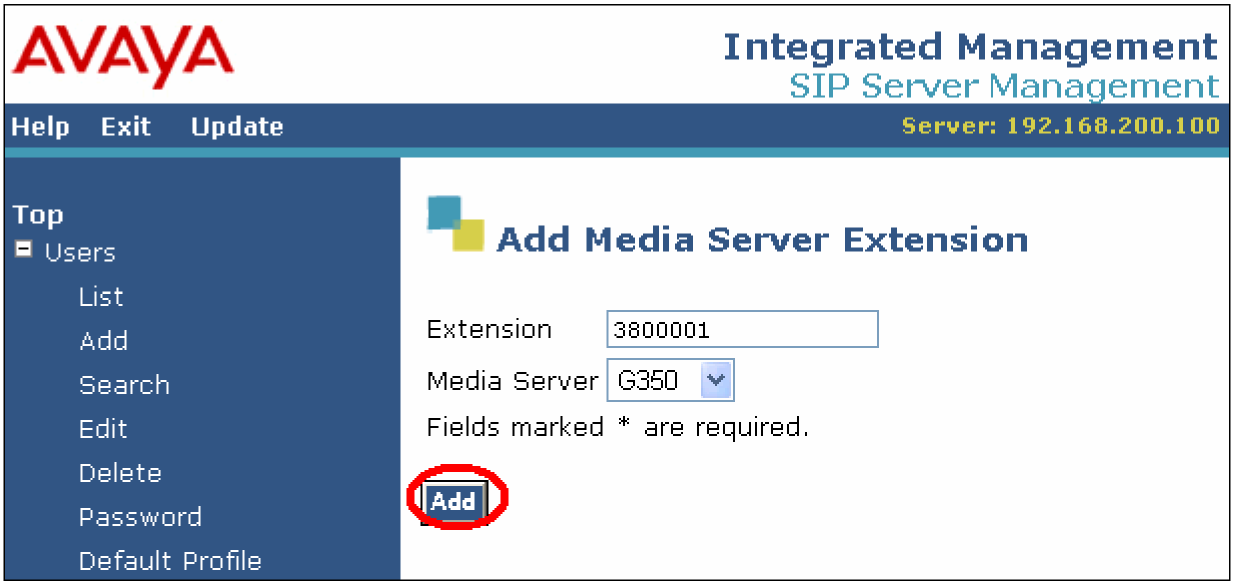


Figure 29: Avaya SES Add Media Server Extension Screen

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

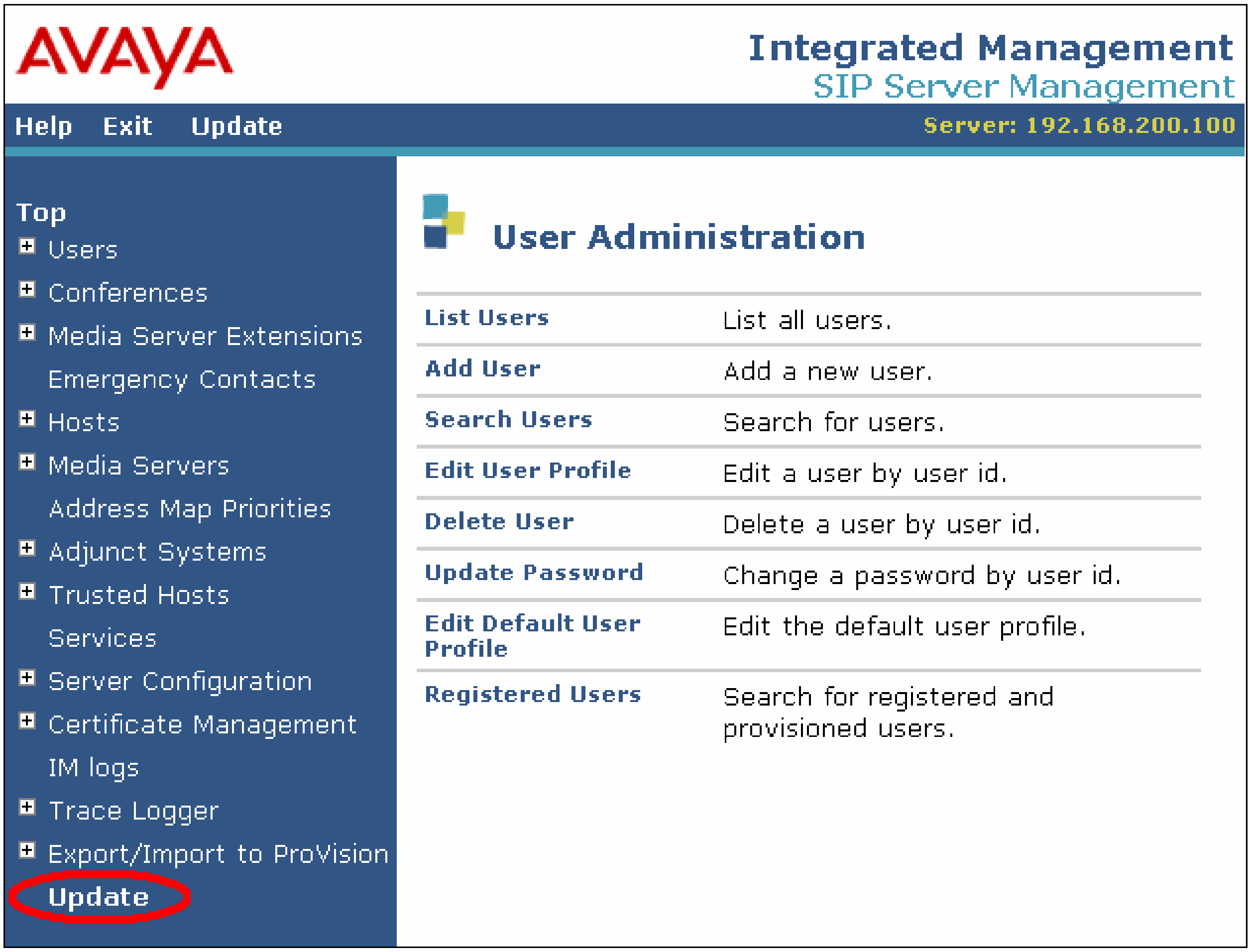


Figure 30: Update from Top Avaya SIP Enablement Services Screen

3.3. Configure iscoord is-phone for IBM Lotus Notes

3.3.1. Configure Lotus Notes

Start the IBM Lotus Notes client from the Windows "start" icon. Click on the "workspace" bookmark (highlighted icon at lower left).

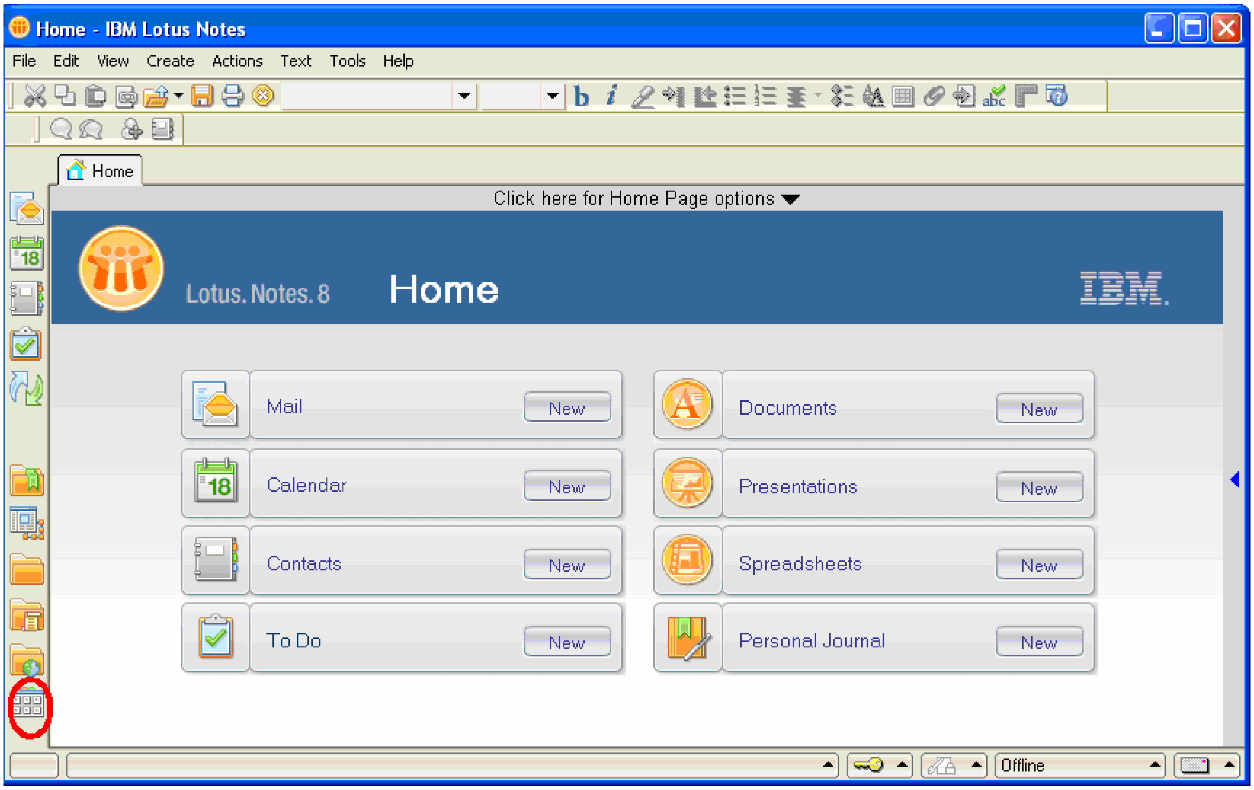


Figure 31: IBM Lotus Notes Introductory Screen

Click “is-phone configuration on Local” button.

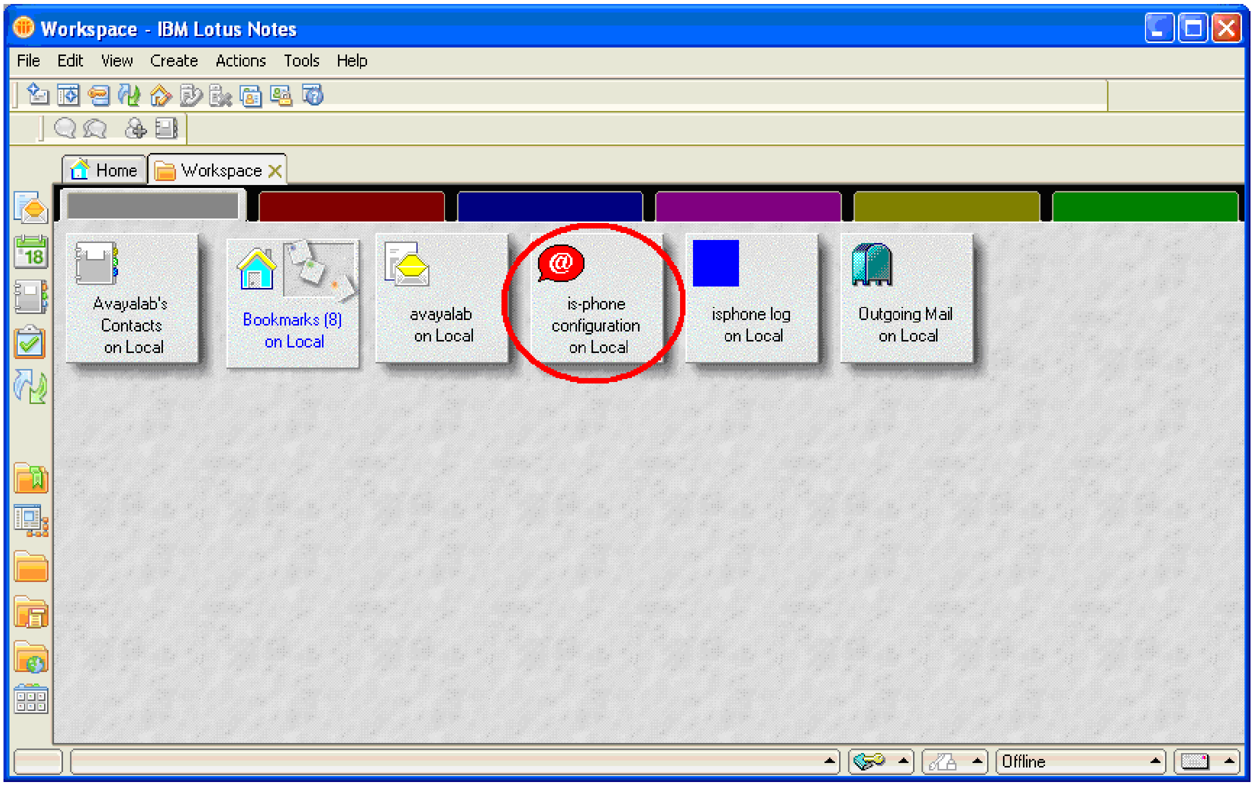


Figure 32: IBM Lotus Notes Workspace Screen

Click on “default” profile name.

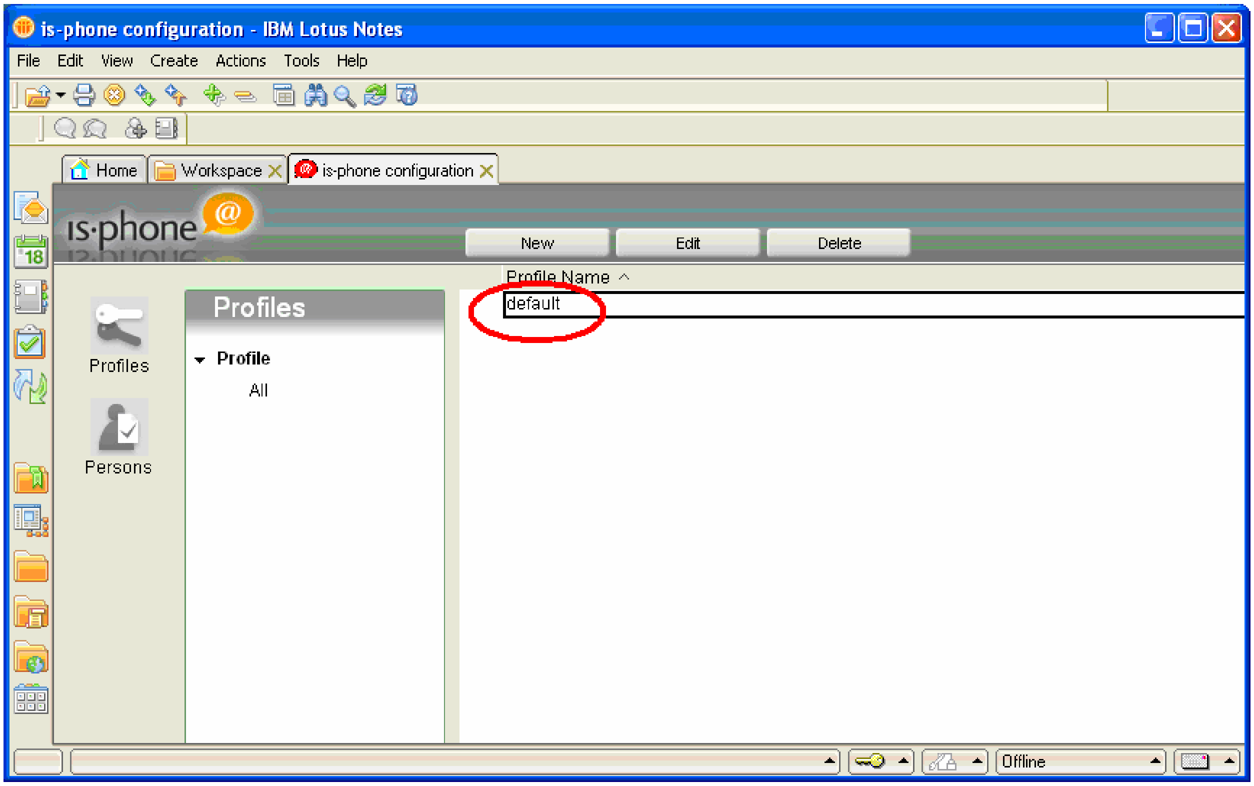


Figure 33: IBM Lotus Notes is-phone Configuration Screen

Click on the "PBX Authentication" tab.

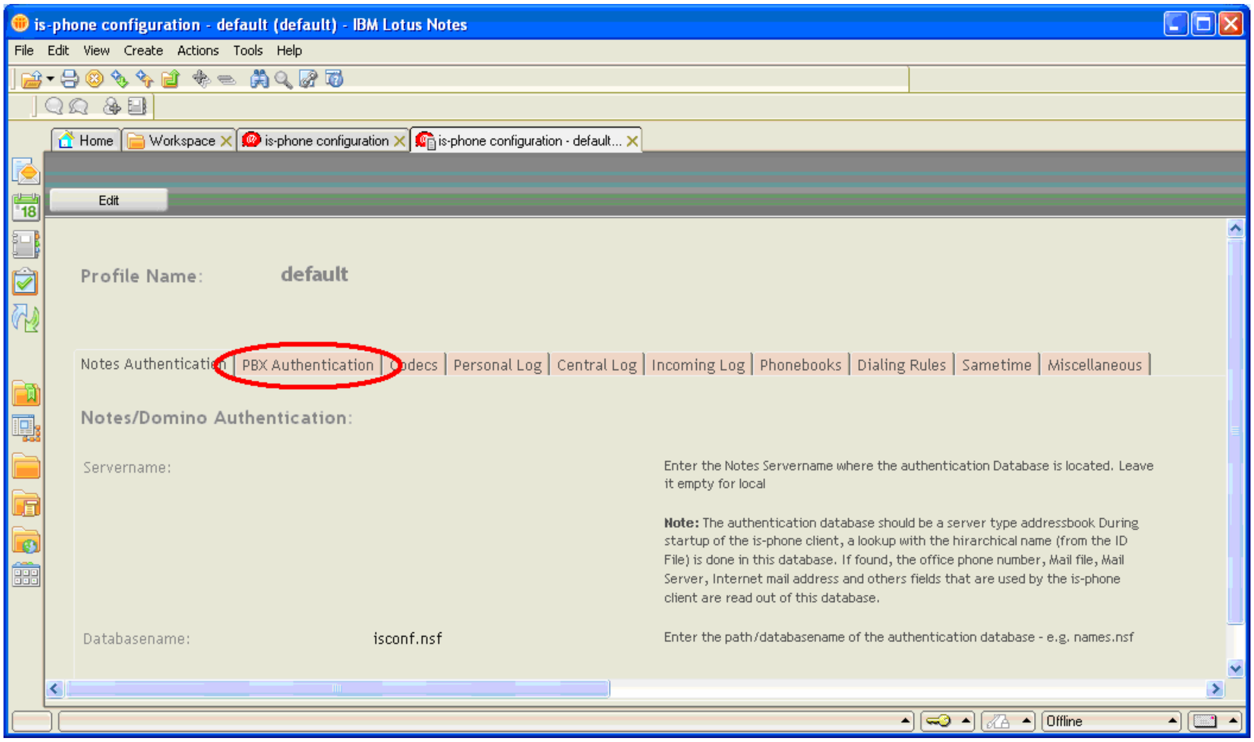


Figure 34: IBM Lotus Notes is-phone Configuration Screen

Enter the values shown in **Table 14** and click the "Codecs" tab.

Parameter	Usage
Registrar Address	Enter the IP address of the SES server (see **Figure 25**).
Realm	Enter the SES SIP Domain (see **Figure 25**).
Number of Digits	Enter "7", the number of digits in the dialing plan.
Password	Enter the password assigned to the is-phone user in **Figure 28**.
Domain	Enter the SES SIP Domain (see **Figure 25**).
DTMF Mode	Select "inbound" from the drop-down list.
Listening Port	Specify "5060", the SIP port used for call signaling.
Reregister Interval	A register interval of 300s was specified for testing purposes.

Table 14: Parameters for Off-PBX-Telephone Station-Mapping

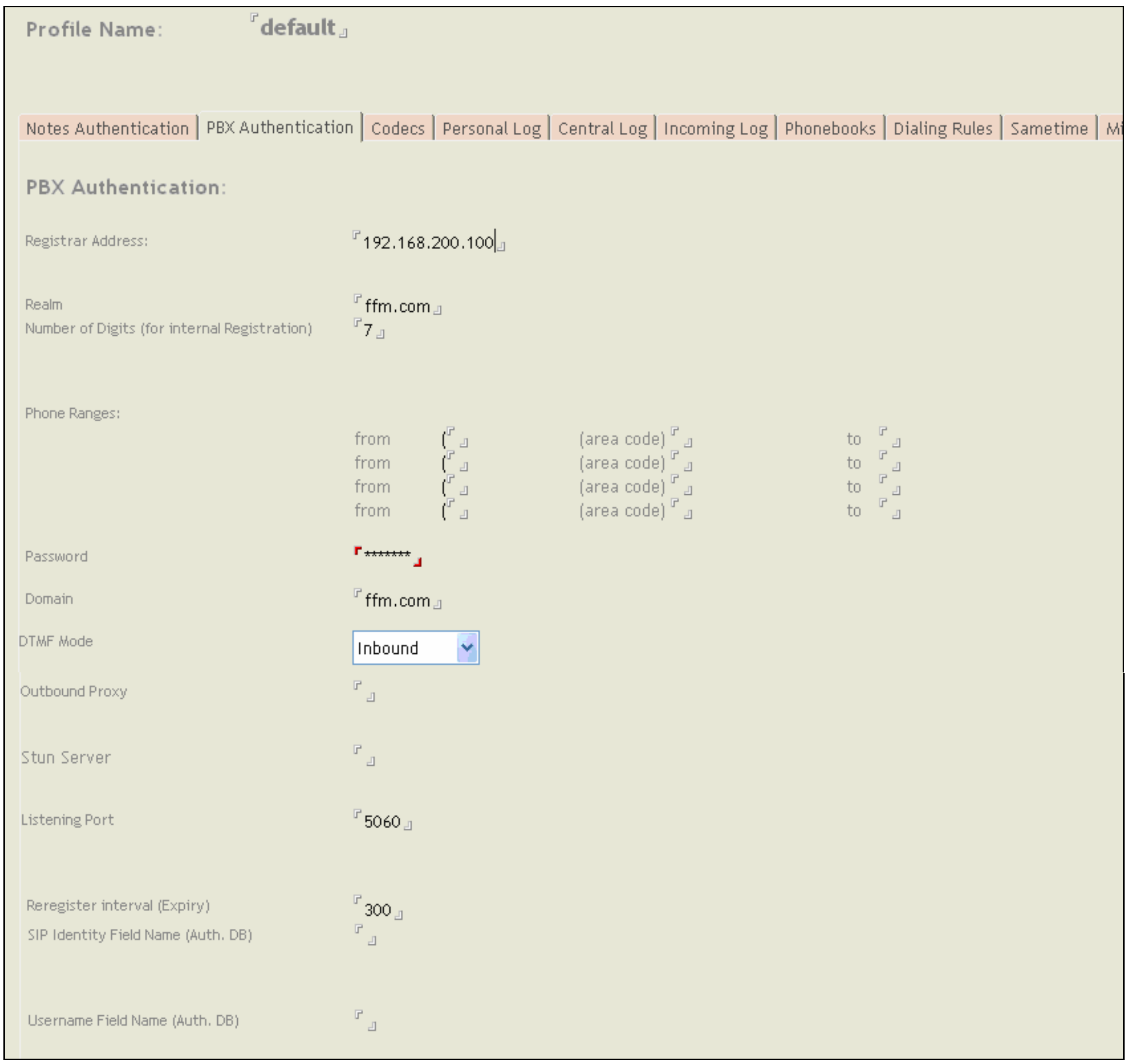


Figure 35: IBM Lotus Notes is-phone PBX-Authentication Screen

Select “G.711alaw” as Codec Priority 1 to match the codec selected in **Figure 9**. Click “Save & Exit”.

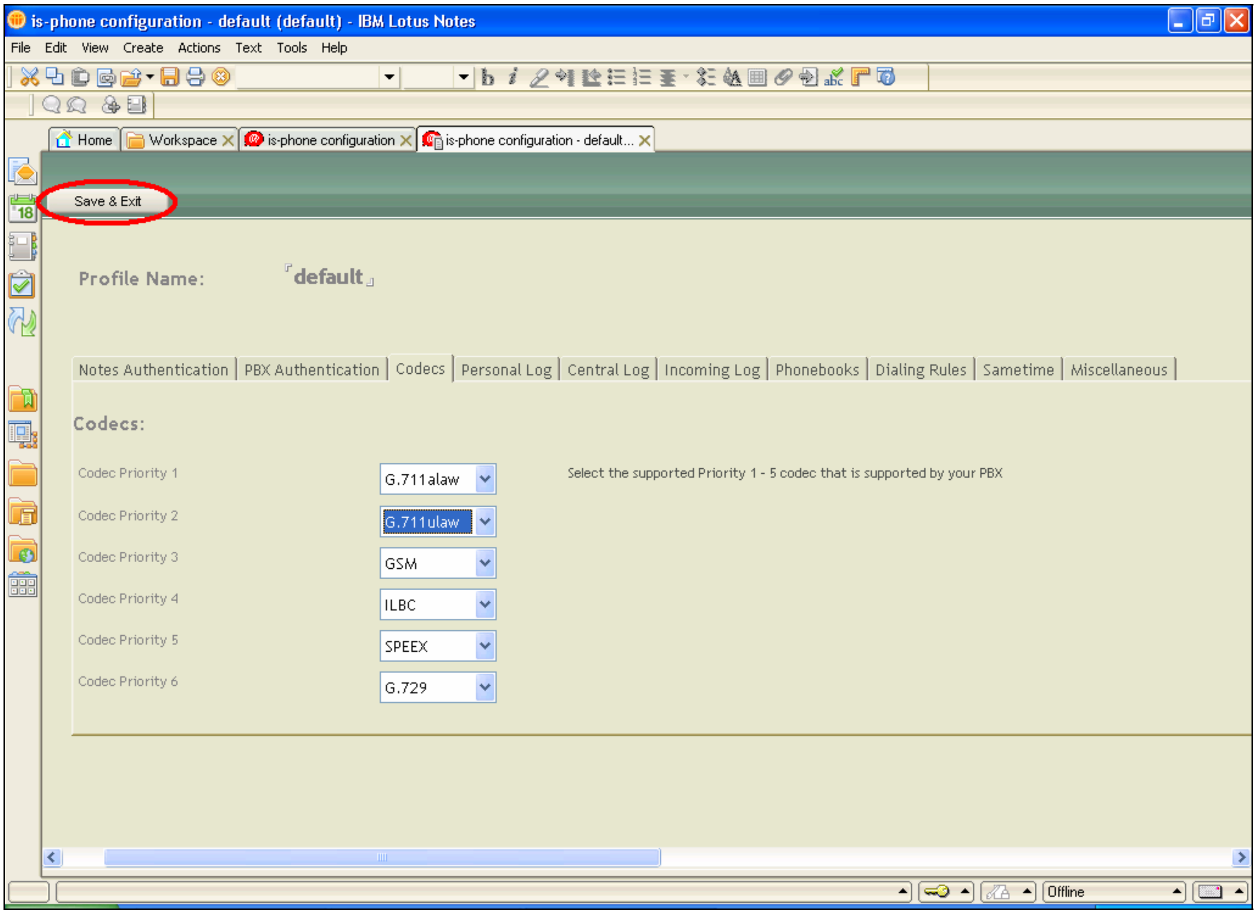


Figure 36: IBM Lotus Notes is-phone Codecs Screen

Click on “Persons” control in left frame:

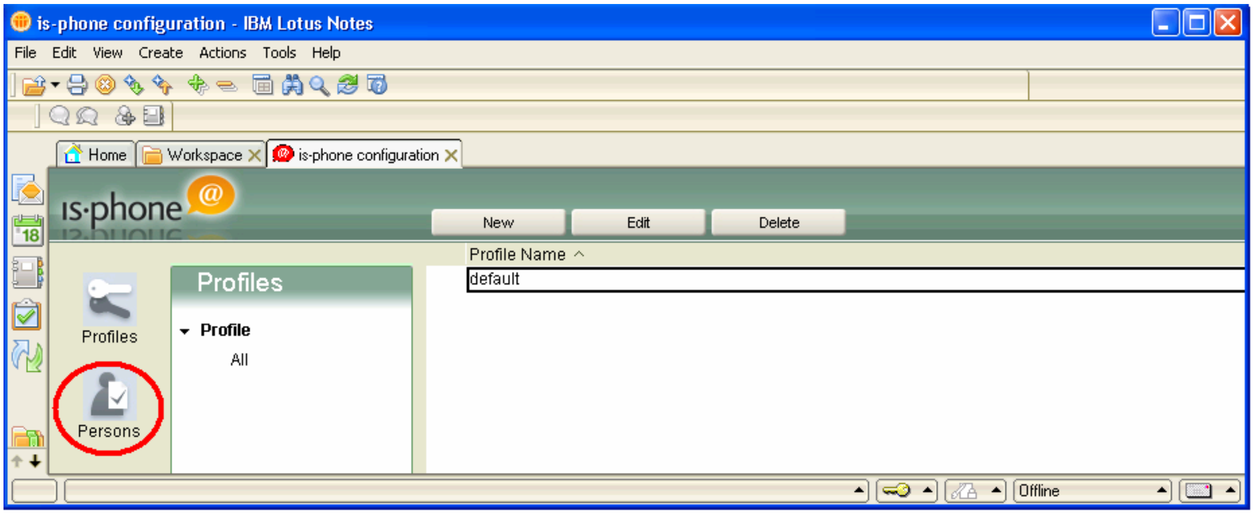


Figure 37: IBM Lotus Notes is-phone User Select Screen

Click “New”.

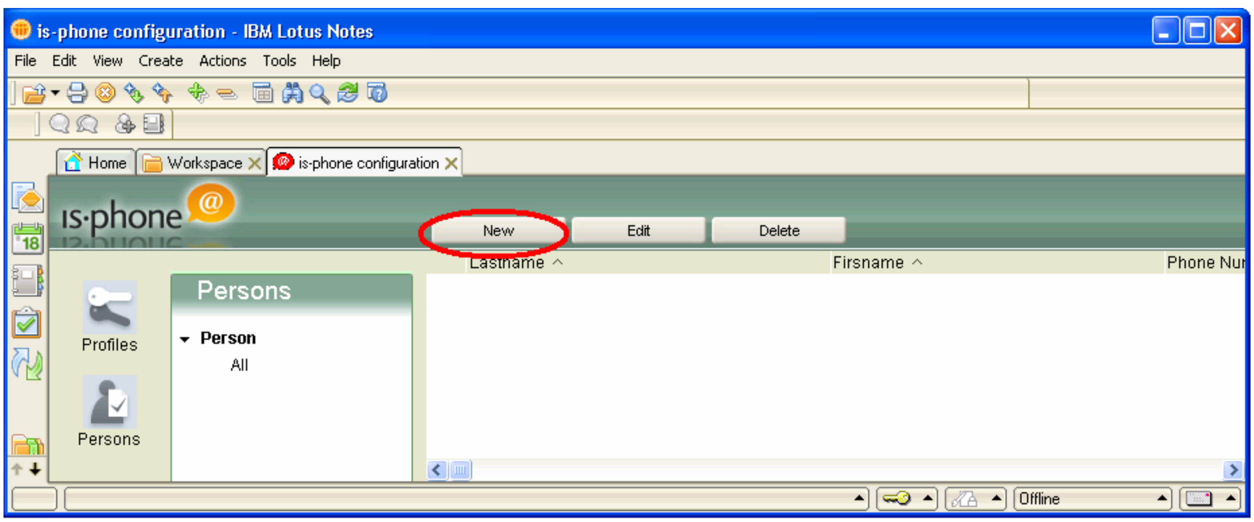


Figure 38: IBM Lotus Notes is-phone New User Screen

Fill in form as shown in **Table 15**. Click "Save & Exit".

Parameter	Usage
Office Phone Number	Enter the extension assigned to the is-phone from **Table 1**.
SIP Identity	Enter the extension assigned to the iscoord is-phone from **Table 1**, followed by "@", followed by the IP address of the SES server, as shown in **Figure 25**.
SIP Username	Enter the extension assigned to the is-phone from **Table 1**.

Table 15: Parameters IBM Lotus User Configuration

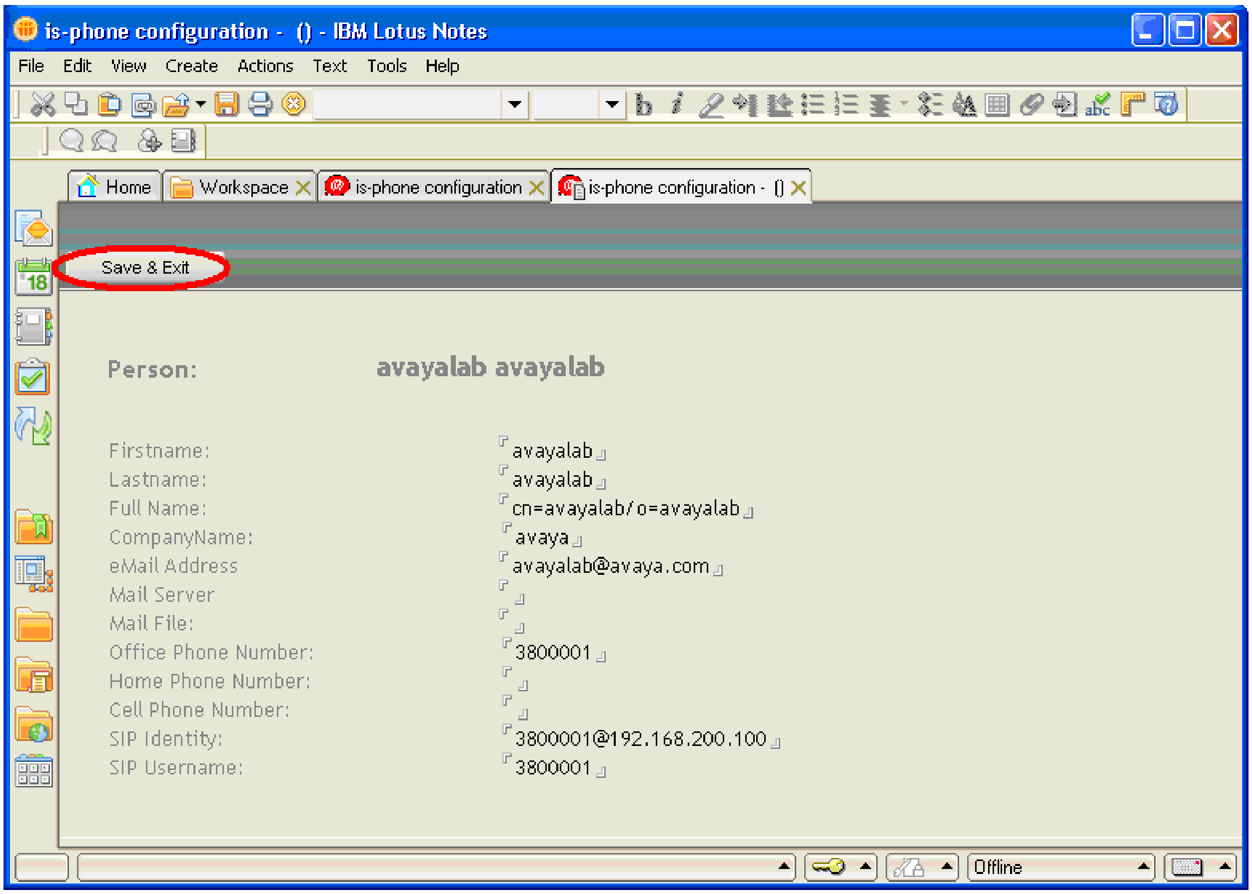


Figure 39: IBM Lotus Notes is-phone User Configuration Screen

3.3.2. Configure iscoord is-phone for IBM Lotus Notes

For detailed information/instructions about is-phone configuration please consult the is-phone for IBM Lotus Notes installation handbook.

Start the is-phone from the Windows "Start" prompt. Enter the "Serial Key" and click "Activate".

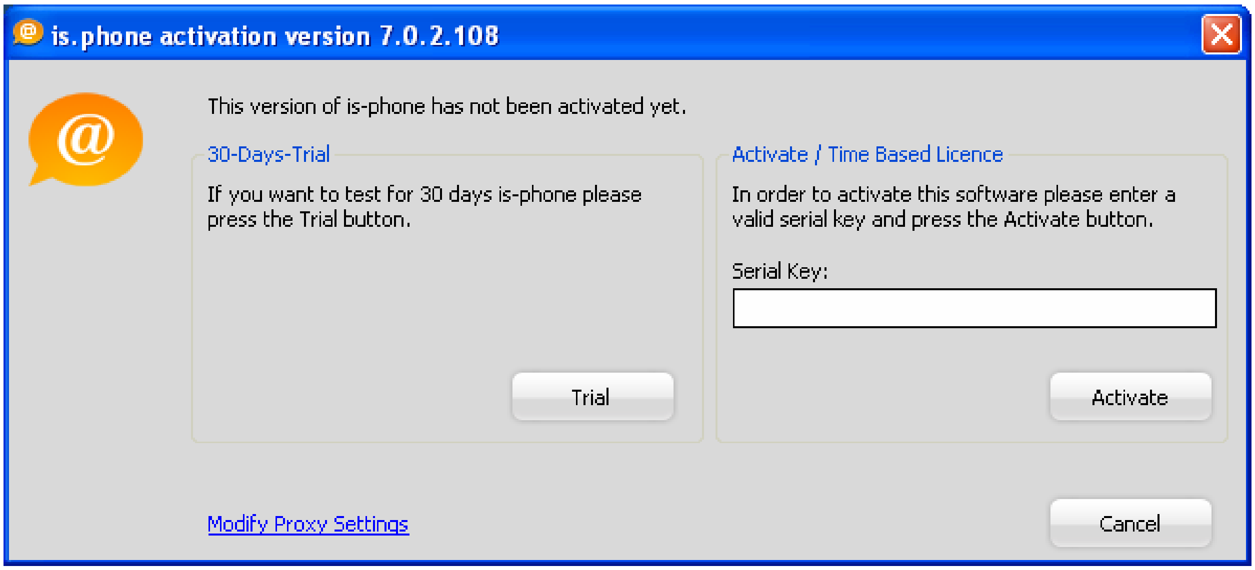


Figure 40: iscoord is-phone for IBM Lotus Notes Introductory Screen

Select "Tools -> Preferences".

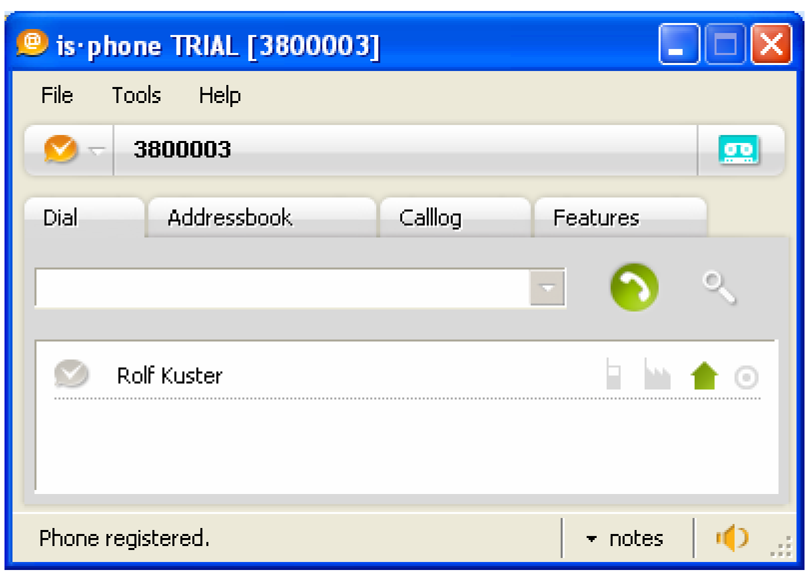
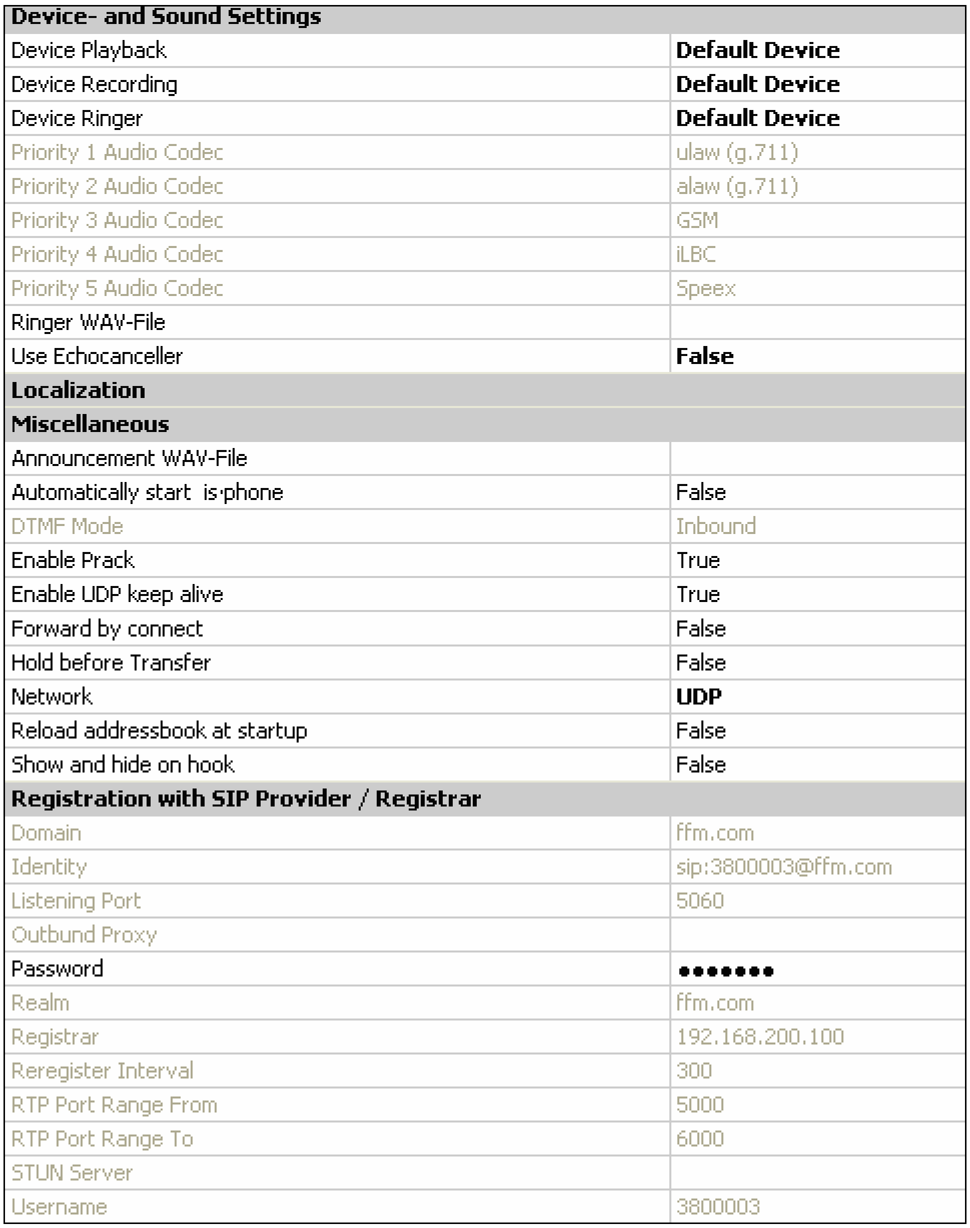


Figure 41: IBM Lotus Notes User Screen

Fill in the form as shown in **Table 16**. Select “Save” from the menu bar (not shown).

Parameter	Usage
Use EchoCanceller	Enter “False”.
Network	Enter “UDP”.
Password	Enter the password which was assigned in **Figure 28**.

Table 16: IBM Lotus Notes User Preferences Parameters

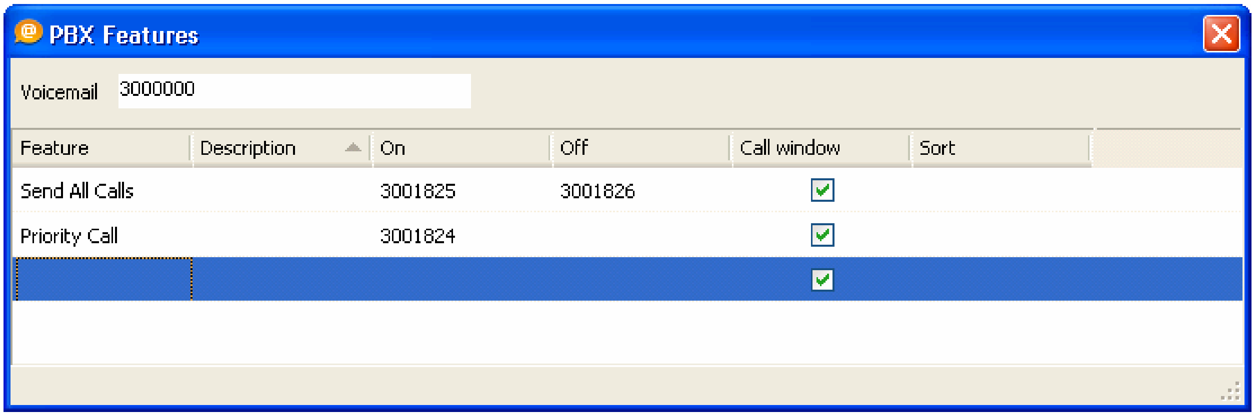


Device- and Sound Settings	
Device Playback	Default Device
Device Recording	Default Device
Device Ringer	Default Device
Priority 1 Audio Codec	ulaw (g.711)
Priority 2 Audio Codec	alaw (g.711)
Priority 3 Audio Codec	GSM
Priority 4 Audio Codec	iLBC
Priority 5 Audio Codec	Speex
Ringer WAV-File	
Use Echocanceller	False
Localization	
Miscellaneous	
Announcement WAV-File	
Automatically start is·phone	False
DTMF Mode	Inbound
Enable Prack	True
Enable UDP keep alive	True
Forward by connect	False
Hold before Transfer	False
Network	UDP
Reload addressbook at startup	False
Show and hide on hook	False
Registration with SIP Provider / Registrar	
Domain	ffm.com
Identity	sip:3800003@ffm.com
Listening Port	5060
Outbund Proxy	
Password	•••••••
Realm	ffm.com
Registrar	192.168.200.100
Reregister Interval	300
RTP Port Range From	5000
RTP Port Range To	6000
STUN Server	
Username	3800003

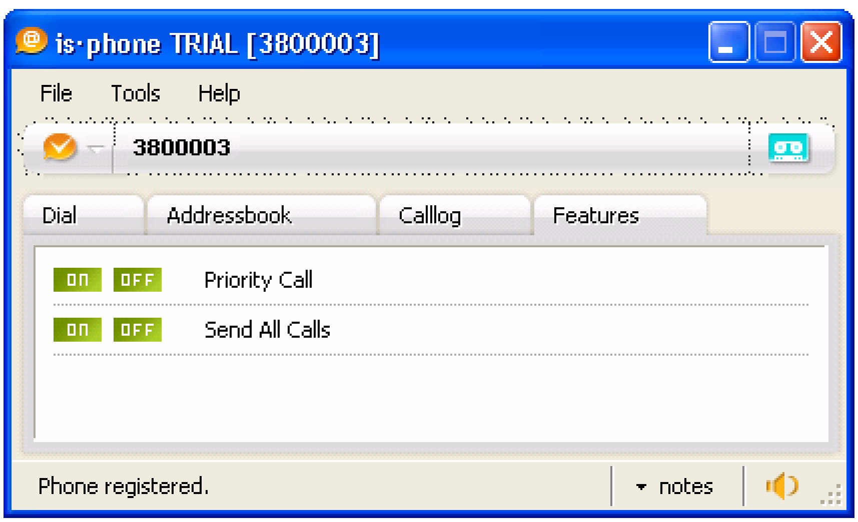
Figure 42: IBM Lotus Notes User Preference Screen

Select "Tools -> Featres".

Enter the names of the FNE features shown in **Table 13** for which ON/OFF buttons are to be created, along with the FNE extensions to activate/deactivate these features. For those features for which there is no "OFF" condition to be defined, i.e. "Priority Call", this field can be left blank.



After this is done "ON", "OFF" buttons appear in the call window which allow these features to be turned on or off. For those features for which there is no "OFF" state, i.e., "Priority Call", the "OFF" button can be ignored.



4. Interoperability Compliance Testing

The objective of the compliance testing performed on the is-phone product was to verify that it is compatible with Avaya Communication Manager. This includes verifying that the essential is-phone features function properly when used with Avaya Communication Manager, and that Avaya Communication Manager features are not hindered by the interaction with is-phone. Furthermore, is-phone'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 is-phone were tested by manually making calls to and from the unit.
- is-phone's robustness was tested by verifying its ability to recover from interruptions to its external LAN.

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 is-phone were tested for proper interoperation with Avaya Communication Manager. All of these capabilities functioned as expected.

- Incoming call
- Outgoing call
- Call hold
- Call hold with consultation
- Unattended transfer
- Attended transfer
- Call forward unconditional
- Call forward busy
- Call forward no answer
- 3-way conference
- Call waiting
- DTMF transmission
- Priority call
- Directed call pickup
- Transfer to voice mail
- Last number dialed
- Send all calls

5. Verification Steps

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

- Verify that the Avaya SES and is-phone systems can ping each other.
- Verify that it is possible to initiate calls from the is-phone.
- Verify that that is-phone can accept incoming calls.

6. Support

Support for is-phone is available at:

iscoord ag
Beustweg 12
CH-8032 Zurich/Switzerland
Phone +41 44 258 88 82
Fax +41 44 258 88 99
Email info@iscoord.com

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
[5] "is-phone for IBM Lotus Notes installation handbook"

8. Conclusion

These Application Notes describe the conformance testing of the iscoord is-phone for IBM Lotus Notes with Avaya Communication Manager and Avaya SES. The various features of the is-phone unit which involve its telephone interface were tested. A detailed description of the configuration required for both the Avaya and the is-phone equipment is documented within these Application Notes. The is-phone 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.